

## SIP-T2 Series/T19(P) E2/T4 Series/T5 Series IP Phones Administrator Guide



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# Introduction

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## About This Guide

Yealink administrator guide is intended for administrators who need to properly configure, customize, manage, and troubleshoot the IP phone system rather than end-users. This guide helps you understand the Voice over Internet Protocol (VoIP) network and Session Initiation Protocol (SIP) components, and provides descriptions of all available phone features.

This guide describes three methods for configuring IP phones: central provisioning, web user interface and phone user interface. It helps you perform the following tasks:

- Configure your IP phone on a provisioning server
- Configure your phone's features and functions via web/phone user interface
- Troubleshoot some common phone issues

Many of the features described in this guide involve network settings, which could affect the IP phone's performance in the network. So an understanding of IP networking and a prior knowledge of IP telephony concepts are necessary.

The information detailed in this guide is applicable to firmware version 82 or higher. The firmware format is like x.x.x.x.rom. The second x from left must be greater than or equal to 82 (for example, the firmware version of SIP-T46G IP phone: 28.82.0.20.rom).

See the [Yealink Products Regulatory Notices guide](#) for all regulatory and safety guidance.

## Chapters in This Guide

This administrator guide includes the following chapters:

- Chapter 1, "[Product Overview](#)" describes the SIP IP phones and expansion modules.
- Chapter 2, "[Getting Started](#)" describes how Yealink phones fit in your network and how to install and connect IP phones, and also gives you an overview of IP phone's initialization process.
- Chapter 3, "[Setting Up Your System](#)" describes some essential information on how to set up your phone network and set up your phone with a provisioning server.
- Chapter 4, "[Configuring Basic Features](#)" describes how to configure the basic features on IP phones.
- Chapter 5, "[Configuring Advanced Features](#)" describes how to configure the advanced features on IP phones.
- Chapter 6, "[Configuring Audio Features](#)" describes how to configure the audio features on IP phones.

- Chapter 7, "[Configuring Security Features](#)" describes how to configure the security features on IP phones.
- Chapter 8, "[Troubleshooting](#)" describes how to troubleshoot IP phones and provides some common troubleshooting solutions.
- Chapter 9, "[Appendix](#)" provides the glossary, time zones, trusted certificates, auto provisioning flowchart, reference information about IP phones compliant with [RFC 3261](#), SIP call flows and some other function lists (for example, DSS keys, reading icons).

## Related Documentations

This guide covers SIP-T54S, SIP-T52S, SIP-T48G/S, SIP-T46G/S, SIP-T42G/S, SIP-T41P/S, SIP-T40P/G, SIP-T29G, SIP-T27P/G, SIP-T23P/G, SIP-T21(P) E2 and SIP-T19(P) E2 IP phones. The following related documents are available:

- Quick Start Guides, which describe how to assemble IP phones and configure the most basic features available on IP phones.
- User Guides, which describe how to configure and use the basic and advanced features available on IP phones via phone user interface or via web user interface.
- Auto Provisioning Guide, which describes how to provision IP phones using the boot file and configuration files.

The purpose of *Auto Provisioning Guide* is to serve as a basic guidance for provisioning Yealink IP phones with a provisioning server. If you are new to this process, it is helpful to read this guide.

- Description of Configuration Parameters in CFG Files, which describes all configuration parameters in configuration files.

Note that Yealink administrator guide contains most of parameters. If you want to find out more parameters which are not listed in this guide, please refer to *Description of Configuration Parameters in CFG Files* guide.

- y000000000000.boot template boot file.
- <y0000000000xx>.cfg and <MAC>.cfg template configuration files.
- IP Phones Deployment Guide for BroadSoft UC-One Environments, which describes how to configure BroadSoft features on the BroadWorks web portal and IP phones.

IP Phone Features Integrated with BroadSoft UC-One User Guide, which describes how to configure and use IP phone features integrated with BroadSoft UC-One on Yealink IP phones.

When the SIP server type is set to BroadSoft, please refer to these two guides to have a better knowledge of configuring and using features integrated with Broadsoft UC-One.

For support or service, please contact your Yealink reseller or go to Yealink Technical Support online: <http://support.yealink.com/>.

## Conventions Used in Yealink Documentations

Yealink documentations contain a few typographic conventions and writing conventions.

You need to know the following basic typographic conventions to distinguish types of in-text information:

Convention	Description
<b>Bold</b>	Highlights the web/phone user interface items such as menus, menu selections, soft keys, or directory names when they are involved in a procedure or user action (for example, Click on <b>Settings</b> -> <b>Upgrade</b> .). Also used to emphasize text (for example, <b>Important!</b> ).
<i>Italics</i>	Used to show the format of examples (for example, <i>http(s)://[IPv6 address]</i> ), or to show the title of a section in the reference documentations available on the Yealink Technical Support Website (for example, <i>Triggering the IP phone to Perform the Auto Provisioning</i> ).
Blue Text	Used for cross references to other sections within this documentation (for example, refer to <a href="#">Ring Tones</a> on page 711), for hyperlinks to non-Yealink websites (for example, <a href="#">RFC 3315</a> ) or for hyperlinks to Yealink Technical Support website.
<i>Blue Text in Italics</i>	Used for hyperlinks to Yealink resources outside of this documentation such as the Yealink documentations (for example, <a href="#">Yealink_SIP_IP_Phones_Auto_Provisioning_Guide_V81</a> ).

You also need to know the following writing conventions to distinguish conditional information:

Convention	Description
<>	Indicates that you must enter information specific to phone or network. For example, when you see <MAC>, enter your phone's 12-digit MAC address. If you see <phoneIPAddress>, enter your phone's IP address.
->	Indicates that you need to select an item from a menu. For example, <b>Settings</b> -> <b>Basic Settings</b> indicates that you need to select <b>Basic Settings</b> from the <b>Settings</b> menu.

## Reading the Configuration Parameter Tables

The feature descriptions discussed in this guide include two tables. One is a summary table of provisioning methods that you can use to configure the features. The other is a table of details of the configuration parameters that you configure to make the features work.

This brief section describes the conventions used in the summary table and configuration parameter table. In order to read the tables and successfully perform configuration changes, an understanding of these conventions is necessary.

## Summary Table Format

The following summary table indicates three provisioning methods (central provisioning, web user interface and phone user interface, refer to [Provisioning Methods](#) for more information) you can use to configure a feature. Note that the type of provisioning methods available for each feature varies; not every feature uses all these three methods.

The central provisioning method requires you to configure parameters located in CFG format configuration files that Yealink provides. For more information on configuration files, refer to [Configuration Files](#) on page 101. As shown below, the table specifies the configuration file name and the corresponding parameters. That is, the <MAC>.cfg file contains the *account.X.auto\_answer* parameter, and the <y0000000000xx>.cfg file contains the *feature.auto\_answer\_delay* parameter.

The web user interface method requires you to configure features by navigating to the specified link. This navigation URL can help you quickly locate the webpage where you can configure the feature.

Provisioning method		Configuration file name	Feature explanation
Provisioning method	Central Provisioning (Configuration File)	<MAC>.cfg	Configure auto answer. <b>Parameter:</b> account.X.auto_answer
		<y0000000000xx>.cfg	Specify a period of delay time for auto answer. <b>Parameter:</b> features.auto_answer_delay
	Web User Interface		Configure auto answer. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?mode=mod_data&p=account-basic&q=load&acc=0
Manual provisioning method		Phone User Interface	Configure auto answer.

## Configuration Parameter Table Format

The following configuration parameter table describes the parameter that you can configure to make the feature (for example, auto answer) work.

Parameter name	Permitted parameter value	
<b>account.X.auto_answer</b>	<b>0 or 1</b>	<b>0</b> — Default Parameter value
<b>Description:</b> Enables or disables auto answer feature for account X. — Feature explanation <b>0-Disabled:</b> — Meaning of the parameter value <b>1-Enabled:</b> If it is set to 1 (Enabled), the IP phone can automatically answer an incoming call. X ranges from 1 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G). X ranges from 1 to 12 (for SIP-T52S/T42G/T42S). X ranges from 1 to 6 (for SIP-T41P/T41S/T27P/T27G). X ranges from 1 to 3 (for SIP-T40P/T40G/T23P/T23G). X ranges from 1 to 2 (for SIP-T21(P)-E2). X is equal to 1 (for SIP-T19(P)-E2/CP860). <b>Note:</b> The IP phone cannot automatically answer the incoming call during a call or while dialing even if auto answer is enabled. — Important information <b>Web User Interface:</b> Account->Basic->Auto Answer — Web path <b>Phone User Interface:</b> Menu->Features->Auto Answer->Line X->Auto Answer — Phone path		

### Note

Sometimes you will see the words "Refer to the following content" in the **Permitted Values** or **Default** field. It means the permitted value or the default value of the parameter has the model difference or there are many permitted values of the parameter, you can get more details from the following **Description** field.

The word "None" in the **Web User Interface** or **Phone User Interface** field means this feature cannot be configured via web/phone user interface.

The above table also indicates three methods for configuring the feature.

### Method 1: Central Provisioning

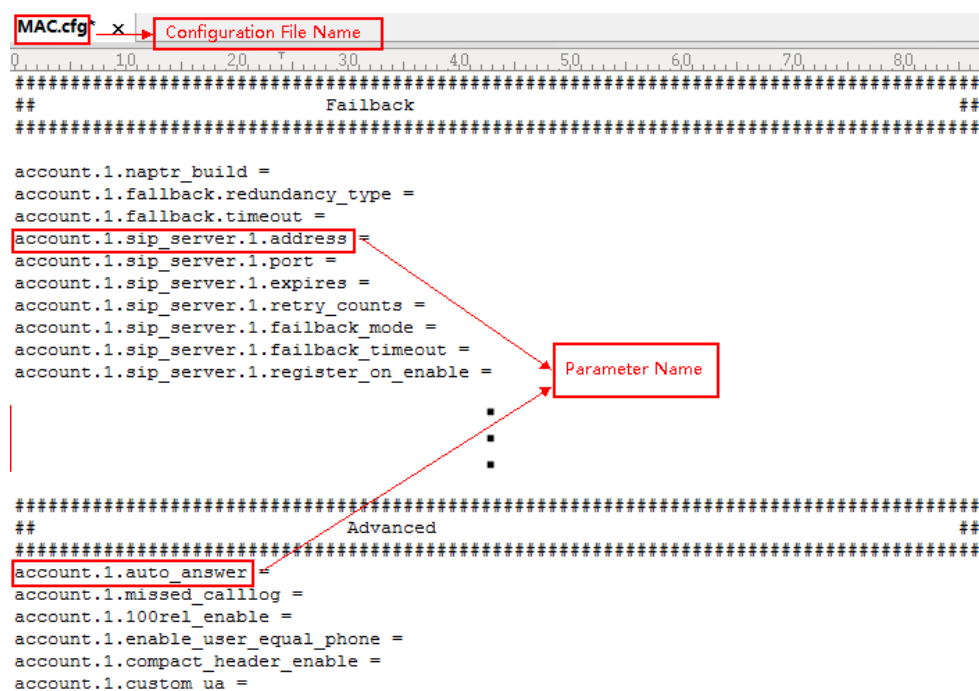
This table specifies the details of *account.X.auto\_answer* parameter, which enables or disables the auto answer feature. This parameter is disabled by default. If you want to enable the auto answer feature, open the MAC.cfg file and locate the parameter name *account.X.auto\_answer*. Set the parameter value to "1" to enable the auto answer feature or "0" to disable the auto answer feature.

Note that some parameters described in this guide contain one or more variables (for example, X or Y). But the variables in the parameters described in the CFG file are all replaced with specific value in the scope of variable. You may need to assign a value to the variable before you search

and locate the specific parameter in the CFG file.

For example, if you want to enable the auto answer feature for account 1, you need to locate the *account.1.auto\_answer* in the MAC.cfg file and then configure it as required (for example, *account.1.auto\_answer* = 1). If you want to configure the SIP server 1 for account 1, you can locate the *account.1.sip\_server.1.address* in the MAC.cfg file and configure it as required (for example, *account.1.sip\_server.1.address* = 10.2.1.48).

The following shows a segment of MAC.cfg file:



```

#####
##                               ##
#####

account.1.naptr_build =
account.1.fallback.redundancy_type =
account.1.fallback.timeout =
account.1.sip_server.1.address =
account.1.sip_server.1.port =
account.1.sip_server.1.expires =
account.1.sip_server.1.retry_counts =
account.1.sip_server.1.fallback_mode =
account.1.sip_server.1.fallback_timeout =
account.1.sip_server.1.register_on_enable =

.
.
.

#####
##                               ##
#####

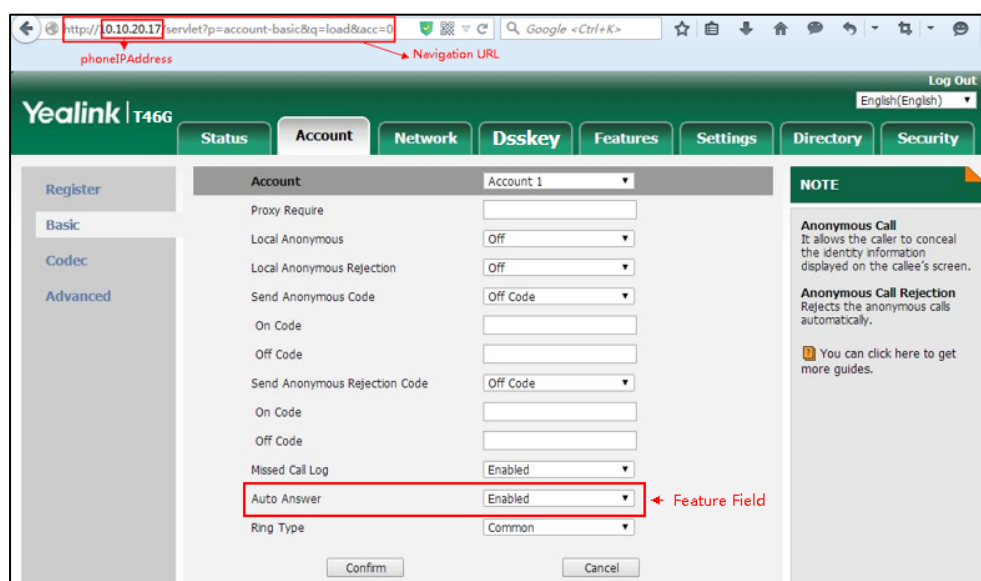
account.1.auto_answer =
account.1.missed_calllog =
account.1.100rel_enable =
account.1.enable_user_equal_phone =
account.1.compact_header_enable =
account.1.custom_ua =

```

## Method 2: Web User Interface

As described in the chapter [Summary Table Format](#), you can directly navigate to the specified webpage to configure the feature. You can also first log into the web user interface, and then locate the feature field according to the web path (for example, **Account->Basic->Auto Answer**) to configure it as required.

As shown in the following illustration:



To successfully log into the web user interface, you may need to enter the user name (default: admin) and password (default: admin). For more information, refer to [Web User Interface](#) on page 98.

### Method 3: Phone User Interface

You can configure features via phone user interface. Access to the desired feature according to the phone path (for example, **Menu->Features->Auto Answer->Line X->Auto Answer**) and then configure it as required.

As shown in the following illustration:



## Recommended References

For more information on configuring and administering other Yealink products not included in this guide, refer to product support page at [Yealink Technical Support](#).

To access the latest Release Notes or other guides for Yealink IP phones, refer to the Document Download page for your phone at [Yealink Technical Support](#).

If you want to find Request for Comments (RFC) documents, type <http://www.ietf.org/rfc/rfcNNNN.txt> (NNNN is the RFC number) into the location field of your browser.

This guide mainly takes the SIP-T46G IP phones as example for reference. For more details on other IP phones, refer to [Yealink phone-specific user guide](#).

For other references, look for the hyperlink or web info throughout this administrator guide.

## Understanding VoIP Principle and SIP Components

This section mainly describes the basic knowledge of VoIP principle and SIP components, which will help you have a better understanding of the phone's deployment scenarios.

### VoIP Principle

#### VoIP

**VoIP** (Voice over Internet Protocol) is a technology using the Internet Protocol instead of traditional Public Switch Telephone Network (PSTN) technology for voice communications.

It is a family of technologies, methodologies, communication protocols, and transmission techniques for the delivery of voice communications and multimedia sessions over IP networks. The H.323 and Session Initiation Protocol (SIP) are two popular VoIP protocols that are found in widespread implementation.

#### H.323

**H.323** is a recommendation from the ITU Telecommunication Standardization Sector (ITU-T) that defines the protocols to provide audio-visual communication sessions on any packet network. The H.323 standard addresses call signaling and control, multimedia transport and control, and bandwidth control for point-to-point and multi-point conferences.

It is widely implemented by voice and video conference equipment manufacturers, is used within various Internet real-time applications such as GnuGK and NetMeeting and is widely deployed by service providers and enterprises for both voice and video services over IP networks.

#### SIP

**SIP** (Session Initiation Protocol) is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. It is an ASCII-based, application-layer control protocol (defined in [RFC 3261](#)) that can be used to establish, maintain, and terminate calls between two or more endpoints. Like other VoIP protocols, SIP is designed to address functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control attributes of an end-to-end call.



SIP provides capabilities to:

- Determine the location of the target endpoint -- SIP supports address resolution, name mapping, and call redirection.
- Determine media capabilities of the target endpoint -- Via Session Description Protocol (SDP), SIP determines the "lowest level" of common services between endpoints. Conferences are established using only media capabilities that can be supported by all endpoints.
- Determine the availability of the target endpoint -- A call cannot be completed because the target endpoint is unavailable, SIP determines whether the called party is already on the IP phone or does not answer in the allotted number of rings. It then returns a message indicating why the target endpoint is unavailable.
- Establish a session between the origin and target endpoint -- The call can be completed, SIP establishes a session between endpoints. SIP also supports mid-call changes, such as the addition of another endpoint to the conference or the change of a media characteristic or codec.
- Handle the transfer and termination of calls -- SIP supports the transfer of calls from one endpoint to another. During a call transfer, SIP simply establishes a session between the transferee and a new endpoint (specified by the transferring party) and terminates the session between the transferee and the transferring party. At the end of a call, SIP terminates the sessions between all parties.

## SIP Components

SIP is a peer-to-peer protocol. The peers in a session are called User Agents (UAs). A user agent can function as one of following roles:

- User Agent Client (UAC) -- A client application that initiates the SIP request.
- User Agent Server (UAS) -- A server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.

### User Agent Client (UAC)

The UAC is an application that initiates up to six feasible SIP requests to the UAS. The six requests issued by the UAC are: INVITE, ACK, OPTIONS, BYE, CANCEL and REGISTER. When the SIP session is being initiated by the UAC SIP component, the UAC determines the information essential for the request, which is the protocol, the port and the IP address of the UAS to which the request is being sent. This information can be dynamic and will make it challenging to put through a firewall. For this reason, it may be recommended to open the specific application type on the firewall. The UAC is also capable of using the information in the request URI to establish the course of the SIP request to its destination, as the request URI always specifies the host which is essential. The port and protocol are not always specified by the request URI. Thus if the request does not specify a port or protocol, a default port or protocol is contacted. It may be preferential to use this method when not using an application layer firewall. Application layer

firewalls like to know what applications are flowing through which ports and it is possible to use content types of other applications other than the one you are trying to let through what has been denied.

## **User Agent Server (UAS)**

UAS is a server that hosts the application responsible for receiving the SIP requests from a UAC, and on reception it returns a response to the request back to the UAC. The UAS may issue multiple responses to the UAC, not necessarily a single response. Communication between UAC and UAS is client/server and peer-to-peer.

Typically, a SIP endpoint is capable of functioning as both a UAC and a UAS, but it functions only as one or the other per transaction. Whether the endpoint functions as a UAC or a UAS depends on the UA that initiates the request.

## **Summary of Changes**

This section describes the changes to this guide for each release and guide version.

### **Changes for Release 82, Guide Version 82.22**

Make minor corrections.

### **Changes for Release 82, Guide Version 82.21**

Make minor corrections.

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## Product Overview

This chapter contains the following information about IP phones:

- [SIP IP Phone Models](#)
- [Expansion Modules](#)

## SIP IP Phone Models

This section introduces SIP-T54S, SIP-T52S, SIP-T48G/S, SIP-T46G/S, SIP-T42G/S, SIP-T41P/S, SIP-T40P/G, SIP-T29G, SIP-T27P/G, SIP-T23P/G, SIP-T21(P) E2 and SIP-T19(P) E2 IP phone models. These IP phones are endpoints in the overall network topology, which are designed to interoperate with other compatible equipments including application servers, media servers, internet-working gateways, voice bridges, and other endpoints. These IP phones are characterized by a large number of functions, which simplify business communication with a high standard of security and can work seamlessly with a large number of SIP PBXs.

These IP phones provide a powerful and flexible IP communication solution for Ethernet TCP/IP networks, delivering excellent voice quality. The high-resolution graphic display supplies content in multiple languages for system status, call log and directory access. IP phones also support advanced functionalities, including LDAP, Busy Lamp Field, Sever Redundancy and Network Conference.

IP phones comply with the SIP standard ([RFC 3261](#)), and they can only be used within a network that supports this model of phone.

## Expansion Modules

This section introduces EXP20, EXP40 and EXP50 expansion modules. The expansion modules are consoles you can connect to Yealink IP phones to add DSS keys, which can be used to assign predefined functionalities for quickly accessing features. If you want to configure the expansion module keys, you have to connect the expansion module(s) to the IP phone in advance.

The following table lists the supported IP phone models for these expansion modules:

Expansion Module	Phone Model
EXP20	SIP-T29G/T27P/T27G
EXP40	SIP-T48G/T48S/T46G/T46S
EXP50	SIP-T54S/T52S

Expansion modules enable you to handle large volume of calls on a regular basis and expand

the functional capability of your IP phone. For more information on how to connect and use the expansion module, refer to [\*Yealink EXP-specific user guide\*](#).

# Getting Started

---

This chapter describes where Yealink IP phones fit in your network and provides basic installation instructions of IP phones.

This chapter provides the following sections:

- [What IP Phones Need to Meet](#)
- [Yealink IP Phones in a Network](#)
- [Initialization Process Overview](#)
- [Verifying Startup](#)

## What IP Phones Need to Meet

In order to operate as SIP endpoints in your network successfully, IP phones must meet the following requirements:

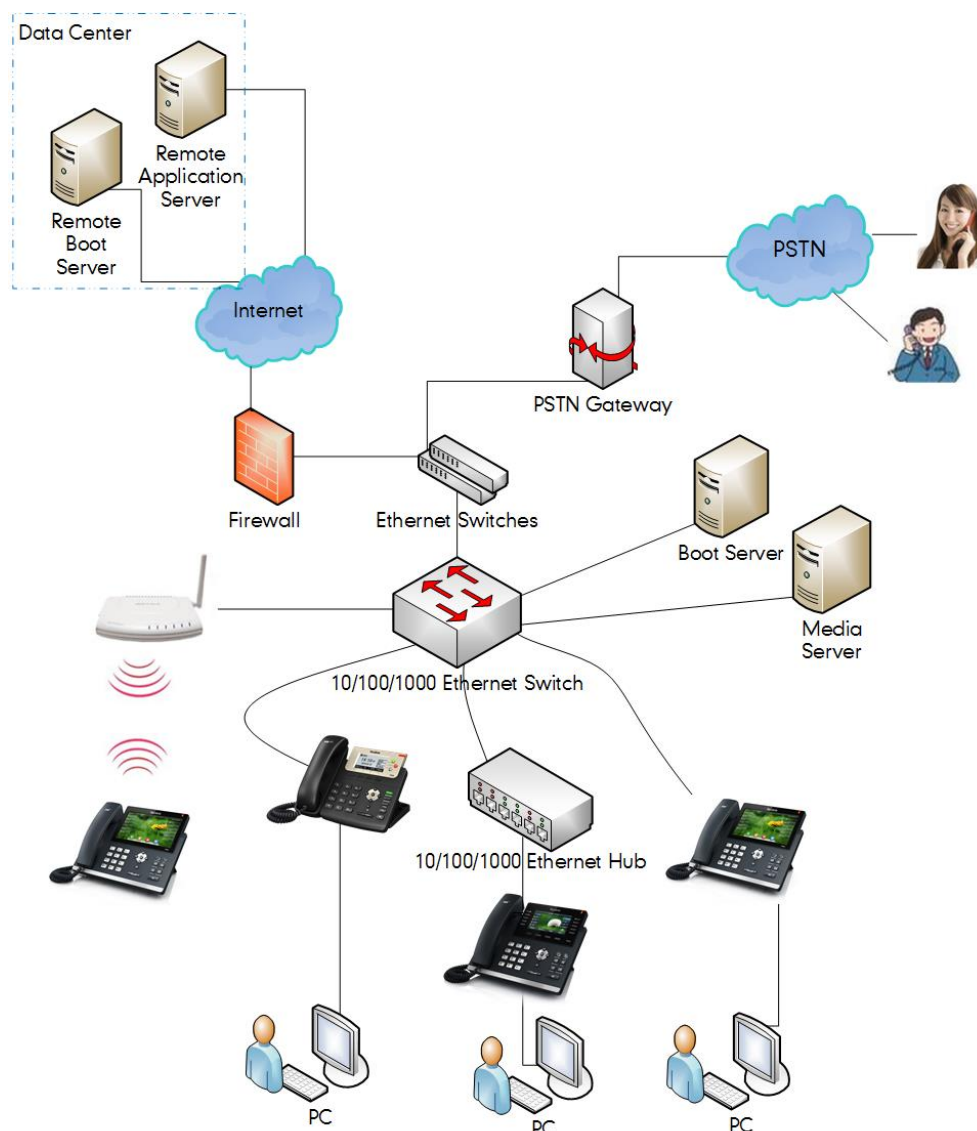
- A working IP network is established.
- VoIP gateways are configured for SIP.
- The latest (or compatible) firmware of IP phones is available.
- A call server is active and configured to receive and send SIP messages.

## Yealink IP Phones in a Network

Most Yealink IP phones connect physically through a Category 5E (CAT5E) cable to a 10/100/1000Mbps Ethernet LAN, and send and receive all data using the same packet-based technology. Some phones (for example, SIP-T48G) can connect to the wireless network.

Since the IP phone is a data terminal, digitized audio being just another type of data from its perspective, the phone is capable of vastly more than traditional business phones. Moreover, Yealink IP phones run the same protocols as your office personal computer, which means that many innovative applications can be developed without resorting to specialized technology.

There are many ways to set up a phone network using Yealink IP phones. The following shows an example of a network setup:



## Initialization Process Overview

The initialization process of the IP phone is responsible for network connectivity and operation of the IP phone in your local network.

Once you connect your IP phone to the network and to an electrical supply, the IP phone begins its initialization process.

During the initialization process, the following events take place:

### **Loading the ROM file**

The ROM file resides in the flash memory of the IP phone. The IP phone comes from the factory with a ROM file preloaded. During initialization, the IP phone runs a bootstrap loader that loads and executes the ROM file.

### **Configuring the VLAN**

If the IP phone is connected to a switch, the switch notifies the IP phone of the VLAN information defined on the switch (if using LLDP or CDP). The IP phone can then proceed with the DHCP request for its network settings (if using DHCP). For more information on VLAN, refer to [VLAN](#) on page 37.

### **Querying the DHCP (Dynamic Host Configuration Protocol) Server**

The IP phone is capable of querying a DHCP server. DHCP is enabled on the IP phone by default. The following network parameters can be obtained from the DHCP server during initialization:

- IP Address
- Subnet Mask
- Default Gateway
- Primary DNS (Domain Name Server)
- Secondary DNS

You need to configure network parameters of the IP phone manually if any of them is not supplied by the DHCP server. For more information on configuring network parameters manually, refer to [Configuring Network Parameters Manually](#) on page 18.

### **Contacting the provisioning server**

If the IP phone is configured to obtain configurations from the provisioning server, it will connect to the provisioning server, download the boot file(s) and configuration file(s) during startup. The IP phone will be able to resolve and update configurations written in the configuration file(s). If the IP phone does not obtain configurations from the provisioning server, the IP phone will use configurations stored in the flash memory. For more information, refer to [Setting Up Your Phones with a Provisioning Server](#) on page 89.

### **Updating firmware**

If the access URL of firmware is defined in the configuration file, the IP phone will download firmware from the provisioning server. If the MD5 value of the downloaded firmware file differs from that of the image stored in the flash memory, the IP phone will perform a firmware update. You can manually upgrade firmware if the IP phone does not download firmware from the provisioning server. For more information, refer to [Upgrading Firmware](#) on page 111.

### **Downloading the resource files**

In addition to configuration file(s), the IP phone may require resource files before it can deliver

service. These resource files are optional, but if some particular features are being deployed, these files are required.

The followings show examples of resource files:

- Language packs
- Ring tones
- Contact files

For more information on resource files, refer to [Resource Files](#) on page 103.

## Verifying Startup

After connected to the power and network, the IP phone begins the initializing process by cycling through the following steps:

1. The power indicator LED illuminates solid red.
2. The message "Welcome Initializing... please wait" appears on the LCD screen when the IP phone starts up.
3. The main LCD screen displays the following:
  - Time and date
  - Soft key labels
4. Press the OK/ ✓ key or press **Menu->Status** to check the IP phone status, the LCD screen displays the valid IP address, MAC address, firmware version, and so on.

If the IP phone has successfully passed through these steps, it starts up properly and is ready for use.



# Setting Up Your System

---

This section describes essential information on how to set up your phone network and set up your phones with a provisioning server. It also provides instructions on how to set up a provisioning server, how to deploy Yealink IP phones from the provisioning server, how to upgrade firmware, and how to keep user personalized settings after auto provisioning.

This chapter provides the following sections:

- [Setting Up Your Phone Network](#)
- [Setting Up Your Phones with a Provisioning Server](#)

## Setting Up Your Phone Network

Yealink IP phones operate on an Ethernet local area network (LAN) or wireless network. Local area network design which varies by organization and Yealink IP phones can be configured to accommodate a number of network designs.

In order to get your IP phones running, you must perform basic network setup, such as IP address and subnet mask configuration. You can configure the IPv4 or IPv6 network parameters for the phone. You can also configure the appropriate security (VLAN and/or 802.1X authentication) and Quality of Service (QoS) settings for the IP phone.

This chapter describes how to configure all the network parameters for IP phones, and it provides the following sections:

- [DHCP](#)
- [DHCP Option](#)
- [Configuring Network Parameters Manually](#)
- [Configuring Transmission Methods of the Internet Port and PC Port](#)
- [Configuring PC Port Mode](#)
- [Web Server Type](#)
- [Wi-Fi](#)
- [VLAN](#)
- [IPv6 Support](#)
- [VPN](#)
- [Network Address Translation \(NAT\)](#)
- [Quality of Service \(QoS\)](#)
- [802.1X Authentication](#)

## DHCP

DHCP (Dynamic Host Configuration Protocol) is a network protocol used to dynamically allocate network parameters to network hosts. The automatic allocation of network parameters to hosts eases the administrative burden of maintaining an IP network. IP phones comply with the DHCP specifications documented in [RFC 2131](#). If using DHCP, IP phones connected to the network become operational without having to be manually assigned IP addresses and additional network parameters.

### Procedure

DHCP can be configured using the following methods.

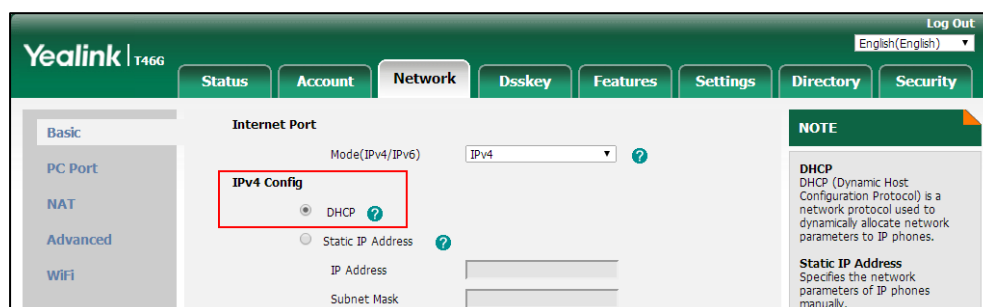
<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure DHCP on the IP phone. <b>Parameter:</b> static.network.internet_port.type
<b>Web User Interface</b>		Configure DHCP on the IP phone. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=network &q=load
<b>Phone User Interface</b>		Configure DHCP on the IP phone.

### Details of Configuration Parameter:

Parameter	Permitted Values	Default
<b>static.network.internet_port.type</b>	<b>0 or 2</b>	<b>0</b>
<p><b>Description:</b> Configures the Internet port type for IPv4.</p> <p><b>0</b>-DHCP <b>2</b>-Static IP Address</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 0 (IPv4) or 2 (IPv4 &amp; IPv6). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Basic-&gt;IPv4 Config</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;WAN Port-&gt;IPv4</p>		

**To configure DHCP via web user interface:**

1. Click on **Network**->**Basic**.
2. In the **IPv4 Config** block, mark the **DHCP** radio box.



3. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
4. Click **OK** to reboot the phone.

**To configure DHCP via phone user interface:**

1. Press **Menu**->**Advanced** (default password: admin) ->**Network**->**WAN Port**->**IPv4**.
2. Press **Left** or **Right** arrow, or the **Switch** soft key to select the **DHCP** from the **Type** field.
3. Press the **Save** soft key to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
4. Press **OK** to reboot the phone.

## Static DNS

Static DNS address(es) can be configured and used even though DHCP is enabled.

### Procedure

Static DNS can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<y0000000000xx>.cfg	Configure the static DNS feature. <b>Parameter:</b> static.network.static_dns_enable
	<MAC>.cfg	Configure static DNS address. <b>Parameters:</b> static.network.primary_dns static.network.secondary_dns
<b>Web User Interface</b>		Configure the static DNS feature. Configure static DNS address. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=netwo">http://&lt;phoneIPAddress&gt;/servlet?p=netwo</a>

	rk&q=load
<b>Phone User Interface</b>	Configure the static DNS feature. Configure static DNS address.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>static.network.static_dns_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Triggers the static DNS feature to on or off.</p> <p><b>0</b>-Off, the IP phone will use the IPv4 DNS obtained from DHCP.</p> <p><b>1</b>-On, the IP phone will use manually configured static IPv4 DNS.</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.internet_port.type" is set to 0 (DHCP). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Basic-&gt;IPv4 Config-&gt;Static DNS</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;WAN Port-&gt;IPv4-&gt;Type (DHCP)-&gt;Static DNS</p>		
<b>static.network.primary_dns</b>	<b>IPv4 Address</b>	<b>Blank</b>
<p><b>Description:</b> Configures the primary IPv4 DNS server.</p> <p><b>Example:</b> static.network.primary_dns = 202.101.103.55</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.static_dns_enable" is set to 1 (On). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Basic-&gt;IPv4 Config-&gt;Static IP Address-&gt;Primary DNS</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;WAN Port-&gt;IPv4-&gt;Type (DHCP)-&gt;Static DNS (Enabled) -&gt;Pri.DNS</p>		
<b>static.network.secondary_dns</b>	<b>IPv4 Address</b>	<b>Blank</b>
<p><b>Description:</b></p>		

Parameters	Permitted Values	Default
<p>Configures the secondary IPv4 DNS server.</p> <p><b>Example:</b></p> <p>static.network.secondary_dns = 202.101.103.54</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.static_dns_enable" is set to 1 (On). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Basic-&gt;IPv4 Config-&gt;Static IP Address-&gt;Secondary DNS</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;WAN Port-&gt;IPv4-&gt;Type (DHCP)-&gt;Static DNS (Enabled) -&gt;Sec.DNS</p>		

**To configure static DNS address when DHCP is used via web user interface:**

1. Click on **Network->Basic**.
2. In the **IPv4 Config** block, mark the **DHCP** radio box.
3. In the **Static DNS** block, mark the **On** radio box.
4. Enter the desired values in the **Primary DNS** and **Secondary DNS** fields.

5. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
6. Click **OK** to reboot the phone.

**To configure static DNS when DHCP is used via phone user interface:**

1. Press **Menu->Advanced** (default password: admin) ->**Network->WAN Port->IPv4**.
2. Press **Left** or **Right** arrow, or the **Switch** soft key to select the **DHCP** from the **Type** field.
3. Press **Left** or **Right** arrow, or the **Switch** soft key to select the **Enabled** from the **Static DNS** field.
4. Enter the desired value in the **Pri.DNS** and **Sec.DNS** field respectively.
5. Press the **Save** soft key to accept the change.

A dialog box pops up to prompt that the settings will take effect after a reboot.

6. Press **OK** to reboot the phone.

## DHCP Option

DHCP provides a framework for passing information to TCP/IP network devices. Network and other control information are carried in tagged data items that are stored in the options field of the DHCP message. The data items themselves are also called options.

DHCP can be initiated by simply connecting the IP phone with the network. IP phones broadcast DISCOVER messages to request the network information carried in DHCP options, and the DHCP server responds with specific values in corresponding options.

The following table lists common DHCP options supported by IP phones.

Parameter	DHCP Option	Description
Subnet Mask	1	Specify the client's subnet mask.
Time Offset	2	Specify the offset of the client's subnet in seconds from Coordinated Universal Time (UTC).
Router	3	Specify a list of IP addresses for routers on the client's subnet.
Time Server	4	Specify a list of time servers available to the client.
Domain Name Server	6	Specify a list of domain name servers available to the client.
Host Name	12	Specify the name of the client.
Domain Server	15	Specify the domain name that client should use when resolving hostnames via DNS.
Network Time Protocol Servers	42	Specify a list of NTP servers available to the client by IP address.
Vendor-Specific Information	43	Identify the vendor-specific information.
Vendor Class Identifier	60	Identify the vendor type.
TFTP Server Name	66	Identify a TFTP server when the 'sname' field in the DHCP header has been used for DHCP options.

For more information on DHCP options, refer to [RFC 2131](#) or [RFC 2132](#).

If you do not have the ability to configure the DHCP options for discovering the provisioning server on the DHCP server, an alternate method of automatically discovering the provisioning

server address is required. Connecting to the secondary DHCP server that responds to DHCP INFORM queries with a requested provisioning server address is one possibility. For more information, refer to [RFC 3925](#). If a single alternate DHCP server responds, this is functionally equivalent to the scenario where the primary DHCP server responds with a valid provisioning server address. If no DHCP servers respond, the INFORM query process will retry and eventually time out.

## DHCP Option 66 and Option 43

During the startup, the phone will automatically detect the custom option, option 66 or option 43 for obtaining the provisioning server address. The priority of obtaining the provisioning server address is as follows: custom option->option 66 (identify the TFTP server) ->option 43. The IP phone can obtain the Auto Configuration Server (ACS) address by detecting option 43 during startup.

To obtain the server address via DHCP option, make sure the DHCP option is properly configured on the phone. The option must be in accordance with the one defined in the DHCP server.

### Procedure

DHCP Active can be configured using the following methods.

Central Provisioning (Configuration File)	<y0000000000xx>.cfg	Configure DHCP Active. <b>Parameter:</b> static.auto_provision.dhcp_option.enable
		Configure the custom DHCP option. <b>Parameter:</b> static.auto_provision.dhcp_option.list_user_opt ions
Web User Interface		Configure DHCP Active. Configure the custom DHCP option. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=settings-autop&q=load

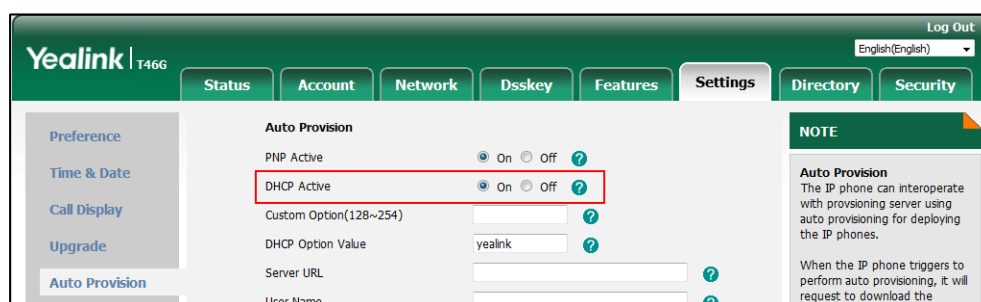
### Details of Configuration Parameters:

Parameters	Permitted Values	Default
static.auto_provision.dhcp_option.enable	0 or 1	1
<b>Description:</b> Triggers the DHCP Active feature to on or off.		

Parameters	Permitted Values	Default
<b>0-Off</b> <b>1-On</b> , the IP phone will obtain the provisioning server address by detecting DHCP options. <b>Web User Interface:</b> Settings->Auto Provision->DHCP Active <b>Phone User Interface:</b> None		
<b>static.auto_provision.dhcp_option.list_user_options</b>	<b>Integer from 128 to 254</b>	<b>Blank</b>
<b>Description:</b> Configures the custom DHCP option for requesting provisioning server address. Multiple options are separated by commas. <b>Note:</b> It works only if the value of the parameter "static.auto_provision.dhcp_option.enable" is set to 1 (On). <b>Web User Interface:</b> Settings->Auto Provision->Custom Option(128-254) <b>Phone User Interface:</b> None		

**To configure the DHCP Active feature via web user interface:**

1. Click on **Settings->Auto Provision**.
2. Mark the **On** radio box in the **DHCP Active** field.



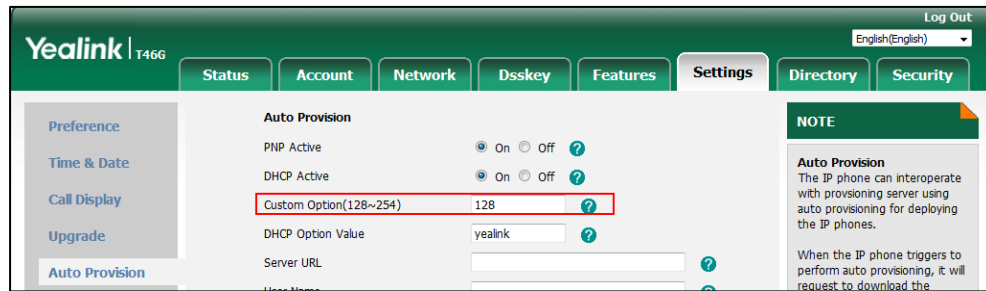
3. Click **Confirm** to accept the change.

**To configure the custom DHCP option via web user interface:**

1. Click on **Settings->Auto Provision**.



2. Enter the desired value in the **Custom Option(128~254)** field.



3. Click **Confirm** to accept the change.

## DHCP Option 42 and Option 2

Yealink IP phones support using the NTP server address offered by DHCP.

DHCP option 42 is used to specify a list of NTP servers available to the client by IP address. NTP servers should be listed in order of preference. DHCP option 2 is used to specify the offset of the client's subnet in seconds from Coordinated Universal Time (UTC).

To update time with the offset time offered by the DHCP server, make sure the DHCP Time feature is enabled at the web path **Settings->Time & Date->DHCP Time**. For more information on how to configure DHCP time feature, refer to [NTP Time Server](#) on page 214.

## DHCP Option 12 Hostname on the IP Phone

This option specifies the host name of the client. The name may or may not be qualified with the local domain name (based on [RFC 2132](#)). See [RFC 1035](#) for character restrictions.

### Procedure

DHCP Option 12 Hostname can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the DHCP Option 12 Hostname. <b>Parameter:</b> static.network.dhcp_host_name
<b>Web User Interface</b>		Configure the DHCP Option 12 Hostname. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-general&q=load

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>static.network.dhcp_host_name</b>	<b>String within 99 characters</b>	<b>Refer to the following content</b>
<p><b>Description:</b></p> <p>Configures the DHCP Option 12 Hostname on the IP phone.</p> <p><b>For SIP-T54S IP phones:</b></p> <p>The default value is SIP-T54S.</p> <p><b>For SIP-T52S IP phones:</b></p> <p>The default value is SIP-T52S.</p> <p><b>For SIP-T48G IP phones:</b></p> <p>The default value is SIP-T48G.</p> <p><b>For SIP-T48S IP phones:</b></p> <p>The default value is SIP-T48S.</p> <p><b>For SIP-T46G IP phones:</b></p> <p>The default value is SIP-T46G.</p> <p><b>For SIP-T46S IP phones:</b></p> <p>The default value is SIP-T46S.</p> <p><b>For SIP-T42G IP phones:</b></p> <p>The default value is SIP-T42G.</p> <p><b>For SIP-T42S IP phones:</b></p> <p>The default value is SIP-T42S.</p> <p><b>For SIP-T41P IP phones:</b></p> <p>The default value is SIP-T41P.</p> <p><b>For SIP-T41S IP phones:</b></p> <p>The default value is SIP-T41S.</p> <p><b>For SIP-T40P IP phones:</b></p> <p>The default value is SIP-T40P.</p> <p><b>For SIP-T40G IP phones:</b></p> <p>The default value is SIP-T40G.</p> <p><b>For SIP-T29G IP phones:</b></p> <p>The default value is SIP-T29G.</p> <p><b>For SIP-T27P IP phones:</b></p> <p>The default value is SIP-T27P.</p> <p><b>For SIP-T27G IP phones:</b></p>		

Parameter	Permitted Values	Default
<p>The default value is SIP-T27G.</p> <p><b>For SIP-T23P IP phones:</b></p> <p>The default value is SIP-T23P.</p> <p><b>For SIP-T23G IP phones:</b></p> <p>The default value is SIP-T23G.</p> <p><b>For SIP-T21P E2 IP phones:</b></p> <p>The default value is SIP-T21P_E2.</p> <p><b>For SIP-T21 E2 IP phones:</b></p> <p>The default value is SIP-T21_E2.</p> <p><b>For SIP-T19P E2 IP phones:</b></p> <p>The default value is SIP-T19P_E2.</p> <p><b>For SIP-T19 E2 IP phones:</b></p> <p>The default value is SIP-T19_E2.</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;DHCP Hostname</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure DHCP Option 12 Hostname on the IP phone via web user interface:**

1. Click on **Features->General Information**.
2. Enter the desired host name in the **DHCP Hostname** field.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'DHCP Hostname' field is highlighted with a red box and contains the value 'SIP-T46G'. Other settings visible include 'Call Waiting' (Enabled), 'Auto Redial' (Disabled), 'Voice Mail Tone' (Enabled), 'Reboot in Talking' (Disabled), 'Hide Feature Access Codes' (Disabled), 'Display Method on Dialing' (User Name), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right provides additional information about various features like Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

3. Click **Confirm** to accept the change.

A dialog box pops up to prompt that the settings will take effect after a reboot.

4. Click **OK** to reboot the phone.

## Configuring Network Parameters Manually

If DHCP is disabled or IP phones cannot obtain network parameters from the DHCP server, you need to configure them manually. The following parameters should be configured for IP phones to establish network connectivity:

- IP Address
- Subnet Mask
- Default Gateway
- Primary DNS
- Secondary DNS

### Procedure

Network parameters can be configured manually using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	<p>Configure network parameters of the IP phone manually.</p> <p><b>Parameters:</b></p> <p>static.network.internet_port.type</p> <p>static.network.ip_address_mode</p> <p>static.network.internet_port.ip</p> <p>static.network.internet_port.mask</p> <p>static.network.internet_port.gateway</p> <p>static.network.primary_dns</p> <p>static.network.secondary_dns</p>
<b>Web User Interface</b>		<p>Configure network parameters of the IP phone manually.</p> <p><b>Navigate to:</b></p> <p><a href="http://&lt;phoneIPAddress&gt;/servlet?p=network&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=network&amp;q=load</a></p>
<b>Phone User Interface</b>		<p>Configure network parameters of the IP phone manually.</p>

**Details of Configuration Parameters:**

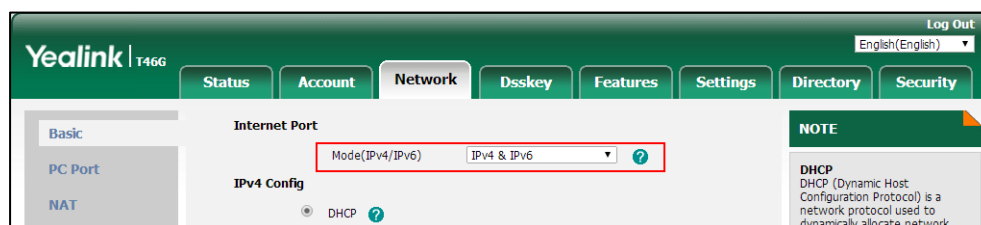
Parameters	Permitted Values	Default
<b>static.network.internet_port.type</b>	<b>0 or 2</b>	<b>0</b>
<p><b>Description:</b> Configures the Internet port type for IPv4.</p> <p><b>0</b>-DHCP <b>2</b>-Static IP Address</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 0 (IPv4) or 2 (IPv4 &amp; IPv6). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Basic-&gt;IPv4 Config</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;WAN Port-&gt;IPv4-&gt;Type</p>		
<b>static.network.ip_address_mode</b>	<b>0, 1 or 2</b>	<b>0</b>
<p><b>Description:</b> Configures the IP address mode.</p> <p><b>0</b>-IPv4 <b>1</b>-IPv6 <b>2</b>-IPv4 &amp; IPv6</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Basic-&gt;Internet Port-&gt;Mode(IPv4/IPv6)</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;WAN Port-&gt;IP Mode</p>		
<b>static.network.internet_port.ip</b>	<b>IPv4 Address</b>	<b>Blank</b>
<p><b>Description:</b> Configures the IPv4 address.</p> <p><b>Example:</b> static.network.internet_port.ip = 192.168.1.20</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 0 (IPv4) or 2 (IPv4 &amp; IPv6), and "static.network.internet_port.type" is set to 2 (Static IP Address). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
Network->Basic->IPv4 Config->Static IP Address->IP Address <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type (Static IP)->IP Address		
<b>static.network.internet_port.mask</b>	<b>Subnet Mask</b>	<b>Blank</b>
<b>Description:</b> Configures the IPv4 subnet mask. <b>Example:</b> static.network.internet_port.mask = 255.255.255.0 <b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 0 (IPv4) or 2 (IPv4 & IPv6), and "static.network.internet_port.type" is set to 2 (Static IP Address). If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Basic->IPv4 Config->Static IP Address->Subnet Mask <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type (Static IP)->Subnet Mask		
<b>static.network.internet_port.gateway</b>	<b>IPv4 Address</b>	<b>Blank</b>
<b>Description:</b> Configures the IPv4 default gateway. <b>Example:</b> static.network.internet_port.gateway = 192.168.1.254 <b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 0 (IPv4) or 2 (IPv4 & IPv6), and "static.network.internet_port.type" is set to 2 (Static IP Address). If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Basic->IPv4 Config->Static IP Address->Default Gateway <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type (Static IP)->Default Gateway		
<b>static.network.primary_dns</b>	<b>IPv4 Address</b>	<b>Blank</b>
<b>Description:</b> Configures the primary IPv4 DNS server.		

Parameters	Permitted Values	Default
<b>Example:</b> static.network.primary_dns = 202.101.103.55  <b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 0 (IPv4) or 2 (IPv4 & IPv6), and "static.network.internet_port.type" is set to 2 (Static IP Address). If you change this parameter, the IP phone will reboot to make the change take effect.  <b>Web User Interface:</b> Network->Basic->IPv4 Config->Static IP Address->Primary DNS  <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type (Static IP)->Pri.DNS		
<b>static.network.secondary_dns</b>	<b>IPv4 Address</b>	<b>Blank</b>
<b>Description:</b> Configures the secondary IPv4 DNS server.  <b>Example:</b> static.network.secondary_dns = 202.101.103.54  <b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 0 (IPv4) or 2 (IPv4 & IPv6), and "static.network.internet_port.type" is set to 2 (Static IP Address). If you change this parameter, the IP phone will reboot to make the change take effect.  <b>Web User Interface:</b> Network->Basic->IPv4 Config->Static IP Address->Secondary DNS  <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type (Static IP)->Sec.DNS		

**To configure the IP address mode via web user interface:**

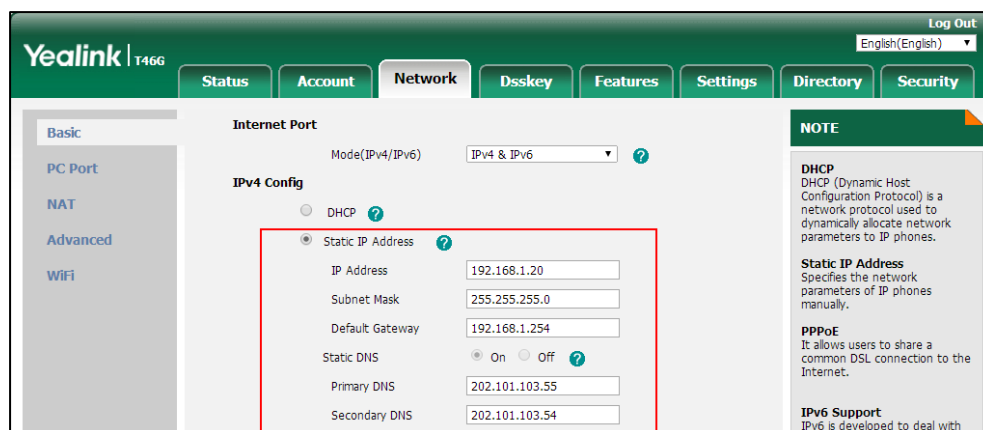
1. Click on **Network->Basic**.
2. Select desired value from the pull-down list of **Mode(IPv4/IPv6)**.



3. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
4. Click **OK** to reboot the phone.

### To configure a static IPv4 address via web user interface:

1. Click on **Network->Basic**.
2. In the **IPv4 Config** block, mark the **Static IP Address** radio box.
3. Enter the desired values in the **IP Address**, **Subnet Mask**, **Default Gateway**, **Primary DNS** and **Secondary DNS** fields.



4. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
5. Click **OK** to reboot the phone.

### To configure the IP address mode via phone user interface:

1. Press **Menu->Advanced** (default password: admin) -> **Network->WAN Port**.
2. Press **◀** or **▶** to select **IPv4** or **IPv4 & IPv6** from the **IP Mode** field.
3. Press the **Save** soft key to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
4. Press **OK** to reboot the phone.

### To configure a static IPv4 address via phone user interface:

1. Press **Menu->Advanced** (default password: admin) -> **Network->WAN Port->IPv4**.
2. Press **◀** or **▶**, or the **Switch** soft key to select the **Static IP** from the **Type** field.
3. Enter the desired value in the **IP Address**, **Subnet Mask**, **Default Gateway**, **Pri.DNS** and **Sec.DNS** field respectively.
4. Press the **Save** soft key to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
5. Press **OK** to reboot the phone.



## Configuring Transmission Methods of the Internet Port and PC Port

Yealink IP phones support two Ethernet ports: Internet port and PC port. You can enable or disable the PC port on the IP phones. Three optional methods of transmission configuration for IP phone Internet port or PC port:

- Auto-negotiate
- Half-duplex
- Full-duplex

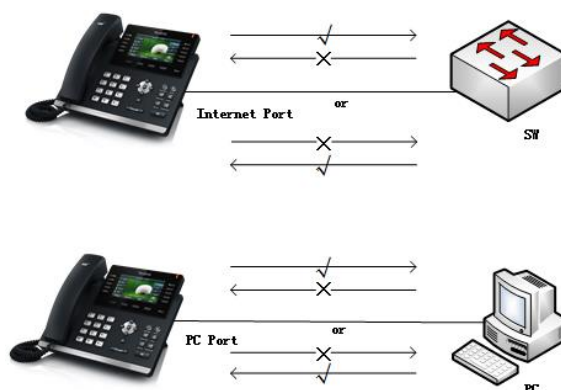
Auto-negotiate is configured for both Internet and PC ports on the IP phone by default.

### Auto-negotiate

Auto-negotiate means that two connected devices choose common transmission parameters (for example, speed and duplex mode) to transmit voice or data over Ethernet. This process entails devices first sharing transmission capabilities and then selecting the highest performance transmission mode supported by both. You can configure the Internet port and PC port on the IP phone to automatically negotiate during the transmission.

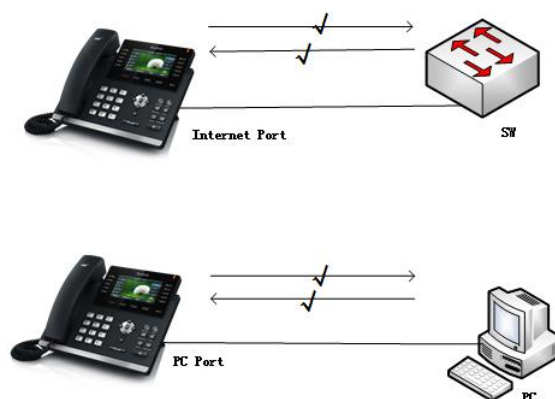
### Half-duplex

Half-duplex transmission refers to transmitting voice or data in both directions, but in one direction at a time; this means one device can send data on the line, but not receive data simultaneously. You can configure the half-duplex transmission on both Internet port and PC port for the IP phone to transmit in 10Mbps or 100Mbps.



## Full-duplex

Full-duplex transmission refers to transmitting voice or data in both directions at the same time; this means one device can send data on the line while receiving data. You can configure the full-duplex transmission on both Internet port and PC port for the IP phone to transmit in 10Mbps, 100Mbps or 1000Mbps (1000Mbps is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T40G/T29G/T27G/T23G IP phones).



## Procedure

The transmission methods of Ethernet ports can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.c fg	Configure the transmission methods of the Ethernet ports. <b>Parameters:</b> static.network.internet_port.speed_duplex static.network.pc_port.speed_duplex
<b>Web User Interface</b>		Configure the transmission methods of the Ethernet ports. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=network-adv&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=network-adv&amp;q=load</a>

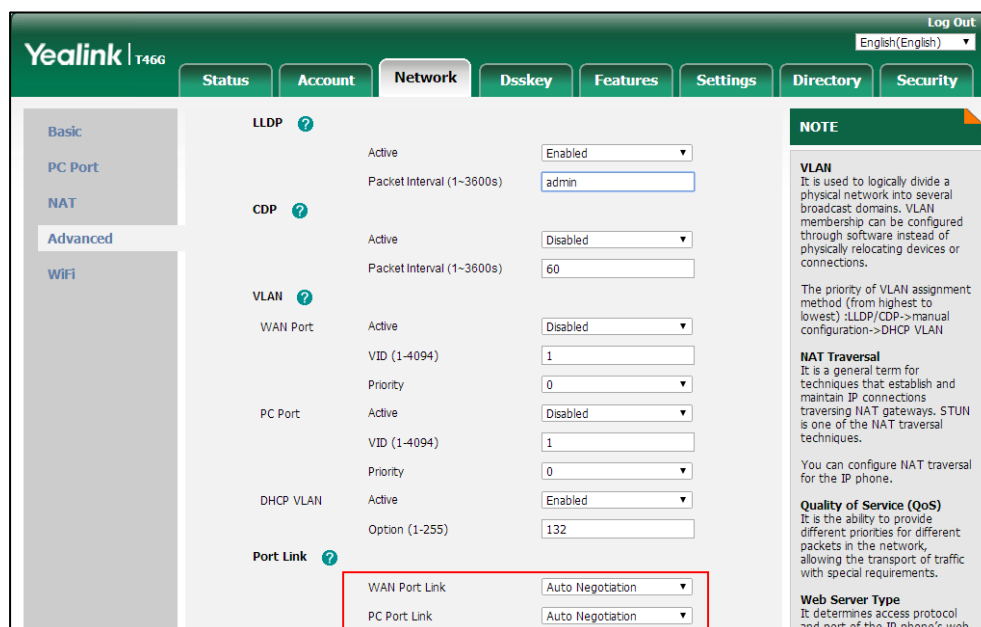
## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.network.internet_port.speed_duplex</b>	<b>0, 1, 2, 3, 4 or 5</b>	<b>0</b>
<b>Description:</b> Configures the transmission method of the Internet port. <b>0</b> -Auto Negotiation <b>1</b> -Full Duplex 10Mbps		

Parameters	Permitted Values	Default
<p><b>2</b>-Full Duplex 100Mbps</p> <p><b>3</b>-Half Duplex 10Mbps</p> <p><b>4</b>-Half Duplex 100Mbps</p> <p><b>5</b>-Full Duplex 1000Mbps (only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T40G/T29G/T27G/T23G IP phones)</p> <p><b>Note:</b> For SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T40G/T29G/T27G/T23G IP phones, you can set the transmission speed to 1000Mbps/Auto Negotiation to transmit in 1000Mbps if the IP phone is connected to the switch supports Gigabit Ethernet. We recommend that you do not change this parameter. If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;Port Link-&gt;WAN Port Link</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.network.pc_port.speed_duplex</b>	<b>0, 1, 2, 3, 4 or 5</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the transmission method of the PC port.</p> <p><b>0</b>-Auto Negotiation</p> <p><b>1</b>-Full Duplex 10Mbps</p> <p><b>2</b>-Full Duplex 100Mbps</p> <p><b>3</b>-Half Duplex 10Mbps</p> <p><b>4</b>-Half Duplex 100Mbps</p> <p><b>5</b>-Full Duplex 1000Mbps (only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T40G/T29G/T27G/T23G IP phones)</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.pc_port.enable" is set to 1 (Auto Negotiation). For SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T40G/T29G/T27G/T23G IP phones, you can set the transmission speed to 1000Mbps/Auto Negotiation to transmit in 1000Mbps if the IP phone is connected to the switch supports Gigabit Ethernet. We recommend that you do not change this parameter. If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;Port Link-&gt;PC Port Link</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure the transmission methods of Ethernet ports via web user interface:**

1. Click on **Network->Advanced**.
2. Select the desired value from the pull-down list of **WAN Port Link**.
3. Select the desired value from the pull-down list of **PC Port Link**.



4. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
5. Click **OK** to reboot the phone.

## Configuring PC Port Mode

The PC port is used to connect a PC. You can enable or disable the PC port on the IP phone.

### Procedure

PC port mode can be configured using the following methods.

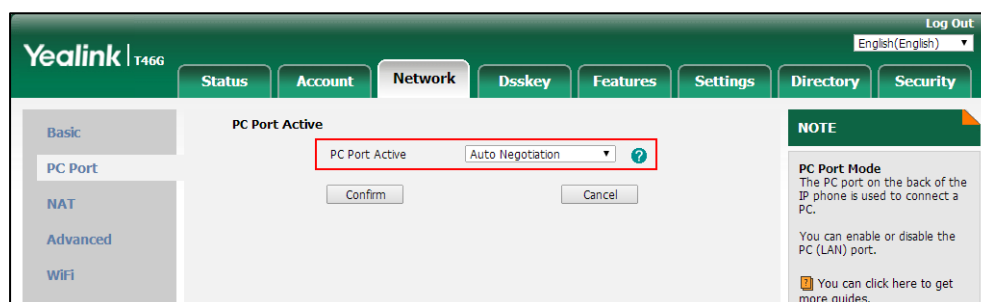
<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the PC port. <b>Parameter:</b> static.network.pc_port.enable
<b>Web User Interface</b>		Configure the PC port. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =network-pcport&q=load

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>static.network.pc_port.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the PC port.</p> <p><b>0</b>-Disabled <b>1</b>-Auto Negotiation</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;PC Port-&gt;PC Port Active</p> <p><b>Phone User Interface:</b> None</p>		

**To enable the PC port via web user interface:**

1. Click on **Network->PC Port**.
2. Select **Auto Negotiation** from the pull-down list of **PC Port Active**.

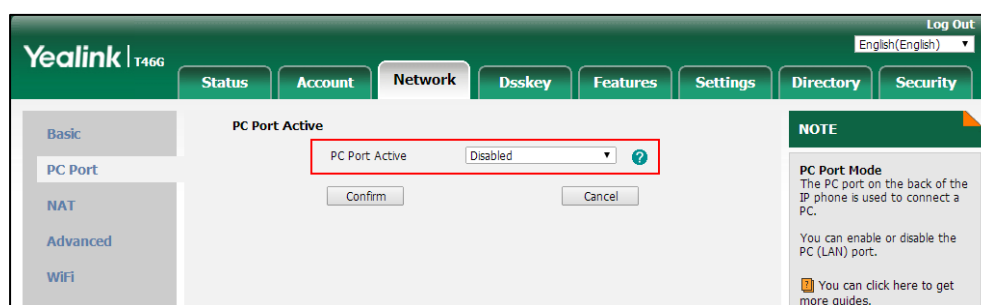


3. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
4. Click **OK** to reboot the phone.

**To disable the PC port via web user interface:**

1. Click on **Network->PC Port**.

2. Select **Disabled** from the pull-down list of **PC Port Active**.



3. Click **Confirm** to accept the change.

A dialog box pops up to prompt that the settings will take effect after a reboot.

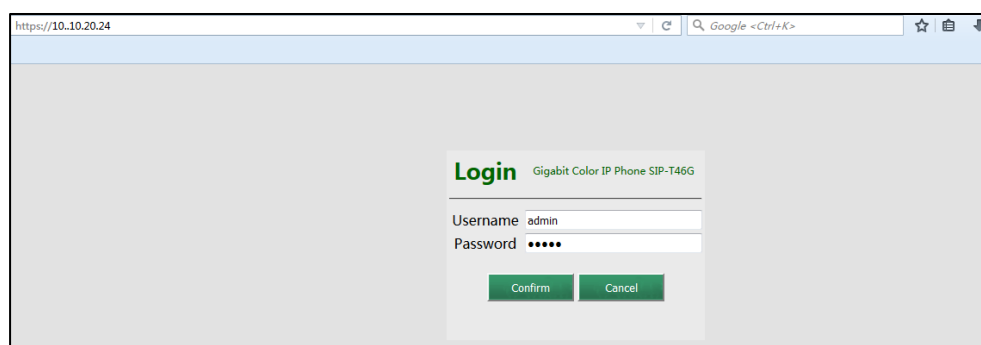
4. Click **OK** to reboot the phone.

## Web Server Type

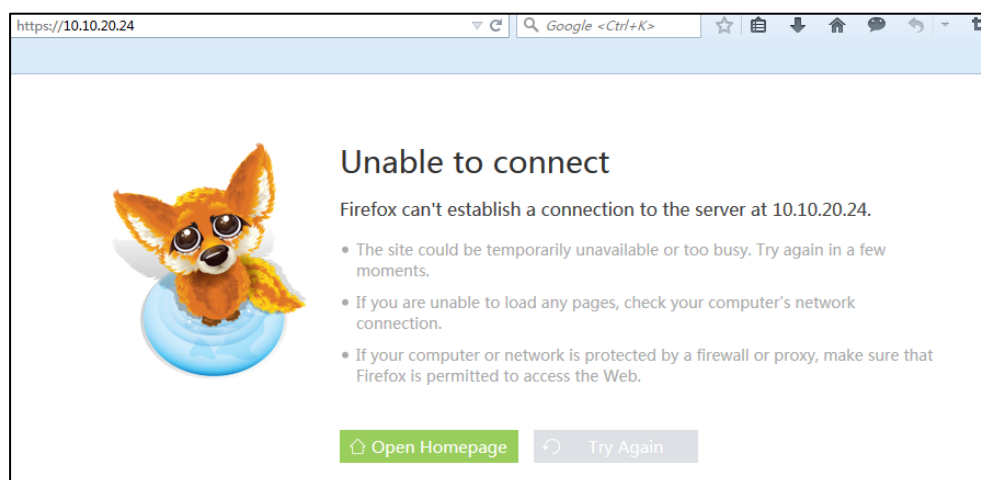
Users can configure the user or administrator features of the phone via web user interface. Web server type determines access protocol of the IP phone's web user interface. IP phones support both HTTP and HTTPS protocols for accessing the web user interface. This can be disabled when it is not needed or when it poses a security threat. For more information on accessing the web user interface, refer to [Web User Interface](#) on page 98.

HTTP is an application protocol that runs on top of the TCP/IP suite of protocols. HTTPS is a web protocol that encrypts and decrypts user page requests as well as pages returned by the web server. Both HTTP and HTTPS port numbers are configurable.

When you enable user to access web user interface of the IP phone using the HTTP/HTTPS protocol (take HTTPS protocol for example):



When you disable user to access web user interface of the IP phone using the HTTP/HTTPS protocol (take HTTPS protocol for example):



## Procedure

Web server type can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the web access type, HTTP port and HTTPS port.  <b>Parameters:</b> static.wui.http_enable static.network.port.http static.wui.https_enable static.network.port.https
<b>Web User Interface</b>		Configure the web access type, HTTP port and HTTPS port.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=network-adv&q=load
<b>Phone User Interface</b>		Configure the web access type, HTTP port and HTTPS port.

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
static.wui.http_enable	0 or 1	1

Parameters	Permitted Values	Default
<b>Description:</b> Enables or disables the user to access web user interface of the IP phone using the HTTP protocol. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->Web Server->HTTP <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->Web Server->HTTP Status		
<b>static.network.port.http</b>	<b>Integer from 1 to 65535</b>	<b>80</b>
<b>Description:</b> Configures the HTTP port for the user to access web user interface of the IP phone using the HTTP protocol. <b>Note:</b> Please take care when choosing an alternate port. If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->Web Server->HTTP Port (1~65535) <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->Web Server->HTTP Port		
<b>static.wui.https_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the user to access web user interface of the IP phone using the HTTPS protocol. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->Web Server->HTTPS <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->Web Server->HTTPS Status		
<b>static.network.port.https</b>	<b>Integer from 1 to 65535</b>	<b>443</b>



Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the HTTPS port for the user to access web user interface of the IP phone using the HTTPS protocol.</p> <p><b>Note:</b> Please take care when choosing an alternate port. If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;Web Server-&gt;HTTPS Port (1~65535)</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;Web Server-&gt;HTTPS Port</p>		

**To configure web server type via web user interface:**

1. Click on **Network->Advanced**.
2. Select the desired value from the pull-down list of **HTTP**.
3. Enter the desired HTTP port number in the **HTTP Port (1~65535)** field.
4. Select the desired value from the pull-down list of **HTTPS**.
5. Enter the desired HTTPS port number in the **HTTPS Port (1~65535)** field.





The screenshot shows the Yealink T46G web interface. The 'Network' tab is selected, and the 'Advanced' sub-tab is active. The 'Web Server' section is highlighted with a red box. It contains the following settings:

- HTTP**: Enabled (dropdown)
- HTTP Port (1~65535)**: 80 (text input)
- HTTPS**: Enabled (dropdown)
- HTTPS Port (1~65535)**: 443 (text input)

Below the 'Web Server' section, the 'VPN' section is visible with 'Active' set to 'Enabled' and an 'Upload VPN Config' button. The 'Confirm' and 'Cancel' buttons are at the bottom.

6. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
7. Click **OK** to reboot the phone.

**To configure web server type via phone user interface:**

1. Press **Menu->Advanced** (default password: admin) -> **Network->Web Server**.
2. Press  or , or the **Switch** soft key to select the desired value from the **HTTP Status** field.
3. Enter the desired HTTP port number in the **HTTP Port** field.
4. Press  or , or the **Switch** soft key to select the desired value from the **HTTPS Status** field.
5. Enter the desired HTTPS port number in the **HTTPS Port** field.
6. Press the **Save** soft key to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
7. Press **OK** to reboot the phone.

## Wi-Fi

Wi-Fi feature enables users to connect their phones to the organization's wireless network. The wireless network is more convenient and cost-effective than wired network. Wi-Fi feature is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.

When the Wi-Fi feature is enabled, the IP phone will automatically scan the available wireless networks. All the available wireless networks will display in scanning list on the LCD screen. You can store up to 5 frequently-used wireless networks on your phone and specify the priority for them.

### Note

To use Wi-Fi feature, make sure the Wi-Fi USB dongle is properly connected to the USB port on the back of the phone. You can check the Wi-Fi dongle MAC address at the path:  
**Menu->Status->Wi-Fi MAC** (phone user interface) or **Status->Status->Network Common->Wi-Fi MAC** (web user interface).

When you connect the Ethernet cable, you can enable the Wi-Fi feature. But you have to disable the Wi-Fi feature if you want to use the wired network.

## Procedure

Wi-Fi feature can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.c fg	Configure Wi-Fi feature. <b>Parameter:</b> static.wifi.enable
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		Configure the Wi-Fi settings. <b>Parameters:</b> static.wifi.X.label static.wifi.X.ssid static.wifi.X.priority static.wifi.X.security_mode static.wifi.X.cipher_type static.wifi.X.password static.wifi.X.eap_type static.wifi.X.eap_user_name static.wifi.X.eap_password static.wifi.show_scan_prompt
<b>Web User Interface</b>		Configure Wi-Fi feature. Configure the Wi-Fi settings. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=network-wifi&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=network-wifi&amp;q=load</a>
<b>Phone User Interface</b>		Configure Wi-Fi feature. Configure the Wi-Fi settings.

### Details of the Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.wifi.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the Wi-Fi feature. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones. <b>Web User Interface:</b> Network->Wi-Fi->Wi-Fi Active <b>Phone User Interface:</b> Menu->Basic->Wi-Fi->Wi-Fi		
<b>static.wifi.X.label</b> (X ranges from 1 to 5)	<b>String within 32 characters</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the profile name of the wireless network X for the IP phone.</p> <p><b>Note:</b> It works only if the value of the parameter "static.wifi.enable" is set to 1 (Enabled). It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Wi-Fi-&gt;Profile Name</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network (press Add)-&gt;Profile Name or Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network-&gt;Option-&gt;Edit-&gt;Profile Name</p>		
<p><b>static.wifi.X.ssid</b></p> <p>(X ranges from 1 to 5)</p>	<p><b>String within 32 characters</b></p>	<p><b>Blank</b></p>
<p><b>Description:</b></p> <p>Configures the Service Set Identifier (SSID) of the wireless network X.</p> <p>SSID is a unique identifier for accessing wireless access points.</p> <p><b>Note:</b> It works only if the value of the parameter "static.wifi.enable" is set to 1 (Enabled). It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Wi-Fi-&gt;SSID</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network (press Add)-&gt;SSID or Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network-&gt;Option-&gt;Edit-&gt;SSID</p>		
<p><b>static.wifi.X.priority</b></p> <p>(X ranges from 1 to 5)</p>	<p><b>Integer from 1 to 5</b></p>	<p><b>1</b></p>
<p><b>Description:</b></p> <p>Configures the priority for the wireless network X for the IP phone.</p> <p>5 is the highest priority, 1 is the lowest priority.</p> <p><b>Note:</b> It works only if the value of the parameter "static.wifi.enable" is set to 1 (Enabled). It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Wi-Fi-&gt;Change Priority</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network-&gt;Option-&gt;Move Up/Move Down</p>		
<p><b>static.wifi.X.security_mode</b></p> <p>(X ranges from 1 to 5)</p>	<p><b>NONE, WEP, WPA-PSK, WPA2-PSK, WPA-EAP or</b></p>	<p><b>NONE</b></p>

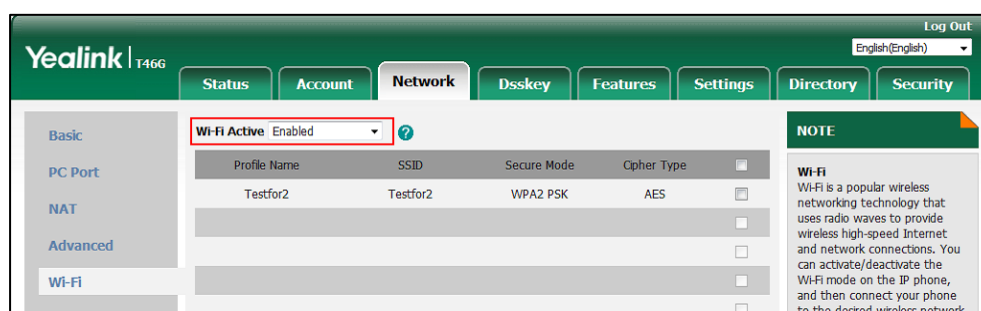
Parameters	Permitted Values	Default
	<b>WPA2-EAP</b>	
<p><b>Description:</b></p> <p>Configures the security mode of the wireless network X.</p> <p><b>Note:</b> It works only if the value of the parameter "static.wifi.enable" is set to 1 (Enabled). It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Wi-Fi-&gt;Secure Mode</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network (press Add)-&gt;Security Mode or Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network-&gt;Option-&gt;Edit-&gt;Security Mode</p>		
<b>static.wifi.X.cipher_type</b> (X ranges from 1 to 5)	<b>NONE, WEP, TKIP, AES or TKIP AES</b>	<b>NONE</b>
<p><b>Description:</b></p> <p>Configures the encryption type of the wireless network X.</p> <p>If the value of the parameter "static.wifi.X.security_mode" is set to <b>NONE</b>, the permitted value of this parameter is <b>NONE</b>.</p> <p>If the value of the parameter "static.wifi.X.security_mode" is set to <b>WEP</b>, the permitted value of this parameter is <b>WEP</b>.</p> <p>If the value of the parameter "static.wifi.X.security_mode" is set to other values, the permitted values of this parameter are <b>TKIP, AES or TKIP AES</b>.</p> <p><b>Note:</b> It works only if the value of the parameter "static.wifi.enable" is set to 1 (Enabled). It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Wi-Fi-&gt;Cipher Type</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network (press Add)-&gt;Cipher Type or Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network-&gt;Option-&gt;Edit-&gt;Cipher Type</p>		
<b>static.wifi.X.password</b> (X ranges from 1 to 5)	<b>String within 64 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the password of the wireless network X.</p> <p><b>Note:</b> It works only if the value of the parameter "static.wifi.enable" is set to 1 (Enabled) and "static.wifi.X.security_mode" is set to <b>WEP, WPA-PSK or WPA2-PSK</b>. It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.</p>		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Network->Wi-Fi->PSK <b>Phone User Interface:</b> Menu->Basic->Wi-Fi->Wi-Fi (On)->Known Network (press Add)->WPA Shared Key or Menu->Basic->Wi-Fi->Wi-Fi (On)->Known Network->Option->Edit->WPA Shared Key		
<b>static.wifi.X.eap_type</b> (X ranges from 1 to 5)	<b>TTLS, PEAP or TLS</b>	<b>Blank</b>
<b>Description:</b> Configures the EAP authentication mode of the wireless network X. <b>Note:</b> It works only if the value of the parameter "static.wifi.enable" is set to 1 (Enabled) and "static.wifi.X.security_mode" is set to <b>WPA-EAP</b> or <b>WPA2-EAP</b> . It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>static.wifi.X.eap_user_name</b> (X ranges from 1 to 5)	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the EAP authentication username of the wireless network X. <b>Note:</b> It works only if the value of the parameter "static.wifi.enable" is set to 1 (Enabled), "static.wifi.X.security_mode" is set to <b>WPA-EAP</b> or <b>WPA2-EAP</b> and the value of the parameter "static.wifi.X.eap_type" is set to <b>TTLS</b> or <b>PEAP</b> . It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones. <b>Web User Interface:</b> Network->Wi-Fi->User Name <b>Phone User Interface:</b> Menu->Basic->Wi-Fi->Wi-Fi (On)->Known Network (press Add)->User Name or Menu->Basic->Wi-Fi->Wi-Fi (On)->Known Network->Option->Edit->User Name		
<b>static.wifi.X.eap_password</b> (X ranges from 1 to 5)	<b>String within 64 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the EAP authentication password of the wireless network X.		

Parameters	Permitted Values	Default
<p><b>Note:</b> It works only if the value of the parameter "static.wifi.enable" is set to 1 (Enabled) and "static.wifi.X.security_mode" is set to <b>WPA-EAP</b> or <b>WPA2-EAP</b>. It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.</p> <p><b>Web User Interface:</b> Network-&gt;Wi-Fi-&gt;PSK</p> <p><b>Phone User Interface:</b> Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network (press Add)-&gt;WPA Shared Key or Menu-&gt;Basic-&gt;Wi-Fi-&gt;Wi-Fi (On)-&gt;Known Network-&gt;Option-&gt;Edit-&gt;WPA Shared Key</p>		
<b>static.wifi.show_scan_prompt</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disable the IP phone to display the Wi-Fi scanning prompt after connecting Wi-Fi USB dongle WF40 to the IP phone.</p> <p><b>0</b>-Disabled, the IP phone will enable the Wi-Fi feature and try to connect to the known wireless network (according to the priority) automatically. But if the IP phone fails to connect to any known wireless network, the IP phone will still display the Wi-Fi scanning prompt when connecting WF40 to the IP phone.</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		

**To enable the Wi-Fi feature via web user interface:**

1. Click on **Network->Wi-Fi**.
2. Select **Enabled** from the pull-down list of **Wi-Fi Active**.



3. Click **Confirm** to accept the change.

### To add a wireless network via web user interface:

1. Click on **Network->Wi-Fi**.
2. Enter the desired value in the **Profile Name** field.
3. Enter the desired value in the **SSID** field.
4. Select the desired value from the pull-down list of **Secure Mode**.
  - If you select **WEP**:
    - 1) Enter the desired password in the **PSK** field.
  - If you select **WPA-PSK** or **WPA2-PSK**:
    - 1) Select the desired value (**TKIP**, **AES** or **TKIP AES**) from the pull-down list of the **Cipher Type**.
    - 2) Enter the desired password in the **PSK** field.
  - If you select **WPA-EAP** or **WPA2-EAP**:
    - 1) Select the desired value (**TKIP**, **AES** or **TKIP AES**) from the pull-down list of the **Cipher Type**.
    - 2) Enter the desired username in the **User Name** field.
    - 3) Enter the desired password in the **PSK** field.

The screenshot shows the Yealink T46G web interface. The 'Network' tab is selected, and the 'Wi-Fi' sub-tab is active. The 'Wi-Fi Active' status is 'Enabled'. Below this, there is a table with columns: Profile Name, SSID, Secure Mode, Cipher Type, and a checkbox. A red box highlights the 'Add' form, which contains the following fields:

Profile Name	Test
SSID	Test
Secure Mode	WPA2 EAP
Cipher Type	AES
User Name	Test1
PSK	*****



Buttons for 'Add', 'Edit', and 'Cancel' are at the bottom of the form. A 'NOTE' section on the right explains Wi-Fi technology and provides a link to guides.

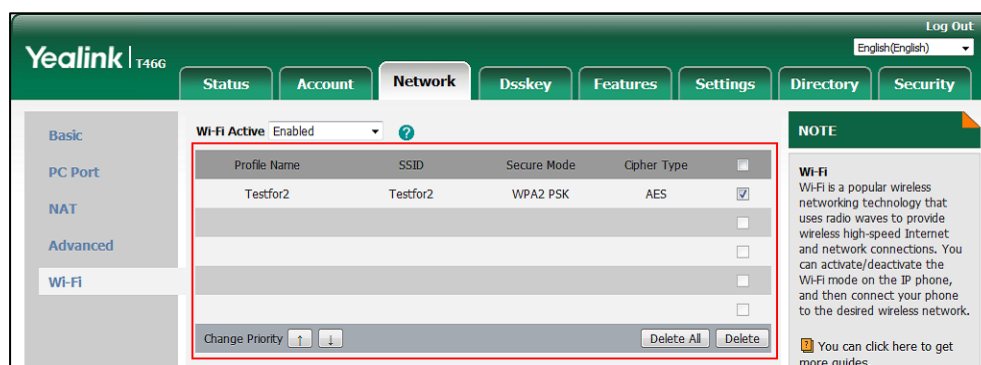
5. Click **Add** to accept the change.
6. Repeat steps 2 to 5 to add more wireless networks.

### To adjust the priority of the added wireless network via web user interface:

1. Click on **Network->Wi-Fi**.





- Click to select the desired wireless network which you want to adjust the priority, and then click  or .





- Repeat the step 2 to adjust the priority of more wireless networks.

#### To enable the Wi-Fi feature via phone user interface:

- Press **Menu->Basic->Wi-Fi**.
- Press  or , or the **Switch** soft key to select **On** from the **Wi-Fi** field.
- Press the **Save** soft key.

The IP phone scans the available wireless network automatically.

#### To add a wireless network:

- Press **Menu->Basic->Wi-Fi**.
- Activate the Wi-Fi mode.
- Press  or  to highlight **Known Network(s)**, and then press the **Add** soft key.
- Use the WLAN settings obtained from your gateway/router to configure this WLAN Profile on the IP phone. Do the following:
  - If you select **None** or **WEP** from the pull-down list of **Security Mode**:  
Enter the profile name, SSID and WPA shared key in the corresponding fields.
  - If you select **WPA-PSK** or **WPA2-PSK** from the pull-down list of **Security Mode**:  
Select the desired Cipher type (**TKIP**, **AES** or **TKIP AES**) from the pull-down list of **Cipher Type**.  
Enter the profile name, SSID and WPA shared key in the corresponding fields.
  - If you select **WPA-EAP** or **WPA2-EAP** from the pull-down list of **Security Mode**:  
Select the desired Cipher type (**TKIP**, **AES** or **TKIP AES**) from the pull-down list of **Cipher Type**.  
Enter the profile name, SSID, username and WPA shared key in the corresponding fields.
- Press the **Save** soft key to accept the change.

## VLAN

VLAN (Virtual Local Area Network) is used to logically divide a physical network into several broadcast domains. VLAN membership can be configured through software instead of physically relocating devices or connections. Grouping devices with a common set of requirements regardless of their physical location can greatly simplify network design. VLANs can address issues such as scalability, security and network management.

The purpose of VLAN configurations on the IP phone is to insert tag with VLAN information to the packets generated by the IP phone. When VLAN is properly configured for the ports (Internet port and PC port) on the IP phone, the IP phone will tag all packets from these ports with the VLAN ID. The switch receives and forwards the tagged packets to the corresponding VLAN according to the VLAN ID in the tag as described in IEEE Std 802.3.

VLAN on IP phones allows simultaneous access for a regular PC. This feature allows a PC to be daisy chained to an IP phone and the connection for both PC and IP phone to be trunked through the same physical Ethernet cable.

In addition to manual configuration, the IP phone also supports automatic discovery of VLAN via LLDP, CDP or DHCP. The assignment takes effect in this order: assignment via LLDP/CDP, manual configuration, then assignment via DHCP.

For more information on VLAN, refer to [VLAN Feature on Yealink IP Phones](#).

### Procedure

VLAN assignment method can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the VLAN assignment method. <b>Parameter:</b> static.network.vlan.vlan_change.enable
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### Details of Configuration Parameter:

Parameter	Permitted Values	Default
<b>static.network.vlan.vlan_change.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to obtain VLAN ID using lower priority of VLAN assignment method or disable VLAN feature when the IP phone cannot obtain VLAN ID using the current VLAN assignment method. <b>0</b> -Disabled <b>1</b> -Enabled, the IP phone will attempt to use the lower priority of VLAN assignment method when failing to obtain the VLAN ID using higher priority of VLAN assignment method. If all the methods are attempted, the phone will disable VLAN feature.		

Parameter	Permitted Values	Default
<p>The priority of each method is: LLDP/CDP&gt;Manual&gt;DHCP VLAN.</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## LLDP

LLDP (Linker Layer Discovery Protocol) is a vendor-neutral Link Layer protocol, which allows IP phones to receive and/or transmit device-related information from/to directly connected devices on the network that are also using the protocol, and store the information about other devices.

When LLDP feature is enabled on IP phones, the IP phones periodically advertise their own information to the directly connected LLDP-enabled switch. The IP phones can also receive LLDP packets from the connected switch. When the application type is "voice", IP phones decide whether to update the VLAN configurations obtained from the LLDP packets. When the VLAN configurations on the IP phones are different from the ones sent by the switch, the IP phones perform an update and reboot. This allows the IP phones to be plugged into any switch, obtain their VLAN IDs, and then start communications with the call control.

### Procedure

LLDP can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	<p>Configure LLDP feature.</p> <p><b>Parameters:</b></p> <p>static.network.lldp.enable</p> <p>static.network.lldp.packet_interval</p>
<b>Web User Interface</b>		<p>Configure LLDP feature.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=network-adv&amp;q=load</p>
<b>Phone User Interface</b>		Configure LLDP feature.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>static.network.lldp.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the LLDP (Linker Layer Discovery Protocol) feature on the IP phone.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, the IP phone will attempt to determine its VLAN ID through LLDP.</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Advanced-&gt;LLDP-&gt;Active</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;LLDP-&gt;LLDP Status</p>		
<b>static.network.lldp.packet_interval</b>	<b>Integer from 1 to 3600</b>	<b>60</b>
<p><b>Description:</b> Configures the interval (in seconds) for the IP phone to send the LLDP (Linker Layer Discovery Protocol) request.</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.lldp.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Advanced-&gt;LLDP-&gt;Packet Interval (1~3600s)</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;LLDP-&gt;Packet Interval</p>		

**To configure LLDP feature via web user interface:**

1. Click on **Network->Advanced**.
2. In the **LLDP** block, select the desired value from the pull-down list of **Active**.

- Enter the desired time interval in the **Packet Interval (1~3600s)** field.

The screenshot shows the Yealink T466 web interface. The 'Network' tab is selected, and the 'LLDP' sub-tab is active. In the 'LLDP' section, the 'Active' dropdown is set to 'Enabled' and the 'Packet Interval (1~3600s)' field is set to '60'. A red box highlights these two fields. Below this, the 'CDP' section is also visible, with 'Active' set to 'Disabled' and 'Packet Interval' set to '60'. On the right, a 'NOTE' box explains that VLAN is used to logically divide a physical network into several broadcast domains.

- Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
- Click **OK** to reboot the phone.

#### To configure LLDP feature via phone user interface:

- Press **Menu->Advanced** (default password: admin) -> **Network->LLDP->LLDP Status**.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **LLDP Status** field.
- Enter the priority value (1-3600s) in the **Packet Interval** field.
- Press the **Save** soft key to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
- Press **OK** to reboot the phone.

## CDP

CDP (Cisco Discovery Protocol) allows IP phones to receive and/or transmit device-related information from/to directly connected devices on the network that are also using the protocol, and store the information about other devices.

If the CDP feature is enabled on IP phones, the IP phones will periodically advertise their own information to the directly connected CDP-enabled switch. The IP phones can also receive CDP packets from the connected switch. If the VLAN configurations on the IP phones are different from the ones sent by the switch, the IP phones will perform an update and reboot. This allows you to plug the IP phones into any switch, obtain their VLAN IDs, and then start communications with the call control.

### Procedure

CDP can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure CDP feature. <b>Parameters:</b> static.network.cdp.enable static.network.cdp.packet_interval
<b>Web User Interface</b>		Configure CDP feature.

	<b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=network-adv&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=network-adv&amp;q=load</a>
<b>Phone User Interface</b>	Configure CDP feature.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>static.network.cdp.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the CDP (Cisco Discovery Protocol) feature on the IP phone. <b>0</b> -Disabled <b>1</b> -Enabled, the IP phone will attempt to determine its VLAN ID through CDP. <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->CDP->Active <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->CDP->CDP Status		
<b>static.network.cdp.packet_interval</b>	<b>Integer from 1 to 3600</b>	<b>60</b>
<b>Description:</b> Configures the interval (in seconds) for the IP phone to send the CDP (Cisco Discovery Protocol) request. <b>Note:</b> It works only if the value of the parameter "static.network.cdp.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->CDP->Packet Interval (1~3600s) <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->CDP->CDP Interval		

**To configure CDP via web user interface:**

1. Click on **Network->Advanced**.
2. In the **CDP** block, select the desired value from the pull-down list of **Active**.

- Enter the desired time interval in the **Packet Interval (1~3600s)** field.

- Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
- Click **OK** to reboot the phone.

#### To configure CDP feature via phone user interface:

- Press **Menu**->**Advanced** (default password: admin) ->**Network**->**CDP**->**CDP Status**.
- Press **Left** or **Right** , or the **Switch** soft key to select the desired value from the **CDP Status** field.
- Enter the priority value (1-3600s) in the **CDP Interval** field.
- Press the **Save** soft key to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
- Press **OK** to reboot the phone.

## Manual Configuration for VLAN in the Wired Network

VLAN is disabled on IP phones by default. You can configure VLAN for the Internet port and PC port manually. Before configuring VLAN on the IP phone, you need to obtain the VLAN ID from your network administrator.

### Procedure

VLAN can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<pre>&lt;y0000000000xx&gt; .cfg</pre>	Configure VLAN for the Internet port and PC port manually.  <b>Parameters:</b> static.network.vlan.internet_port_enable static.network.vlan.internet_port_vid static.network.vlan.internet_port_priority static.network.vlan.pc_port_enable static.network.vlan.pc_port_vid static.network.vlan.pc_port_priority
------------------------------------------------------	---------------------------------------	------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

<b>Web User Interface</b>	<p>Configure VLAN for the Internet port and PC port manually.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=network-adv&amp;q=load</p>
<b>Phone User Interface</b>	<p>Configure VLAN for the Internet port and PC port manually.</p>

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.network.vlan.internet_port_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables VLAN for the Internet port.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;VLAN-&gt;WAN Port-&gt;Active</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;VLAN-&gt;WAN Port-&gt;VLAN Status</p>		
<b>static.network.vlan.internet_port_vid</b>	<b>Integer from 1 to 4094</b>	<b>1</b>
<p><b>Description:</b></p> <p>Configures VLAN ID for the Internet port.</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.vlan.internet_port_enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;VLAN-&gt;WAN Port-&gt;VID (1-4094)</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;VLAN-&gt;WAN Port-&gt;VID Number</p>		



Parameters	Permitted Values	Default
<b>static.network.vlan.internet_port_priority</b>	<b>Integer from 0 to 7</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures VLAN priority for the Internet port.</p> <p>7 is the highest priority, 0 is the lowest priority.</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.vlan.internet_port_enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;VLAN-&gt;WAN Port-&gt;Priority</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;VLAN-&gt;WAN Port-&gt;Priority</p>		
<b>static.network.vlan.pc_port_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables VLAN for the PC port.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.pc_port.enable" is set to 1 (Auto Negotiation). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;VLAN-&gt;PC Port-&gt;Active</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;VLAN-&gt;PC Port-&gt;VLAN Status</p>		
<b>static.network.vlan.pc_port_vid</b>	<b>Integer from 1 to 4094</b>	<b>1</b>
<p><b>Description:</b></p> <p>Configures VLAN ID for the PC port.</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.pc_port.enable" is set to 1 (Auto Negotiation) and the value of the parameter "static.network.vlan.pc_port_enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;VLAN-&gt;PC Port-&gt;VID (1-4094)</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->VLAN->PC Port->VID Number		
<b>static.network.vlan.pc_port_priority</b>	<b>Integer from 0 to 7</b>	<b>0</b>
<b>Description:</b> Configures VLAN priority for the PC port. 7 is the highest priority, 0 is the lowest priority. <b>Note:</b> It works only if the value of the parameter "static.network.pc_port.enable" is set to 1 (Auto Negotiation) and the value of the parameter "static.network.vlan.pc_port_enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->VLAN >PC Port->Priority <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->VLAN->PC Port->Priority		

**To configure VLAN for Internet port via web user interface:**

1. Click on **Network->Advanced**.
2. In the **WAN Port** block, select the desired value from the pull-down list of **Active**.
3. Enter the VLAN ID in the **VID (1-4094)** field.
4. Select the desired value (0-7) from the pull-down list of **Priority**.

The screenshot shows the Yealink T46G web interface. The 'Network' tab is selected, and the 'Advanced' sub-tab is active. Under the 'VLAN' section, the 'WAN Port' configuration is highlighted with a red box. It shows the 'Active' state with a dropdown menu set to 'Enabled'. Below this, the 'VID (1-4094)' field is set to '1' and the 'Priority' dropdown is set to '0'. The 'PC Port' section below it shows the 'Active' state with a dropdown menu set to 'Disabled'. On the right side, there is a 'NOTE' section explaining VLAN and NAT Traversal.

5. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
6. Click **OK** to reboot the phone.

**To configure VLAN for PC port via web user interface:**

1. Click on **Network->Advanced**.
2. In the **PC Port** block, select the desired value from the pull-down list of **Active**.
3. Enter the VLAN ID in the **VID (1-4094)** field.
4. Select the desired value (0-7) from the pull-down list of **Priority**.

The screenshot shows the Yealink T46G web interface. The 'Network' tab is selected, and the 'Advanced' sub-tab is active. Under the 'Advanced' section, the 'PC Port' configuration is visible. The 'PC Port' section is highlighted with a red box. It contains three fields: 'Active' (set to 'Enabled'), 'VID (1-4094)' (set to '1'), and 'Priority' (set to '0'). To the right of the configuration area, there is a 'NOTE' section with text about VLAN and NAT Traversal.

5. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
6. Click **OK** to reboot the phone.

**To configure VLAN for Internet port (or PC port) via phone user interface:**

1. Press **Menu->Advanced** (default password: admin) ->**Network->VLAN->WAN Port** (or **PC Port**).
2. Press **Left** or **Right** arrow, or the **Switch** soft key to select the desired value from the **VLAN Status** field.
3. Enter the VLAN ID (1-4094) in the **VID Number** field.
4. Enter the priority value (0-7) in the **Priority** field.
5. Press the **Save** soft key to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
5. Press **OK** to reboot the phone.

**DHCP VLAN**

IP phones support VLAN discovery via DHCP. When the VLAN discovery method is set to DHCP, the IP phone will examine DHCP option for a valid VLAN ID. The predefined option 132 is used to supply the VLAN ID by default. You can customize the DHCP option used to request the VLAN ID.

## Procedure

DHCP VLAN can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure DHCP VLAN discovery feature. <b>Parameters:</b> static.network.vlan.dhcp_enable static.network.vlan.dhcp_option
<b>Web User Interface</b>		Configure DHCP VLAN discovery feature. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=network-adv&q=load
<b>Phone User Interface</b>		Configure DHCP VLAN discovery feature.

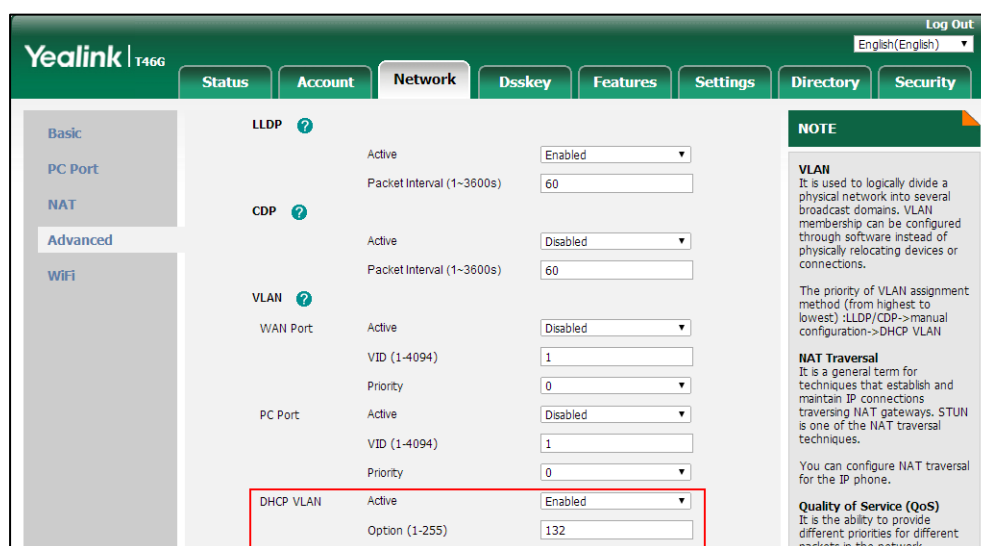
## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.network.vlan.dhcp_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables DHCP VLAN discovery feature on the IP phone. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->VLAN->DHCP VLAN->Active <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->VLAN->DHCP VLAN->DHCP VLAN		
<b>static.network.vlan.dhcp_option</b>	<b>Integer from 1 to 255</b>	<b>132</b>
<b>Description:</b> Configures the DHCP option from which the IP phone will obtain the VLAN settings. You can configure at most five DHCP options and separate them by commas. <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take		

Parameters	Permitted Values	Default
<p>effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;VLAN-&gt;DHCP VLAN-&gt;Option (1-255)</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;VLAN-&gt;DHCP VLAN-&gt;Option</p>		

**To configure DHCP VLAN discovery via web user interface:**

1. Click on **Network->Advanced**.
2. In the **DHCP VLAN** block, select the desired value from the pull-down list of **Active**.
3. Enter the desired option in the **Option (1-255)** field.



4. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
5. Click **OK** to reboot the phone.

**To configure DHCP VLAN discovery via phone user interface:**

1. Press **Menu->Advanced** (default password: admin) -> **Network->VLAN->DHCP VLAN**.
2. Press **Left** or **Right** arrow, or the **Switch** soft key to select the desired value from the **DHCP VLAN** field.
3. Enter the desired option in the **Option** field.
4. Press the **Save** soft key to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
5. Press **OK** to reboot the phone.

## IPv6 Support

Because Internet Protocol version 4 (IPv4) uses a 32-bit address, it cannot meet the increased demands for unique IP addresses for all devices that connect to the Internet. Therefore, Internet Protocol version 6 (IPv6) is the next generation network layer protocol, which designed as a replacement for the current IPv4 protocol.

IPv6 is developed by the Internet Engineering Task Force (IETF) to deal with the long-anticipated problem of IPv4 address exhaustion. Yealink IP Phone supports IPv4 addressing mode, IPv6 addressing mode, as well as an IPv4&IPv6 dual stack addressing mode. IPv4 uses a 32-bit address, consisting of four groups of three decimal digits separated by dots; for example, 192.168.1.100. IPv6 uses a 128-bit address, consisting of eight groups of four hexadecimal digits separated by colons; for example, 2026:1234:1:1:215:65ff:fe1f:caa.

VoIP network based on IPv6 can provide end-to-end security capabilities, enhanced Quality of Service (QoS), a set of service requirements to deliver performance guarantee while transporting traffic over the network.

If you configure the network settings on the phone for an IPv6 network, you can set up an IP address for the phone either by using SLAAC (ICMPv6), DHCPv6 or by manually entering an IP address. Ensure that your network environment supports IPv6. Contact your ISP for more information.

### IPv6 Address Assignment Method

Supported IPv6 address assignment methods:

- **Manual Assignment:** An IPv6 address and other configuration parameters (for example, DNS server) for the IP phone can be statically configured by an administrator.
- **Stateless Address Autoconfiguration (SLAAC)/ICMPv6:** SLAAC is one of the most convenient methods to assign IP addresses to IPv6 nodes. SLAAC requires no manual configuration of the IP phone, minimal (if any) configuration of routers, and no additional servers. To use IPv6 SLAAC, the IP phone must be connected to a network with at least one IPv6 router connected. This router is configured by the network administrator and sends out Router Advertisement announcements onto the link. These announcements can allow the on-link connected IP phone to configure itself with IPv6 address, as specified in RFC 4862.
- **Stateful DHCPv6:** The Dynamic Host Configuration Protocol for IPv6 (DHCPv6) has been standardized by the IETF through RFC 3315. DHCPv6 enables DHCP servers to pass configuration parameters such as IPv6 network addresses to IPv6 nodes. It offers the capability of automatic allocation of reusable network addresses and additional configuration flexibility. This protocol is a stateful counterpart to "IPv6 Stateless Address Autoconfiguration" ([RFC 2462](#)), and can be used separately or concurrently with the latter to obtain configuration parameters.

## How the IP phone obtains the IPv6 address and network settings?

The following table lists where the IP phone obtains the IPv6 address and other network settings:

DHCPv6	SLAAC (ICMPv6)	How the IP phone obtains the IPv6 address and network settings?
Disabled	Disabled	You have to manually configure the static IPv6 address and other network settings.
Enabled	Disabled	The IP phone can obtain the IPv6 address and other network settings via DHCPv6.
Enabled	Enabled	If the SLAAC server is working, the server can specify the IP phone to obtain the IPv6 address and other network settings either from DHCPv6 or SLAAC.  If the SLAAC server is not working, the IP phone will try to obtain the IPv6 address and other network settings via DHCPv6.

## Procedure

IPv6 can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<MAC>.cfg	Configure the IPv6 address assignment method.  <b>Parameters:</b> static.network.ip_address_mode static.network.ipv6_internet_port.type static.network.ipv6_internet_port.ip static.network.ipv6_prefix static.network.ipv6_internet_port.gateway static.network.ipv6_icmp_v6.enable
		Configure the IPv6 static DNS address.  <b>Parameters:</b> static.network.ipv6_primary_dns static.network.ipv6_secondary_dns
	<y0000000000xx>.cfg	Configure the IPv6 static DNS.  <b>Parameter:</b> static.network.ipv6_static_dns_enable
<b>Web User Interface</b>		Configure the IPv6 address assignment method.  Configure the IPv6 static DNS address.

	Configure the IPv6 static DNS.  <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=network&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=network&amp;q=load</a>
<b>Phone User Interface</b>	Configure the IPv6 address assignment method.  Configure the IPv6 static DNS address.  Configure the IPv6 static DNS.

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.network.ip_address_mode</b>	<b>0, 1 or 2</b>	<b>0</b>
<b>Description:</b> Configures the IP address mode.  <b>0</b> -IPv4 <b>1</b> -IPv6 <b>2</b> -IPv4 & IPv6  <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.  <b>Web User Interface:</b> Network->Basic->Internet Port->Mode (IPv4/IPv6)  <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->WAN Port->IP Mode		
<b>static.network.ipv6_internet_port.type</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Configures the Internet port type for IPv6.  <b>0</b> -DHCP <b>1</b> -Static IP Address  <b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 & IPv6). If you change this parameter, the IP phone will reboot to make the change take effect.  <b>Web User Interface:</b> Network->Basic->IPv6 Config  <b>Phone User Interface:</b>		



Parameters	Permitted Values	Default
Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type		
<b>static.network.ipv6_static_dns_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Triggers the static IPv6 DNS feature to on or off.</p> <p><b>0</b>-Off, the IP phone will use the IPv6 DNS obtained from DHCP.</p> <p><b>1</b>-On, the IP phone will use manually configured static IPv6 DNS.</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.ipv6_internet_port.type" is set to 0 (DHCP). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Basic-&gt;IPv6 Config-&gt;IPv6 Static DNS</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;WAN Port-&gt;IPv6-&gt;Type (DHCP) -&gt;Static DNS</p>		
<b>static.network.ipv6_internet_port.ip</b>	<b>IPv6 address</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the IPv6 address.</p> <p><b>Example:</b></p> <p>static.network.ipv6_internet_port.ip = 2026:1234:1:1:215:65ff:fe1f:caa</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 &amp; IPv6), and "static.network.ipv6_internet_port.type" is set to 1 (Static IP Address). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Basic-&gt;IPv6 Config-&gt;Static IP Address-&gt;IP Address</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;WAN Port-&gt;IPv6-&gt;Type (Static IP) -&gt;IP Address</p>		
<b>static.network.ipv6_prefix</b>	<b>Integer from 0 to 128</b>	<b>64</b>
<p><b>Description:</b></p> <p>Configures the IPv6 prefix.</p>		

Parameters	Permitted Values	Default
<p><b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 &amp; IPv6), and "static.network.ipv6_internet_port.type" is set to 1 (Static IP Address). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Basic-&gt;IPv6 Config-&gt;Static IP Address-&gt;IPv6 Prefix(0~128)</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;WAN Port-&gt;IPv6-&gt; Type (Static IP) -&gt;IPv6 IP Prefix</p>		
<b>static.network.ipv6_internet_port.gateway</b>	<b>IPv6 address</b>	<b>Blank</b>
<p><b>Description:</b> Configures the IPv6 default gateway.</p> <p><b>Example:</b> static.network.ipv6_internet_port.gateway = 3036:1:1:c3c7:c11c:5447:23a6:255</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 &amp; IPv6), and "static.network.ipv6_internet_port.type" is set to 1 (Static IP Address). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Basic-&gt;IPv6 Config-&gt;Static IP Address-&gt;Default Gateway</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;WAN Port-&gt;IPv6-&gt; Type (Static IP) -&gt;Default Gateway</p>		
<b>static.network.ipv6_primary_dns</b>	<b>IPv6 address</b>	<b>Blank</b>
<p><b>Description:</b> Configures the primary IPv6 DNS server.</p> <p><b>Example:</b> static.network.ipv6_primary_dns = 3036:1:1:c3c7: c11c:5447:23a6:256</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 &amp; IPv6). In DHCP environment, you also need to make sure the value of the parameter "static.network.ipv6_static_dns_enable" is set to 1 (On). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
Network->Basic->IPv6 Config->Static IP Address->Primary DNS <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type (Static IP) ->Pri.DNS <b>Or</b> Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type (DHCP) ->Static DNS (Enabled) ->Pri.DNS		
<b>static.network.ipv6_secondary_dns</b>	<b>IPv6 address</b>	<b>Blank</b>
<b>Description:</b> Configures the secondary IPv6 DNS server. <b>Example:</b> static.network.ipv6_secondary_dns = 2026:1234:1:1:c3c7:c11c:5447:23a6 <b>Note:</b> It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 & IPv6). In DHCP environment, you also need to make sure the value of the parameter "static.network.ipv6_static_dns_enable" is set to 1 (On). If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Basic->IPv6 Config->Static IP Address->Secondary DNS <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type (Static IP) ->Sec.DNS <b>Or</b> Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type (DHCP) ->Static DNS(Enabled) ->Sec.DNS		
<b>static.network.ipv6_icmp_v6.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to obtain IPv6 network settings via SLAAC (Stateless Address Autoconfiguration) method. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "static.network.ipv6_internet_port.type" is set to 0 (DHCP). If you change this parameter, the IP phone will reboot to make the change take effect. It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones. SLAAC is enabled on SIP-T42G/T42S/T41P/T41S/T40P/T40G/T27P/T27G/T23P/T23G/T21(P) E2/T19(P) E2 IP phones by default. You are not allowed to configure this parameter for these IP phones.		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Network->Advanced->ICMPv6 Status->Active <b>Phone User Interface:</b> None		

**To configure IPv6 address assignment method via web user interface:**

1. Click on **Network->Basic**.
2. Select the desired address mode (**IPv6** or **IPv4 & IPv6**) from the pull-down list of **Mode(IPv4/IPv6)**.
3. In the **IPv6 Config** block, mark the **DHCP** or the **Static IP Address** radio box.
  - If you mark the **Static IP Address** radio box, configure the IPv6 address and other configuration parameters in the corresponding fields.

**Yealink T46G** Log Out English(English)

Status Account **Network** Dsskey Features Settings Directory Security

Basic PC Port NAT Advanced WiFi

**Internet Port**

Mode(IPv4/IPv6) IPv4 & IPv6 ?

**IPv4 Config**

☒ DHCP ?

☐ Static IP Address ?

IP Address

Subnet Mask

Default Gateway

Static DNS ☐ On ☒ Off ?

Primary DNS

Secondary DNS

**IPv6 Config**

☐ DHCP ?

☒ Static IP Address ?

IP Address 2026:1234:1:1:215:65ff:fe1

IPv6 Prefix(0~128) 64

Default Gateway 3036:1:1:c3c7:c11c:5447:2

IPv6 Static DNS ☒ On ☐ Off ?

Primary DNS 3036:1:1:c3c7: c11c:5447:.

Secondary DNS 2026:1234:1:1:c3c7:c11c:5

**NOTE**

**DHCP**  
DHCP (Dynamic Host Configuration Protocol) is a network protocol used to dynamically allocate network parameters to IP phones.

**Static IP Address**  
Specifies the network parameters of IP phones manually.

**PPPoE**  
It allows users to share a common DSL connection to the Internet.

**IPv6 Support**  
IPv6 is developed to deal with the long-anticipated problem of IPv4 address exhaustion.

You can click here to get more guides.

- (Optional.) If you mark the **DHCP** radio box, you can configure the static DNS address in the corresponding fields.

**Yealink T466** Log Out English(English)

Status Account **Network** Dsskey Features Settings Directory Security

Basic PC Port NAT Advanced **WiFi**

**Internet Port** Mode(IPv4/IPv6) IPv4 & IPv6 ?

**IPv4 Config**

- ☒ DHCP ?
- ☐ Static IP Address ?
- IP Address
- Subnet Mask
- Default Gateway
- Static DNS ☐ On ☒ Off ?
- Primary DNS
- Secondary DNS

**IPv6 Config**

- ☒ DHCP ?
- ☐ Static IP Address ?
- IP Address 2026:1234:1:1:215:65ff:fe1
- IPv6 Prefix(0~128) 64
- Default Gateway 3036:1:1:c3c7:c11c:5447:2
- IPv6 Static DNS ☒ On ☐ Off ?
- Primary DNS 3036:1:1:c3c7:c11c:5447::
- Secondary DNS 2026:1234:1:1:c3c7:c11c:5

**NOTE**

**DHCP**  
DHCP (Dynamic Host Configuration Protocol) is a network protocol used to dynamically allocate network parameters to IP phones.

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Specifies the network parameters of IP phones manually.

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It allows users to share a common DSL connection to the Internet.

**IPv6 Support**  
IPv6 is developed to deal with the long-anticipated problem of IPv4 address exhaustion.

**?** You can click here to get more guides.

4. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
5. Click **OK** to reboot the phone.

**To configure SLAAC feature via web user interface:**

1. Click on **Network->Advanced**.

- In the **ICMPv6 Status** block, select the desired value from the pull-down list of **Active**.

The screenshot shows the Yealink T46G web interface with the 'Network' tab selected. The 'ICMPv6 Status' is set to 'Enabled'. The interface includes a sidebar with 'Basic', 'PC Port', 'NAT', 'Advanced', and 'WiFi' sections. The main content area shows various network settings like LLDP, CDP, Voice QoS, Local RTP Port, Span to PC, and VPN. A red box highlights the 'ICMPv6 Status' dropdown menu, which is currently set to 'Enabled'. The right sidebar contains a 'NOTE' section with information about VLAN, NAT Traversal, Quality of Service (QoS), Web Server Type, and 802.1X Authentication.

- Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
- Click **OK** to reboot the phone.

#### To configure IPv6 address assignment method via phone user interface:

- Press **Menu->Advanced** (default password: admin) -> **Network->WAN Port**.
- Press **◀** or **▶** to select **IPv4 & IPv6** or **IPv6** from the **IP Mode** field.
- Press **▲** or **▼** to highlight **IPv6** and then press the **Enter** soft key.
- Press **▲** or **▼**, or the **Switch** soft key to desired IPv6 address assignment method from the **Type** field.

If you select the **Static IP**, configure the IPv6 address and other network parameters in the corresponding fields.

- Press the **Save** soft key to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
- Press **OK** to reboot the phone.

#### To configure static DNS when DHCP is used via phone user interface:

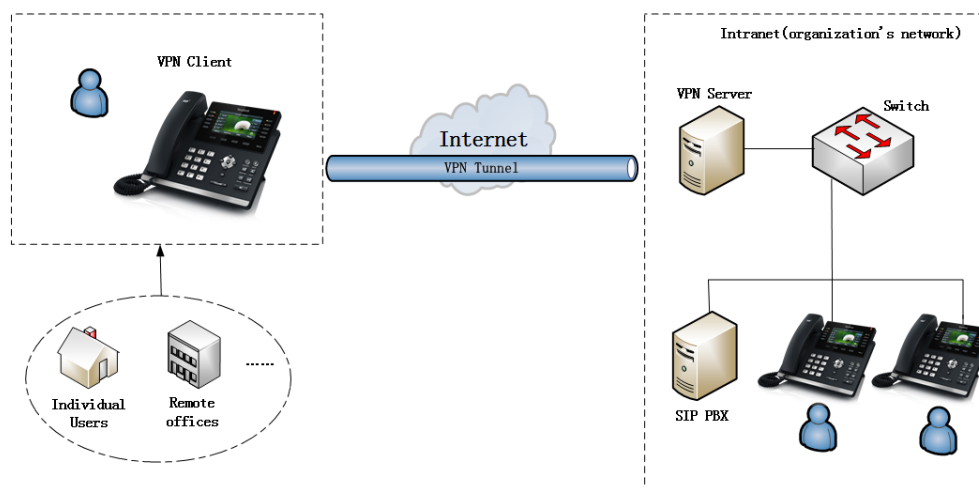
- Press **Menu->Advanced** (default password: admin) -> **Network->WAN Port->IPv6**.
- Press **◀** or **▶**, or the **Switch** soft key to select the **DHCP** from the **Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select **Enabled** from the **Static DNS** field.
- Enter the desired value in the **Pri.DNS** and **Sec.DNS** field respectively.
- Press the **Save** soft key to accept the change.

A dialog box pops up to prompt that the settings will take effect after a reboot.

6. Press **OK** to reboot the phone.

## VPN

VPN (Virtual Private Network) is a secured private network connection built on top of public telecommunication infrastructure, such as the Internet. It has become more prevalent due to benefits of scalability, reliability, convenience and security. VPN provides remote offices or individual users with secure access to their organization's network.



### VPN Technology

IP phones support SSL VPN, which provides remote-access VPN capabilities through SSL. OpenVPN is a full featured SSL VPN software solution that creates secure connections in remote access facilities, designed to work with the TUN/TAP virtual network interface. TUN and TAP are virtual network kernel devices. TAP simulates a link layer device and provides a virtual point-to-point connection, while TUN simulates a network layer device and provides a virtual network segment.

IP phones use OpenVPN to achieve VPN feature. To prevent disclosure of private information, tunnel endpoints must authenticate each other before secure VPN tunnel is established. After VPN feature is configured properly on the IP phone, the IP phone acts as a VPN client and uses the certificates to authenticate the VPN server.

To use VPN, the compressed package of VPN-related files should be uploaded to the IP phone in advance. The file format of the compressed package must be \*.tar. The related VPN files are: certificates (ca.crt and client.crt), key (client.key) and the configuration file (vpn.cnf) of the VPN client.

The following table lists the unified directories of the OpenVPN certificates and key in the configuration file (vpn.cnf) for Yealink IP phones:

VPN files	Description	Unified Directories
ca.crt	CA certificate	/config/openvpn/keys/ca.crt

VPN files	Description	Unified Directories
client.crt	Client certificate	/config/openvpn/keys/client.crt
client.key	Private key of the client	/config/openvpn/keys/client.key

For more information, refer to [OpenVPN Feature on Yealink IP phones](#).

## Procedure

VPN can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cf g	Configure VPN feature and upload a TAR file to the IP phone.  <b>Parameters:</b> static.network.vpn_enable static.openvpn.url
<b>Web User Interface</b>		Configure VPN feature and upload a TAR file to the IP phone.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=network-adv&q=load
<b>Phone User Interface</b>		Configure VPN feature.

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.network.vpn_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables OpenVPN feature on the IP phone. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->VPN->Active <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->VPN->VPN Active		
<b>static.openvpn.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>



Parameters	Permitted Values	Default
<p><b>Description:</b> Configures the access URL of the *.tar file for OpenVPN.</p> <p><b>Example:</b> static.openvpn.url = http://192.168.10.25/OpenVPN.tar</p> <p><b>Web User Interface:</b> Network-&gt;Advanced-&gt;VPN-&gt;Upload VPN Config</p> <p><b>Phone User Interface:</b> None</p>		

To upload a TAR file and configure VPN via web user interface:

1. Click on **Network->Advanced**.
2. Click **Browse** to locate the TAR file from the local system.
3. Click **Upload** to upload the TAR file.

The screenshot shows the Yealink T46G web user interface. The 'Network' tab is selected, and the 'Advanced' sub-tab is active. The 'VPN' section is highlighted with a red box, showing the 'Upload VPN Config' button and the 'Browse...' button. The 'Web Server' section is also visible, showing 'HTTP' and 'HTTPS' settings. A 'NOTE' section on the right provides information about VLAN, NAT Traversal, Quality of Service (QoS), and Web Server Type.



The web user interface prompts the message "Import config..."

4. In the **VPN** block, select the desired value from the pull-down list of **Active**.
5. Click **Confirm** to accept the change.

A dialog box pops up to prompt that the settings will take effect after a reboot.

6. Click **OK** to reboot the phone.

**To configure VPN via phone user interface after uploading a TAR file:**

1. Press **Menu**->**Advanced** (default password: admin) ->**Network**->**VPN**.
2. Press  or  , or the **Switch** soft key to select the desired value from the **VPN Active** field.

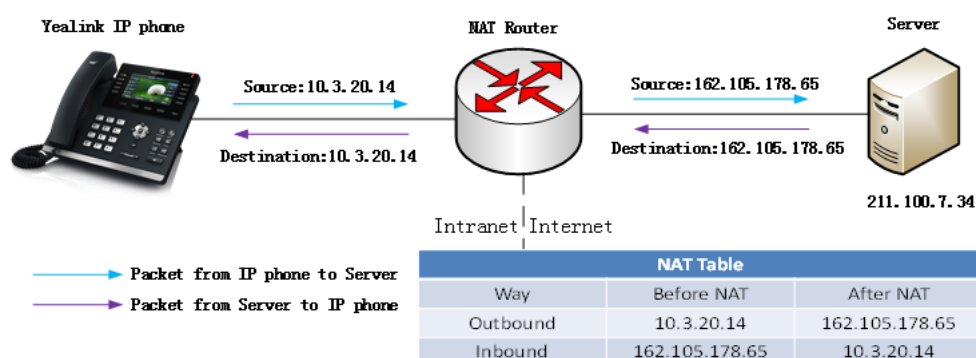
You must upload the OpenVPN TAR file using configuration files or via web user interface in advance.

3. Press the **Save** soft key to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
4. Press **OK** to reboot the phone.

## Network Address Translation (NAT)

Network Address Translation (NAT) is one of the technologies for solving the network problem – the shortage of IP addresses. Many countries provide only one public IP address for complete companies. They configure NAT to advertise the IP address for the entire network to the outside world. This can reduce the need for a large number of public IP addresses.

NAT is essentially a translation table that maps public IP address and port combinations to internal private ones. NAT ensures security since each outgoing or incoming request must first go through a translation process.



## NAT Types

### Symmetrical NAT

In symmetrical NAT, the NAT router stores the address and port where the packet was sent. Only packets coming from this address and port are forwarded back to the private address.

### Full Cone NAT

In full cone NAT, all packets from a private address (for example, iAddr: port1) to public network will be sent through a public address (for example, eAddr: port2). Packets coming from the address of any server to eAddr: port2 will be forwarded back to the private address (for example, iAddr: port1).

## Address Restricted Cone NAT

Restricted cone NAT works similar like full cone NAT. A public host (hAddr: any) can send packets to iAddr: port1 through eAddr: port2 only if iAddr: port1 has previously sent a packet to hAddr: any. "Any" means the port number doesn't matter.

## Port Restricted Cone NAT

Port restricted cone NAT works similar like full cone NAT. A public host (hAddr: hPort) can send packets to iAddr: port1 through eAddr: port2 only if iAddr: port1 has previously sent a packet to hAddr: hPort.

## NAT Traversal

In the VoIP environment, NAT breaks end-to-end connectivity.

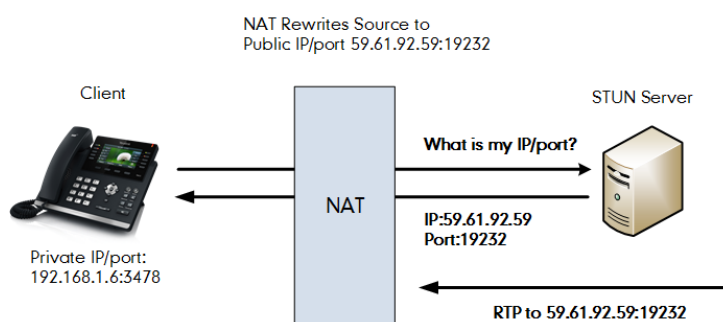
NAT traversal is a general term for techniques that establish and maintain IP connections traversing NAT gateways, typically required for client-to-client networking applications, especially for VoIP deployments. Yealink IP phones support three NAT traversal techniques: manual NAT, STUN and ICE. If manual NAT and STUN are all enabled, the IP phone will use the manually configured external IP address for NAT traversal. The TURN protocol is used as part of the ICE approach to NAT traversal.

### Manual NAT (Static NAT)

Manual NAT helps IP connections traverse NAT gateways without the third-party network server (STUN/TURN server). If manual NAT feature is enabled, the configured public IP address and port can be carried in the SIP requests or RTP packets, in which the other party obtains the phone's public address. It is useful to reduce the cost of the company's network deployment.

## STUN (Simple Traversal of UDP over NATs)

STUN is a network protocol, used in NAT traversal for applications of real-time voice, video, messaging, and other interactive IP communications. The STUN protocol allows entities behind a NAT to first discover the presence of a NAT and the type of NAT (for more information on the NAT types, refer to [NAT Types](#) on page 64) and to obtain the mapped (public) IP address and port number that the NAT has allocated for the UDP connections to remote parties. The protocol requires assistance from a third-party network server (STUN server) usually located on public Internet. The IP phone can be configured to act as a STUN client, to send exploratory STUN messages to the STUN server. The STUN server uses those messages to determine the public IP address and port used, and then informs the client.



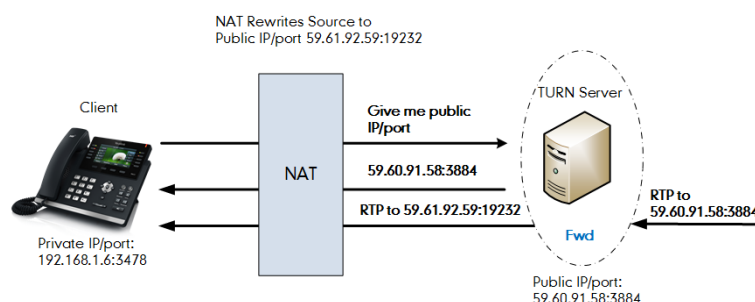
Capturing packets after you enable the STUN feature, you can find that the IP phone sends Binding Request to the STUN server, and then mapped IP address and port is placed in the Binding Response: Binding Success Response MAPPED-ADDRESS: 59.61.92.59:19232.

No.	Time	Source	Destination	Protocol	Length	Info
444	18.587848	192.168.1.6	218.107.220.74	STUN	62	Binding Request
447	18.711349	218.107.220.74	192.168.1.6	STUN	98	Binding Success Response MAPPED-ADDRESS: 59.61.92.59:19232

STUN does not work if the NAT device is symmetric. This may be a drawback in many situations as most enterprise-class firewalls are symmetric.

## TURN (Traversal Using Relays around NAT)

TURN is a network protocol described in [RFC 5766](#), allows a host located behind a NAT (called the TURN client) to communicate and exchange packets with other hosts (peers, called the TURN server) using a relay. In these situations, the host uses the services of an intermediate node to act as a communication relay. It governs the reception of data over a Transmission Control Protocol (TCP) or a UDP connection. This solves the problems of clients behind symmetric NATs which cannot rely on STUN to solve the NAT traversal issue. This method is appropriate in some situations, but it scales poorly since the media must go through the TURN server.



If you configure both STUN and TURN on the phone, it discovers what type of NAT device is between the phone and the public network. If the NAT device is full cone, address restricted cone, or port restricted cone, the phone uses STUN. If the NAT device is symmetric, the phone uses TURN. TURN is compatible with all types of NAT devices but can be costly since all traffic goes through a media relay (which can be slow, can exchange more messages, and requires the TURN server to allocate bandwidth for calls).

Although TURN will almost always provide connectivity to a client, it comes at high cost to the provider of the TURN server. Therefore other mechanisms (such as STUN or direct connectivity) will be preferred when possible.

## ICE (Interactive Communications Establishment)

ICE, described in [RFC 5245](#), is a technique for Network Address Translator (NAT) traversal for UDP-based media streams established by the offer/answer model, not intended for NAT traversal for SIP. It is an extension to the offer/answer model, and works by including a multiplicity of IP addresses and ports in SDP offers and answers, which are then tested for connectivity by peer-to-peer connectivity checks.

ICE makes use of the STUN protocol and its extension, TURN. In an ICE environment, two IP phones communicating at different locations are able to communicate via the SIP protocol by exchanging Session Description Protocol (SDP) messages. At the beginning of the ICE process, the phones are ignorant of their own topologies. In particular, they might or might not be behind a NAT. ICE allows IP phones to discover enough information about their topologies to find the optimal path(s) by which they can communicate.

ICE optimizes the media path. For example, when two IP phones in the same network are calling each other via a long media path through other external networks. With ICE enabled, the short

media path in the same network would be chosen, which will probably have better quality than the long one.

ICE is a complex solution to the problem of NAT traversal. Due to its complexity there is very limited client support for ICE today.

## SIP and TLS Source Ports for NAT Traversal

You can configure the SIP and TLS source ports on the IP Phone. Previously, the IP phone used default values (5060 for UDP/TCP and 5061 for TLS). In the configuration files, you can use the following parameters to configure the SIP and TLS source ports:

- Local SIP Port
- TLS SIP Port

If NAT is disabled, the port number shows in the Via and Contact SIP headers of SIP messages. If NAT is enabled, the phone uses the NAT port number (and NAT IP address) in the Via and Contact SIP headers of SIP messages, but still use the configured source port.

## Procedure

NAT traversal can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<y0000000000xx>.cfg	Configure STUN feature and STUN server on a phone basis.  <b>Parameters:</b> static.sip.nat_stun.enable static.sip.nat_stun.server static.sip.nat_stun.port
		Configure manual NAT feature on a phone basis.  <b>Parameters:</b> static.network.static_nat.enable static.network.static_nat.addr
		Configure ICE feature.  <b>Parameter:</b> static.ice.enable
		Configure TURN feature and TURN server.  <b>Parameters:</b> static.sip.nat_turn.enable static.sip.nat_turn.server static.sip.nat_turn.port static.sip.nat_turn.username

		static.sip.nat_turn.password
		Configure local SIP port and TLS SIP port. <b>Parameters:</b> sip.listen_port sip.tls_listen_port
	<MAC>.cfg	Configure NAT traversal on a per-line basis. <b>Parameter:</b> account.X.nat.nat_traversal
<b>Web User Interface</b>		Configure manual NAT feature on a phone basis. Configure ICE feature. Configure TURN feature and TURN server. Configure STUN feature and STUN server on a phone basis. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=network-nat&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=network-nat&amp;q=load</a>
		Configure local SIP port and TLS SIP port. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=settings-sip&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=settings-sip&amp;q=load</a>
		Configure NAT traversal on a per-line basis. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-register&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-register&amp;q=load&amp;acc=0</a>
<b>Phone User Interface</b>		Configure STUN feature and STUN server on a phone basis. Configure NAT traversal on a per-line basis.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
static.sip.nat_stun.enable	0 or 1	0

Parameters	Permitted Values	Default
<p><b>Description:</b> Enables or disables the STUN (Simple Traversal of UDP over NATs) feature on the IP phone.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;NAT-&gt;STUN-&gt;Active</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;NAT-&gt;NAT Status</p>		
<b>static.sip.nat_stun.server</b>	<b>IP address or domain name</b>	<b>Blank</b>
<p><b>Description:</b> Configures the IP address or the domain name of the STUN (Simple Traversal of UDP over NATs) server.</p> <p><b>Example:</b> sip.nat_stun.server = 218.107.220.201</p> <p><b>Note:</b> It works only if the value of the parameter "static.sip.nat_stun.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;NAT-&gt;STUN-&gt;STUN Server</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;NAT-&gt;STUN Server</p>		
<b>static.sip.nat_stun.port</b>	<b>Integer from 1024 to 65000</b>	<b>3478</b>



Parameters	Permitted Values	Default
<p><b>Description:</b> Configures the port of the STUN (Simple Traversal of UDP over NATs) server.</p> <p><b>Example:</b> sip.nat_stun.port = 3478</p> <p><b>Note:</b> It works only if the value of the parameter "static.sip.nat_stun.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;NAT-&gt;STUN-&gt;STUN Port (1024~65000)</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;NAT-&gt;STUN Port</p>		
<b>account.X.nat.nat_traversal<sup>[1]</sup></b>	<b>0, 1 or 2</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the NAT traversal for a specific account.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-STUN, it works only if the value of the parameter "static.sip.nat_stun.enable" is set to 1 (Enabled).</p> <p><b>2</b>-Manual NAT, it works only if the value of the parameter "network.static_nat.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Account-&gt;Register-&gt;NAT</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Accounts-&gt;NAT Status</p> <p><b>Note:</b> Manual NAT cannot be configured via phone user interface.</p>		
<b>static.network.static_nat.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the manual NAT feature on the IP phone.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;NAT-&gt;Nat Manual-&gt;Active</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>static.network.static_nat.addr</b>	<b>IP address</b>	<b>Blank</b>
<b>Description:</b> Configures the IP address to be advertised in SIP signaling. It should match the external IP address used by the NAT device. <b>Example:</b> network.static_nat.addr = 10.3.5.33 <b>Note:</b> It works only if the value of the parameter "static.network.static_nat.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->NAT->Nat Manual->IP Address <b>Phone User Interface:</b> None		
<b>static.ice.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the ICE (Interactive Connectivity Establishment) feature on the IP phone. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> To use ICE feature, you have to configure the STUN and/or TURN server address in advance. If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->NAT->ICE->Active <b>Phone User Interface:</b> None		
<b>static.sip_nat_turn.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the TURN (Traversal Using Relays around NAT) feature on the IP phone. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take		

Parameters	Permitted Values	Default
<p>effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;NAT-&gt;TURN-&gt;Active</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.sip.nat_turn.server</b>	<b>IP address or domain name</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the IP address or the domain name of the TURN (Traversal Using Relays around NAT) server.</p> <p><b>Example:</b></p> <p>static.sip.nat_turn.server = 218.107.220.202</p> <p><b>Note:</b> It works only if the value of the parameter "static.sip.nat_turn.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;NAT-&gt;TURN-&gt;TURN Server</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.sip.nat_turn.port</b>	<b>Integer from 1 to 65535</b>	<b>3478</b>
<p><b>Description:</b></p> <p>Configures the port of the TURN (Traversal Using Relays around NAT) server.</p> <p><b>Example:</b></p> <p>static.sip.nat_turn.port = 3478</p> <p><b>Note:</b> It works only if the value of the parameter "static.sip.nat_turn.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;NAT-&gt;TURN-&gt;TURN Port (1~65535)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.sip.nat_turn.username</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p>		

Parameters	Permitted Values	Default
<p>Configures the user name to authenticate to TURN (Traversal Using Relays around NAT) server.</p> <p><b>Example:</b></p> <p>sip.nat_turn.username = admin</p> <p><b>Note:</b> It works only if the value of the parameter "static.sip.nat_turn.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;NAT-&gt;TURN-&gt;User Name</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.sip.nat_turn.password</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the password to authenticate to the TURN (Traversal Using Relays around NAT) server.</p> <p><b>Example:</b></p> <p>sip.nat_turn.password = yealink1105</p> <p><b>Note:</b> It works only if the value of the parameter "static.sip.nat_turn.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;NAT-&gt;TURN-&gt;Password</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>sip.listen_port</b>	<b>Integer from 1024 to 65535</b>	<b>5060</b>
<p><b>Description:</b></p> <p>Configures the local SIP port.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;SIP-&gt;Local SIP Port</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>sip.tls_listen_port</b>	<b>0, Integer from 1024 to 65535</b>	<b>5061</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the local TLS listen port.</p> <p>If it is set to 0, the IP phone will not listen the TLS service.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;SIP-&gt;TLS SIP Port</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

#### To configure NAT traversal and STUN server via web user interface:

1. Click on **Network->NAT**.
2. In the **STUN** block, select the desired value from the pull-down list of **Active**.
3. Enter the IP address or the domain name of the STUN server in the **STUN Server** field.
4. Enter the port of the STUN server in the **STUN Port (1024~65000)** field.

The screenshot shows the Yealink T46G web interface. The 'Network' tab is selected, and the 'NAT' sub-tab is active. In the 'STUN' configuration section, the 'Status' is set to 'Active', the 'STUN Server' is '218.107.220.201', and the 'STUN Port (1024~65000)' is '3478'. A red box highlights these three fields. On the right, a 'NOTE' box states: 'Network NAT: You can click here to get more guides.'

5. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
6. Click **OK** to reboot the phone.

#### To configure manual NAT via web user interface:

1. Click on **Network->NAT**.
2. In the **Nat Manual** block, select the desired value from the pull-down list of **Active**.

3. Enter the external IP address in the **IP Address** field.

4. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
5. Click **OK** to reboot the phone.

**To configure ICE feature via web user interface:**

1. Click on **Network->NAT**.
2. In the **ICE** block, select the desired value from the pull-down list of **Active**.

3. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
4. Click **OK** to reboot the phone.

**To configure NAT traversal for account via web user interface:**

1. Click on **Account->Register**.
2. Select the desired account from the pull-down list of **Account**.

3. Select **STUN/Manual NAT** from the pull-down list of **NAT**.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected. The 'NAT' dropdown menu is highlighted with a red box, showing 'STUN' selected. The 'Confirm' button is visible at the bottom.

4. Click **Confirm** to accept the change.



**To configure local SIP port and TLS SIP port via web user interface:**

1. Click on **Settings->SIP**.
2. Enter the desired local SIP port in the **Local SIP Port** field.
3. Enter the desired TLS SIP port in the **TLS SIP Port** field.

The screenshot shows the Yealink T46G web interface. The 'Settings' tab is selected, and the 'SIP Config' section is visible. The 'Local SIP Port' and 'TLS SIP Port' fields are highlighted with a red box, showing values of 5060 and 5061 respectively. The 'Confirm' button is visible at the bottom.

- Click **Confirm** to accept the change.





**To configure NAT traversal and STUN server via phone user interface:**

- Press **Menu**->**Advanced** (default password: admin) ->**Network**->**NAT**->**NAT Status**.
- Press  or , or the **Switch** soft key to select the desired value from the **NAT Status** field.
- Enter the IP address or the domain name of the STUN server in the **STUN Server** field.
- Enter the port of the STUN server in the **STUN Port** field.
- Press the **Save** soft key to accept the change.

A dialog box pops up to prompt that the settings will take effect after a reboot.

- Press **OK** to reboot the phone.

**To configure NAT traversal for a specific account via phone user interface:**

- Press **Menu**->**Advanced** (default password: admin) ->**Accounts**.
- Press  or  to select the desired account and then press the **Enter** soft key.
- Press  or , or the **Switch** soft key to select the desired value from the **NAT Status** field.
- Press the **Save** soft key to accept the change.

## Keep Alive

IP phones can send keep-alive packets to the NAT device for keeping the communication port open.

### Procedure

Keep alive feature can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	<p>Configure the type of keep-alive packets on a per-line basis.</p> <p><b>Parameter:</b></p> <p>account.X.nat.udp_update_enable</p>
		<p>Configure the keep-alive interval on a per-line basis.</p> <p><b>Parameter:</b></p> <p>account.X.nat.udp_update_time</p>
<b>Web User Interface</b>		<p>Configure the type of keep-alive packets on a per-line basis.</p> <p>Configure the keep-alive interval on a per-line basis.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=a</p>



	ccount-adv&q=load&acc=0
--	-------------------------

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>account.X.nat.udp_update_enable<sup>[1]</sup></b>	<b>0, 1, 2 or 3</b>	<b>1</b>
<p><b>Description:</b></p> <p>Configures the type of keep-alive packets sent by the IP phone to the NAT device to keep the communication port open so that NAT can continue to function for a specific account.</p> <p><b>0</b>-Disabled, the IP phone will not send keep-alive packets.</p> <p><b>1</b>-Default (the IP phone sends UDP packets to the server)</p> <p><b>2</b>-Options (the IP phone sends SIP OPTIONS packets to the server)</p> <p><b>3</b>-Notify (the IP phone sends SIP NOTIFY packets to the server)</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;Keep Alive Type</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.nat.udp_update_time<sup>[1]</sup></b>	<b>Integer from 15 to 2147483647</b>	<b>30</b>
<p><b>Description:</b></p> <p>Configures the keep-alive interval (in seconds) for a specific account.</p> <p><b>Example:</b></p> <p>account.1.nat.udp_update_time = 30</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.nat.udp_update_enable" is set to 1, 2 or 3.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;Keep Alive Interval(Seconds)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure the type of keep-alive packets and keep-alive interval via web user interface:**

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.

3. Select the desired value from the pull-down list of **Keep Alive Type**.
4. Enter the keep-alive interval in the **Keep Alive Interval(Seconds)** field.

5. Click **Confirm** to accept the change.

## Rport

The Session Initiation Protocol (SIP) operates over UDP and TCP. When used with UDP, responses to requests are returned to the source address the request came from, and returned to the port written into the topmost "Via" header of the request message. However, this behavior is not desirable when the client is behind a Network Address Translation (NAT) or firewall. So a new parameter "rport" for the "Via" header field is required.

Rport described in [RFC 3581](#), allows a client to request that the server sends the response back to the source port from which the request came.

Rport feature depends on support from a SIP server.

### Procedure

Rport feature can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure NAT Rport feature on a per-line basis. <b>Parameter:</b> account.X.nat.rport
<b>Web User Interface</b>		Configure NAT Rport feature on a per-line basis. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>

### Details of Configuration Parameter:

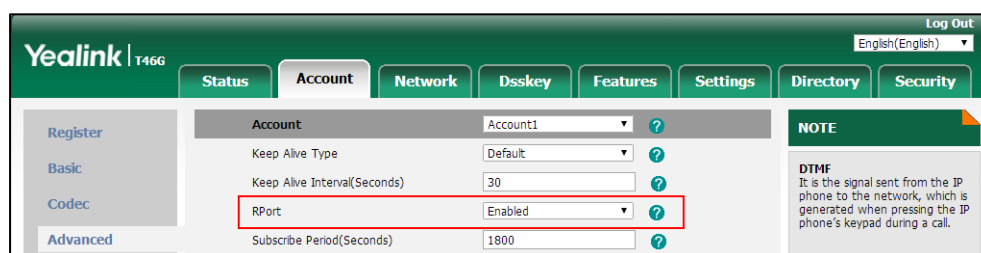
Parameter	Permitted Values	Default
account.X.nat.rport <sup>[1]</sup>	0, 1 or 2	0
<b>Description:</b>		

Parameter	Permitted Values	Default
<p>Enables or disables NAT Rport feature for a specific account.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>2</b>-Enable Direct Process</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;RPort</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

#### To configure Rport feature via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **RPort**.



4. Click **Confirm** to accept the change.

## Quality of Service (QoS)

Quality of Service (QoS) is the ability to provide different priorities for different packets in the network, allowing the transport of traffic with special requirements. QoS guarantees are important for applications that require fixed bit rate and are delay sensitive when the network capacity is insufficient. There are four major QoS factors to be considered when configuring a modern QoS implementation: bandwidth, delay, jitter and loss.

QoS provides better network service through the following features:

- Supporting dedicated bandwidth
- Improving loss characteristics
- Avoiding and managing network congestion

- Shaping network traffic
- Setting traffic priorities across the network

The Best-Effort service is the default QoS model in IP networks. It provides no guarantees for data delivering, which means delay, jitter, packet loss and bandwidth allocation are unpredictable. Differentiated Services (DiffServ or DS) is the most widely used QoS model. It provides a simple and scalable mechanism for classifying and managing network traffic and providing QoS on modern IP networks. Differentiated Services Code Point (DSCP) is used to define DiffServ classes and stored in the first six bits of the ToS (Type of Service) field. Each router on the network can provide QoS simply based on the DiffServ class. The DSCP value ranges from 0 to 63 with each DSCP specifying a particular per-hop behavior (PHB) applicable to a packet. A PHB refers to the packet scheduling, queuing, policing, or shaping behavior of a node on any given packet.

Four standard PHBs available to construct a DiffServ-enabled network and achieve QoS:

- **Class Selector PHB** -- backwards compatible with IP precedence. Class Selector code points are of the form "xxx000". The first three bits are the IP precedence bits. These class selector PHBs retain almost the same forwarding behavior as nodes that implement IP precedence-based classification and forwarding.
- **Expedited Forwarding PHB** -- the key ingredient in DiffServ model for providing a low-loss, low-latency, low-jitter and assured bandwidth service.
- **Assured Forwarding PHB** -- defines a method by which BAs (Bandwidth Allocations) can be given different forwarding assurances.
- **Default PHB** -- specifies that a packet marked with a DSCP value of "000000" gets the traditional best effort service from a DS-compliant node.

VoIP is extremely bandwidth and delay-sensitive. QoS is a major issue in VoIP implementations, regarding how to guarantee that packet traffic not be delayed or dropped due to interference from other lower priority traffic. VoIP can guarantee high-quality QoS only if the voice and the SIP packets are given priority over other kinds of network traffic. IP phones support the DiffServ model of QoS.

## Voice QoS

In order to make VoIP transmissions intelligible to receivers, voice packets should not be dropped, excessively delayed, or made to suffer varying delay. DiffServ model can guarantee high-quality voice transmission when the voice packets are configured to a higher DSCP value.

## SIP QoS

SIP protocol is used for creating, modifying and terminating two-party or multi-party sessions. To ensure good voice quality, SIP packets emanated from IP phones should be configured with a high transmission priority.

DSCPs for voice and SIP packets can be specified respectively.

### Note

For voice and SIP packets, the IP phone obtains DSCP info from the network policy if LLDP feature is enabled, which takes precedence over manual settings. For more information on LLDP, refer to [LLDP](#) on page 41.

## Procedure

QoS can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the DSCPs for voice packets and SIP packets.  <b>Parameters:</b> static.network.qos.audiotos static.network.qos.signalto
<b>Web User Interface</b>		Configure the DSCPs for voice packets and SIP packets.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=network-adv&q=load

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.network.qos.audiotos</b>	<b>Integer from 0 to 63</b>	<b>46</b>
<b>Description:</b> Configures the DSCP (Differentiated Services Code Point) for voice packets. The default DSCP value for RTP packets is 46 (Expedited Forwarding). <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->Voice QoS (0~63) <b>Phone User Interface:</b> None		
<b>static.network.qos.signalto</b>	<b>Integer from 0 to 63</b>	<b>26</b>
<b>Description:</b> Configures the DSCP (Differentiated Services Code Point) for SIP packets. The default DSCP value for SIP packets is 26 (Assured Forwarding). <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->SIP QoS (0~63)		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b>		
None		

To configure DSCPs for voice packets and SIP packets via web user interface:

1. Click on **Network->Advanced**.
2. Enter the desired value in the **Voice QoS (0~63)** field.
3. Enter the desired value in the **SIP QoS (0~63)** field.

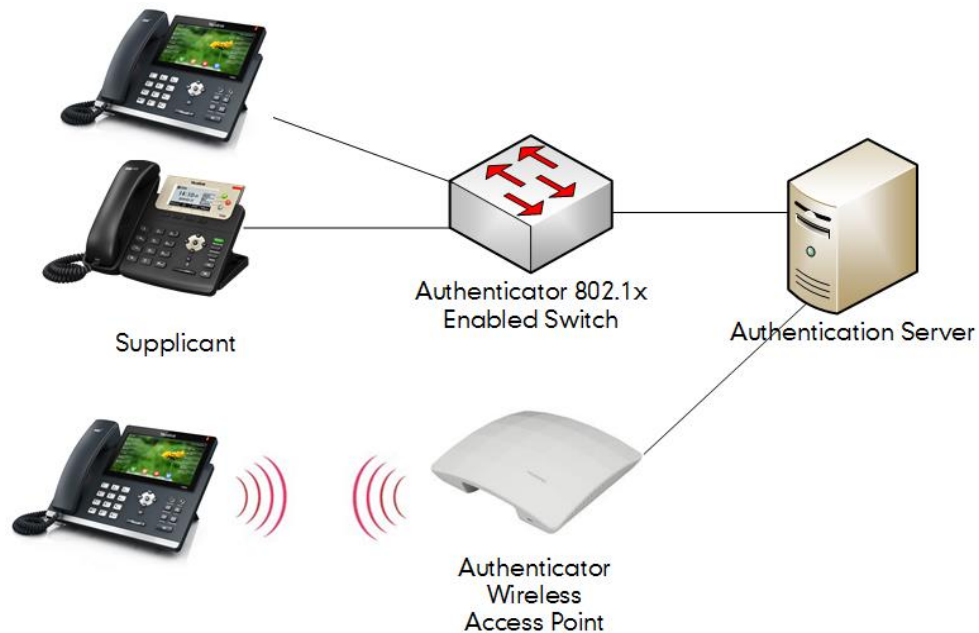
The screenshot shows the Yealink T46G web interface. The 'Network' tab is selected, and the 'Advanced' sub-tab is active. In the 'Voice QoS' section, the 'Voice QoS (0~63)' field is set to 46 and the 'SIP QoS (0~63)' field is set to 26. These fields are highlighted with a red rectangle. Other settings include LLDP (Active: Enabled, Packet Interval: 60), CDP (Active: Disabled, Packet Interval: 60), Local RTP Port (Max: 12780, Min: 11780), Span to PC (Enabled), Registration Random (0), and VPN (Active: Enabled). A 'NOTE' sidebar on the right contains information about VLAN, NAT Traversal, Quality of Service (QoS), Web Server Type, and 802.1X Authentication.

4. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
5. Click **OK** to reboot the phone.

## 802.1X Authentication

IEEE 802.1X authentication is an IEEE standard for Port-based Network Access Control (PNAC), part of the IEEE 802.1 group of networking protocols. It offers an authentication mechanism for devices to connect/link to a LAN or WLAN.

The 802.1X authentication involves three parties: a supplicant, an authenticator and an authentication server. The supplicant is the IP phone that wishes to attach to the LAN or WLAN. With 802.1X port-based authentication, the IP phone provides credentials, such as user name and password, for the authenticator, and then the authenticator forwards the credentials to the authentication server for verification. If the authentication server determines the credentials are valid, the IP phone is allowed to access resources located on the protected side of the network.



Yealink IP phones support the following protocols for 802.1X authentication:

- EAP-MD5
- EAP-TLS (requires Device and CA certificates, requires no password)
- EAP-PEAP/MSCHAPv2 (requires CA certificates)
- EAP-TTLS/EAP-MSCHAPv2 (requires CA certificates)
- EAP-PEAP/GTC (requires CA certificates)
- EAP-TTLS/EAP-GTC (requires CA certificates)
- EAP-FAST (supports EAP In-Band provisioning, requires CA certificates if the provisioning mode is Authenticated Provisioning)

For more information on 802.1X authentication, refer to [Yealink 802.1X Authentication](#).

## Procedure

802.1X authentication can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>. cfg	Configure the 802.1X authentication. <b>Parameters:</b> static.network.802_1x.mode static.network.802_1x.eap_fast_provision_m
------------------------------------------------------	-------------------------	----------------------------------------------------------------------------------------------------------------------------------------

		ode static.network.802_1x.anonymous_identity static.network.802_1x.identity static.network.802_1x.md5_password static.network.802_1x.root_cert_url static.network.802_1x.client_cert_url
<b>Web User Interface</b>		Configure the 802.1X authentication. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=network-adv&q=load
<b>Phone User Interface</b>		Configure the 802.1X authentication.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>static.network.802_1x.mode</b>	<b>0, 1, 2, 3, 4, 5, 6 or 7</b>	<b>0</b>
<b>Description:</b> Configures the 802.1x authentication method. <b>0</b> -EAP-None, 802.1x authentication is not required. <b>1</b> -EAP-MD5 <b>2</b> -EAP-TLS <b>3</b> -EAP-PEAP/MSCHAPv2 <b>4</b> -EAP-TTLS/EAP-MSCHAPv2 <b>5</b> -EAP-PEAP/GTC <b>6</b> -EAP-TTLS/EAP-GTC <b>7</b> -EAP-FAST <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Network->Advanced->802.1x->802.1x Mode <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Network->802.1x->802.1x Mode		
<b>static.network.802_1x.eap_fast_provision_mode</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b>		



Parameters	Permitted Values	Default
<p>Configures the EAP In-Band provisioning method for EAP-FAST.</p> <p><b>0</b>-Unauthenticated Provisioning, EAP In-Band provisioning is enabled by server unauthenticated PAC (Protected Access Credential) provisioning using anonymous Diffie-Hellman key exchange.</p> <p><b>1</b>-Authenticated Provisioning, EAP In-Band provisioning is enabled by server authenticated PAC provisioning using certificate based server authentication.</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.802_1x.mode" is set to 7 (EAP-FAST). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Advanced-&gt;802.1x-&gt;Provisioning Mode</p> <p><b>Phone User Interface:</b> None</p>		
<b>static.network.802_1x.anonymous_identity</b>	<b>String within 512 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the anonymous identity (user name) for 802.1X authentication. It is used for constructing a secure tunnel for 802.1X authentication.</p> <p><b>Example:</b> static.network.802_1x.anonymous_identity = anonymous</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.802_1x.mode" is set to 2, 3, 4, 5, 6 or 7. If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Advanced-&gt;802.1x-&gt;Anonymous Identity</p> <p><b>Phone User Interface:</b> None</p>		
<b>static.network.802_1x.identity</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the identity (or user name) for 802.1x authentication.</p> <p><b>Example:</b> static.network.802_1x.identity = yealink</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.802_1x.mode" is set to 1, 2, 3, 4, 5, 6 or 7. If you change this parameter, the IP phone will reboot to make the change</p>		

Parameters	Permitted Values	Default
<p>take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Advanced-&gt;802.1x-&gt;Identity</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;802.1x-&gt;Identity</p>		
<b>static.network.802_1x.md5_password</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the password for 802.1x authentication.</p> <p><b>Example:</b> static.network.802_1x.md5_password = admin123</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.802_1x.mode" is set to 1, 3, 4, 5, 6 or 7. If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Advanced-&gt;802.1x-&gt;MD5 Password</p> <p><b>Phone User Interface:</b> Menu-&gt;Advanced (default password: admin) -&gt;Network-&gt;802.1x-&gt;MD5 Password</p>		
<b>static.network.802_1x.root_cert_url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of the CA certificate.</p> <p><b>Example:</b> static.network.802_1x.root_cert_url = http://192.168.1.10/ca.pem</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.802_1x.mode" is set to 2, 3, 4, 5, 6 or 7. If the authentication method is EAP-FAST, you also need to set the value of the parameter "static.network.802_1x.eap_fast_provision_mode" to 1 (Authenticated Provisioning). The format of the CA certificate must be *.pem, *.crt, *.cer or *.der.</p> <p><b>Web User Interface:</b> Network-&gt;Advanced-&gt;802.1x-&gt;CA Certificates</p> <p><b>Phone User Interface:</b> None</p>		
<b>static.network.802_1x.client_cert_url</b>	<b>URL within 511 characters</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the access URL of the device certificate.</p> <p><b>Example:</b></p> <p>static.network.802_1x.client_cert_url = http://192.168.1.10/client.pem</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.802_1x.mode" is set to 2 (EAP-TLS). The format of the device certificate must be *.pem.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;802.1x-&gt;Device Certificates</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure the 802.1X authentication via web user interface:**

1. Click on **Network->Advanced**.
2. In the **802.1x** block, select the desired protocol from the pull-down list of **802.1x Mode**.
  - a) If you select **EAP-MD5**:
    - 1) Enter the user name for authentication in the **Identity** field.
    - 2) Enter the password for authentication in the **MD5 Password** field.

The screenshot shows the Yealink T46G web interface. The 'Network' tab is selected, and the 'Advanced' sub-tab is active. Under the '802.1x' section, the '802.1x Mode' is set to 'EAP-MD5'. The 'Identity' field is filled with 'yealink', and the 'MD5 Password' field is masked with dots. The 'Provisioning Mode' is set to 'Unauthenticated Provisic'. The 'CA Certificates' and 'Device Certificates' fields have 'Upload' buttons and 'Browse...' links. The 'Confirm' and 'Cancel' buttons are at the bottom.

- b) If you select **EAP-TLS**:
  - 1) (Optional.) Enter the anonymous user name for authentication in the **Anonymous Identity** field.
  - 2) Enter the user name for authentication in the **Identity** field.

- 3) Leave the **MD5 Password** field blank.
- 4) In the **CA Certificates** field, click **Browse** to select the desired CA certificate (\*.pem, \*.crt, \*.cer or \*.der) from your local system.
- 5) In the **Device Certificates** field, click **Browse** to select the desired client (\*.pem or \*.cer) certificate from your local system.
- 6) Click **Upload** to upload the certificates.

The screenshot shows the Yealink T46G web interface. The 'Network' tab is selected, and the '802.1x' configuration page is displayed. The '802.1x Mode' is set to 'EAP-TLS'. The 'Provisioning Mode' is 'Unauthenticated Provisioning'. The 'Anonymous Identity' field is empty. The 'Identity' field contains 'yealink'. The 'MD5 Password' field is empty. The 'CA Certificates' and 'Device Certificates' fields both have 'Upload' and 'Browse...' buttons. A red box highlights the 802.1x configuration section. On the right, there is a 'NOTE' section with information about VLAN, NAT Traversal, and Quality of Service (QoS).

- c) If you select **EAP-PEAP/MSCHAPv2**:
- 1) (Optional.) Enter the anonymous user name for authentication in the **Anonymous Identity** field.
  - 2) Enter the user name for authentication in the **Identity** field.
  - 3) Enter the password for authentication in the **MD5 Password** field.
  - 4) In the **CA Certificates** field, click **Browse** to select the desired CA certificate (\*.pem, \*.crt, \*.cer or \*.der) from your local system.

- 5) Click **Upload** to upload the certificate.

The screenshot shows the Yealink T46G configuration interface. The 'Network' tab is selected, and the '802.1x' section is expanded. The '802.1x Mode' is set to 'EAP-PEAP/MSCHAPv2'. The 'CA Certificates' field has an 'Upload' button highlighted with a red box. Other fields include 'Provisioning Mode' (Unauthenticated Provisioning), 'Anonymous Identity' (Anonymous), 'Identity' (yealink), and 'MD5 Password' (masked). There are also 'Browse...' buttons for CA and Device Certificates.

- d) If you select **EAP-TTLS/EAP-MSCHAPv2**:

- 1) (Optional.) Enter the anonymous user name for authentication in the **Anonymous Identity** field.
- 2) Enter the user name for authentication in the **Identity** field.
- 3) Enter the password for authentication in the **MD5 Password** field.
- 4) In the **CA Certificates** field, click **Browse** to select the desired CA certificate (\*.pem, \*.crt, \*.cer or \*.der) from your local system.
- 5) Click **Upload** to upload the certificate.

The screenshot shows the Yealink T46G configuration interface. The 'Network' tab is selected, and the '802.1x' section is expanded. The '802.1x Mode' is set to 'EAP-TTLS/EAP-MSCHAPv2'. The 'CA Certificates' field has an 'Upload' button highlighted with a red box. Other fields include 'Provisioning Mode' (Unauthenticated Provisioning), 'Anonymous Identity' (Anonymous), 'Identity' (yealink), and 'MD5 Password' (masked). There are also 'Browse...' buttons for CA and Device Certificates.

- e) If you select **EAP-PEAP/GTC**:
- 1) (Optional.) Enter the anonymous user name for authentication in the **Anonymous Identity** field.
  - 2) Enter the user name for authentication in the **Identity** field.
  - 3) Enter the password for authentication in the **MD5 Password** field.
  - 4) In the **CA Certificates** field, click **Browse** to select the desired CA certificate (\*.pem, \*.crt, \*.cer or \*.der) from your local system.

The screenshot shows the Yealink T46G web interface. The 'Network' tab is selected. Under the '802.1x' section, the '802.1x Mode' is set to 'EAP-PEAP/GTC'. The 'Provisioning Mode' is 'Unauthenticated Provisic'. The 'Anonymous Identity' is 'Anonymous'. The 'Identity' is 'yealink'. The 'MD5 Password' is masked with dots. The 'CA Certificates' field has an 'Upload' button and a 'Browse...' button. The 'Device Certificates' field also has an 'Upload' button and a 'Browse...' button. A red box highlights the '802.1x Mode' and 'CA Certificates' fields. On the right, there is a 'NOTE' section with information about VLAN, NAT Traversal, and Quality of Service (QoS).

- 5) Click **Upload** to upload the certificate.
- f) If you select **EAP-TTLS/EAP-GTC**:
- 1) (Optional.) Enter the anonymous user name for authentication in the **Anonymous Identity** field.
  - 2) Enter the user name for authentication in the **Identity** field.
  - 3) Enter the password for authentication in the **MD5 Password** field.

- 4) In the **CA Certificates** field, click **Browse** to select the desired CA certificate (\*.pem, \*.crt, \*.cer or \*.der) from your local system.

The screenshot shows the Yealink T46G web interface with the 'Network' tab selected. The '802.1x' configuration section is highlighted with a red box. The '802.1x Mode' is set to 'EAP-TTLS/EAP-GTC'. The 'Provisioning Mode' is set to 'Unauthenticated Provisioning'. The 'Anonymous Identity' is set to 'Anonymous'. The 'Identity' field contains 'yealink'. The 'MD5 Password' field contains a masked password. The 'CA Certificates' field has an 'Upload' button and a 'Browse...' button. The 'Device Certificates' field also has an 'Upload' button and a 'Browse...' button. The 'Confirm' and 'Cancel' buttons are at the bottom.

- 5) Click **Upload** to upload the certificate.
- g) If you select **EAP-FAST**:
- 1) Select the desired value from the pull-down list of **Provisioning Mode**.
  - 2) (Optional.) Enter the anonymous user name for authentication in the **Anonymous Identity** field.
  - 3) Enter the user name for authentication in the **Identity** field.
  - 4) Enter the password for authentication in the **MD5 Password** field.
  - 5) (Optional.) In the **CA Certificates** field, click **Browse** to select the desired CA certificate (\*.pem, \*.crt, \*.cer or \*.der) from your local system.

The CA certificate needs to be uploaded only when **Authenticated Provisioning** mode is selected from the **Provisioning Mode** field.

The screenshot shows the Yealink T46G web interface. The 'Network' tab is selected, and the '802.1x' configuration page is displayed. The '802.1x Mode' is set to 'EAP-FAST'. The 'Provisioning Mode' is set to 'Unauthenticated Provisic'. The 'Identity' field is 'yealink' and the 'MD5 Password' is masked. The 'CA Certificates' and 'Device Certificates' fields have 'Upload' buttons. A red box highlights the '802.1x Mode', 'Provisioning Mode', 'Identity', and 'MD5 Password' fields.

- 6) Click **Upload** to upload the certificate.
3. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that settings will take effect after a reboot.
4. Click **OK** to reboot the phone.

#### To configure the 802.1X authentication via phone user interface:

1. Press **Menu->Advanced** (default password: admin) -> **Network->802.1x**.
2. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **802.1x Mode** field.
  - a) If you select **EAP-MD5**:
    - 1) Enter the user name for authentication in the **Identity** field.
    - 2) Enter the password for authentication in the **MD5 Password** field.
  - b) If you select **EAP-TLS**:
    - 1) Enter the user name for authentication in the **Identity** field.
    - 2) Leave the **MD5 Password** field blank.
  - c) If you select **EAP-PEAP/MSCHAPv2**:
    - 1) Enter the user name for authentication in the **Identity** field.
    - 2) Enter the password for authentication in the **MD5 Password** field.
  - d) If you select **EAP-TTLS/EAP-MSCHAPv2**:
    - 1) Enter the user name for authentication in the **Identity** field.
    - 2) Enter the password for authentication in the **MD5 Password** field.



- e) If you select **EAP-PEAP/GTC**:
    - 1) Enter the user name for authentication in the **Identity** field.
    - 2) Enter the password for authentication in the **MD5 Password** field.
  - f) If you select **EAP-TTLS/EAP-GTC**:
    - 1) Enter the user name for authentication in the **Identity** field.
    - 2) Enter the password for authentication in the **MD5 Password** field.
  - g) If you select **EAP-FAST**:
    - 1) Enter the user name for authentication in the **Identity** field.
    - 2) Enter the password for authentication in the **MD5 Password** field.
3. Click **Save** to accept the change.

A dialog box pops up to prompt that the settings will take effect after a reboot.
  4. Press **OK** to reboot the phone.

## Setting Up Your Phones with a Provisioning Server

This chapter provides basic instructions for setting up your IP phones with a provisioning server.

This chapter consists of the following sections:

- [Provisioning Points to Consider](#)
- [Provisioning Methods](#)
- [Boot Files, Configuration Files and Resource Files](#)
- [Setting Up a Provisioning Server](#)
- [Upgrading Firmware](#)
- [Keeping User Personalized Settings after Auto Provisioning](#)

## Provisioning Points to Consider

- If you are provisioning a mass of IP phones, we recommend you to use central provisioning method as your primary configuration method. For more information on central provisioning, refer to [Central Provisioning](#) on page 97.
- A provisioning server maximizes the flexibility you have when installing, configuring, upgrading, and managing the IP phones, and enables you to store boot, configuration, log, and contact files on the server. You can set up a provisioning server on the local area network (LAN) or anywhere on the Internet. For more information, refer to [Setting Up a Provisioning Server](#) on page 108.
- If the IP phone cannot obtain the address of a provisioning server during startup, and has not been configured with settings from any other source, the IP phone will use configurations stored in the flash memory. If the phone cannot obtain the address of a provisioning server and has previously been configured with settings, it will use those

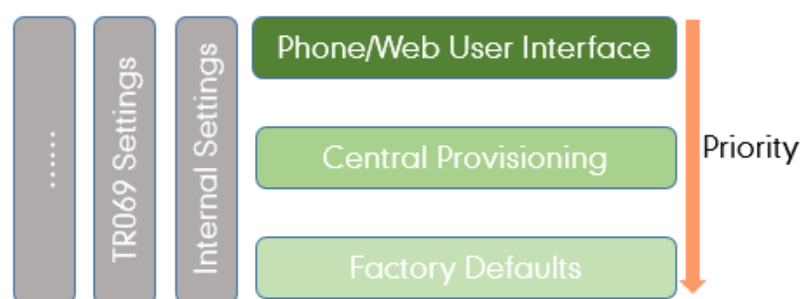
previous settings.

## Provisioning Methods

IP phones can be configured automatically through configuration files stored on a central provisioning server, manually via web user interface or phone user interface, or by a combination of the automatic and manual methods. If a central provisioning server is not available, you can configure most features using manual method.

There may be a configuration priority among the provisioning methods - settings you make using a higher priority provisioning method override settings made using a lower priority provisioning method.

The precedence order for configuration parameter changes is as follows (highest to lowest):



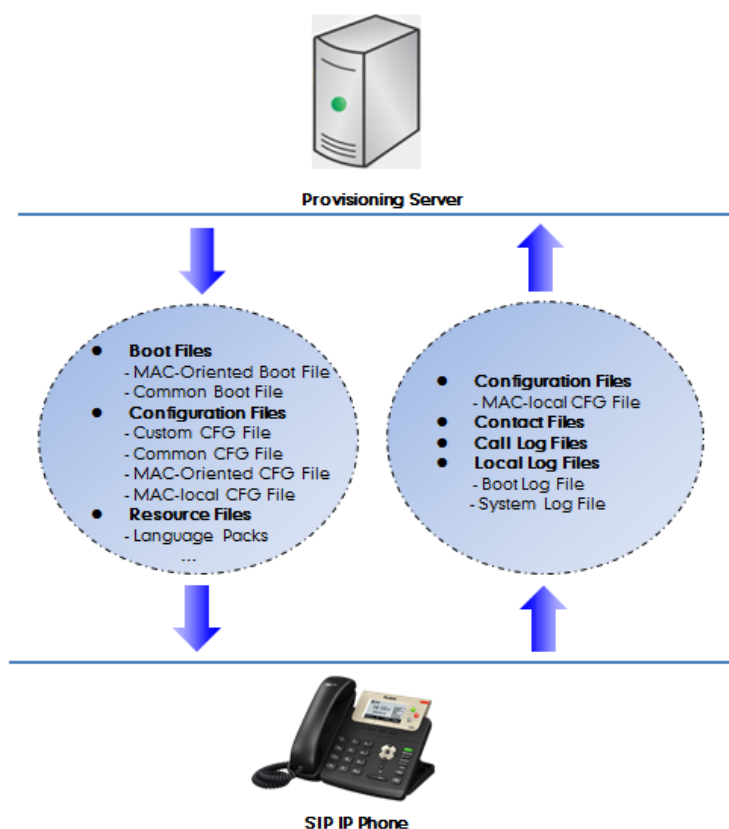
### Note

The priority mechanism takes effect only if the value of the parameter "static.auto\_provision.custom.protect" is set to 1. For more information on this parameter, refer to [Configuration Parameters](#) on page 122.

Static settings have no priority. For example, settings associated with auto provisioning/network/syslog, TR069 settings and internal settings (e.g., the temporary configurations to be used for program running). For more information, refer to [Appendix F: Static Settings](#) on page 898.

## Central Provisioning

The following figure shows how the phone interoperates with provisioning server when you use the centralized provisioning method:



Using the boot files and configuration files to provision the phones and to modify features and configurations is called the central provisioning method. You can use a text-based editing application to edit boot files and configuration files, and then store boot files and configuration files to a provisioning server. IP phones can be centrally provisioned from a provisioning server. For more information on the provisioning server, refer to [Setting Up a Provisioning Server](#) on page 108. For more information on boot files, refer to [Boot Files](#) on page 99. For more information on configuration files, refer to [Configuration Files](#) on page 101.

IP phones can obtain the provisioning server address during startup. Then IP phones download boot files and configuration files from the provisioning server, resolve and update the configurations written in configuration files. This entire process is called auto provisioning. For more information on auto provisioning, refer to [Yealink SIP IP Phones Auto Provisioning Guide V81](#). In addition to the boot files and configuration files, the IP phones also download resource files during auto provisioning. For more information on resource files, refer to [Resource Files](#) on page 103.

Yealink IP phones support keeping user personalized configuration settings using the MAC-local CFG file. For more information on this file, refer to [MAC-local CFG File](#) on page 102.

The IP phones can be configured to upload log files (log files provide a history of phone events),

call log files and contact files to the provisioning server. You can configure a separate directory for each of these files to help organize: a log file directory, a call log file directory and a contact file directory. For more information, refer to [Viewing Log Files](#) on page 827, [Backing up the Call Log](#) on page 308 and [Backing up the Local Contacts](#) on page 332.

## Manual Provisioning

When you manually configure a phone via web user interface or phone user interface, the changes associated with non-static settings you make will be stored in the MAC-local CFG file. For more information on MAC-local CFG file, refer to [MAC-local CFG File](#) on page 102. This file is stored on the phone, but a copy can be also uploaded to the provisioning server or a specific URL (if configured).

There are two ways to manually provision IP phones:

- [Web User Interface](#)
- [Phone User Interface](#)

### Web User Interface

You can configure IP phones via web user interface, a web-based interface that is especially useful for remote configuration. Because features and configurations vary by phone model and firmware version, options available on each page of the web user interface can vary.

An administrator or a user can configure IP phones via web user interface; but accessing the web user interface requires password. The default user name and password for the administrator are both "admin" (case-sensitive). The default user name and password for the user are both "user" (case-sensitive). For more information on configuring passwords, refer to [User and Administrator Passwords](#) on page 791.

This method enables you to perform configuration changes on a per-phone basis. Note that the features configured via web user interface are limited. So, you can use the web user interface method as the sole configuration method or in conjunction with central provisioning method and phone user interface method.

IP phones support both HTTP and HTTPS protocols for accessing the web user interface. For more information, refer to [Web Server Type](#) on page 28.

### Phone User Interface

You can configure IP phones via phone user interface on a per-phone basis. As with the web user interface, phone user interface makes configurations available to users and administrators; but the **Advanced/Advanced Settings** option is only available to administrators and requires an administrator password (default: admin). For more information on configuring password, refer to [User and Administrator Passwords](#) on page 791.

If you want to reset all settings made from the phone user interface to default, refer to [Yealink phone-specific user guide](#).

## Boot Files, Configuration Files and Resource Files

When IP phones are configured with central provisioning method, they will request to download the boot files, configuration files and resource files from the provisioning server.

The following sections describe the details of boot files, configuration files and resource files:

- [Boot Files](#)
- [Configuration Files](#)
- [Resource Files](#)
- [Obtaining Boot Files/Configuration Files/Resource Files](#)

### Boot Files

Yealink IP phones running firmware version 81 or later support a new boot file in which you can customize the download sequence of configuration files. It is efficiently for you to provision your IP phones in different deployment scenarios, especially when you want to apply a set of features or settings to a group of phones.

#### Note

You can select whether to use the boot file or not for auto provisioning according to your deployment scenario. If you do not use the boot file, proceed to [Configuration Files](#) on page 101. That is, you can also use the old mechanism for auto provisioning.

The boot files are valid BOOT files that can be created or edited using a text editor such as UltraEdit. The boot files are first downloaded when you provision the phones using centralized provisioning (refer to [Central Provisioning](#)). The configuration parameters are not included in the boot file. You can reference some configuration files that contain parameters in the boot files to be acquired by all your phones and specify the download sequence of these configuration files.

Yealink supports two types of boot files: common boot file and MAC-Oriented boot file.

During auto provisioning, the IP phone first tries to download the MAC-Oriented boot file (refer to [MAC-Oriented Boot File](#)), and then download configuration files referenced in the MAC-Oriented boot file in sequence from the provisioning server. If no matched MAC-Oriented boot file is found, the IP phone tries to download the common boot file (refer to [Common Boot File](#)) and then downloads configuration files referenced in the common boot file in sequence. If no common boot file is found, the IP phone downloads the common CFG file (refer to [Common CFG File](#)) and MAC-Oriented CFG file (refer to [MAC-Oriented CFG File](#)) in sequence.

The following figure shows an example of common boot file:

```
#!version:1.0.0.1
#The header above must appear as-is in the first line
include:config <configure/sip.cfg>
```

```
include:config "http://10.2.5.206/configure/account.cfg"
overwrite_mode = 1
```

**Learn the following:**

- The line beginning with “#” is considered to be a comment.
- The file header “#!version:1.0.0.1” is not a comment and must be placed in the first line. It cannot be edited or deleted.
- Each “include” statement can reference a configuration file. The referenced configuration file format must be \*.cfg.
- The contents in the angle brackets or double quotation marks represent the download paths of the referenced configuration files (for example, http://10.2.5.206/configure/account.cfg). The download path must point to a specific CFG file. The sip.cfg and account.cfg are the specified configuration files to be downloaded during auto provisioning.
- The CFG files are downloaded in the order listed (top to bottom).

The IP phone downloads the boot file first, and then downloads the sip.cfg and account.cfg configuration files from the “configure” directory on the provisioning server in sequence. The parameters in the new downloaded configuration files will override the duplicate parameters in files downloaded earlier.

- “overwrite\_mode = 1” means overwrite mode is enabled. The overwrite mode will be applied to the configuration files specified to download. If the value of a parameter in configuration files is left blank or a parameter in configuration files is deleted or commented out, the factory default value can take effect immediately after auto provisioning.

**Note**

Overwrite mode only affects the non-static settings configured using configuration file. If you do not use the boot file for auto provisioning, overwrite mode is disabled by default and you are not allowed to enable it.

For more information on how to customize boot file, refer to

[Yealink\\_SIP\\_IP\\_Phones\\_Auto\\_Provisioning\\_Guide\\_V81](#).

**Common Boot File**

Common boot file, named y000000000000.boot, is effectual for all phones.

**MAC-Oriented Boot File**

MAC-Oriented boot file, named <MAC>.boot. It will only be effectual for a specific IP phone. The MAC-Oriented boot file should be created using template boot file in advance.

The MAC-Oriented boot file is named after the MAC address of the IP phone. MAC address, a

unique 12-digit serial number assigned to each phone, can be obtained from the bar code on the back of the IP phone. For example, if the MAC address of an IP phone is 00156574B150, the name of the MAC-Oriented boot file is 00156574b150.boot (lowercase).

## Configuration Files

The configuration files are valid CFG files that can be created or edited using a text editor such as UltraEdit. An administrator can deploy and maintain a mass of Yealink IP phones automatically through configuration files stored on a provisioning server.

Yealink configuration files consist of:

- [Common CFG File](#)
- [MAC-Oriented CFG File](#)
- [MAC-local CFG File](#)
- [Custom CFG File](#)

### Common CFG File

Common CFG file, named <y0000000000xx>.cfg, contains parameters that affect the basic operation of the IP phone, such as language and volume. It will be effectual for all IP phones of the same model. The common CFG file has a fixed name.

The following table lists the name of the common CFG file for each IP phone model:

IP Phone Model	Common CFG file
SIP-T54S	y000000000070.cfg
SIP-T52S	y000000000074.cfg
SIP-T48G	y000000000035.cfg
SIP-T48S	y000000000065.cfg
SIP-T46G	y000000000028.cfg
SIP-T46S	y000000000066.cfg
SIP-T42G	y000000000029.cfg
SIP-T42S	y000000000067.cfg
SIP-T41P	y000000000036.cfg
SIP-T41S	y000000000068.cfg
SIP-T40P	y000000000054.cfg
SIP-T40G	y000000000076.cfg
SIP-T29G	y000000000046.cfg

IP Phone Model	Common CFG file
SIP-T27P	y000000000045.cfg
SIP-T27G	y000000000069.cfg
SIP-T23P/G	y000000000044.cfg
SIP-T21(P) E2	y000000000052.cfg
SIP-T19(P) E2	y000000000053.cfg

## MAC-Oriented CFG File

MAC-Oriented CFG file, named <MAC>.cfg, contains parameters unique to a particular phone, such as account registration. It will only be effectual for a specific IP phone.

The MAC-Oriented CFG file is named after the MAC address of the IP phone. MAC address, a unique 12-digit serial number assigned to each phone, can be obtained from the bar code on the back of the IP phone. For example, if the MAC address of an IP phone is 00156574B150, the name of the MAC-Oriented CFG file is 00156574b150.cfg (lowercase).

## MAC-local CFG File

MAC-local CFG file, named <MAC>-local.cfg, contains changes associated with non-static settings that users make via web user interface and phone user interface (for example, updates to time and date formats, ring tones, dial plan and DSS keys). This file generates only if the value of the parameter "static.auto\_provision.custom.protect" is set to 1.

The MAC-local CFG file is also named after the MAC address (the bar code label on the back of the IP phone or on the outside of the box) of the IP phone. For example, if the MAC address of an IP phone is 00156574B150, the name of the MAC-local CFG file is 00156574b150-local.cfg (lowercase).

### Note

After the provisioning priority mechanism is enabled (configured by the parameter "static.auto\_provision.custom.protect"), all older changes made via web/phone user interface will not be saved in the <MAC>-local.cfg file. But the older settings still take effect on the phone. For more information on this parameter, refer to [Configuration Parameters](#) on page 122.

## Keeping User Personalized Settings

The MAC-local CFG file is stored locally on the IP phone and can also be uploaded to the provisioning server/a specific URL (if configured, refer to [Configuration Parameters](#)). This file enables users to keep their personalized configuration settings, even though the IP phone reboots or upgrades. For more information on how to keep user personalized settings, refer to [Keeping User Personalized Settings after Auto Provisioning](#) on page 121.



Users can also select to clear the user personalized configuration settings. Users can clear the MAC-local CFG file using the following methods:

- To clear the MAC-local CFG file, reset the IP phone to factory configuration settings by selecting **Reset local settings** via phone user interface (navigate to **Menu->Advanced** (default password: admin) -> **Reset Config**).
- To clear the MAC-local CFG file, reset the IP phone to factory configuration settings by navigating to the **Upgrade** menu via web user interface and clicking **Reset local setting**.

### Configurations defined never be saved to the <MAC>-local.cfg file

Most configurations made by users via phone user interface and web user interface can be saved to the <MAC>-local.cfg file, but some static settings will never be saved to the <MAC>-local.cfg file. For more information, refer to [Appendix F: Static Settings](#) on page 898.

You need to reset the phone configurations not saved in the <MAC>-local.cfg file separately. For more information, refer to [Resetting Issues](#) on page 857.

By default, the MAC-local.cfg file will be stored on the IP phone. The IP phone can be configured to upload this file to the provisioning server each time the file updates. For more information, refer to the parameter "static.auto\_provision.custom.sync" described in the section [Configuration Parameters](#) on page 122.

## Custom CFG File

You can create some new CFG files (for example, sip.cfg, account.cfg) containing any combination of configuration parameters. This especially useful when you want to apply a set of features or settings to a group of phones using the boot file.

For more information on how to create a new CFG file, refer to [Yealink\\_SIP\\_IP\\_Phones\\_Auto\\_Provisioning\\_Guide\\_V81](#).

## Resource Files

When configuring some particular features, you may need to upload resource files to IP phones. Resource files are optional, but if the particular feature is being employed, these files are required.

If the resource file is to be used for all IP phones of the same model, the access URL of resource file is best specified in the common CFG file. However, if you want to specify the desired phone to use the resource file, the access URL of resource file should be specified in the MAC-Oriented CFG file. During provisioning, the IP phones will request the resource files in addition to the configuration files. For more information on the access URL of resource file, refer to the corresponding section in this guide.

The followings show examples of resource files:

- Language packs
- Ring tones

- Local contact file
- Input method file

For more information on resource files, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

If you want to delete resource files from a phone at a later date - for example, if you are giving the phone to a new user - you can reset the IP phone to factory configuration settings. For more information, refer to [Resetting Issues](#) on page 857.

## Obtaining Boot Files/Configuration Files/Resource Files

Yealink supplies some template configuration files and resource files for you, so you can directly edit and customize the files as required. You can ask the distributor or Yealink FAE for template files. You can also obtain the template files online: <http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>.

The names of the Yealink-supplied template files are:

Template File		File Name	Description
Boot File		y000000000000.boot	Allows you to customize the download sequence of the configuration files during auto provisioning. For more information, refer to <a href="#">Boot Files</a> on page 99.
Configuration Files	Common CFG File	<Common>.cfg	Allow you to deploy and maintain a mass of Yealink IP phones. For more information, refer to <a href="#">Common CFG File</a> and <a href="#">MAC-Oriented CFG File</a> on page 102.
	MAC-Oriented CFG File	<MAC>.cfg	
	Custom CFG Files	For example, sip.cfg account.cfg	Allow you to apply a set of features or settings to a group of Yealink IP phones. For more information, refer to <a href="#">Custom CFG File</a> on page 103.
Resource Files	AutoDST Template	AutoDST.xml	Allows you to add or modify time zone and DST settings for your area. For more information, refer to <a href="#">Customizing an AutoDST Template File</a> on page 228.

Template File		File Name	Description
	Language Packs	For example, 000.GUI.English.lang 1.English_note.xml 1.English.js	Allow you to customize the translation of the existing language on the phone/web user interface. For more information, refer to <a href="#">Loading Language Packs</a> on page 232.
	Keypad Input Method File	ime.txt Russian_ime.txt Hebrew_ime.txt	Allow you to customize the existing input method on Yealink IP phones. For more information, refer to <a href="#">Keypad Input Method Customization</a> on page 242.
	Replace Rule Template	dialplan.xml	Allows you to customize multiple replace rules for IP phone dial plan. For more information, refer to <a href="#">Customizing Replace Rule Template File</a> on page 269.
	Dial Now Template	dialnow.xml	Allows you to customize multiple dial now rules for IP phone dial plan. For more information, refer to <a href="#">Customizing Dial Now Template File</a> on page 275.
	Softkey Layout Template	CallFailed.xml CallIn.xml Connecting.xml Dialing.xml (not applicable to SIP-T48G/S IP phones) RingBack.xml Talking.xml	Allow you to customize soft key layout for different call states. For more information, refer to <a href="#">Customizing Softkey Layout Template File</a> on page 253.
	Directory Template	favorite_setting.xml	Allows you to customize the directory list for your IP phone. For more

Template File		File Name	Description
			information, refer to <a href="#">Customizing a Directory Template File</a> on page 299.
	Super Search Template	super_search.xml	Allows you to customize the search source list for your IP phone. For more information, refer to <a href="#">Customizing a Super Search Template File</a> on page 303.
	Local Contact File	contact.xml	Allows you to add or modify multiple contacts at a time for your IP phone. For more information, refer to <a href="#">Customizing a Local Contact File</a> on page 315.
	Remote Phone Book Template	Department.xml Menu.xml	Allows you to add or modify multiple remote contacts for your IP phone. For more information, refer to <a href="#">Customizing Remote Phone Book Template File</a> on page 512.
	Screen Saver Template	CustomScreenSaver.xml	Allows you to customize the XML items (for example, notifications, company logo) which will be displayed on screen saver. For more information, refer to <a href="#">Customizing a Screen Saver XML Template File</a> on page 153.

**To download template files:**

1. Go to Yealink [Document Download](#) page and select the desired phone model.
2. Download and extract the combined files to your local system.

For example, the following illustration shows the template files available for SIP-T46G IP phones running firmware version 81.

<b>User Documents</b>	Yealink_SIP-T46G_User_Guide_V81_20.pdf
	Yealink_SIP-T2_Series_T19(P) E2_T4_Series_IP_Phones_Administrator_Guide_V81_20.pdf
	Yealink_SIP-T2_Series_T19(P) E2_T4_Series_CP860_IP_Phones_XML_Browser_Developer's_Guide_V81_20.pdf
	Yealink_SIP-T2_Series_T19(P) E2_T4_Series_IP_Phones_Auto_Provisioning_Guide_V81_20.pdf
	<b>Yealink_SIP_Phones_AutoProvisioning_Template_V81_20.zip</b>
	Yealink_IP_Phones_Configuration_Guide_for_User_Access_Level_V81_20.pdf
	Yealink_IP_Phones_Description_of_Configuration_Parameters_in_CFG_Files_V81_20.zip
	Yealink_Phonebook_Generation_Tool_V81_20.zip
	Yealink_Configuration_Encryption_Tool_User_Guide_V81_20.pdf

3. Open the folder you extracted and identify the template file you will edit according to the table introduced above.

For some features, you can customize the filename as required. The following table lists the special characters supported by Yealink IP phones:

<b>Platform</b>	<b>Server</b>	<b>HTTP/HTTPS</b>	<b>TFTP/FTP</b>
<b>Windows</b>		<b>Support:</b> ~ ` ! @ \$ ^ ( ) _ - , . ' ; [ ] { } (including space) <b>Not Support:</b>   < > : " / \ * ? # % & = +	<b>Support:</b> ~ ` ! @ \$ ^ ( ) _ - , . ' ; [ ] { } % & = + (including space) <b>Not Support:</b>   < > : " / \ * ? #
<b>Linux</b>		<b>Support:</b> ~ ` ! @ \$ ^ ( ) _ - , . ' ; [ ] { }   < > : " (including space) <b>Not Support:</b> / \ * ? # % & = +	<b>Support:</b> ~ ` ! @ \$ ^ ( ) _ - , . ' ; [ ] { }   < > : " % & = + (including space) <b>Not Support:</b> / \ * ? #

## Setting Up a Provisioning Server

This chapter provides basic instructions for setting up a provisioning server and deploying phones from the provisioning server.

This chapter consists of the following sections:

- [Why Using a Provisioning Server?](#)

- [Supported Provisioning Protocols](#)
- [Configuring a Provisioning Server](#)
- [Deploying Phones from the Provisioning Server](#)

## Why Using a Provisioning Server?

You can use a provisioning server to configure your IP phones. A provisioning server allows for flexibility in upgrading, maintaining and configuring the phone. Boot files, configuration files and resource files are normally located on this server.

When IP phones are triggered to perform auto provisioning, it will request to download the boot files and configuration files from the provisioning server. During the auto provisioning process, the IP phone will download and update configuration files to the phone flash. For more information on auto provisioning, refer to [Yealink SIP IP Phones Auto Provisioning Guide V81](#).

The IP phones can be configured to periodically upload the log files to the provisioning server or specific server, which can help an administrator more easily find the system problem and fix it. For more information on log files, refer to [Viewing Log Files](#) on page 827. The IP phones can also be configured to upload the contact file to the provisioning server or specific server, avoiding the loss of the local contacts. For more information, refer to [Backing up the Local Contacts](#) on page 332.

## Supported Provisioning Protocols

IP phones perform the auto provisioning function of uploading log files (if configured), uploading contact files (if configured), downloading boot files, downloading configuration files, downloading resource files and upgrading firmware. The transfer protocol is used to download files from the provisioning server. IP phones support several transport protocols for provisioning, including FTP, TFTP, HTTP, and HTTPS protocols. And you can specify the transport protocol in the provisioning server address, for example, `http://xxxxxxx`. If not specified, the TFTP protocol is used. The provisioning server address can be IP address, domain name or URL. If a user name and password are specified as part of the provisioning server address, for example, `http://user:pwd@server/dir`, they will be used only if the server supports them.

### Note

A URL should contain forward slashes instead of back slashes and should not contain spaces. Escape characters are not supported.

If a user name and password are not specified as part of the provisioning server address, the User Name and Password of the provisioning server configured on the phone will be used.

There are two types of FTP methods—active and passive. IP phones are not compatible with active FTP.

## Configuring a Provisioning Server

The provisioning server can be set up on the local LAN or anywhere on the Internet. Use the following procedure as a recommendation if this is your first provisioning server setup. For more

information on how to set up a provisioning server, refer to [Yealink\\_SIP\\_IP\\_Phones\\_Auto\\_Provisioning\\_Guide\\_V81](#).

**To set up the provisioning server:**

1. Install a provisioning server application or locate a suitable existing server.
2. Create an account and home directory.
3. Set security permissions for the account.
4. Create boot files and then edit them as desired.
5. Create configuration files and then edit them as desired.
6. Copy the boot files, configuration files and resource files to the provisioning server.

For more information on how to deploy IP phones using boot files and configuration files, refer to [Deploying Phones from the Provisioning Server](#) on page 110.

**Note**

Typically all phones are configured with the same server account, but the server account provides a means of conveniently partitioning the configuration. Give each account a unique home directory on the server and change the configuration on a per-line basis.

## Deploying Phones from the Provisioning Server

During auto provisioning, IP phones download the boot file first, and then download the configuration files referenced in the boot file in sequence. The parameters in the new downloaded configuration files will override the duplicate parameters in files downloaded earlier. For more information on boot files and configuration files, refer to [Boot Files](#) on page 99 and [Configuration Files](#) on page 101.

The boot files can only be used by the IP phones running firmware version 81 or later. The configuration files, supplied with each firmware release, must be used with that release. Otherwise, configurations may not take effect, and the IP phone will behave without exception. Before you configure parameters in the configuration files, Yealink recommends that you create new configuration files containing only those parameters that require changes.

**To deploy IP phones from the provisioning server:**

1. Create per-phone boot files by performing the following steps:
  - a) Obtain a list of phone MAC addresses (the bar code label on the back of the IP phone or on the outside of the box).
  - b) Create per-phone <MAC>.boot files by using the template boot file.
  - c) Specify the configuration files paths in the file as desired.
2. Edit the common boot file by performing the following step:
  - a) Specify the configuration files paths in the file as desired.
3. Create per-phone configuration files by performing the following steps:
  - a) Create per-phone <MAC>.cfg files by using the MAC-Oriented CFG file from the distribution as templates.



- b) Edit the parameters in the file as desired.
- 4. Create new common configuration files by performing the following steps:
  - a) Create <y0000000000xx>.cfg files by using the Common CFG file from the distribution as templates.
  - b) Edit the parameters in the file as desired.
- 5. Copy boot files and configuration files to the home directory of the provisioning server.
- 6. Reboot IP phones to trigger the auto provisioning process.

IP phones discover the provisioning server address, and then download the boot files and configuration files from the provisioning server.

For protecting against unauthorized access, you can encrypt configuration files. For more information on encrypting configuration files, refer to [Encrypting and Decrypting Files](#) on page 814.

#### Note

During auto provisioning, the IP phone tries to download the MAC-Oriented boot file first. If no matched MAC-Oriented boot file is found on the server, the IP phone tries to download the common boot file. If the MAC-Oriented boot file and common boot file exist simultaneously on the provisioning server, the common boot file will be ignored after the IP phone successfully downloads the matched MAC-Oriented boot file.

During the auto provisioning process, the IP phone supports the following methods to discover the provisioning server address:

- **Zero Touch:** Zero Touch feature guides you to configure network settings and the provisioning server address via phone user interface after startup.
- **PnP:** PnP feature allows IP phones to discover the provisioning server address by broadcasting the PnP SUBSCRIBE message during startup.
- **DHCP:** DHCP option can be used to provide the address or URL of the provisioning server to IP phones. When the IP phone requests an IP address using the DHCP protocol, the resulting response may contain option 66 or the custom option (if configured) that contains the provisioning server address.
- **Static:** You can manually configure the server address via phone user interface or web user interface.

For more information on the above methods, refer to

[Yealink\\_SIP\\_IP\\_Phones\\_Auto\\_Provisioning\\_Guide\\_V81](#).

## Upgrading Firmware

This section provides information on upgrading the IP phone firmware. Two methods of firmware upgrade:

- Manually, from the local system for a single phone.

- Automatically, from the provisioning server for a mass of phones.

The following table lists the associated and latest firmware name for each IP phone model (X is replaced by the actual firmware version).

IP Phone Model	Associated Firmware Name	Firmware Name Example
SIP-T54S/T52S (T5S firmware unified)	70.x.x.x.rom	70.82.0.20.rom
SIP-T48S/T46S/T42S/T41S (T4S firmware unified)	66.x.x.x.rom	66.82.0.20.rom
SIP-T48G	35.x.x.x.rom	35.82.0.20.rom
SIP-T46G	28.x.x.x.rom	28.82.0.20.rom
SIP-T42G	29.x.x.x.rom	29.82.0.20.rom
SIP-T41P	36.x.x.x.rom	36.82.0.20.rom
SIP-T40P	54.x.x.x.rom	54.82.0.20.rom
SIP-T40G	76.x.x.x.rom	76.82.0.20.rom
SIP-T29G	46.x.x.x.rom	46.82.0.20.rom
SIP-T27P	45.x.x.x.rom	45.82.0.20.rom
SIP-T27G	69.x.x.x.rom	69.82.0.20.rom
SIP-T23P/G	44.x.x.x.rom	44.82.0.20.rom
SIP-T21(P) E2	52.x.x.x.rom	52.82.0.20.rom
SIP-T19(P) E2	53.x.x.x.rom	53.82.0.20.rom

#### Note

You can download the latest firmware online:  
<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>.

Do not unplug the network and power cables when the IP phone is upgrading firmware.

## Upgrading Firmware from the Provisioning Server

IP phones support using FTP, TFTP, HTTP and HTTPS protocols to download configuration files and firmware from the provisioning server, and then upgrade firmware automatically.

IP phones can download firmware stored on the provisioning server in one of two ways:

- Check for configuration files and then download firmware during startup.
- Automatically check for configuration files and then download firmware at a fixed interval or specific time.

Method of checking for configuration files is configurable.

## Procedure

Configuration changes can be performed using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	<p>Configure the way for the IP phone to check for configuration files.</p> <p><b>Parameters:</b></p> <p>static.auto_provision.power_on</p> <p>static.auto_provision.repeat.enable</p> <p>static.auto_provision.repeat.minutes</p> <p>static.auto_provision.weekly.enable</p> <p>static.auto_provision.weekly_upgrade_interval</p> <p>static.auto_provision.inactivity_time_expire</p> <p>static.auto_provision.weekly.begin_time</p> <p>static.auto_provision.weekly.end_time</p> <p>static.auto_provision.weekly.dayofweek</p> <p>static.auto_provision.flexible.enable</p> <p>static.auto_provision.flexible.interval</p> <p>static.auto_provision.flexible.begin_time</p> <p>static.auto_provision.flexible.end_time</p> <p>static.auto_provision.prompt.enable</p>
		<p>Specify the access URL of firmware.</p> <p><b>Parameter:</b></p> <p>static.firmware.url</p>
<b>Web User Interface</b>		<p>Configure the way for the IP phone to check for configuration files.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=settings-autop&amp;q=load</p>
		<p>Upgrade firmware.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=settings-upgrade&amp;q=load</p>

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
static.auto_provision.power_on	0 or 1	1

Parameters	Permitted Values	Default
<b>Description:</b> Triggers the power on feature to on or off. <b>0</b> -Off <b>1</b> -On, the IP phone will perform an auto provisioning process when powered on. <b>Web User Interface:</b> Settings->Auto Provision->Power On <b>Phone User Interface:</b> None		
<b>static.auto_provision.repeat.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Triggers the repeatedly feature to on or off. <b>0</b> -Off <b>1</b> -On <b>Web User Interface:</b> Settings->Auto Provision->Repeatedly <b>Phone User Interface:</b> None		
<b>static.auto_provision.repeat.minutes</b>	<b>Integer from 1 to 43200</b>	<b>1440</b>
<b>Description:</b> Configures the interval (in minutes) for the IP phone to perform an auto provisioning process repeatedly. <b>Note:</b> It works only if the value of the parameter "static.auto_provision.repeat.enable" is set to 1 (On). <b>Web User Interface:</b> Settings->Auto Provision->Interval(Minutes) <b>Phone User Interface:</b> None		
<b>static.auto_provision.weekly.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Triggers the weekly feature to on or off. <b>0</b> -Off		

Parameters	Permitted Values	Default
<p><b>1-On</b>, the IP phone will perform an auto provisioning process weekly.</p> <p><b>Web User Interface:</b> Settings-&gt;Auto Provision-&gt;Weekly</p> <p><b>Phone User Interface:</b> None</p>		
<b>static.auto_provision.weekly_upgrade_interval</b>	<b>Integer from 0 to 12</b>	<b>0</b>
<p><b>Description:</b> Configures the time interval (in weeks) for the IP phone to perform an auto provisioning. If it is set to 0, the IP phone will perform an auto provisioning process at the specific day(s) configured by the parameter "static.auto_provision.weekly.dayofweek" every week. If it is set to other values (for example, 3), the IP phone will perform an auto provisioning process at a random day between the specific day(s) configured by the parameter "static.auto_provision.weekly.dayofweek" every three weeks.</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.weekly.enable" is set to 1 (On).</p> <p><b>Web User Interface:</b> Settings-&gt;Auto Provision-&gt;Weekly Upgrade Interval(0~12week)</p> <p><b>Phone User Interface:</b> None</p>		
<b>static.auto_provision.inactivity_time_expire</b>	<b>Integer from 0 to 120</b>	<b>0</b>
<p><b>Description:</b> Configures the delay time (in minutes) to perform an auto provisioning process when the IP phone is inactive at regular week. If it is set to 0, the IP phone will perform an auto provisioning process at random between a starting time configured by the parameter "static.auto_provision.weekly.begin_time" and an ending time configured by the parameter "static.auto_provision.weekly.end_time". If it is set to other values (for example, 60), the IP phone will perform an auto provisioning process only when the IP phone has been inactivated for 60 minutes (1 hour) between the starting time and ending time.</p> <p><b>Note:</b> The IP phone may perform an auto provisioning process when you are using the IP phone on office hour. It works only if the value of the parameter "static.auto_provision.weekly.enable" is set to 1 (On).</p> <p><b>Web User Interface:</b> Settings-&gt;Auto Provision-&gt;Inactivity Time Expire(0~120min)</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>static.auto_provision.weekly.begin_time</b>	Time from 00:00 to 23:59	00:00
<b>Description:</b> Configures the starting time of the day for the IP phone to perform an auto provisioning process weekly. <b>Note:</b> It works only if the value of the parameter "static.auto_provision.weekly.enable" is set to 1 (On). <b>Web User Interface:</b> Settings->Auto Provision->Time <b>Phone User Interface:</b> None		
<b>static.auto_provision.weekly.end_time</b>	Time from 00:00 to 23:59	00:00
<b>Description:</b> Configures the ending time of the day for the IP phone to perform an auto provisioning process weekly. <b>Note:</b> It works only if the value of the parameter "static.auto_provision.weekly.enable" is set to 1 (On). <b>Web User Interface:</b> Settings->Auto Provision->Time <b>Phone User Interface:</b> None		
<b>static.auto_provision.weekly.dayofweek</b>	0, 1, 2, 3, 4, 5, 6 or a combination of these digits	0123456
<b>Description:</b> Configures the days of the week for the IP phone to perform an auto provisioning process weekly. <b>0</b> -Sunday <b>1</b> -Monday <b>2</b> -Tuesday <b>3</b> -Wednesday		

Parameters	Permitted Values	Default
<p><b>4</b>-Thursday</p> <p><b>5</b>-Friday</p> <p><b>6</b>-Saturday</p> <p><b>Example:</b></p> <p>static.auto_provision.weekly.dayofweek = 01</p> <p>If the value of the parameter "static.auto_provision.weekly_upgrade_interval" is set to 0, it means the IP phone will perform an auto provisioning process every Sunday and Monday.</p> <p>If the value of the parameter "static.auto_provision.weekly_upgrade_interval" is set to other value (for example, 3), it means the IP phone will perform an auto provisioning process by randomly selecting a day from Sunday and Monday every three weeks.</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.weekly.enable" is set to 1 (On).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Auto Provision-&gt;Day of Week</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.flexible.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Triggers the flexible feature to on or off.</p> <p><b>0</b>-Off</p> <p><b>1</b>-On, the IP phone will perform an auto provisioning process at random between a starting time configured by the parameter "static.auto_provision.flexible.begin_time" and an ending time configured by the parameter "static.auto_provision.flexible.end_time" on a random day within the period configured by the parameter "static.auto_provision.flexible.interval".</p> <p><b>Note:</b> The day within the period is based upon the phone's MAC address and does not change with a reboot whereas the time within the start and end is calculated again with every reboot.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Auto Provision-&gt;Flexible Auto Provision</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.flexible.interval</b>	<b>Integer from 1 to 1000</b>	<b>30</b>
<p><b>Description:</b></p> <p>Configures the interval (in days) for the IP phone to perform an auto provisioning process.</p>		

Parameters	Permitted Values	Default
<p>The auto provisioning occurs on a random day within this period based on the phone's MAC address.</p> <p><b>Example:</b></p> <p>static.auto_provision.flexible.interval = 30</p> <p>The IP phone will perform an auto provisioning process on a random day (for example, 18) based on the phone's MAC address.</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.flexible.enable" is set to 1 (On).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Auto Provision-&gt;Flexible Interval Days</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.flexible.begin_time</b>	<b>Time from 00:00 to 23:59</b>	<b>02:00</b>
<p><b>Description:</b></p> <p>Configures the starting time of the day for the IP phone to perform an auto provisioning process at random.</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.flexible.enable" is set to 1 (On).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Auto Provision-&gt;Flexible Time</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.flexible.end_time</b>	<b>Time from 00:00 to 23:59</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the ending time of the day for the IP phone to perform an auto provisioning process at random.</p> <p>If it is left blank or set to a specific value equal to starting time configured by the parameter "static.auto_provision.weekly.begin_time", the IP phone will perform an auto provisioning process at the starting time.</p> <p>If it is set to a specific value greater than starting time configured by the parameter "static.auto_provision.weekly.begin_time", the IP phone will perform an auto provisioning process at random between the starting time and ending time.</p> <p>If it is set to a specific value less than starting time configured by the parameter "static.auto_provision.weekly.begin_time", the IP phone will perform an auto provisioning</p>		



Parameters	Permitted Values	Default
<p>process at random between the starting time on that day and ending time in the next day.</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.flexible.enable" is set to 1 (On).</p> <p><b>Web User Interface:</b> Settings-&gt;Auto Provision-&gt;Flexible Time</p> <p><b>Phone User Interface:</b> None</p>		
<b>static.auto_provision.prompt.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the IP phone to prompt you for the configuration update and the result (if any configuration changes) during auto provisioning.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> If the IP phone performs the auto provision when receiving a SIP NOTIFY message which contains the header "Event: check-sync", the IP phone will display the prompt message no matter whether the configuration is updated.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>static.firmware.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of the firmware file.</p> <p><b>Example:</b> static.firmware.url = http://192.168.1.20/28.82.0.20.rom</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Settings-&gt;Upgrade-&gt;Select And Upgrade Firmware</p> <p><b>Phone User Interface:</b> None</p>		

**To configure the way for the IP phone to check for configuration files via web user interface:**

1. Click on **Settings->Auto Provision**.

2. Make the desired change.

3. Click **Confirm** to accept the change.

When the "Power On" is set to **On**, the IP phone will check boot files and configuration files stored on the provisioning server during startup and then will download firmware from the server.

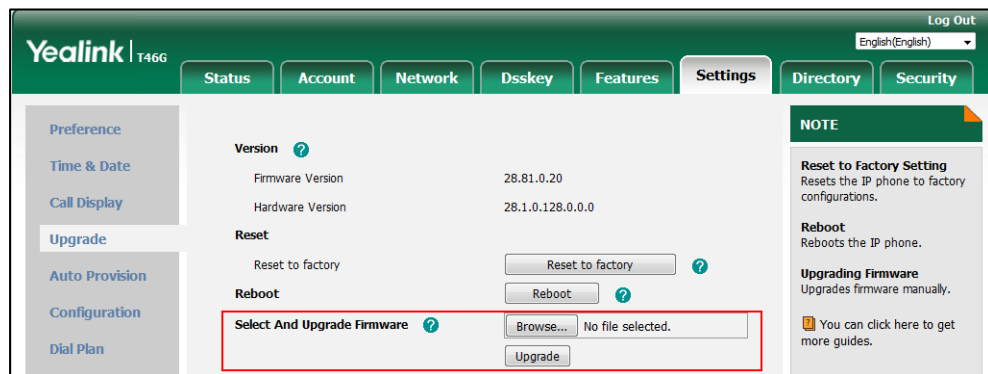
## Upgrading Firmware via Web User Interface

To manually upgrade firmware via web user interface, you need to store firmware to your local system in advance.

### To upgrade firmware manually via web user interface:

1. Click on **Settings->Upgrade**.
2. Click **Browse** to locate the required firmware from your local system.

3. Click **Upgrade**.



A dialog box pops up to prompt "Firmware of the SIP Phone will be updated. It will take 5 minutes to complete. Please don't power off!".

4. Click **OK** to confirm the upgrade.

**Note**

Do not close and refresh the browser when the IP phone is upgrading firmware via web user interface.

## Keeping User Personalized Settings after Auto Provisioning

Generally, the administrator deploys phones in batch and timely maintains company phones via auto provisioning, yet some users would like to keep the personalized settings (for example, ring tones, wallpaper, dial plan, time format or DSS keys), after auto provisioning. These specific scenarios are applicable to Yealink IP phones. The following demonstrated specific scenarios are taking SIP-T46G IP phones as example for reference.

**Note**

Yealink IP phones support FTP, TFTP, HTTP and HTTPS protocols for uploading the <MAC>-local.cfg file. This section takes the TFTP protocol as an example. Before performing the following, make sure the provisioning server supports uploading.

If you are using the HTTP/HTTPS server, you can specify the way the IP phone uploads the <MAC>-local.cfg file to the provisioning server. It is determined by the value of the parameter "static.auto\_provision.custom.upload\_method".

## Configuration Parameters

The following table lists the configuration parameters used to determine the phone behavior for keeping user personalized settings:

Parameters	Permitted Values	Default
<b>static.auto_provision.custom.protect</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to keep user personalized settings after auto provisioning.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, &lt;MAC&gt;-local.cfg file generates and personalized non-static settings configured via web or phone user interface will be kept after auto provisioning.</p> <p><b>Note:</b> The provisioning priority mechanism (phone/web user interface &gt;central provisioning &gt;factory defaults) takes effect only if the value of this parameter is set to 1 (Enabled). If the value of the parameter "overwrite_mode" is set to 1 in the boot file, the value of this parameter will be forced to set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.custom.sync</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to upload the &lt;MAC&gt;-local.cfg file to the server each time the file updates, and download the &lt;MAC&gt;-local.cfg file from the server during auto provisioning.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.custom.protect" is set to 1 (Enabled). The upload/download path is configured by the parameter "static.auto_provision.custom.sync.path".</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.custom.sync.path</b>	<b>URL</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the URL for uploading/downloading the &lt;MAC&gt;-local.cfg file.</p> <p>If it is left blank, the IP phone will try to upload/download the &lt;MAC&gt;-local.cfg file to/from the root directory of provisioning server.</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.custom.sync" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.custom.upload_method</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the way the IP phone uploads the &lt;MAC&gt;-local.cfg file to the provisioning server (for HTTP/HTTPS server only).</p> <p><b>0</b>-PUT</p> <p><b>1</b>-POST</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.custom.sync" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

For more information on how to configure these parameters in different scenarios, refer to the following introduced scenarios.

## Scenario A Keep user personalized configuration settings

**The administrator wishes to upgrade firmware from the old version to the latest version. Meanwhile, keep user personalized settings after auto provisioning and upgrade.**

For more information on the flowchart of keep user personalized configuration settings, refer to [Appendix E: Auto Provisioning Flowchart \(Keep User Personalized Configuration Settings\)](#) on page 897.

### Scenario Conditions:

- SIP-T46G IP phone current firmware version: 28.82.0.10. This firmware supports keeping personalized settings and generating a <MAC>-local.cfg file.

- SIP-T46G IP phone target firmware version: 28.82.0.20. This firmware supports keeping personalized settings and generating a <MAC>-local.cfg file.
- SIP-T46G IP phone MAC: 001565770984
- Provisioning server URL: tftp://192.168.1.211
- Place the target firmware to the root directory of the provisioning server.

The old firmware version supports keeping personalized settings and generating a <MAC>-local.cfg file. To keep user personalized settings after auto provisioning and upgrade, you need to configure the value of the parameter "static.auto\_provision.custom.protect" to 1 in the configuration file.

**Do one of the following operations:**

**Scenario Operations I:**

1. Edit the following parameters in the y000000000028.cfg file you want the IP phone to download:

```
static.auto_provision.custom.protect = 1  
  
static.auto_provision.custom.sync = 1  
  
static.firmware.url = tftp://192.168.1.211/28.82.0.20.rom
```

2. Trigger the IP phone to perform the auto provisioning process. For more information on how to trigger auto provisioning process, refer to *Triggering the IP Phone to Perform the Auto Provisioning* section in [Yealink SIP IP Phones Auto Provisioning Guide V81](#).

During auto provisioning, the IP phone first downloads the y000000000028.cfg file, and then downloads firmware from the root directory of the provisioning server.

The IP phone reboots to complete firmware upgrade, and then starts auto provisioning process again which is triggered by phone reboot (the power on mode is enabled by default). It downloads the y000000000028.cfg, 001565770984.cfg and the 001565770984-local.cfg file in sequence from the provisioning server, and then updates configurations in these downloaded configuration files orderly to the IP phone system. The IP phone starts up successfully, and the personalized settings in the 001565770984-local.cfg file are kept after auto provisioning.

When a user customizes feature configurations via web/phone user interface, the IP phone will save the personalized configuration settings to the 001565770984-local.cfg file on the IP phone, and then upload this file to the provisioning server each time the file updates.

**Note**

If a configuration item is both in the downloaded <MAC>-local.cfg file and Common CFG file/MAC-Oriented CFG file, setting of the configuration item in the <MAC>-local.cfg file will be written and saved to the IP phone system.

**Scenario Operations II:**

1. Edit the following parameters in the y000000000028.cfg file you want the IP phone to download:

```
static.auto_provision.custom.protect = 1

static.auto_provision.custom.sync = 0

static.firmware.url = tftp://192.168.1.211/28.82.0.20.rom
```

2. Trigger the IP phone to perform the auto provisioning process. For more information on how to trigger auto provisioning process, refer to *Triggering the IP Phone to Perform the Auto Provisioning* section in [Yealink SIP IP Phones Auto Provisioning Guide\\_V81](#).

During auto provisioning, the IP phone first downloads the y000000000028.cfg file, and then downloads firmware from the root directory of the provisioning server.

The IP phone reboots to complete firmware upgrade, and then starts auto provisioning process again which is triggered by phone reboot (the power on mode is enabled by default). It downloads the y000000000028.cfg and 001565770984.cfg files in sequence, and then updates configurations in the downloaded configuration files orderly to the IP phone system. As the value of the parameter "static.auto\_provision.custom.protect" is set to 1, configurations in the 001565770984-local.cfg file saved on the IP phone are also updated. The IP phone starts up successfully, and personalized settings are kept after auto provisioning.

When a user customizes feature configurations via web/phone user interface, the IP phone will save the personalized settings to the 001565770984-local.cfg file on the IP phone only.

**Note**

In this scenario, the IP phone will not upload the <MAC>-local.cfg file to provisioning server and request to download the <MAC>-local.cfg file from provisioning server during auto provisioning. If a configuration item is both in the <MAC>-local.cfg file on the IP phone and Common CFG file/MAC-Oriented CFG file downloaded from auto provisioning server, setting of the configuration item in the <MAC>-local CFG file will be written and saved to the IP phone system.

If the value of the parameter "static.auto\_provision.custom.protect" is set to 0, the personalized settings in the 001565770984-local.cfg file will be overridden after auto provisioning, no matter what the value of the parameter "static.auto\_provision.custom.sync" is.

**Scenario B Clear user personalized configuration settings**

**When the IP phone is given to a new user but many personalized configurations settings of last user are saved on the phone; or when the end user encounters some problems because of the wrong configurations, the administrator or user may wish to clear user personalized configuration settings via phone/web user interface.**

### Scenario Conditions:

- SIP-T46G IP phone MAC: 001565770984
- The current firmware of the phone is 28.82.0.20 or later.
- Provisioning server URL: tftp://192.168.1.211
- static.auto\_provision.custom.protect = 1

### Note

The **Reset local settings** option on the web/phone user interface appears only if the value of the parameter "static.auto\_provision.custom.protect" was set to 1.

If the value of the parameter "static.auto\_provision.custom.sync" is set to 1, the 001565770984-local.cfg file on the provisioning server will be cleared.

### To clear personalized configuration settings via phone user interface:

1. Press **Menu->Advanced** (default password: admin) ->**Reset Config**.
2. Select **Reset local settings**.

The LCD screen prompts "Reset local.cfg settings?".



3. Press the **OK** soft key.

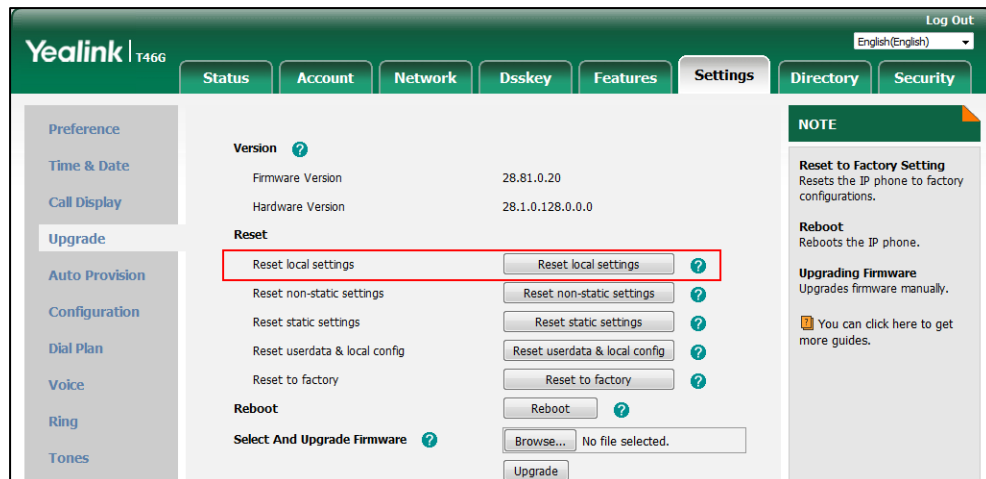
The LCD screen prompts "Reset local settings, Please wait...".

### To clear personalized configuration settings via web user interface:

1. Click on **Settings->Upgrade**.



2. Click **Reset local settings**.



The web user interface prompts "Clear local.cfg settings?".

3. Click **OK**.

Configurations in the 001565770984-local.cfg file saved on the phone will be cleared. If the IP phone is triggered to perform auto provisioning after resetting local configuration, it will download the configuration files from the provisioning server and update the configurations to the phone system. As there is no configuration in the 001565770984-local.cfg file, configurations in the y000000000028.cfg/001565770984.cfg file will take effect. If there are no configuration files on the provisioning server, the IP phone will be reset to factory defaults.

**Note**

As the static settings are never saved in the <MAC>-local.cfg file, you need to reset the static settings separately by clicking **Reset static settings** option.

## Scenario C Keep user personalized settings after factory reset

**The IP phone requires factory reset when it has a breakdown, but the user wishes to keep personalized settings of the phone after factory reset.**

**Scenario Conditions:**

- SIP-T46G IP phone MAC: 001565770984
- Provisioning server URL: tftp://192.168.1.211
- static.auto\_provision.custom.sync = 1

**Note**

As the parameter "static.auto\_provision.custom.sync" was set to 1, the 001565770984-local.cfg file on the IP phone will be uploaded to the provisioning server at tftp://192.168.1.211.

You can keep the personalized settings of the phone after factory reset via phone or web user interface.

**To reset the phone to factory via phone user interface:**

1. Press **Menu**->**Advanced** (default password: admin) ->**Reset Config**.
2. Select **Reset to Factory Settings**.

The LCD screen prompts "Reset to factory setting?".



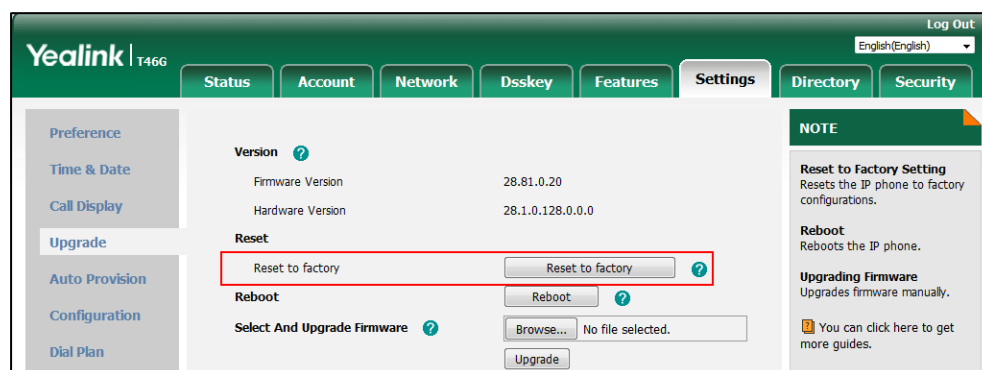
3. Press the **OK** soft key.

The LCD screen prompts "Resetting to factory, please wait...".

The LCD screen prompts "Welcome Initializing...please wait".

**To reset the phone to factory via web user interface:**

1. Click on **Settings**->**Upgrade**.
2. Click **Reset to factory** to reset the phone.



The web user interface prompts "Do you want to reset to factory?".

3. Click **OK**.

After startup, all configurations of the phone will be reset to factory defaults. So the value of the parameter "static.auto\_provision.custom.sync" will be reset to 0. Configurations in the 001565770984-local.cfg file saved on the IP phone will also be cleared. But configurations in the 001565770984-local.cfg file stored on the provisioning server (tftp://192.168.1.211) will not be cleared after reset.

**To retrieve personalized settings of the phone after factory reset:**

1. Set the values of the parameters "static.auto\_provision.custom.sync" and "static.auto\_provision.custom.protect" to be 1 in the configuration file (y000000000028.cfg

or 001565770984.cfg).

2. Trigger the phone to perform the auto provisioning process.

As the value of the parameter "static.auto\_provision.custom.sync" is set to 1, the IP phone will download the 001565770984-local.cfg file from the provisioning server to override the one stored on the phone. So the configurations in 001565770984-local.cfg file will be updated and stored on the IP phone during auto provisioning. As the value of the parameter "static.auto\_provision.custom.protect" is set to 1, the personalized configuration settings will be kept after auto provisioning. As a result, the personalized configuration settings of the phone are retrieved after factory reset.

## Scenario D Import or export the local configuration file

**The administrator or user can export the local configuration file to check the personalized settings of the phone configured by the user, or import the local configuration file to configure or change settings of the phone.**

### Scenario Conditions:

- SIP-T46G IP phone MAC: 001565770984
- The current firmware of the phone is 28.82.0.20 or later.
- Provisioning server URL: tftp://192.168.1.211

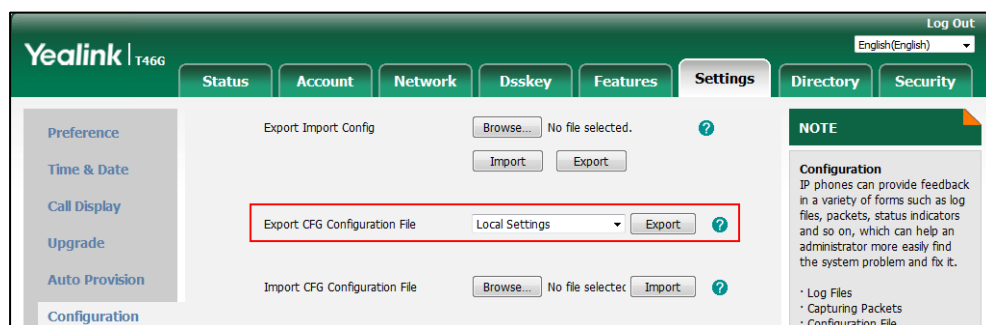
### Note

As the personalized settings of the phone cannot be changed via auto provisioning when the value of the parameter "static.auto\_provision.custom.protect" is set to 1, it is cautious to change the settings in the <MAC>-local.cfg file before importing it.

### Scenario Operations:

#### To export local configuration file via web user interface:

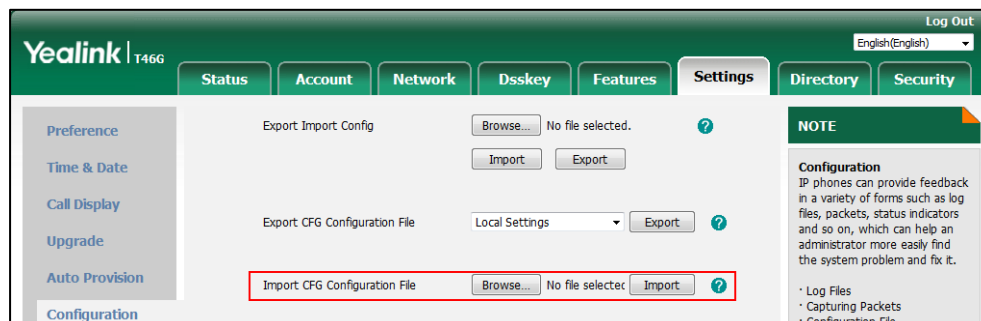
1. Click on **Settings->Configuration**.
2. Select **Local Settings** from the pull-down list of **Export CFG Configuration File**, and then click **Export** to open file download window, and then save the 001565770984-local.cfg file to the local system.



The administrator or user can edit the 001565770984-local.cfg file after exporting.

**To import local configuration file via web user interface:**

1. Click on **Settings->Configuration**.
2. In the **Import CFG Configuration File** field, click **Browse** to locate the 001565770984-local.cfg file from your local system.



3. Click **Import**.

The configurations in the imported 001565770984-local.cfg file will override the one in the existing local configuration file. The configurations only in the existing local configuration file will not be cleared. As a result, the configurations in the new 001565770984-local.cfg file contain the configurations only in the existing local configuration file and those in the imported 001565770984-local.cfg file. And this new 001565770984-local.cfg file will be saved to the phone flash and take effect.

**Note**

If the value of the parameter "static.auto\_provision.custom.sync" is set to 1, and the 001565770984-local.cfg file is successfully imported, the new 001565770984-local.cfg file will be uploaded to the provisioning server and overrides the existing one on the server.

# Configuring Basic Features

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This chapter provides information for making configuration changes for the following basic features:

- [Power Indicator LED](#)
- [Notification Popups](#)
- [Contrast](#)
- [Wallpaper](#)
- [Transparency](#)
- [Screen Saver](#)
- [Power Saving](#)
- [Backlight](#)
- [Idle Clock](#)
- [Bluetooth](#)
- [Page Switch Key](#)
- [Enable Page Tips](#)
- [Page Tips for Expansion Module](#)
- [Label Length](#)
- [Account Registration](#)
- [Multiple Line Keys per Account](#)
- [Multiple Call Appearances](#)
- [Call Display](#)
- [Display Method on Dialing](#)
- [Web Server Type](#)
- [Time and Date](#)
- [Language](#)
- [Input Method](#)
- [Logo Customization](#)
- [Softkey Layout](#)
- [Key As Send](#)
- [Send Tone](#)
- [Key Tone](#)
- [Dial Plan](#)

- [Emergency Dialplan](#)
- [Hotline](#)
- [Off Hook Hot Line Dialing](#)
- [Directory List](#)
- [Search Source List In Dialing](#)
- [Call Log](#)
- [Call List Show Number](#)
- [Missed Call Log](#)
- [Local Directory](#)
- [Live Dialpad](#)
- [Speed Dial](#)
- [Call Waiting](#)
- [Auto Redial](#)
- [Auto Answer](#)
- [IP Direct Auto Answer](#)
- [Allow IP Call](#)
- [Accept SIP Trust Server Only](#)
- [Call Completion](#)
- [Anonymous Call](#)
- [Anonymous Call Rejection](#)
- [Do Not Disturb \(DND\)](#)
- [Busy Tone Delay](#)
- [Return Code When Refuse](#)
- [Early Media](#)
- [180 Ring Workaround](#)
- [Use Outbound Proxy in Dialog](#)
- [SIP Session Timer](#)
- [Session Timer](#)
- [Ignoring Incoming Calls](#)
- [Call Hold](#)
- [Call Forward](#)
- [Call Transfer](#)
- [Conference](#)
- [Transfer on Conference Hang Up](#)

- [Transfer Mode via Dsskey](#)
- [Allow Trans Exist Call](#)
- [Directed Call Pickup](#)
- [Group Call Pickup](#)
- [Dialog Info Call Pickup](#)
- [Recent Call In Dialing](#)
- [ReCall](#)
- [Call Number Filter](#)
- [Call Park](#)
- [Calling Line Identification Presentation \(CLIP\)](#)
- [Connected Line Identification Presentation \(COLP\)](#)
- [Mute](#)
- [Intercom](#)
- [Call Timeout](#)
- [Ringing Timeout](#)
- [Send user=phone](#)
- [SIP Send MAC](#)
- [SIP Send Line](#)
- [Reserve # in User Name](#)
- [Password Dial](#)
- [Unregister When Reboot](#)
- [100 Reliable Retransmission](#)
- [Reboot in Talking](#)
- [Answer By Hand](#)
- [USB Recording](#)
- [CSTA Control](#)
- [Quick Login](#)
- [CFG File Version Information Customization](#)

## Power Indicator LED

Power indicator LED indicates power status and phone status.

You can configure the power indicator LED light behavior in the following scenarios:

- When the IP phone receives an incoming call.
- When the IP phone receives a voice mail or a text message.

- When a call is muted.
- When a call is placed on hold or is held.
- When the IP phone is busy.
- When the IP phone misses a call.

## Procedure

Power indicator LED can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000 xx>.cfg	Configure the power indicator LED.  <b>Parameters:</b>  phone_setting.common_power_led_enable phone_setting.ring_power_led_flash_enable phone_setting.mail_power_led_flash_enable phone_setting.mute_power_led_flash_enable phone_setting.hold_and_held_power_led_flash_enable phone_setting.talk_and_dial_power_led_enable phone_setting.missed_call_power_led_flash.enable
<b>Web User Interface</b>		Configure the power indicator LED.  <b>Navigate to:</b>  <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-powerled&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-powerled&amp;q=load</a>

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
phone_setting.common_power_led_enable	0 or 1	0
<b>Description:</b> Enables or disables the power indicator LED to be turned on.  <b>For</b> <b>SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones:</b> <b>0</b> -Disabled (power indicator LED is off) <b>1</b> -Enabled (power indicator LED is solid red)  <b>For SIP-T19(P) E2 IP phones:</b> <b>0</b> -Disabled (power indicator LED is off) <b>1</b> -Enabled (power indicator LED is solid yellow)  <b>Web User Interface:</b>		



Parameters	Permitted Values	Default
Features->Power LED->Common Power Light On <b>Phone User Interface:</b> None		
<b>phone_setting.ring_power_led_flash_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the power indicator LED to flash when the IP phone receives an incoming call. <b>For</b> <b>SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones:</b> <b>0</b> -Disabled (power indicator LED does not flash) <b>1</b> -Enabled (power indicator LED fast flashes (300ms) red) <b>For SIP-T19(P) E2 IP phones:</b> <b>0</b> -Disabled (power indicator LED does not flash) <b>1</b> -Enabled (power indicator LED fast flashes (300ms) yellow) <b>Web User Interface:</b> Features->Power LED->Ringing Power Light Flash <b>Phone User Interface:</b> None		
<b>phone_setting.mail_power_led_flash_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the power indicator LED to flash when the IP phone receives a voice mail or a text message. <b>For</b> <b>SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones:</b> <b>0</b> -Disabled (power indicator LED does not flash) <b>1</b> -Enabled (power indicator LED slowly flashes (1000ms) red) <b>For SIP-T19(P) E2 IP phones:</b> <b>0</b> -Disabled (power indicator LED does not flash) <b>1</b> -Enabled (power indicator LED slowly flashes (1000ms) yellow) <b>Note:</b> It works only if the value of the parameter "account.X.display_mwi.enable" is set to 1 (Enabled). <b>Web User Interface:</b>		

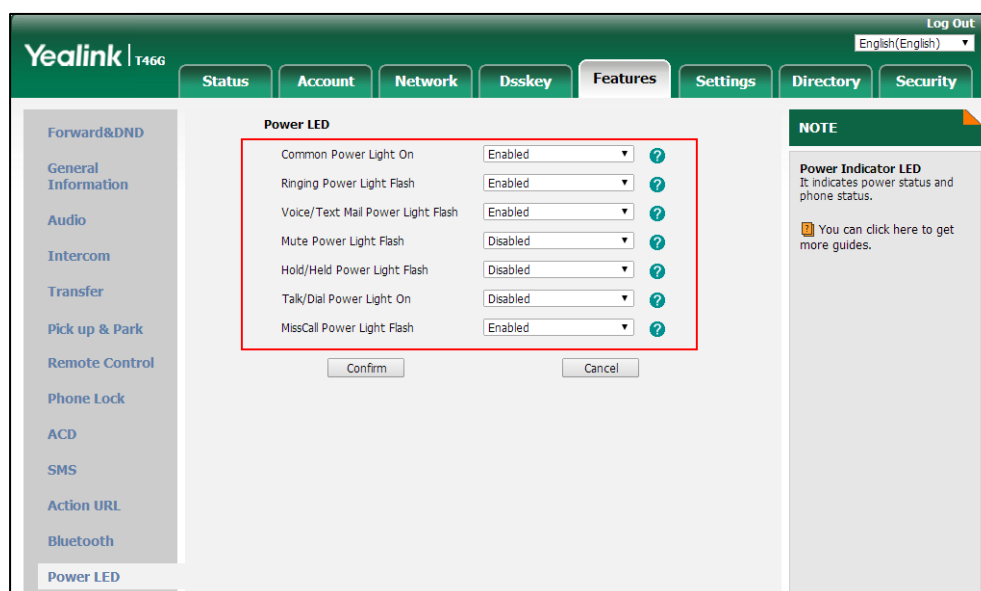
Parameters	Permitted Values	Default
Features->Power LED->Voice/Text Mail Power Light Flash <b>Phone User Interface:</b> None		
<b>phone_setting.mute_power_led_flash_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the power indicator LED to flash when a call is muted. <b>For</b> <b>SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones:</b> <b>0</b> -Disabled (power indicator LED does not flash) <b>1</b> -Enabled (power indicator LED fast flashes (300ms) red) <b>For SIP-T19(P) E2 IP phones:</b> <b>0</b> -Disabled (power indicator LED does not flash) <b>1</b> -Enabled (power indicator LED fast flashes (300ms) yellow) <b>Web User Interface:</b> Features->Power LED->Mute Power Light Flash <b>Phone User Interface:</b> None		
<b>phone_setting.hold_and_held_power_led_flash_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the power indicator LED to flash when a call is placed on hold or is held. <b>For</b> <b>SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones:</b> <b>0</b> -Disabled (power indicator LED does not flash) <b>1</b> -Enabled ( power indicator LED fast flashes (500ms) red) <b>For SIP-T19(P) E2 IP phones:</b> <b>0</b> -Disabled (power indicator LED does not flash) <b>1</b> -Enabled ( power indicator LED fast flashes (500ms) yellow) <b>Web User Interface:</b> Features->Power LED->Hold/Held Power Light Flash <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>phone_setting.talk_and_dial_power_led_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the power indicator LED to be turned on when the IP phone is busy.</p> <p><b>For</b> <b>SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones:</b>  <b>0</b>-Disabled (power indicator LED is off)  <b>1</b>-Enabled (power indicator LED is solid red)</p> <p><b>For SIP-T19(P) E2 IP phones:</b>  <b>0</b>-Disabled (power indicator LED is off)  <b>1</b>-Enabled (power indicator LED is solid yellow)</p> <p><b>Web User Interface:</b> Features-&gt;Power LED-&gt;Talk/Dial Power Light On</p> <p><b>Phone User Interface:</b> None</p>		
<b>phone_setting.missed_call_power_led_flash.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the power indicator LED to flash when the IP phone misses a call.</p> <p><b>For</b> <b>SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones:</b>  <b>0</b>-Disabled (power indicator LED does not flash)  <b>1</b>-Enabled (power indicator LED slowly flashes (1000ms) red)</p> <p><b>For SIP-T19(P) E2 IP phones:</b>  <b>0</b>-Disabled (power indicator LED does not flash)  <b>1</b>-Enabled (power indicator LED slowly flashes (1000ms) yellow)</p> <p><b>Web User Interface:</b> Features-&gt;Power LED-&gt;MissCall Power Light Flash</p> <p><b>Phone User Interface:</b> None</p>		

**To configure the power Indicator LED via web user interface:**

1. Click on **Features->Power LED**.
2. Select the desired value from the pull-down list of **Common Power Light On**.

3. Select the desired value from the pull-down list of **Ringing Power Light Flash**.
4. Select the desired value from the pull-down list of **Voice/Text Mail Power Light Flash**.
5. Select the desired value from the pull-down list of **Mute Power Light Flash**.
6. Select the desired value from the pull-down list of **Hold/Held Power Light Flash**.
7. Select the desired value from the pull-down list of **Talk/Dial Power Light On**.
8. Select the desired value from the pull-down list of **MissCall Power Light Flash**.

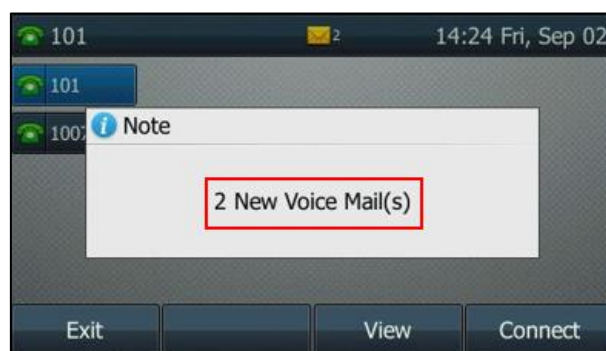


9. Click **Confirm** to accept the change.

## Notification Popups

Notification popups feature allows the IP phone to display the pop-up message box when it misses a call, forwards an incoming call to other party or receives a new voice mail or a new text message.

The following shows an example of receiving a new voice mail:



## Procedure

Notification popups can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.c fg	Configure notification popups. <b>Parameters:</b> features.voice_mail_popup.enable features.missed_call_popup.enable features.forward_call_popup.enable features.text_message_popup.enable
<b>Web User Interface</b>		Configure notification popups. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=feature s-notifypop&q=load

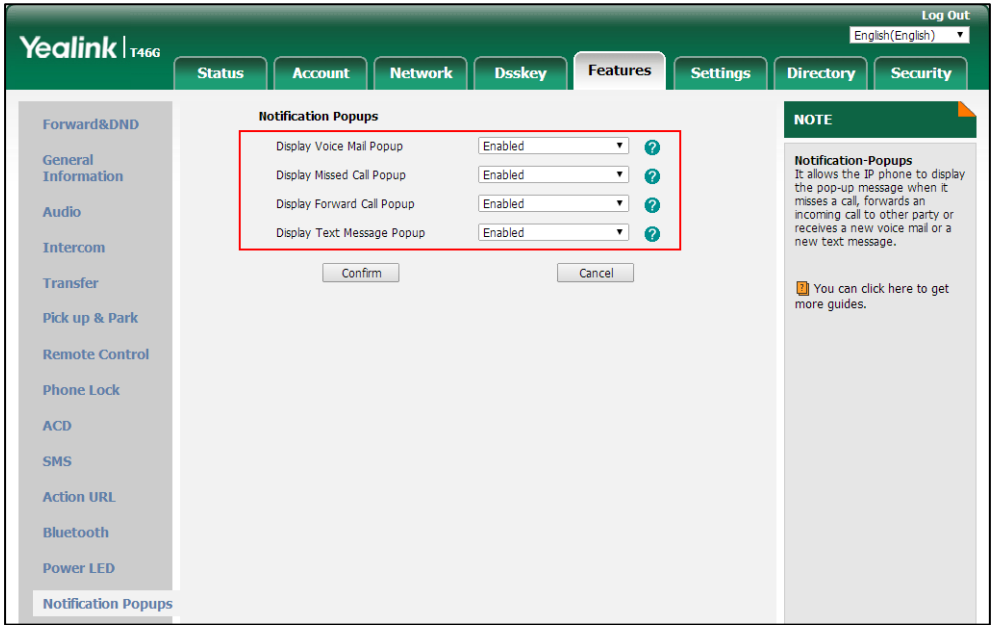
### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>features.voice_mail_popup.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to display the pop-up message box when it receives a new voice mail. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "account.X.display_mwi.enable" is set to 1 (Enabled). If the voice mail pop-up message box disappears, it won't pop up again unless the user receives a new voice mail or the user re-registers the account that has unread voice mail(s). <b>Web User Interface:</b> Features->Notification Popups->Display Voice Mail Popup <b>Phone User Interface:</b> None		
<b>features.missed_call_popup.enable</b>	<b>0 or 1</b>	<b>1</b>

Parameters	Permitted Values	Default
<b>Description:</b> Enables or disables the IP phone to display the pop-up message box when it misses a call. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "account.X.missed_callog" is set to 1 (Enabled). <b>Web User Interface:</b> Features->Notification Popups->Display Missed Call Popup <b>Phone User Interface:</b> None		
<b>features.forward_call_popup.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to display the pop-up message box when it forwards an incoming call to other party. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Features->Notification Popups->Display Forward Call Popup <b>Phone User Interface:</b> None		
<b>features.text_message_popup.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to display the pop-up message box when it receives a new text message. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "features.text_message.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Features->Notification Popups->Display Text Message Popup <b>Phone User Interface:</b> None		

**To configure the notification popups via web user interface:**

1. Click on **Features->Notification Popups**.
2. Select the desired value from the pull-down list of **Display Voice Mail Popup**.
3. Select the desired value from the pull-down list of **Display Missed Call Popup**.
4. Select the desired value from the pull-down list of **Display Forward Call Popup**.
5. Select the desired value from the pull-down list of **Display Text Message Popup**.



6. Click **Confirm** to accept the change.

## Contrast

Contrast determines the readability of the texts displayed on the LCD screen. Adjusting the contrast to a comfortable level can optimize the screen viewing experience. When configured properly, contrast allows users to read the LCD's display with minimal eyestrain. You can configure the LCD's contrast of SIP-T40P/G, SIP-T27P/G, SIP-T23P/G, SIP-T21(P) E2 and SIP-T19(P) E2 IP phones, EXP20 (connected to SIP-T29G/T27P/T27G IP phones) and EXP40 (connected to SIP-T48G/T48S/T46G/T46S IP phones). Make sure the expansion module has been connected to the IP phone before adjustment.

It is not applicable to SIP-T54S/T52S/T42G/T42S/T41P/T41S IP phones.

### Procedure

Contrast can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the contrast of the LCD screen. <b>Parameter:</b>
------------------------------------------------------	---------------------	----------------------------------------------------------------

	phone_setting.contrast
<b>Web User Interface</b>	Configure the contrast of the LCD screen.  <b>Navigate to:</b>  http://<phoneIPAddress>/servlet?p=settings-preference&q=load
<b>Phone User Interface</b>	Configure the contrast of the LCD screen.

**Details of the Configuration Parameter:**

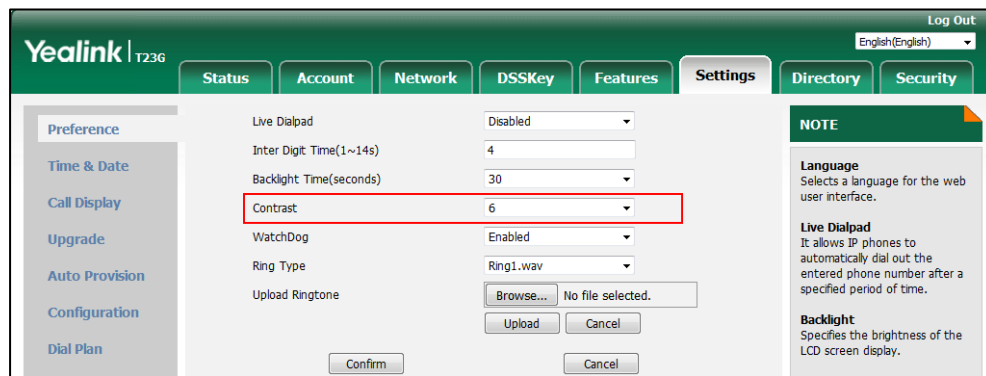
Parameter	Permitted Values	Default
phone_setting.contrast	Integer from 1 to 10	6
<p><b>Description:</b></p> <p>Configures the contrast of the LCD screen.</p> <p>For T48G/T48S/T46G/T46S IP phones, it configures the LCD's contrast of the connected EXP40 only.</p> <p>For T29G IP phones, it configures the LCD's contrast of the connected EXP20 only.</p> <p>For T27P/G IP phones, it configures the LCD's contrast of the IP phone and the connected EXP20.</p> <p>For T40P/T40G/T23P/T23G/T21(P) E2/T19(P) E2 IP phones, it configures the LCD's contrast of the IP phone.</p> <p><b>Note:</b> We recommend that you set the contrast of the LCD screen to 6 as a more comfortable level. It is not applicable to SIP-T54S/T52S/T42G/T42S/T41P/T41S IP phones.</p> <p><b>Web User Interface:</b></p> <p><b>For SIP-T48G/T48S/T46G/T46S/T29G:</b></p> <p>None</p> <p><b>For SIP-T40P/T40G/T27P/T27G/T23P/T23G/T21(P) E2/T19(P) E2:</b></p> <p>Settings-&gt;Preference-&gt;Contrast</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Display-&gt;Contrast</p>		

**To configure contrast via web user interface (take SIP-T23G IP phones for example):**

1. Click on **Settings->Preference**.



2. Select the desired value from the pull-down list of **Contrast**.



3. Click **Confirm** to accept the change.

**To configure contrast via phone user interface (take SIP-T23G IP phones for example):**

1. Press **Menu->Settings->Basic Settings->Display->Contrast**.
2. Press ◀ or ▶, or the **Switch** soft key to increase or decrease the intensity of contrast.
3. Press the **Save** soft key to accept the change.

**To configure the contrast via phone user interface (take SIP-T46G IP phones for example):**

1. Press **Menu->Basic->Display->Contrast**.  
If EXP40 is not connected to the phone, the Contrast Setting screen displays "No EXP".
2. Press ◀ or ▶, or the **Switch** soft key to increase or decrease the intensity of contrast.
3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

## Wallpaper

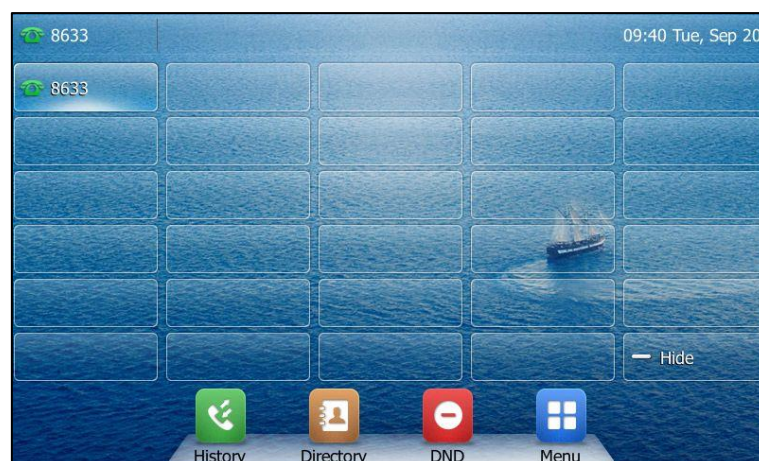
Wallpaper is an image used as the background of the IP phone and EXP50 (if connected). Users can select an image from IP phone's built-in background or customize wallpaper from personal pictures. To set the custom wallpaper as the IP phone/EXP50 background, you need to upload the custom wallpaper to the IP phone in advance. The wallpaper is only applicable to SIP-T54S, SIP-T52S, SIP-T48G/S, SIP-T46G/S, SIP-T29G IP phones and EXP50 (connected to the SIP-T54S/T52S IP phones) expansion module.

For SIP-T54S/T52S IP phones, you can also set a custom picture stored in your USB flash drive as the wallpaper. In order to do this, make sure the USB flash drive containing pictures is connected to your phone. For more information, refer to [Yealink phone-specific user guide](#).

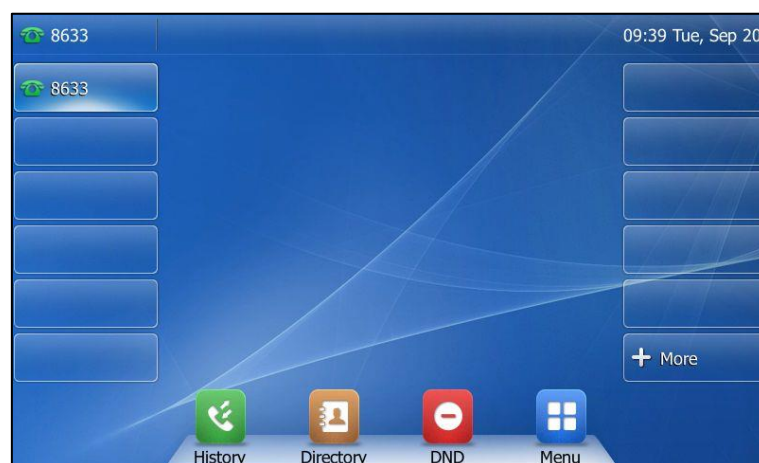
You can set the wallpaper for the SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones idle screen (for T48G/S, the DSS key list is folded) and EXP50 (connected to the SIP-T54S/T52S IP phones) expansion module.

You can also separately set the wallpaper for the SIP-T48G/S IP phones when the DSS key list is unfolded, avoiding the display of the line key labels to be blocked by the background image.

When the DSS key list is unfolded (tap **More** key when the phone is idle):



When the DSS key list is folded (tap **Hide** key when the DSS key list is unfolded):



### Note

The wallpaper will display on the entire screen. Note that the line key labels, time and date, icons, and soft keys will display over the wallpaper, while they are invisible on the image logo screen.

The wallpaper image format must meet the following:

Phone Model	Format	Resolution	Single File Size	Note
SIP-T54S/T52S	*.jpg/*.png/*.bmp/*.jpeg	<=4.2 megapixels	<=5MB	2MB of space should be reserved for the phone
SIP-T48G/S		<=2.0 megapixels	<=5MB	

Phone Model	Format	Resolution	Single File Size	Note
SIP-T46G/T46S /T29G		<=1.8 megapixels	<=5MB	

## Procedure

Wallpaper can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>. cfg	Configure the wallpaper displayed on the IP phone.  <b>Parameter:</b> phone_setting.backgrounds
		Configure the wallpaper displayed on the expansion module.  <b>Parameter:</b> expansion_module.backgrounds
		Specify the access URL of the custom wallpaper.  <b>Parameter:</b> wallpaper_upload.url
		Configure the wallpaper with DSSkey unfold.  <b>Parameter:</b> phone_setting.backgrounds_with_dsskey_unfold
<b>Web User Interface</b>		Configure the wallpaper displayed on the IP phone.  Configure the wallpaper displayed on the expansion module.  Upload the custom wallpaper.  Configure the wallpaper with DSSkey unfold.  <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=settings-preference&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=settings-preference&amp;q=load</a>
<b>Phone User Interface</b>		Configure the wallpaper displayed on the IP phone.  Configure the wallpaper displayed on the

	<p>expansion module.</p> <p>Configure the wallpaper with DSSkey unfold.</p>
--	-----------------------------------------------------------------------------

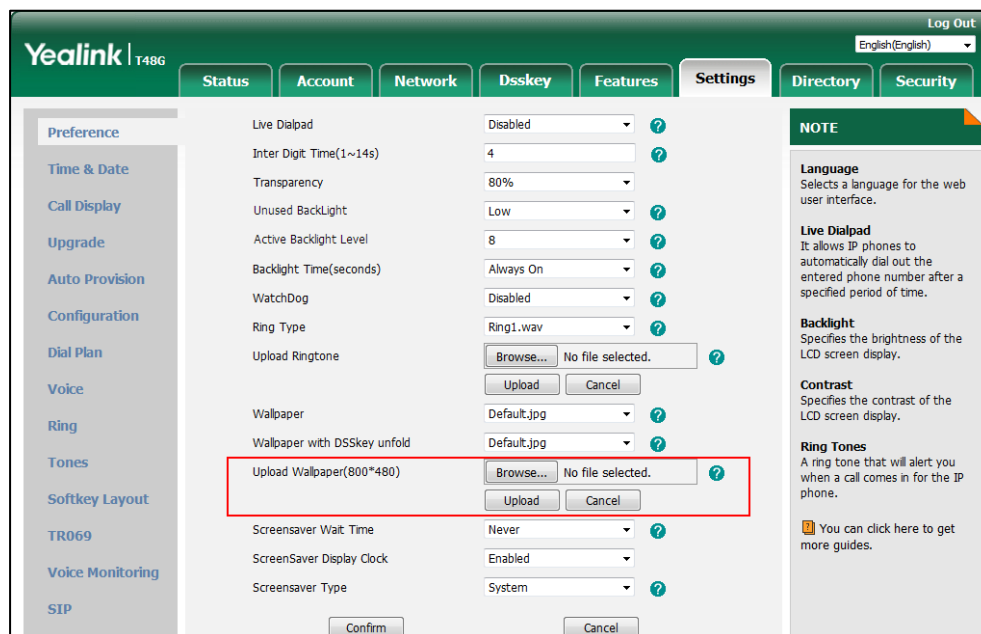
**Details of the Configuration Parameters:**

Parameters	Permitted Values	Default
<b>phone_setting.backgrounds</b>	<b>Refer to the following content</b>	<b>Default.jpg</b>
<p><b>Description:</b></p> <p>Configures the wallpaper displayed on the IP phone idle screen.</p> <p>For SIP-T48G/S IP phones, it configures the wallpaper when the DSS key list is folded.</p> <p><b>Permitted Values:</b></p> <p>Default.jpg, 01.jpg, 02.jpg, 03.jpg, 04.jpg, 05.jpg, 06.jpg, 07.jpg, 08.jpg, 09.jpg or 10.jpg or custom wallpaper name (for example, wallpaper.jpg)</p> <p><b>Example:</b></p> <p>phone_setting.backgrounds = Default.jpg</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Preference-&gt;Wallpaper</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Display-&gt;Wallpaper</p>		
<b>expansion_module.backgrounds</b>	<b>Refer to the following content</b>	<b>Default-exp50.jpg</b>
<p><b>Description:</b></p> <p>Configures the wallpaper displayed on the expansion module.</p> <p><b>Permitted Values:</b></p> <p>Default-exp50.jpg, 01-exp50.jpg, 02-exp50.jpg, 03-exp50.jpg, 04-exp50.jpg, 05-exp50.jpg, 06-exp50.jpg, 07-exp50.jpg, 08-exp50.jpg, 09-exp50.jpg or 10-exp50.jpg or custom wallpaper name (for example, wallpaper.jpg)</p> <p><b>Example:</b></p> <p>expansion_module.backgrounds = Default-exp50.jpg</p> <p><b>Note:</b> It is only applicable to EXP50 expansion modules connected to the SIP-T54S/T52S IP phones.</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
Settings->Preference->Wallpaper for expansion modules <b>Phone User Interface:</b> Menu->Basic->Display->EXP Wallpaper->Wallpaper		
<b>wallpaper_upload.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the access URL of the wallpaper image. <b>Example:</b> wallpaper_upload.url = http://192.168.10.25/wallpaper.jpg <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones. The format of the wallpaper image must be *.jpg, *.png, *.bmp or *.jpeg. <b>Web User Interface:</b> Settings->Preference->Upload Wallpaper(800*480) <b>Phone User Interface:</b> None		
<b>phone_setting.backgrounds_with_dsskey_unfold</b>	<b>Refer to the following content</b>	<b>Auto</b>
<b>Description:</b> Configures the wallpaper displayed on the IP phone when the DSS key list is unfolded. If it is set to Auto, the IP phone will use the wallpaper selected for the idle screen configured by the parameter "phone_setting.backgrounds". <b>Permitted Values:</b> Auto, Default.jpg, 01.jpg, 02.jpg, 03.jpg, 04.jpg, 05.jpg, 06.jpg, 07.jpg, 08.jpg, 09.jpg or 10.jpg or custom wallpaper name (for example, wallpaper.jpg) <b>Example:</b> phone_setting.backgrounds_with_dsskey_unfold = 01.jpg <b>Note:</b> It is only applicable to SIP-T48G/S IP phones. <b>Web User Interface:</b> Settings->Preference->Wallpaper with Dsskey unfold <b>Phone User Interface:</b> Menu->Basic->Display->Dsskey Wallpaper		

**To upload custom wallpaper via web user interface (take SIP-T48G IP phones for example):**

1. Click on **Settings->Preference**.
2. In the **Upload Wallpaper(800\*480)** field, click **Browse** to locate the wallpaper image from your local system.
3. Click **Upload** to upload the file.

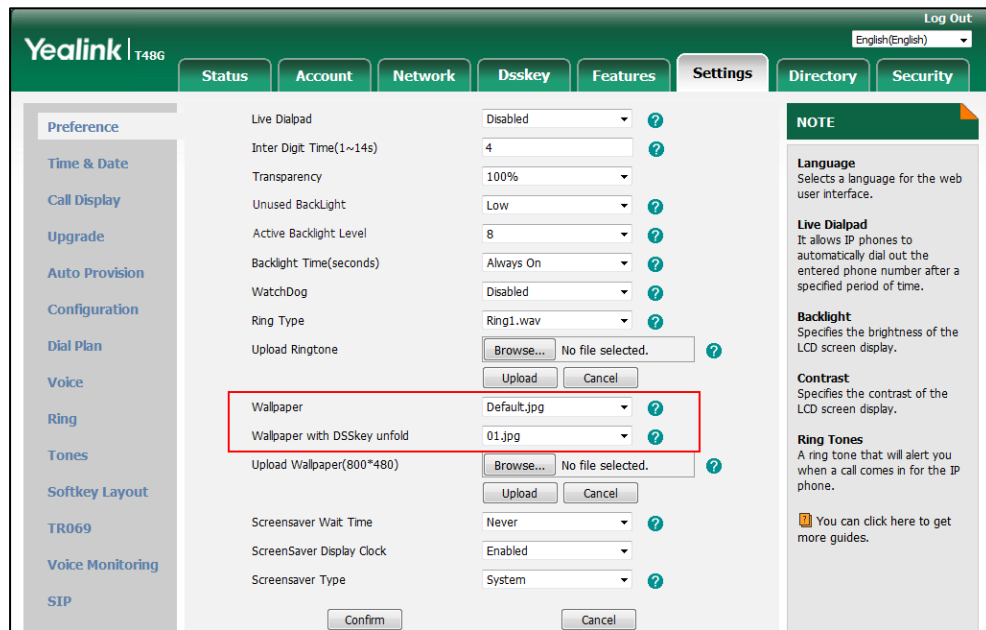


The custom wallpaper appears in the pull-down lists of **Wallpaper** and **Wallpaper with DSSkey unfold**.

**To change the wallpaper via web user interface (take SIP-T48G IP phones for example):**

1. Click on **Settings->Preference**.
2. Select the desired wallpaper from the pull-down list of **Wallpaper**.

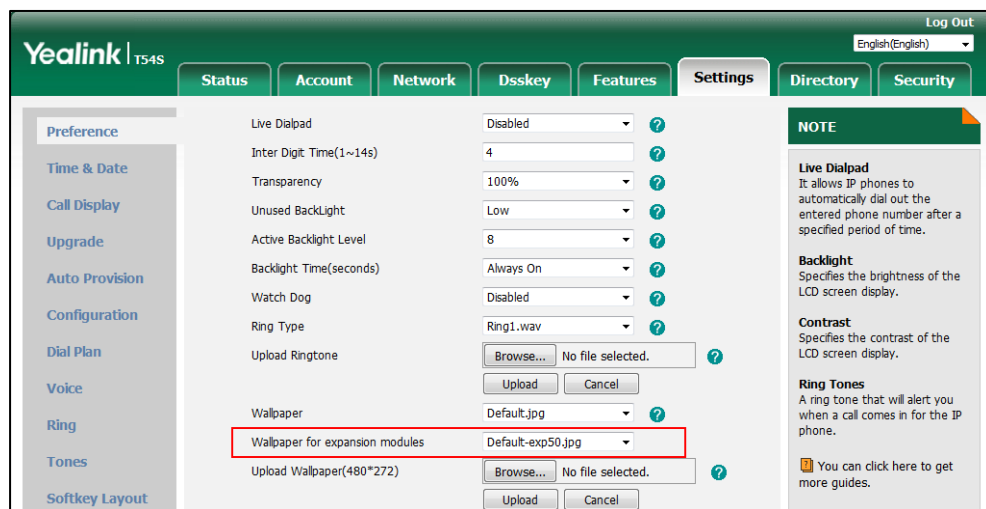
3. Select the desired wallpaper from the pull-down list of **Wallpaper with DSSkey unfold**.



4. Click **Confirm** to accept the change.

**To change the wallpaper for expansion modules via web user interface (take SIP-T54S IP phones for example):**

1. Click on **Settings->Preference**.
2. Select the desired wallpaper from the pull-down list of **Wallpaper for expansion modules**.



3. Click **Confirm** to accept the change.

**To change the wallpaper via phone user interface (take SIP-T48G IP phones for example):**

1. Tap  -> **Basic->Display->Wallpaper/Dsskey Wallpaper**.
2. Tap  or  , or press  or  to select the desired wallpaper image.

3. Tap the **Save** soft key to accept the change.

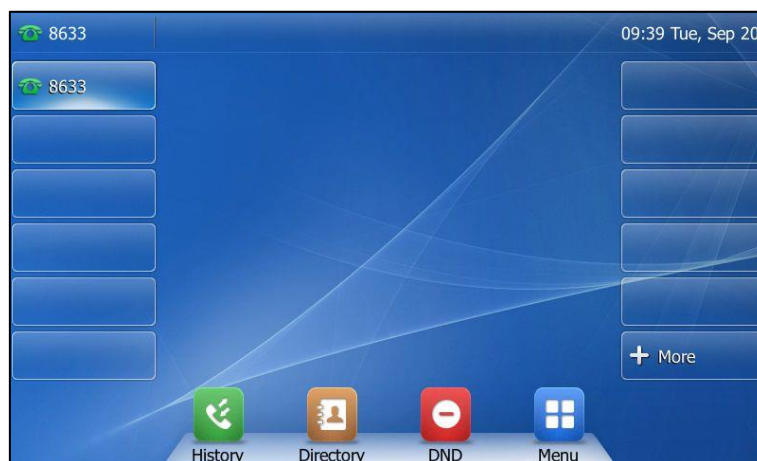
**To change the wallpaper for expansion modules via phone user interface (take SIP-T54S IP phones for example):**

1. Press **Menu**->**Basic**->**Display**->**EXP Wallpaper**.
2. Press ◀ or ▶, or the **Switch** soft key to select the desired wait time from the **Wallpaper** field.
3. Press the **Save** soft key to accept the change.

## Transparency

If you are using a custom image with a single color or complex background, it may affect your experience of the idle screen display. Users can choose an appropriate transparency for DSS key labels and status bar on the idle screen as required. The transparency is only applicable to SIP-T54S/T52S/T48G/T48S IP phones.

When the transparency is set to 100%:



When the transparency is set to 0%:





## Procedure

Transparency can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the transparency of the LCD screen. <b>Parameter:</b> phone_setting.idle_dsskey_and_title.transparency
<b>Web User Interface</b>		Configure the transparency of the LCD screen. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?path=settings-preference&q=load
<b>Phone User Interface</b>		Configure the transparency of the LCD screen.

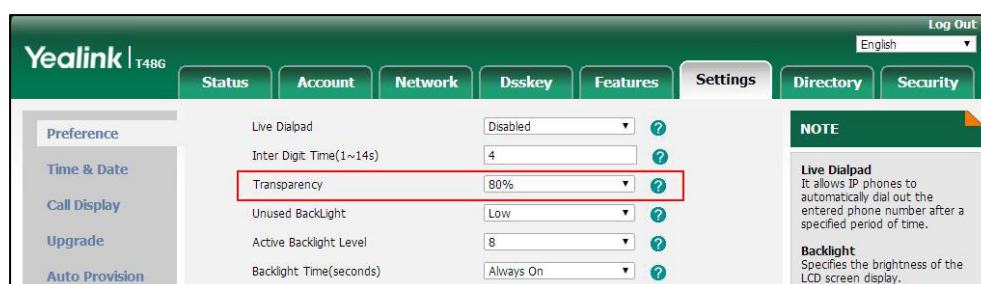
### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
phone_setting.idle_dsskey_and_title.transparency	0%, 20%, 40%, 60%, 80% or 100%	100%
<p><b>Description:</b></p> <p>Configures the transparency of the DSS key labels and status bar on the idle screen for the IP phone.</p> <p>If it is set to 0%, the DSS key labels and status bar are non-transparent.</p> <p><b>Example:</b></p> <p>phone_setting.idle_dsskey_and_title.transparency = 80%</p> <p><b>Note:</b> It is also applicable when the DSS key list is unfolded. It is only applicable to SIP-T54S/T52S/T48G/T48S IP phones.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Preference-&gt;Transparency</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Display-&gt;Transparency</p>		

**To change the transparency via web user interface (take SIP-T48G IP phones for example):**


1. Click on **Settings->Preference**.

2. Select the desired value from the pull-down list of **Transparency**.



3. Click **Confirm** to accept the change.

#### To change the transparency via phone user interface:

1. Tap  -> **Basic**-> **Display**-> **Transparency**.
2. Tap the gray box of the **Transparency** field, and then select the desired value from the pull-down list.
3. Tap the **Save** soft key to accept the change.

## Screen Saver

The screen saver will automatically start when the IP phone has been idle for a certain amount of time if you have configured the screensaver wait time. You can stop the screen saver and return to the idle screen at any time by pressing a key on the phone or tapping the touch screen (touch screen is only applicable to SIP-T48G/S IP phones).

For SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones, you can set the built-in or custom picture(s) as screen saver. To set the custom picture(s) as screen saver for the IP phone, you need to upload the custom picture(s) in advance. If multiple pictures are uploaded, all pictures are displayed like a slide show when screen saver starts.

For SIP-T54S/T52S IP phones, you can also set a custom picture stored in your USB flash drive as the screen saver. In order to do this, make sure the USB flash drive containing pictures is connected to your phone. For more information, refer to [Yealink phone-specific user guide](#).

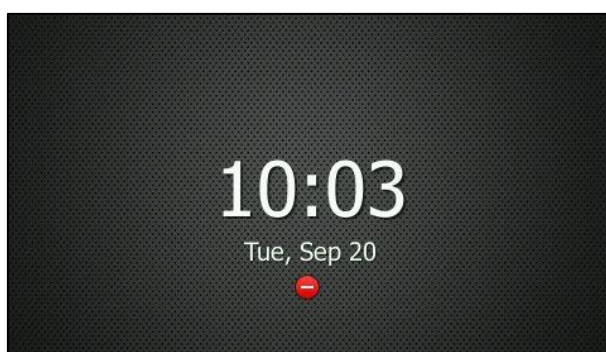
Yealink IP phones support displaying XML items (for example, notifications or company logo) on the screen saver. You can configure a screen saver XML template file to customize the XML items. For SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones, the XML items are displayed on the built-in picture or custom pictures (if uploaded). For more information on screen server XML template file, refer to [Customizing a Screen Saver XML Template File](#) on page 153.

The screen saver image format must meet the following:

Phone Model	Format	Resolution	Single File Size	Note
SIP-T54S/T52S	*.jpg/*.png/*.bmp/*.jpeg	<=4.2 megapixels	<=5MB	2MB of space should be reserved for the

Phone Model	Format	Resolution	Single File Size	Note
SIP-T48G/S		<=2.0 megapixels	<=5MB	phone
SIP-T46G/T46S /T29G		<=1.8 megapixels	<=5MB	

The following shows that the built-in screen saver is displaying on the phone:



## Customizing a Screen Saver XML Template File

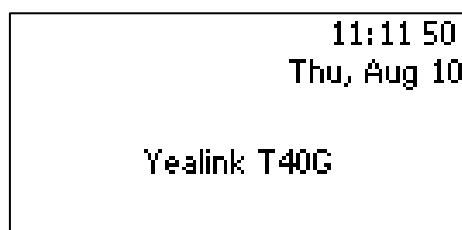
You can customize the screen saver XML template file to configure the IP phone whether to display following XML items on the screen saver:

- Time and date
- Status icons
- Notifications
- Image (only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones)

The display position of the XML items is configurable. For

SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones, you can also customize the color of the time & date and notification texts.

The following shows an example:



SIP-T40G



SIP-T46G

If you update the screen saver xml file, you can do one of the following to trigger the IP phone to update the configuration:

- Reset the screensaver type to **Server XML**.
- Change the access URL of the screen saver xml file.
- Press the corresponding XML Browser DSS key.

Before customizing, you need to obtain the screen saver XML template file. You can ask the distributor or Yealink FAE for screen saver XML template. You can also obtain the screen saver XML template online:

<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>.

The following table lists description of each element in the template file:

Parameter	Position	Value	Description
YealinkIPPhoneCustomScreenSaver	Root tag	none	The root element of the xml file.
LineSpacing	Root tag	integer	The vertical distance between different lines.
InsertImageLineNum	Root tag	For SIP-T48G/S IP phones: Integer from 0 to 11	Specify which line to insert the image (configured by the <b>Image</b> parameter). <b>Note:</b> It is only applicable to

Parameter	Position	Value	Description
		For SIP-T54S/T52S/T 46G/T46S/T29G IP phones: Integer from 0 to 8	SIP-T54S/T52S/T48G/T48S/T46G/T4 6S/T29G IP phones.
SystemTime	Body	"show" "hide"	Whether to display the time and date.
Size	SystemTime tag	"normal" "small" "double" "large"	Font size of the text. "small": 12 pt "normal": 18 pt "double": 24 pt "large": 28 pt
horizontalAlign	SystemTime tag	"left" "middle" "right"	Horizontal position of the time and date.
verticalAlign	SystemTime tag	"top" "middle" "bottom"	Vertical position of the time and date.
Color	SystemTime tag	"white" "black" "red" "green" "brown" "blue" "magenta" "cyan" "lightgray" "darkgray" "lightred" "lightgreen" "yellow" "lightblue" "lightmagenta" "lightcyan" or RGB values (for example, 123,4,40 or	Color of the text. <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T4 6S/T29G IP phones.

Parameter	Position	Value	Description
		RGB(123,4,140))	
StatusIcons	Body	"show" "hide"	Whether to display the status icons.
horizontalAlign	StatusIcons tag	"left" "middle" "right"	Horizontal position of the status icons.
verticalAlign	StatusIcons tag	"top" "middle" "bottom"	Vertical position of the status icons.
Line	Body	string	Text to be displayed on the line. If the length of the text is too long to be displayed on the LCD screen, the exceeding characters will be displayed as ellipsis.  <b>Note:</b> The SIP-T48G/S IP phones support displaying up to 11 lines of content; SIP-T54S/T52S/T46G/T46S/T29G/T27G/T27P IP phones support displaying up to 8 lines of content; SIP-T42G/T42S/T41P/T41S/T23P/T23G/T21(P) E2/T19(P) E2 IP phones support displaying up to 4 lines of content.
Size	Line tag	"normal" "small" "double" "large"	Font size of the text. "small": 12 pt "normal": 18 pt "double": 24 pt "large": 28 pt
Align	Line tag	"center" "left" "right"	Alignment of the text.
Color	Line tag	"white" "black" "red" "green" "brown" "blue"	Color of the text.  <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.

Parameter	Position	Value	Description
		"magenta" "cyan" "lightgray" "darkgray" "lightred" "lightgreen" "yellow" "lightblue" "lightmagenta" "lightcyan" or RGB values (for example, 123,4,40 or RGB(123,4,140))	
Image	Body	string	The access URL of the image. <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T4 6S/T29G IP phones.
horizontalAlign	Image tag	"left" "middle" "right"	Horizontal position of the image. <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T4 6S/T29G IP phones.
verticalAlign	Image tag	"top" "middle" "bottom"	Vertical position of the image. <b>Note:</b> It works only if you do not configure <b>InsertImageLineNum</b> parameter or set the <b>InsertImageLineNum</b> parameter to 0. It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T4 6S/T29G IP phones.
height	Image tag	integer	The height (in pixels) of the company logo displayed on the IP phone. <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T4 6S/T29G IP phones.
width	Image tag	integer	The width (in pixels) of the company logo displayed on the IP phone. <b>Note:</b> It is only applicable to

Parameter	Position	Value	Description
			SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.

#### Customizing a screen saver xml file:

1. Open the screen saver xml file using an ASCII editor.
2. Modify settings as you want in the screen saver xml file.

```
CustomScreenSaver.xml x
0 10 20 30 40 50 60 70 80 90 100 110 120
<?xml version="1.0" encoding="ISO-8859-1"?>
<YalinkIPPhoneCustomScreenSaver LineSpacing = "9" InsertImageLineNum = "1" >

<SystemTime Size="Large" horizontalAlign="right" verticalAlign="top" Color="black">show</SystemTime>

<StatusIcons horizontalAlign="middle" verticalAlign="top">show</StatusIcons>

<Line Size="large" Align="center" Color="blue">Yalink</Line>
<Line Size="large" Align="center" Color="RGB" >SIP Phone</Line>
<Line Size="large" Align="center" Color="RGB">Test</Line>

<Image horizontalAlign="middle" verticalAlign="bottom" height="30" width="30">http://192.168.1.1/Yalink.jpeg</Image>

</YalinkIPPhoneCustomScreenSaver>
```

3. Save this file and place it to the provisioning server (for example, 192.168.1.100).
4. Specify the access URL of the screen saver xml file in the configuration file.

#### Procedure

Screen saver can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the time to wait in the idle state before the screen saver starts.  <b>Parameter:</b> screensaver.wait_time
		Configure the type of screen saver to display.  <b>Parameter:</b> screensaver.type
		Specify the access URL of the custom screen saver image.  <b>Parameter:</b> screensaver.upload_url
		Specify the access URL of the screen saver xml file.  <b>Parameter:</b> screensaver.xml_browser.url
		Delete custom screen saver image.  <b>Parameter:</b> screensaver.delete



		<p>Configure the IP phone to display the clock and icons when the screen saver starts.</p> <p><b>Parameter:</b></p> <p>screensaver.display_clock.enable</p>
		<p>Configure the interval for the IP phone to change the picture when the screen saver starts.</p> <p><b>Parameter:</b></p> <p>screensaver.picture_change_interval</p>
		<p>Configure the interval for the IP phone to move the clock and icons when the screen saver starts.</p> <p><b>Parameter:</b></p> <p>screensaver.clock_move_interval</p>
		<p>Assign a XML Browser DSS key.</p> <p><b>Parameters:</b></p> <p>linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type</p> <p>linekey.X.line/ programablekey.X.line/ expansion_module.X.key.Y.line</p> <p>linekey.X.value/ programablekey.X.value/ expansion_module.X.key.Y.value</p> <p>linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label</p> <p>linekey.X.shortlabel</p>
<b>Web User Interface</b>	<p>Configure the time to wait in the idle state before the screen saver starts.</p> <p>Configure the type of screen saver to display.</p> <p>Upload the custom screen saver image.</p> <p>Delete custom screen saver image.</p> <p>Specify the access URL of the screen saver xml file.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?m=mod_data&amp;p=settings-preference&amp;q=load</p>	
	<p>Assign a XML Browser DSS key.</p> <p><b>Navigate to:</b></p>	

	http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0
<b>Phone User Interface</b>	Configure the screen saver wait time on the IP phone.
	Assign a XML Browser DSS key.

**Details of the Configuration Parameters:**

Parameters	Permitted Values	Default
<b>screensaver.wait_time</b>	<b>15, 30, 60, 120, 300, 600, 1800, 3600, 7200, 10800, 21600</b>	<b>21600</b>
<b>Description:</b> Configures the time (in seconds) to wait in the idle state before the screen saver starts. <b>15-15s</b> <b>30-30s</b> <b>60-1min</b> <b>120-2min</b> <b>300-5min</b> <b>600-10min</b> <b>1800-30min</b> <b>3600-1h</b> <b>3200-2h</b> <b>10800-3h</b> <b>21600-6h</b> <b>Web User Interface:</b> Settings->Preference->Screensaver Wait Time <b>Phone User Interface:</b> Menu->Basic->Display->Screensaver->Wait Time		
<b>screensaver.type</b>	<b>0, 1 or 2</b>	<b>0</b>
<b>Description:</b> Configures the type of screen saver to display. <b>0</b> -System, the LCD screen will display the built-in picture. <b>1</b> -Custom, the LCD screen will display the custom screen saver images (configured by the parameter "screensaver.upload_url"). If multiple images are uploaded, the IP phone will		

Parameters	Permitted Values	Default
<p>display all images alternately. The time interval is configured by the parameter "screensaver.picture_change_interval".</p> <p><b>2-Server XML</b>, the LCD screen will display XML items (configured by the parameter "screensaver.xml_browser.url") when screen saver starts.</p> <p><b>Note:</b> It is only applicable to SIP-T54S, SIP-T52S, SIP-T48G/S, SIP-T46G/S and SIP-T29G IP phones.</p> <p><b>Web User Interface:</b> Settings-&gt;Preference-&gt;Screensaver Type</p> <p><b>Phone User Interface:</b> Menu-&gt;Basic-&gt;Display-&gt;Screensaver-&gt;Screensaver Type</p> <p><b>Note:</b> It is configurable only if you have uploaded custom image file(s) to the IP phone.</p>		
<b>screensaver.upload_url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of the custom screen saver image.</p> <p><b>Example:</b> screensaver.upload_url = http://192.168.10.25/Screencapture.jpg</p> <p>During the auto provisioning process, the IP phone connects to the HTTP provisioning server "192.168.10.25", and downloads the screen saver image "Screencapture.jpg".</p> <p>If you want to upload multiple screen saver images to the phone simultaneously, you can configure as following: screensaver.upload_url = http://192.168.10.25/Screencapture.jpg screensaver.upload_url = http://192.168.10.25/Screensaver.jpg</p> <p><b>Note:</b> It is only applicable to SIP-T54S, SIP-T52S, SIP-T48G/S, SIP-T46G/S and SIP-T29G IP phones.</p> <p><b>Web User Interface:</b> Settings-&gt;Preference-&gt;Screensaver Type (Custom)-&gt;Upload Screensaver</p> <p><b>Phone User Interface:</b> None</p>		
<b>screensaver.xml_browser.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of the screen saver xml file.</p> <p><b>Example:</b> screensaver.xml_browser.url = http://192.168.10.25/ScreenSaver.xml</p>		

Parameters	Permitted Values	Default
<p>During the auto provisioning process, the IP phone connects to the HTTP provisioning server "192.168.10.25", and downloads the screen saver xml file "ScreenSaver.xml".</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Preference-&gt;Screensaver Type (Server XML)-&gt;XML Browser URL</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>screensaver.delete</b>	<b>http://localhost/all</b> or <b>http://localhost/na me.(jpg/png/bmp)</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Deletes the specified or all custom screen saver images.</p> <p><b>Example:</b></p> <p>Delete all custom screen saver images:</p> <p>screensaver.delete = http://localhost/all</p> <p>Delete a custom screen saver image (for example, Screenshot.jpg):</p> <p>screensaver.delete = http://localhost/Screenshot.jpg</p> <p><b>Note:</b> It is only applicable to SIP-T54S, SIP-T52S, SIP-T48G/S, SIP-T46G/S and SIP-T29G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>screensaver.display_clock.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to display the clock and icons when the screen saver starts.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "screensaver.type" is set to 0 (System) or 1 (Custom). It is only applicable to SIP-T54S, SIP-T52S, SIP-T48G/S, SIP-T46G/S and SIP-T29G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Preference-&gt;Screensaver Type(System/Custom)-&gt;Screensaver Display Clock</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
Menu->Basic->Display->Screensaver(System/Custom)->Display Clock		
<b>screensaver.picture_change_interval</b>	<b>Integer from 5 to 1200</b>	<b>60</b>
<p><b>Description:</b></p> <p>Configures the interval (in seconds) for the IP phone to change the picture when the screen saver starts.</p> <p><b>Note:</b> It works only if the value of the parameter "screensaver.type" is set to 1 (Custom) and the parameter "screensaver.upload_url" should be configured in advance. It is only applicable to SIP-T54S, SIP-T52S, SIP-T48G/S, SIP-T46G/S and SIP-T29G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>screensaver.clock_move_interval</b>	<b>Integer from 5 to 1200</b>	<b>600</b>
<p><b>Description:</b></p> <p>Configures the interval (in seconds) for the IP phone to move the clock and icons when the screen saver starts.</p> <p><b>Note:</b> It works only if the value of the parameter "screensaver.display_clock.enable" is set to 1 (Enabled). It is only applicable to SIP-T54S, SIP-T52S, SIP-T48G/S, SIP-T46G/S and SIP-T29G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## XML Browser DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

**Details of Configuration Parameters:**

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>programmablekey.X.type<sup>[2]</sup></b> <b>expansion_module.X.key.Y.type<sup>[3]</sup></b>	27
<b>Description:</b> Configures a DSS key as a directed call pickup DSS key on the IP phone. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.value<sup>[1]</sup></b> <b>programmablekey.X.value<sup>[2]</sup></b> <b>expansion_module.X.key.Y.value<sup>[3]</sup></b>	String within 99 characters
<b>Description:</b> Configures the access URL of the screen saver xml file. <b>Example:</b> linekey.3.value = http://192.168.1.1/CustomScreenSaver.xml <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Value <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Value	
<b>linekey.X.label<sup>[1]</sup></b> <b>programmablekey.X.label<sup>[2]</sup></b> <b>expansion_module.X.key.Y.label<sup>[3]</sup></b>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b>	

Parameters	Permitted Values
Dsskey->Line Key/Programable Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	<b>String within 99 characters</b>
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Short <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To upload custom screen saver via web user interface:

1. Click on **Settings->Preference**.
2. Select **Custom** from the pull-down list of **Screensaver Type**.
3. In the **Upload Screensaver** field, click **Browse** to locate the custom picture from your local system.
4. Click **Upload** to upload the file.

The **Upload Screensaver** field appears only if **Screensaver Type** is set to **Custom**.

The screenshot shows the Yealink T46G web interface. The 'Settings' tab is selected, and the 'Preference' sub-tab is active. The 'Screensaver Type' is set to 'Custom'. The 'Upload Screensaver' field is visible and highlighted with a red box. The 'Upload Screensaver' field has a 'Browse...' button and a 'No file selected.' message. The 'Upload' and 'Cancel' buttons are also visible.

The custom screen saver appears in the pull-down list of **Screensaver**. The **Screensaver** field appears only if **Screensaver Type** is set to **Custom**.

**To configure the access URL of the screen saver xml file via web user interface:**

1. Click on **Settings**->**Preference**.
2. Select **Server XML** from the pull-down list of **Screensaver Type**.
3. Enter the URL of the screen saver xml file in the **XML Browser URL** field.

The screenshot shows the Yealink T46G web interface. The 'Settings' tab is selected, and the 'Preference' sub-tab is active. The 'Screensaver Type' is set to 'Server XML'. The 'XML Browser URL' field is visible and highlighted with a red box. The 'XML Browser URL' field contains the text 'http://192.168.1.1/CustomScreenSav'. The 'Upload' and 'Cancel' buttons are also visible.



- Click **Confirm** to accept the change.

**To set the system screen saver via web user interface:**

- Click on **Settings->Preference**.
- Select **System** from the pull-down list of **Screensaver Type**.

The screenshot shows the Yealink T46G web interface. The 'Settings' tab is selected, and the 'Preference' sub-tab is active. On the left sidebar, 'Preference' is highlighted. The main content area lists various settings. The 'Screensaver Type' dropdown is highlighted with a red box and set to 'System'. Other settings include Live Dialpad (Disabled), Inter Digit Time (4), Unused Backlight (Low), Active Backlight Level (8), Backlight Time (Always On), Watch Dog (Enabled), Ring Type (Ring1.wav), Upload Ringtone (No file selected), Wallpaper (Default.jpg), Upload Wallpaper (No file selected), Screensaver Wait Time (6h), and Screensaver Display Clock (Enabled). A 'NOTE' section on the right provides details about Live Dialpad, Backlight, Contrast, and Ring Tones.

- Click **Confirm** to accept the change.

**To configure the screen saver wait time and screensaver display clock via web user interface:**

- Click on **Settings->Preference**.
- Select the desired time from the pull-down list of **Screensaver Wait Time**.
- Select the desired value from the pull-down list of **Screensaver Display Clock**.

This screenshot is similar to the previous one, showing the same Yealink T46G web interface. In this view, the 'Screensaver Wait Time' and 'Screensaver Display Clock' dropdowns are highlighted with a red box. The 'Screensaver Type' dropdown remains set to 'System'. The rest of the interface, including the sidebar and the 'NOTE' section, is identical to the previous screenshot.

- Click **Confirm** to accept the change.

**To configure a XML Browser DSS key via web user interface:**

1. Click on **Dsskey**->**Line Key** (or **Programable Key**).
2. In the desired DSS key field, select **Direct Pickup** from the pull-down list of **Type**.
3. Enter the access URL of the screen saver xml file in the **Value** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

The screenshot shows the Yealink T46G web interface. The 'Dsskey' tab is selected in the top navigation bar. Under 'Dsskey', the 'Line Key' sub-tab is active. A table displays the configuration for Line Keys 1 through 9. Line Key 3 is highlighted with a red box. The table has columns for Key, Type, Value, Label, Line, and Extension. Line Key 3 is configured with Type 'XML Browser', Value '8/CustomScreenSaver.xml', and Label 'N/A'. The interface also includes a sidebar with navigation links, a top navigation bar, and a 'NOTE' section on the right.

Key	Type	Value	Label	Line	Extension
Line Key1	Line	Default	4604	Line 1	
Line Key2	Line	Default		Line 2	
Line Key3	XML Browser	8/CustomScreenSaver.xml	N/A		
Line Key4	Line	Default		Line 4	
Line Key5	Line	Default		Line 5	
Line Key6	Line	Default		Line 6	
Line Key7	Line	Default		Line 7	
Line Key8	Line	Default		Line 8	
Line Key9	Line	Default		Line 9	

5. Click **Confirm** to accept the change.

**To configure the screen saver via phone user interface:**

1. Press **Menu**->**Basic**->**Display**->**Screensaver**.
2. Press **Left** or **Right** arrow, or the **Switch** soft key to select the desired wait time from the **Wait Time** field.
3. (Optional.) Press **Left** or **Right** arrow, or the **Switch** soft key to select the desired value from the **Display Clock** field.
4. Press **Left** or **Right** arrow, or the **Switch** soft key to select the desired value from the **Screensaver Type** field.
5. Press the **Save** soft key to accept the change.

**To configure a XML Browser DSS key via phone user interface:**

1. Press **Menu**->**Features**->**DSS Keys**.
2. Select the desired DSS key.
3. Press **Left** or **Right** arrow, or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press **Left** or **Right** arrow, or the **Switch** soft key to select **XML Browser** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the access URL of the screen saver xml file in the **Value** field.
7. Press the **Save** soft key to accept the change.

# Power Saving

The power-saving feature is used to turn off the backlight and screen (turning off the screen is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones and EXP50 expansion module (connected to the SIP-T54S/T52S IP phones) to conserve energy. The IP phone and EXP50 (if connected) enter power-saving mode after the phone has been idle for a certain period of time. And the IP phone and EXP50 (if connected) will exit power-saving mode if a phone/EXP50 event occurs - for example, if the phone has an incoming call or message, or you press a key on the phone/EXP50 or tap the touch screen (touch screen is only applicable to SIP-T48G/S IP phones). The power saving is not applicable to SIP-T19(P) E2 IP phones.

If the screen saver (refer to [Screen Saver](#)) is enabled on your phone, power-saving mode will still occur. For example, if a screen saver is configured to display after the phone has been idle for 5 minutes, and power-saving mode is configured to turn off the backlight and screen after the phone has been idle for 15 minutes, the backlight and screen will be turned off after the screen saver displays for 10 minutes.

You can configure the following power-saving settings:

- **Office Hour:** Configures the starting time and ending time of the day's office hour for each day of the week. You can configure power saving around your work schedule.
- **Idle TimeOut (minutes):** Configures the period of time before the IP phone enters power-saving mode. You can configure different idle timeouts for office hours and off hours (evenings and weekends). You can also specify a separate timeout period that applies after you use the phone.

By default, the Office Hours Idle TimeOut is much longer than the Off Hours Idle TimeOut. If you use the IP phone, the idle timeout that applies (User Input Extension Idle TimeOut or Office Hours/Off Hours Idle TimeOut) is the timeout with the highest value. If the phone has an incoming call or message, the User Input Extension Idle TimeOut is ignored.

For SIP-T54S/T52S/T48S/T48G/T46S/T46G/T29G IP phones, you can also configure the flashing period for the power indicator LED when the IP phone enters power-saving mode.

**Note**

If you disable the power saving feature, the IP phone will automatically enter power-saving mode to protect the screen when the phone is inactive for 72 hours. That is, the color screen phones will turn off the backlight and screen, and the black-and-white screen phones will only turn off the backlight. Image persistence may be caused on LCD if power saving is disabled.

## Procedure

Power saving can be configured using the following methods.

<b>Central Provisioning (Configuration</b>	<code>&lt;y0000000000 xx&gt;.cfg</code>	Configure the power saving intelligent mode. <b>Parameter:</b> features.power_saving.intelligent_mode
--------------------------------------------	---------------------------------------------	-------------------------------------------------------------------------------------------------------------

<b>File)</b>		<p>Configure the power saving feature.</p> <p><b>Parameter:</b></p> <p>features.power_saving.enable</p>
		<p>Configure the office hour.</p> <p><b>Parameters:</b></p> <p>features.power_saving.office_hour.monday  features.power_saving.office_hour.tuesday  features.power_saving.office_hour.wednesday  features.power_saving.office_hour.thursday  features.power_saving.office_hour.friday  features.power_saving.office_hour.saturday  features.power_saving.office_hour.sunday</p>
		<p>Configure the idle timeout.</p> <p><b>Parameters:</b></p> <p>features.power_saving.office_hour.idle_timeout  features.power_saving.off_hour.idle_timeout  features.power_saving.user_input_ext.idle_timeout</p>
		<p>Configure the flashing period of the power indicator LED in the power-saving mode.</p> <p><b>Parameters:</b></p> <p>features.power_saving.power_led_flash.on_time  features.power_saving.power_led_flash.off_time</p>
<b>Web User Interface</b>		<p>Configure the power saving feature.</p> <p>Configure the office hour.</p> <p>Configure the idle timeout.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=settings-powersaving&amp;q=load</p>

**Details of the Configuration Parameters:**

Parameters	Permitted Values	Default
features.power_saving.intelligent_mode	0 or 1	1
<p><b>Description:</b></p> <p>Enables or disables the power saving intelligent mode.</p>		

Parameters	Permitted Values	Default
<p><b>0</b>-Disabled, the IP phone stays in power-saving mode even if the office hour arrives the next day.</p> <p><b>1</b>-Enabled, the IP phone will automatically identify the office hour and exit power-saving mode once the office hour arrives the next day.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.power_saving.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the power saving feature.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Power Saving-&gt;Power Saving</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.power_saving.office_hour.idle_timeout</b>	<b>Integer from 1 to 960</b>	<b>960</b>
<p><b>Description:</b></p> <p>Configures the time (in minutes) to wait in the idle state before the IP phone enters power-saving mode during the office hours.</p> <p><b>Example:</b></p> <p>features.power_saving.office_hour.idle_timeout = 600</p> <p>The IP phone will enter power-saving mode when it has been inactivated for 600 minutes (10 hour) during the office hours.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Power Saving-&gt;Office Hour Idle TimeOut</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.power_saving.off_hour.idle_timeout</b>	<b>Integer from 1 to</b>	<b>10</b>

Parameters	Permitted Values	Default
	10	
<p><b>Description:</b></p> <p>Configures the time (in minutes) to wait in the idle state before the IP phone enters power-saving mode during the non-office hours.</p> <p><b>Example:</b></p> <p>features.power_saving.off_hour.idle_timeout = 5</p> <p>The IP phone will enter power-saving mode when it has been inactivated for 5 minutes during the non-office hours.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Power Saving-&gt;Off Hour Idle TimeOut</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
features.power_saving.user_input_ext.idle_timeout	Integer from 1 to 30	10
<p><b>Description:</b></p> <p>Configures the minimum time (in minutes) to wait in the idle state - after using the phone - before the IP phone enters power-saving mode.</p> <p><b>Example:</b></p> <p>features.power_saving.user_input_ext.idle_timeout = 5</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Power Saving-&gt;User Input Extension Idle TimeOut</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
features.power_saving.office_hour.monday	Integer from 0 to 23, Integer from 0 to 23	7,19
features.power_saving.office_hour.tuesday		7,19
features.power_saving.office_hour.wednesday		7,19
features.power_saving.office_hour.thursday		7,19
features.power_saving.office_hour.friday		7,19
features.power_saving.office_hour.saturday		7,7
features.power_saving.office_hour.sunday		7,7
<p><b>Description:</b></p> <p>Configures the starting time and ending time of the day's office hour.</p>		

Parameters	Permitted Values	Default
<p>Starting time and ending time are separated by a comma.</p> <p><b>Example:</b></p> <p>features.power_saving.office_hour.monday = 7,19</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Power Saving-&gt;Monday/Tuesday/Wednesday/Thursday/Friday/Saturday/Sunday</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.power_saving.power_led_flash.on_time</b>	<b>0, Integer from 100 to 10000</b>	<b>500</b>
<p><b>Description:</b></p> <p>Configures the period of time (in milliseconds) that the power indicator LED is on when the IP phone enters the power-saving mode.</p> <p>If it is set to 0 and the value of the parameter "features.power_saving.power_led_flash.off_time" is not set to 0, the power indicator LED will be off when the IP phone enters the power-saving mode.</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.power_saving.power_led_flash.off_time</b>	<b>0, Integer from 100 to 10000</b>	<b>3000</b>
<p><b>Description:</b></p> <p>Configures the period of time (in milliseconds) that the power indicator LED is off when the IP phone enters the power-saving mode.</p> <p>If it is set to 0, the power indicator LED will be on when the IP phone enters the power-saving mode.</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure the power saving feature via web user interface:**

1. Click on **Settings->Power Saving**.
2. Select **Enabled** from the pull-down list of **Power Saving**.
3. Enter the starting time and ending time respectively in the desired day field.
4. Enter the desired value (1-960) in the **Office Hours Idle TimeOut** field.
5. Enter the desired value (1-10) in the **Off Hours Idle TimeOut** field.
6. Enter the desired value (1-30) in the **User Input Extension Idle TimeOut** field.

The screenshot displays the Yealink T46G web interface. The 'Settings' tab is selected, and the 'Power Saving' sub-tab is active. The 'Power Saving' dropdown menu is set to 'Enabled'. Below this, the 'Office Hour' section is visible, showing time ranges for each day of the week (Monday through Sunday). The 'Idle TimeOut (minutes)' section is also visible, with fields for 'Office Hour Idle TimeOut' (set to 960), 'Off Hour Idle TimeOut' (set to 10), and 'User Input Extension Idle TimeOut' (set to 10). A red box highlights the 'Office Hour' and 'Idle TimeOut' sections. At the bottom of the form, there are 'Confirm' and 'Cancel' buttons. On the right side, a 'NOTE' sidebar is present, titled 'Settings Powersaving', with a message: 'You can click here to get more guides.'

7. Click **Confirm** to accept the change.

## Backlight

Backlight determines the brightness of the LCD screen display, allowing users to read easily in dark environments. Backlight time specifies the delay time to change the intensity of the LCD screen when the IP phone is inactive. Backlight turns off quickly if a short backlight time is configured, this may not give users enough time to read messages. Backlight time is applicable to

SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones, EXP40 (connected to SIP-T48G/T48S/T46G/T46S IP phones) and EXP20 (connected to SIP-T29G/T27P/T27G IP phones).

Active Backlight Level/Active Level is used to adjust the backlight intensity of the LCD screen when the phone is active. Unused BackLight/Inactive Level is used to adjust the backlight intensity of the LCD screen when the phone is inactive. Active Backlight Level/Active Level is applicable to SIP-T54S/T52S IP phones and the connected EXP50, SIP-T48G/T48S/T46G/T46S IP phones and the connected EXP40, SIP-T29G/T27P/T27G IP phones and the connected EXP20. Unused BackLight/Inactive Level is only applicable to SIP-T54S/T52S IP phones,



SIP-T48G/T48S/T46G/T46S IP phones and the connected EXP40, SIP-T29G IP phones and the connected EXP20.

**Note**

It is not applicable to SIP-T19(P) E2 IP phones.

Before you adjust the LCD's backlight of expansion module, make sure the expansion module has been connected to the IP phone.

The following table lists available methods and configuration options to configure the backlight of phone models/expansion modules.

Phone Model (and the connected expansion module)	Configuration Methods	Configuration Options
T54S/T52S(EXP50) T48G/S(EXP40) T46G/S(EXP40) T29G(EXP20)	Configuration Files Web User Interface Phone User Interface	Unused BackLight/Inactive Level (not applicable to EXP50) Active Backlight Level/Active Level Backlight Time (not applicable to EXP50)
T27P/G(EXP20)	Configuration Files Web User Interface Phone User Interface	Active Backlight Level/Active Level Backlight Time
T42G/S T41P/S T40P/G T23P/G T21(P) E2	Configuration Files Web User Interface Phone User Interface	Backlight Time

**Procedure**

Backlight can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the backlight of the LCD screen. <b>Parameters:</b> phone_setting.active_backlight_level phone_setting.inactive_backlight_level phone_setting.backlight_time
<b>Web User Interface</b>		Configure the backlight of the LCD screen. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=setting">http://&lt;phoneIPAddress&gt;/servlet?p=setting</a>

	gs-preference&q=load
<b>Phone User Interface</b>	Configure the backlight of the LCD screen.

**Details of Configuration Parameters:**

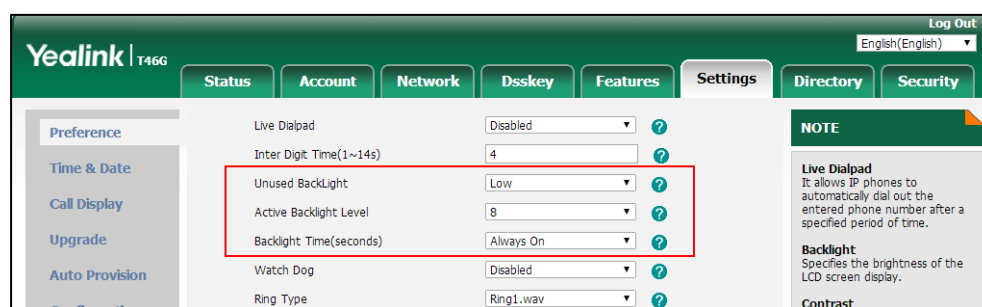
Parameters	Permitted Values	Default
<b>phone_setting.active_backlight_level</b>	<b>Integer from 1 to 10</b>	<b>8</b>
<p><b>Description:</b></p> <p>Configures the intensity of the LCD screen when the phone is active.</p> <p>10 is the highest intensity.</p> <p>For T54S/T52S IP phones, it configures the LCD's intensity of the IP phone and the connected EXP50.</p> <p>For T48G/T48S/T46G/T46S IP phones, it configures the LCD's intensity of the IP phone and the connected EXP40.</p> <p>For T29G/T27P/T27G IP phones, it configures the LCD's intensity of the IP phone and the connected EXP20.</p> <p><b>Note:</b> It is applicable to SIP-T54S/T52S IP phones and the connected EXP50, SIP-T48G/T48S/T46G/T46S IP phones and the connected EXP40, SIP-T29G/T27P/T27G IP phones and the connected EXP20.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Preference-&gt;Active Backlight Level</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Display-&gt;Backlight-&gt;Active Level</p>		
<b>phone_setting.inactive_backlight_level</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Configures the intensity of the LCD screen when the phone is inactive.</p> <p><b>0</b>-Off, it works only if the value of the parameter "phone_setting.backlight_time" is not set to 1 (Always On).</p> <p><b>1</b>-Low</p> <p>For T54S/T52S IP phones, it configures the LCD's intensity of the IP phone.</p> <p>For T48G/T48S/T46G/T46S IP phones, it configures the LCD's intensity of the IP phone and the connected EXP40.</p> <p>For T29G IP phones, it configures the LCD's intensity of the IP phone and the connected EXP20.</p> <p><b>Note:</b> It is applicable to SIP-T54S/T52S IP phones, SIP-T48G/T48S/T46G/T46S IP phones and the connected EXP40, SIP-T29G IP phones and the connected EXP20.</p>		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Settings->Preference->Unused BackLight <b>Phone User Interface:</b> Menu->Basic->Display->Backlight->Inactive Level		
<b>phone_setting.backlight_time</b>	<b>0, 1, 15, 30, 60, 120, 300, 600 or 1800</b>	<b>Refer to the following content</b>
<b>Description:</b> Configures the delay time (in seconds) to change the intensity of the LCD screen when the IP phone is inactive. <b>0</b> -Always On <b>1</b> -Always Off (not applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones) <b>15</b> -15s <b>30</b> -30s <b>60</b> -1min, the intensity of the LCD screen will be changed when the IP phone has been inactivated for 60 seconds. <b>120</b> -2min <b>300</b> -5min <b>600</b> -10min <b>1800</b> -30min <b>For SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G:</b> The default value is 0. <b>For SIP-T42G/T42S/T41P/T41S/T40P/T40G/T27P/T27G/T23P/T23G/T21(P) E2:</b> The default value is 30. <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> Settings->Preference->Backlight Time(seconds) <b>Phone User Interface:</b> Menu->Basic->Display->Backlight->Backlight Time		

**To configure the backlight via web user interface:**

1. Click on **Settings->Preference**.
2. Select the desired value from the pull-down list of **Unused BackLight**.
3. Select the desired value from the pull-down list of **Active Backlight Level**.

4. Select the desired value from the pull-down list of **Backlight Time(seconds)**.



5. Click **Confirm** to accept the change.

#### To configure the backlight via phone user interface:

1. Press **Menu->Basic->Display->Backlight**.
2. Press **◀** or **▶**, or the **Switch** soft key to select the desired level from the **Active Level** field.
3. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Inactive Level** field.
4. Press **◀** or **▶**, or the **Switch** soft key to select the desired time from the **Backlight Time** field.
5. Press the **Save** soft key to accept the change.

## Idle Clock Display

Idle clock is displayed on the center of the idle screen by default. You can hide the idle clock, and the time and date are displayed on the right of the status bar. It is only applicable to SIP-T54S IP phones.

### Procedure

Idle clock can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the idle clock. <b>Parameter:</b> phone_setting.idle_clock_display.enable
<b>Phone User Interface</b>		Configure the idle clock.

### Details of Configuration Parameter:

Parameter	Permitted Values	Default
phone_setting.idle_clock_display.enable	0 or 1	1

**Description:**

Enables or disables the IP phone to display the idle clock.

**0**-Disabled

**1**-Enabled

**Note:** It is only applicable to SIP-T54S IP phones.



**Web User Interface:**

None

**Phone User Interface:**

Menu->Basic->Display->Idle Clock->Idle Clock Display

**To configure the idle clock via phone user interface:**

1. Press **Menu->Basic->Display->Idle Clock**.
2. Press  or  , or the **Switch** soft key to select the desired value from the **Idle Clock Display** field.
3. Press the **Save** soft key to accept the change.

## Bluetooth

Bluetooth enables low-bandwidth wireless connections within a range of 10 meters (32 feet). The best performance is in the 1 to 2 meters (3 to 6 feet) range. You can activate/deactivate the Bluetooth mode and then pair and connect the Bluetooth headset/Bluetooth-Enabled mobile phone with your phone. After connecting the Bluetooth-Enabled mobile phone, you can synchronize the mobile contacts to the IP phone. For SIP-T54S/T52S IP phones, you can also make and receive mobile calls on IP phone. For more information, refer to the [Yealink phone-specific user guide](#). It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.

You can personalize the Bluetooth device name for the IP phone. The pre-configured Bluetooth device name will display in scanning list of other devices. It is helpful for the other Bluetooth devices to identify and pair with your IP phone.

### Note

To use this feature on SIP-T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones, make sure the Bluetooth USB dongle is properly connected to the USB port on the back of the phone. SIP-T52S/T54S IP phones have the built-in Bluetooth dongle.

You can check the Bluetooth dongle MAC address at the path:

**Menu->Status->More->Phone->Bluetooth MAC** (phone user interface) or  
**Status->Status->Network Common->Bluetooth MAC** (web user interface).

## Procedure

Bluetooth mode can be configured using the following methods.

Central Provisioning (Configuration File)	<y0000000000xx>.cfg	Configure Bluetooth mode. <b>Parameter:</b> features.bluetooth_enable
		Configure the Bluetooth device name. <b>Parameter:</b> features.bluetooth_adapter_name
Web User Interface		Configure Bluetooth mode. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-bluetooth&q=load
Phone User Interface		Configure Bluetooth mode. Configure the Bluetooth device name.

## Details of the Configuration Parameters:

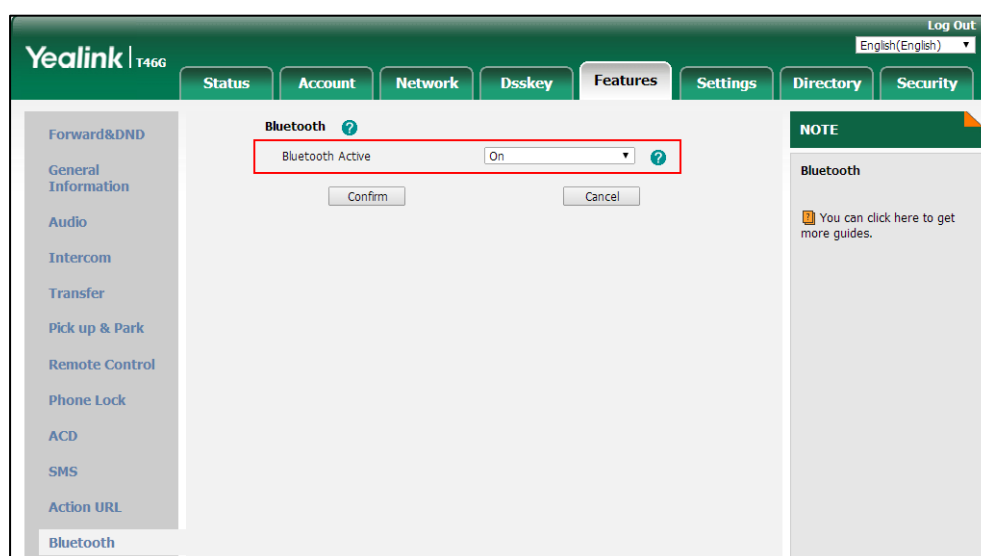
Parameters	Permitted Values	Default
<b>features.bluetooth_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Triggers Bluetooth mode to on or off. <b>0</b> -Off <b>1</b> -On <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones. <b>Web User Interface:</b> Features->Bluetooth->Bluetooth Active <b>Phone User Interface:</b> Menu->Basic->Bluetooth->Bluetooth		
<b>features.bluetooth_adapter_name</b>	<b>String within 64 characters</b>	<b>Refer to the following content</b>
<b>Description:</b> Configures the Bluetooth device name. <b>For SIP-T54S IP phones:</b>		

Parameters	Permitted Values	Default
<p>The default value is Yealink T54S.</p> <p><b>For SIP-T52S IP phones:</b></p> <p>The default value is Yealink T52S.</p> <p><b>For SIP-T48G IP phones:</b></p> <p>The default value is Yealink T48G.</p> <p><b>For SIP-T48S IP phones:</b></p> <p>The default value is Yealink T48S.</p> <p><b>For SIP-T46G IP phones:</b></p> <p>The default value is Yealink T46G.</p> <p><b>For SIP-T46S IP phones:</b></p> <p>The default value is Yealink T46S.</p> <p><b>For SIP-T42S IP phones:</b></p> <p>The default value is Yealink T42S.</p> <p><b>For SIP-T41S IP phones:</b></p> <p>The default value is Yealink T41S.</p> <p><b>For SIP-T29G IP phones:</b></p> <p>The default value is Yealink T29G.</p> <p><b>For SIP-T27G IP phones:</b></p> <p>The default value is Yealink T27G.</p> <p><b>Note:</b> It works only if the value of the parameter "features.bluetooth_enable" is set to 1 (On). It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Bluetooth-&gt;Bluetooth (On)-&gt;Edit My Device Information-&gt;Device Name</p>		

**To activate the Bluetooth mode via web user interface:**

1. Click on **Features->Bluetooth**.

2. Select the desired value from the pull-down list of **Bluetooth Active**.



3. Click **Confirm** to accept the change.

#### To activate the Bluetooth mode via phone user interface:

1. Press **Menu->Basic->Bluetooth**.
2. Press **◀** or **▶**, or the **Switch** soft key to select **On** from the **Bluetooth** field.
3. Press the **Save** soft key to accept the change.

#### To edit device information via phone user interface:

1. Press **Menu->Basic->Bluetooth**.
2. Press **◀** or **▶**, or the **Switch** soft key to select **On** from the **Bluetooth** field.
3. Press the **Save** soft key to accept the change.
4. Select **Edit My Device Information** and then press the **Enter** soft key.

The LCD screen displays the device name and MAC address. The MAC address cannot be edited.

5. Enter the desired name in the **Device Name** field.
6. Press the **Save** soft key to accept the change.

## Page Switch Key

The page switch key allows you to conveniently switch between pages of line keys on the phone.

Up to three pages are supported. By default, you can use the page switch key only if a line key not located on the first page is assigned functionality. The maximum of the line keys displaying on the first page depends on how many dedicated hard line keys your phone supports. For example, a SIP-T46G phone supports 10 dedicated hard line keys. It can display a maximum of 10 line keys on the first page. If the line key X (X is greater than 10) is assigned functionality, the 10th line key in the right bottom will automatically act as a page switch key.



You can configure the line key in the right bottom to be fixed as a switch page key. Then you can always use this key. It is especially useful for users who need to configure multi-page line keys.

Each page of the line keys has a corresponding page icon. For more information on page icon, refer to [Enable Page Tips](#) on page 183.

It is only applicable to SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T29G/T27P/T27G IP phones.

## Procedure

Page switch key can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the page switch key. <b>Parameter:</b> features.keep_switch_page_key.enable
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## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.keep_switch_page_key.enable	0 or 1	0
<p><b>Description:</b></p> <p>Enables or disables the line key in the bottom right of the phone to be fixed as a page switch key.</p> <p><b>0</b>-Disabled, the page icons appear only if a line key not located on the first page is assigned functionality. Then you can use the line key in the bottom right to switch pages.</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## Enable Page Tips




Enable page tips feature allows users to enable the page icon and page switch key LED on the phone to indicate different statuses. It is mainly used in the scenario of configuring multi-page line key. It is only applicable to

SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T29G/T27P/T27G IP phones.

The following shows the page icons:



The following table lists the page icons to indicate different statuses:

Phone Models	Icons	Description
SIP-T54S/T52S/T46G/T46S/T29G		Fast-flashing: the BLF monitored user receives an incoming call on the non-current page. Solid: there is a parked call to the line on the non-current page.
		Fast-flashing: the line receives an incoming call on the non-current page.
SIP-T42G/T42S/T41P/T41S/T27P/T27G		Fast-flashing: The BLF monitored user receives an incoming call on the non-current page. The line receives an incoming call on the non-current page. Solid: There is a parked call on the non-current page.

## Procedure

Enable page tips can be configured using the following methods.

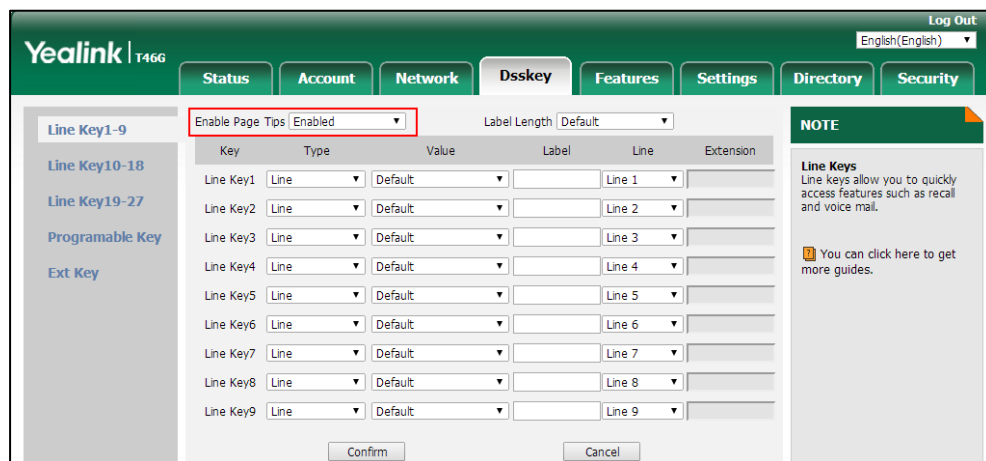
<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure enable page tips. <b>Parameter:</b> phone_setting.page_tip
<b>Web User Interface</b>		Configure enable page tips. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?parameter=dsskey&amp;model=1&amp;q=load&amp;linepage=1">http://&lt;phoneIPAddress&gt;/servlet?parameter=dsskey&amp;model=1&amp;q=load&amp;linepage=1</a>

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>phone_setting.page_tip</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the page icon and page switch key LED to indicate different states of line keys on the non-current page.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key-&gt;Enable Page Tips</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure the page icon to indicate status via web user interface:**

1. Click on **Dsskey->Line Key**.
2. Select **Enabled** from the pull-down list of **Enable Page Tips**.



3. Click **Confirm** to accept the change.

## Page Tips for Expansion Module

You are allowed to configure the page switch key LED on the expansion module to indicate when BLF monitored user receives an incoming call on the non-current page. It is only applicable to EXP50 connected to the SIP-T54S/T52S IP phones.

The following table lists the page switch key LED to indicate different statuses:

LED Status	Description
Off (non-configurable)	Indicates non-current pages.
Solid green (non-configurable)	Indicates current page.
Flashing red	The BLF monitored user receives an incoming call on the non-current pages.

You can also customize the page switch key LED status when BLF monitored user receives an incoming call on the non-current pages.

## Procedure

Page tips for expansion module can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx> .cfg	Configure page tips for the page switch keys for the expansion modules. <b>Parameters:</b> expansion_module.page_tip.blf_call_in.enable expansion_module.page_tip.blf_call_in.led
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## Details of the Configuration Parameters:

Parameters	Permitted Values	Default
<b>expansion_module.page_tip.blf_call_in.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the page switch key LED on the expansion module to indicate when BLF monitored user receives an incoming call on the non-current pages. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It is only applicable to EXP50 expansion modules connected to the SIP-T54S/T52S IP phones. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>expansion_module.page_tip.blf_call_in.led</b>	<b>String</b>	<b>\$LEDr30 0o300\$</b>
<b>Description:</b>		

Parameters	Permitted Values	Default
<p>Configures the page switch key LED status on the expansion module when BLF monitored user receives an incoming call on the non-current pages.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p>If it is left blank, the default value takes effect.</p> <p><b>Example:</b></p> <pre>expansion_module.page_tip.blf_call_in.led = \$LED300o300\$</pre> <p>It means an infinite loop for page switch key LED status: illuminates solid red for 300ms and then goes out for 300ms.</p> <p><b>Note:</b> It works only if the value of the parameter "expansion_module.page_tip.blf_call_in.enable" is set to 1 (Enabled). It is only applicable to EXP50 expansion modules connected to the SIP-T54S/T52S IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## Label Length

Label length allows IP phones to extend the display length of the line key label on the idle screen. If the label length feature is enabled, more characters will be displayed on the idle LCD screen. Label length is only applicable to SIP-T54S/T48G/T48S/T46G/T46S/T29G IP phones.

When label length feature is set to **Default**:



SIP-T46G



SIP-T48G

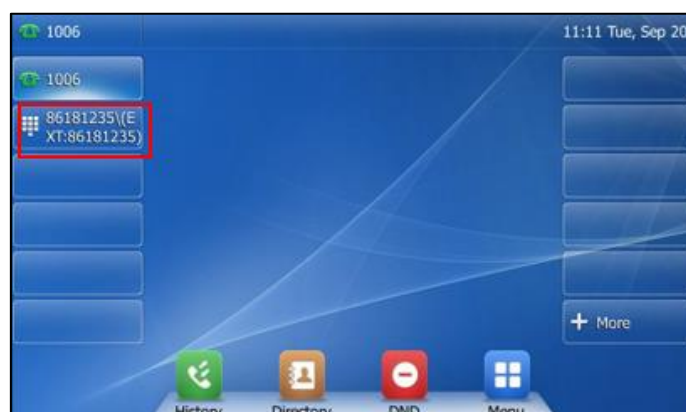
When label length feature is set to **Extended**:

For SIP-T54S/T46G/T46S/T29G IP phones, the display length of the line key label is extended and the characters are displayed in one line.



SIP-T46G

For SIP-T48G/S IP phones, the characters are displayed in two lines.



SIP-T48G

When label length feature is set to **Mid-range** (not applicable to SIP-T48G IP phones):



SIP-T46G

## Procedure

Label length can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure label length. <b>Parameter:</b> features.config_dsskey_length
<b>Web User Interface</b>		Configure label length. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;model=1&amp;q=load&amp;linepage=1">http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;model=1&amp;q=load&amp;linepage=1</a>

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.config_dsskey_length	0, 1 or 2	0
<b>Description:</b> Configures the label length displayed on the idle LCD screen for the line key. <b>0</b> -Default <b>1</b> -Extended <b>2</b> -Mid-range (only applicable to SIP-T54S/T46G/T46S/T29G IP phones) <b>Note:</b> It is only applicable to SIP-T54S/T48G/T48S/T46G/T46S/T29G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Label Length <b>Phone User Interface:</b> None		

**To configure the label length via web user interface:**

1. Click on **Dsskey**->**Line Key**.
2. Select **Extended** from the pull-down list of **Label Length**.

The screenshot shows the Yealink T46G web interface. The top navigation bar includes tabs for Status, Account, Network, Dsskey, Features, Settings, Directory, and Security. The 'Dsskey' tab is active. On the left, a sidebar lists 'Line Key1-9', 'Line Key10-18', 'Line Key19-27', 'Programable Key', and 'Ext Key'. The main area shows a configuration table for Line Keys. At the top of this table, there is a dropdown menu for 'Label Length' which is currently set to 'Extended'. The table has columns for Key, Type, Value, Label, Line, and Extension. Below the table are 'Confirm' and 'Cancel' buttons. On the right, a 'NOTE' section explains that Line Keys allow quick access to features like recall and voice mail, and provides a link to more guides.

3. Click **Confirm** to accept the change.

## Linekey Length in Short

Linekey length in short allows IP phones to shorten the line key label when the phone places a call, receives a call, is during a call, or is idle (if large DND icon is enabled). You can configure the phone to display 2 characters or nothing via linekey length in short feature. It is only applicable to SIP-T52S IP phones.

If you configure the phone to display 2 characters, the display will match the rules below:

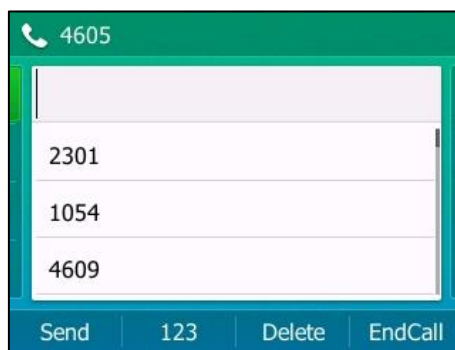
Label	Short Label	Description
Any string or Blank	Configured (for example, aLine1)	"Short Label" has higher priority than "Label". If the value of "Short Label" is configured, the value of "Label" is ignored. Display the first two characters of the short label (for example, aL).
Configured (for example, aLine1) (for example, sunmy)	Blank	• If the label contains uppercase characters or digits, only the uppercase characters and digits are displayed. It displays two characters from left to right (for example, L1).
Blank	Blank	• else, it displays the first two characters of the label (for example, su). If both left blank, the default label is used. The display rule is the same as above.



The following figure shows an example of label display when it is set to 2 chars:



The following figure shows an example of label display when it is set to No Label:



## Procedure

Linekey length in short can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure linekey length in short. <b>Parameter:</b> features.config_dsskey_length.shor ten
<b>Web User Interface</b>		Configure linekey length in short. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;model=1&amp;q=load&amp;linepage=1">http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;model=1&amp;q=load&amp;linepage=1</a>

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.config_dsskey_length.shorten	0 or 1	1
<b>Description:</b> Configures the label length in short of line key displayed on LCD screen when the phone		

Parameter	Permitted Values	Default
<p>places a call, receives a call, is during a call, or is idle (if large DND icon is enabled).</p> <p><b>0</b>-No Label, you can press the line key to display the line key label when the label length is shorten.</p> <p><b>1-2</b> chars, the exceeding characters will be ignored.</p> <p><b>Note:</b> It is only applicable to SIP-T52S IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key-&gt;Linekey length in short</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure the linekey length in short via web user interface:**

1. Click on **Dsskey->Line Key**.
2. Select the desired value from the pull-down list of **Linekey Length in short**.

The screenshot shows the Yealink T52S web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'Dsskey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Dsskey' tab is selected. On the left sidebar, 'Line Key1-7' is selected. The main area shows a configuration for 'Linekey length in short' with a dropdown menu set to '2 chars'. Below this is a table for configuring Line Keys (Line Key1 to Line Key7). The table has columns for Key, Type, Value, Label, Short, Line, and Extension. The 'Short' column is currently empty. At the bottom of the table are 'Confirm' and 'Cancel' buttons. On the right sidebar, there is a 'NOTE' section titled 'Line Keys' with a brief description and a link to get more guides.

3. Click **Confirm** to accept the change.

## Account Registration

Registering an account makes it easier for the IP phones to receive an incoming call or dial an outgoing call. Yealink IP phones support registering multiple accounts on a phone; each account requires an extension or phone number.

The number of the registered accounts must meet the following:

Phone Model	Accounts
SIP-T54S/T48G/T48S/T46G/T46S/T29G	<=16
SIP-T52S/T42G/T42S	<=12
SIP-T41P/T41S/T27P/T27G	<=6
SIP-T40P/T40G/T23P/T23G	<=3

Phone Model	Accounts
SIP-T21(P) E2	<=2
SIP-T19(P) E2	1

The IP phones support SIP server redundancy for account registration. For more information, refer to [Server Redundancy](#) on page 650. If you want to customize multiple DSS keys to associate with an account, refer to [Multiple Line Keys per Account](#) on page 199. If you want to configure the maximum number of concurrent calls per line key, refer to [Multiple Call Appearances](#) on page 206.

## Procedure

SIP Account registration can be configured using the following methods.

Central Provisioning (Configuration File)	<MAC>.cfg	<p>Configure the SIP account registration information.</p> <p><b>Parameters:</b></p> <p>account.X.enable</p> <p>account.X.label</p> <p>account.X.display_name</p> <p>account.X.auth_name</p> <p>account.X.user_name</p> <p>account.X.password</p> <p>account.X.sip_server.Y.address</p> <p>account.X.sip_server.Y.port</p> <p>account.X.outbound_proxy_enable</p> <p>account.X.outbound_proxy.Y.address</p> <p>account.X.outbound_proxy.Y.port</p>
		<p>Configure the interval for the IP phone to retry to re-register when registration fails.</p> <p><b>Parameter:</b></p> <p>account.X.reg_fail_retry_interval</p>
Web User Interface		<p>Configure the SIP account registration information.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=account-register&amp;q=load&amp;acc=0</p>

	<p>Configure the interval for the IP phone to retry to register when registration fails.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</p>
<b>Phone User Interface</b>	Configure the SIP account registration information.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>account.X.enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the account X.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Register-&gt;Line Active</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Accounts-&gt;Activation</p>		
<b>account.X.label<sup>[1]</sup></b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>(Optional.) Configures the label to be displayed on the LCD screen for a specific account.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Register-&gt;Label</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Accounts-&gt;Label</p>		
<b>account.X.display_name<sup>[1]</sup></b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the display name to be displayed on the called party's LCD screen for a specific account.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Register-&gt;Display Name</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Accounts->Display Name		
<b>account.X.auth_name<sup>[1]</sup></b>	<b>String within 99 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the user name for register authentication for a specific account. <b>Note:</b> The user name for register authentication is provided by ITSP. It is always matched with a password (configured by the parameter "account.X.password") used for register authentication, if required by the server. <b>Web User Interface:</b> Account->Register->Register Name <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Accounts->Register Name		
<b>account.X.user_name<sup>[1]</sup></b>	<b>String within 99 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the register user name for a specific account. <b>Note:</b> The register user name is provided by ITSP. It is used to identify the account. <b>Web User Interface:</b> Account->Register->User Name <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Accounts->User Name		
<b>account.X.password<sup>[1]</sup></b>	<b>String within 99 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the password for register authentication for a specific account. <b>Note:</b> The password for register authentication is provided by ITSP. <b>Web User Interface:</b> Account->Register->Password <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Accounts->Password		
<b>account.X.sip_server.Y.address<sup>[1]</sup></b> (X ranges from 1 to 16, Y ranges from 1 to 2)	<b>String within 256 characters</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<b>Description:</b> Configures the IP address or domain name of the SIP server Y that accepts registrations for a specific account. <b>Example:</b> account.1.sip_server.1.address = 10.2.1.48 <b>Web User Interface:</b> Account->Register->SIP Server Y->Server Host <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Accounts->SIP ServerY		
<b>account.X.sip_server.Y.port<sup>[1]</sup></b> (X ranges from 1 to 16, Y ranges from 1 to 2)	<b>Integer from 0 to 65535</b>	<b>5060</b>
<b>Description:</b> Configures the port of the SIP server Y that specifies registrations for a specific account. <b>Example:</b> account.1.sip_server.1.port = 5060 <b>Note:</b> If the value of this parameter is set to 0, the port used depends on the value specified by the parameter "account.X.sip_server.Y.transport_type". <b>Web User Interface:</b> Account->Register->SIP Server Y->Port <b>Phone User Interface:</b> None		
<b>account.X.outbound_proxy_enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to send requests to the outbound proxy server for a specific account. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Account->Register->Enable Outbound Proxy Server <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Accounts->Outbound Status		
<b>account.X.outbound_proxy.Y.address<sup>[1]</sup></b>	<b>IP address or domain name</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the IP address or domain name of the outbound proxy server Y for a specific account.</p> <p><b>Example:</b></p> <p>account.1.outbound_proxy.1.address = 10.1.8.11</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.outbound_proxy_enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Register-&gt;Outbound Proxy Server Y</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Accounts-&gt;Outbound ProxyY</p>		
<b>account.X.outbound_proxy.Y.port<sup>[1]</sup></b>	<b>Integer from 0 to 65535</b>	<b>5060</b>
<p><b>Description:</b></p> <p>Configures the port of the outbound proxy server Y for a specific account.</p> <p><b>Example:</b></p> <p>account.1.outbound_proxy.1.port = 5060</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.outbound_proxy_enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Register-&gt;Outbound Proxy Server Y-&gt;Port</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.reg_fail_retry_interval<sup>[1]</sup></b>	<b>Integer from 0 to 1800</b>	<b>30</b>
<p><b>Description:</b></p> <p>Configures the interval (in seconds) for the IP phone to retry to re-register account X when registration fails.</p> <p><b>Example:</b></p> <p>account.1.reg_fail_retry_interval = 30</p> <p><b>Note:</b> It works only if the values of the parameters "account.X.reg_failed_retry_min_time" and "account.X.reg_failed_retry_max_time" are set to 0.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;SIP Registration Retry Timer(0~1800s)</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b>		
None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

#### To register an account via web user interface:

1. Click **Account->Register**.
2. Select the desired account from the pull-down list of **Account**.
3. Select **Enabled** from the pull-down list of **Line Active**.
4. Enter the desired value in **Label**, **Display Name**, **Register Name**, **User Name**, **Password** and **SIP Server1/2** field respectively.

The screenshot shows the Yealink T46G web interface for account registration. The 'Account' tab is selected, and the 'Register' sub-tab is active. The 'Account' dropdown is set to 'Account1'. The 'Line Active' dropdown is set to 'Enabled'. The 'Label', 'Display Name', 'Register Name', 'User Name', and 'Password' fields are all set to '101'. The 'SIP Server 1' section shows 'Server Host' as '10.2.1.48' and 'Port' as '5060'. The 'SIP Server 2' section is also visible with 'Server Host' and 'Port' fields. The 'Enable Outbound Proxy Server' dropdown is set to 'Disabled'. The 'NAT' dropdown is set to 'STUN'. A 'NOTE' section on the right provides information about Account Registration, Server Redundancy, and NAT Traversal.

5. If you use outbound proxy servers, do the following:
  - 1) Select **Enabled** from the pull-down list of **Enable Outbound Proxy Server**.
  - 2) Enter the desired IP address or domain name in the **Outbound Proxy Server 1/2** field and the desired port of the outbound proxy server 1/2 in the **Port** field respectively.
6. Click **Confirm** to accept the change.



To configure the interval for re-register when registration fails via web user interface:

1. Click **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Enter the desired interval in the **SIP Registration Retry Timer(0~1800s)** field.

4. Click **Confirm** to accept the change.

To register an account via phone user interface:

1. Press **Menu->Advanced** (default password: admin) -> **Accounts**.
2. Select the desired account and then press the **Enter** soft key.
3. Select **Enabled** from the **Activation** field.
4. Enter the desired value in **Label**, **Display Name**, **Register Name**, **User Name**, **Password** and **SIP Server1/2** field respectively. Contact your system administrator for more information.
5. If you use outbound proxy servers, do the following:
  - 1) Select **Enabled** from the **Outbound Status** field.
  - 2) Enter the desired IP address or domain name in the **Outbound Proxy1/2** field.
6. Press the **Save** soft key to accept the change.

## Multiple Line Keys per Account

You can customize the number of DSS keys to be automatically assigned with Line type. It means multiple DSS keys will associate with an account. It is useful for managing a high volume of calls to a line. These line keys use its Account Label (refer to [Account Registration](#)) as default label to display on the screen. In order to distinguish these line keys, you can customize the line key labels one by one or configure the Auto Label rule for these line keys.

For more information on how to register accounts, refer to [Account Registration](#) on page 192. If

you want to configure maximum number of concurrent calls per line key, refer to [Multiple Call Appearances](#) on page 206. It is not applicable to SIP-T19(P) E2 IP phones.

The number of the DSS keys associated with an account must meet the following:

Phone Model	Line Key	Ext Key (with expansion modules connected)
SIP-T54S	<=27	<=180
SIP-T52S	<=21	<=180
SIP-T48G/S	<=29	<=240
SIP-T46G/S	<=27	<=240
SIP-T29G	<=27	<=228
SIP-T27P/G	<=21	<=228
SIP-T42G/T42S/T41P/T41S	<=15	/
SIP-T40P/T40G/T23P/T23G	<=3	/
SIP-T21(P) E2	<=2	/

The following shows two line keys associated with a registered account 101:



When you customize multiple line keys to be associated with an account, you can configure the IP phone whether to transfer a call or set up a conference call using a new line key.

## Procedure

Multiple line keys per account can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<y0000000000xx>.cfg	Configure auto linekeys. <b>Parameter:</b> features.auto_linekeys.enable
		Configure whether to transfer a call using a new line key on a phone basis. <b>Parameter:</b> phone_setting.call_appearance.transfer_vi

		a_new_linekey
		<p>Configure whether to set up a conference call using a new line key on a phone basis.</p> <p><b>Parameter:</b></p> <p>phone_setting.call_appearance.conference_via_new_linekey</p>
	<MAC>.cfg	<p>Configure the number of DSS keys to be assigned automatically.</p> <p><b>Parameter:</b></p> <p>account.X.number_of_linekey</p>
		<p>Configure the Auto Label feature.</p> <p><b>Parameters:</b></p> <p>account.X.auto_label.enable</p> <p>account.X.auto_label.rule</p>
		<p>Configure whether to transfer a call using a new line key on a per-line basis.</p> <p><b>Parameter:</b></p> <p>account.X.phone_setting.call_appearance.transfer_via_new_linekey</p>
<b>Web User Interface</b>		<p>Configure auto linekeys.</p> <p><b>Navigate to:</b></p> <p><a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load</a></p>
		<p>Configure the number of DSS keys to be assigned automatically.</p> <p><b>Navigate to:</b></p> <p><a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a></p>

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>features.auto_linekeys.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the DSS keys to be assigned with Line type automatically.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> The number of the DSS keys is determined by the value of the parameter "account.X.number_of_linekey". It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b> Features-&gt;General Information-&gt;Auto Linekeys</p> <p><b>Phone User Interface:</b> None</p>		
<b>account.X.number_of_linekey<sup>[1]</sup></b>	<b>Integer from 1 to 999</b>	<b>1</b>
<p><b>Description:</b> Configures the number of DSS keys to be assigned with Line type automatically from the first unused one (unused one means the DSS key is configured as N/A or Line). If a DSS key is used, the IP phone will skip to the next unused DSS key.</p> <p>The order of DSS key assigned automatically is Line Key-&gt;Ext Key.</p> <p><b>Example:</b> account.1.number_of_linekey = 1</p> <p><b>Note:</b> It works only if the value of the parameter "features.auto_linekeys.enable" is set to 1 (Enabled). To assign Ext Key, make sure the expansion module has been connected to the phone in advance. It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b> Account-&gt;Advanced-&gt;Number of line key</p> <p><b>Phone User Interface:</b> None</p>		
<b>account.X.auto_label.enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the Auto Label feature. It is only applicable to the automatically assigned line DSS keys.</p>		

<p><b>0</b>-Disabled, the line keys labels are determined by the value of the parameter "account.X.label".</p> <p><b>1</b>-Enabled, the line keys labels are determined by the custom label rule (configure by the parameter "account.X.auto_label.rule").</p> <p><b>Note:</b> It works only if the value of the parameter "features.auto_linekeys.enable" is set to 1 (Enabled). It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>account.X.auto_label.rule<sup>[1]</sup></b>	<b>String within 99 characters</b>	<b>{L}_{1}</b>
<p><b>Description:</b> Configures the Auto Label rule.</p> <p>You need to know the following basic regular expression syntax:</p> <p><b>{L}</b>: The value is configured by the parameter "account.X.label".</p> <p><b>{N}</b>: An increasing number from N. For example, abc{1}{5} represents the following labels: abc15, abc26, abc37, and so on.</p> <p>Multiple labels are separated by " ". For example, Yea Yea Yea Tom_{2} means to display "Yea" for first three line keys, and from the fourth one, display label Tom_2, Tom_3, and so on in turn.</p> <p><b>Other Characters:</b> for example, ABC, will display ABC same as what you have configured.</p> <p><b>Note:</b> It works only if the values of the parameters "features.auto_linekeys.enable" and "account.X.auto_label.enable" are set to 1 (Enabled). It is not applicable to SIP-T19(P) E2 IP phones. The number of valid labels is configured by the parameter "account.X.number_of_linekey".</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>phone_setting.call_appearance.transfer_via_new_linekey</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the IP phone to transfer a call using a new line key when multiple line keys are associated with an account.</p> <p><b>0</b>-Disabled, the IP phone will transfer a call using the current line key.</p> <p><b>1</b>-Enabled, the IP phone will transfer a call by automatically selecting a new line key (the corresponding line key is not seized) instead of the current line key. If all line keys are seized,</p>		

the current line key will be used.

**Note:** The number of the line keys is determined by the value of the parameter "account.X.number\_of\_linekey". The value configured by the parameter "account.X.phone\_setting.call\_appearance.transfer\_via\_new\_linekey" takes precedence over that configured by this parameter. It is not applicable to SIP-T19(P) E2 IP phones.

**Web User Interface:**

None

**Phone User Interface:**

None

<b>phone_setting.call_appearance.conference_via_new_linekey</b>	<b>0 or 1</b>	<b>1</b>
-----------------------------------------------------------------	---------------	----------

**Description:**

Enables or disables the IP phone to set up a conference call using a new line key when multiple line keys are associated with an account.

**0**-Disabled, the IP phone will place a new call using the current line key when pressing the **Conf/Conference** soft key.

**1**-Enabled, the IP phone will place a new call by automatically selecting a new line key (the corresponding line key is not seized) when pressing the **Conf/Conference** soft key. If all line keys are seized, the current line key will be used.

**Note:** The number of the line keys is determined by the value of the parameter "account.X.number\_of\_linekey". The value configured by the parameter "account.X.phone\_setting.call\_appearance.conference\_via\_new\_linekey" takes precedence over that configured by this parameter. It is not applicable to SIP-T19(P) E2 IP phones.

**Web User Interface:**

None

**Phone User Interface:**

None

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2.

**Per-Line Parameters:**

Two parameters listed in the above table have a per-line equivalent that you can configure. The per-line parameters are listed in the following table. Note that the per-line parameter takes precedence over the global parameter.

X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

Per-Line Parameters	Global Parameters
account.X.phone_setting.call_appearance.transfer_via_new_linekey	phone_setting.call_appearance.transfer_via_new_linekey
account.X.phone_setting.call_appearance.conference_via_new_linekey	phone_setting.call_appearance.transfer_via_new_linekey

To configure auto linekeys feature via web user interface:

1. Click on **Features->General Information**.
2. Select **Enabled** from the pull-down list of **Auto Linekeys**.

If **Auto LineKeys** is enabled, you can automatically assign multiple DSS keys with Line type for a registered line on the phone.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'Auto Linekeys' setting is set to 'Enabled' and is highlighted with a red rectangular box. Other settings visible include 'Call Waiting' (Enabled), 'Auto Redial' (Disabled), 'Voice Mail Tone' (Enabled), 'DHCP Hostname' (SIP-T46G), 'Reboot in Talking' (Disabled), 'Hide Feature Access Codes' (Disabled), and 'Display Method on Dialing' (User Name). A 'NOTE' section on the right provides additional information about various features.

3. Click **Confirm** to accept the change.

To configure the number of line keys via web user interface:

1. Click **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Enter the desired number in the **Number of line key** field.

This field appears only if **Auto Linekeys** is enabled.

4. Click **Confirm** to accept the change.

## Multiple Call Appearances

You can enable each registered line on the phone to support multiple concurrent calls. For example, you can place one call on hold, switch to another call on the same registered line, and have both calls display.

You can set the maximum number of concurrent calls per line key on a phone or a per-line basis. For example, if you configure the maximum number of concurrent calls per line key for account 1 to three, you can only have three call appearances on line 1. The additional incoming calls will be rejected. If you assign a registered line to multiple line keys (refer to [Multiple Line Keys per Account](#)), the number of concurrent calls applies to all line keys. That is, you can have three call appearances per line key.

### Procedure

Multiple call appearances can be configured using the configuration file.

<b>Central Provisioning (Configuration Files)</b>	<y0000000000xx>.cf g	Specify the number of concurrent calls for each line key on a phone basis. <b>Parameter:</b> phone_setting.call_appearance.calls_per_linekey
	<MAC>.cfg	Specify the number of concurrent calls for each line key on a per-line basis. <b>Parameter:</b> account.X.phone_setting.call_appearance.ca



		lls_per_linekey
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**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>phone_setting.call_appearance.calls_per_linekey</b>	<b>Integer from 0 to 24</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the maximum number of concurrent calls per line key for the IP phone. It applies to all registered lines.</p> <p>If it is set to 0, there is no limit for the number of concurrent calls.</p> <p><b>Example:</b></p> <p>phone_setting.call_appearance.calls_per_linekey = 2</p> <p>It means that you can have up to two concurrent calls per line key on the IP phone.</p> <p><b>Note:</b> The value configured by the parameter "account.X.phone_setting.call_appearance.calls_per_linekey" takes precedence over that configured by this parameter. It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**Per-Line Parameter:**

"account.X.phone\_setting.call\_appearance.calls\_per\_linekey" is the per-line parameter of the global parameter "phone\_setting.call\_appearance.calls\_per\_linekey", and it takes precedence over the global parameter.

X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2.

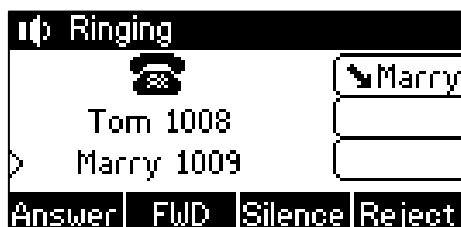
## Call Display

Display contact photo allows the IP phone to present the contact avatar when it receives an incoming call, dials an outgoing call or engages in a call. To use this feature, make sure that you have uploaded the avatar for the contact in advance. For more information, refer to [Customizing a Local Contact File \(Color Screen Phones\)](#) on page 324. Display contact photo feature is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.

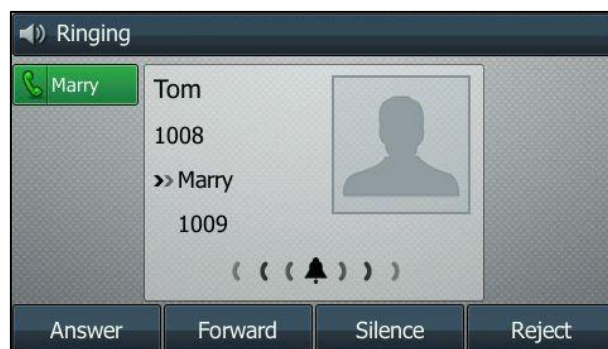
Display called party information allows the IP phone to present the callee identity in addition to the presentation of caller identity when it receives an incoming call.

The following figure shows an example of screen display when Display Called Party Information feature is enabled on the phone (a call from Tom (phone number: 1008) to Marry (phone number: 1009)).

For SIP-T42G/T42S/T41P/T41S/T40P/T40G/T27P/T27G/T23P/T23G/T21(P) E2/T19(P) E2 IP phones:



For SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones:



You can customize the call information to be displayed on the IP phone as required. IP phones support six call information display methods: Number+Name, Name, Name+Number, Number, Full Contact Info (display name< sip:xxx@domain.com>) or Null.

The caller information may be too long to be entirely displayed on the phone in a short time. You can customize the horizontal scroll speed of the caller information when the phone is ringing, so that you can check the caller information more quickly before picking up the call.

#### Note

SIP-T42G/T42S/T41P/T41S/T40P/T40G/T23P/T23G/T21(P) E2/T19(P) E2 IP phones have a limited display (up to three lines) due to their smaller screen size.

## Procedure

Call Display can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure display contact photo feature. <b>Parameter:</b> phone_setting.contact_photo_display.enable
		Configure display called party information feature.

		<b>Parameter:</b> phone_setting.called_party_info_display.enable
		Specify the call information display method. <b>Parameter:</b> phone_setting.call_info_display_method
		Specify the horizontal scroll speed of caller information. <b>Parameter:</b> phone_setting.incoming_call.horizontal_roll_interval
<b>Web User Interface</b>	Configure display contact photo feature. Configure display called party information feature. Specify the call information display method. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=settings-calldisplay&q=load	

### Details of Configuration Parameters:

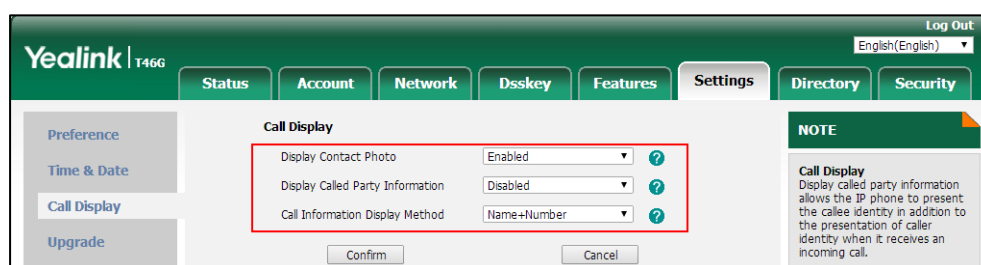
Parameters	Permitted Values	Default
phone_setting.contact_photo_display.enable	0 or 1	1
<b>Description:</b> Enables or disables the IP phone to display contact avatar when it receives an incoming call, dials an outgoing call or engages in a call. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones. <b>Web User Interface:</b> Settings->Call Display->Display Contact Photo <b>Phone User Interface:</b> None		
phone_setting.called_party_info_display.enable	0 or 1	0

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Enables or disables the IP phone to display the called party information when receiving an incoming call.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> The information display method is configured by the parameter "phone_setting.call_info_display_method".</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Call Display-&gt;Display Called Party Information</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>phone_setting.call_info_display_method</b>	<b>0, 1, 2, 3, 4 or 5</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the call information display method when the IP phone receives an incoming call, dials an outgoing call or is during an active call.</p> <p><b>0</b>-Name+Number <b>1</b>-Number+Name <b>2</b>-Name <b>3</b>-Number <b>4</b>-Full Contact Info (display name&lt;sip:xxx@domain.com&gt;) <b>5</b>-Null</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Call Display-&gt;Call Information Display Method</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>phone_setting.incoming_call.horizontal_roll_interval</b>	<b>Integer from 100 to 2000</b>	<b>500</b>
<p><b>Description:</b></p> <p>Configures the interval (in milliseconds) for the IP phone to horizontally scroll the caller information when the phone is ringing.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
None		

To configure call display features via web user interface:

1. Click on **Settings->Call Display**.
2. Select the desired value from the pull-down list of **Display Contact Photo**.
3. Select the desired value from the pull-down list of **Display Called Party Information**.
4. Select the desired value from the pull-down list of **Call Information Display Method**.



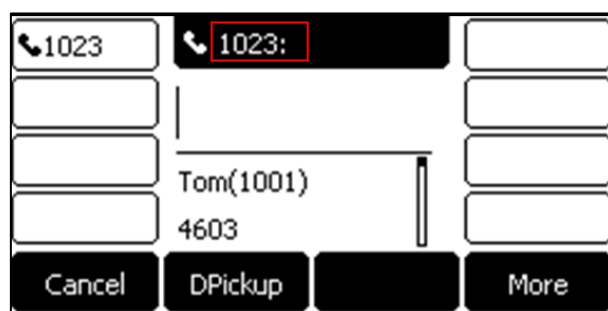
5. Click **Confirm** to accept the change.

## Display Method on Dialing

When the IP phone is on the pre-dialing or dialing screen, the account information will be displayed on the LCD screen.



SIP-T46G



SIP-T27G

You can customize the account information to be displayed on the IP phone as required. IP phones support three account information display methods: Label, Display Name or User Name.

### Procedure

Display method on dialing can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure display method on dialing. <b>Parameter:</b> features.caller_name_type_on_dialing
<b>Web User Interface</b>		Configure display method on dialing. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load</a>

### Details of Configuration Parameter:

Parameter	Permitted Values	Default
features.caller_name_type_on_dialing	1, 2 or 3	3
<b>Description:</b> Configures the selected line information displayed on the pre-dialing or dialing screen. <b>1-Label</b> <b>2-Display Name</b> <b>3-User Name</b> <b>Web User Interface:</b> Features->General Information->Display Method on Dialing <b>Phone User Interface:</b> None		

### To configure display method on dialing via web user interface:

1. Click on **Features->General Information**.

- Select the desired value from the pull-down list of **Display Method on Dialing**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected. In the 'General Information' section, the 'Display Method on Dialing' dropdown menu is highlighted with a red box and set to 'User Name'. Other settings include Call Waiting (Enabled), Auto Redial (Disabled), and various intervals. A 'NOTE' sidebar on the right provides additional information about features like Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

- Click **Confirm** to accept the change.

## Time and Date

IP phones maintain a local clock. The time and date can be displayed in several formats on the idle screen of IP phones. You can select one of the default time/date formats or customize the date format.

There are 2 available time formats: "12 Hour" or "24 Hour".

There are 7 available date formats by default. For example, for the date format "WWW DD MMM", "WWW" represents the abbreviation of the weekday, "DD" represents the two-digit day, and "MMM" represents the first three letters of the month.

The date formats available:

Date Format	Example (2016-09-02)
WWW MMM DD	Fri, Sep 02
DD-MMM-YY	02-Sep-16
YYYY-MM-DD	2016-09-02
DD/MM/YYYY	02/09/2016
MM/DD/YY	09/02/16
DD MMM YYYY	02 Sep, 2016
WWW DD MMM	Fri, 02 Sep

Yealink IP phones also support customizing date format. For example, YYYY-MMM-DDD-WWW, and W,MD, and so on. For more information, refer to [Time and Date Settings](#) on page 219.

The following table lists available configuration methods for time and date.

Option	Configuration Methods
NTP time server	Configuration Files Web User Interface Phone User Interface
Time Zone	Configuration Files Web User Interface Phone User Interface
Time	Web User Interface Phone User Interface
Time Format	Configuration Files Web User Interface Phone User Interface
Date	Web User Interface Phone User Interface
Date Format	Configuration Files Web User Interface Phone User Interface
Date Format (custom)	Configuration Files
Daylight Saving Time	Configuration Files Web User Interface

## NTP Time Server

A time server is a computer server that reads the actual time from a reference clock and distributes this information to the clients in a network. The Network Time Protocol (NTP) is the most widely used protocol that distributes and synchronizes time in the network.

The IP phones synchronize the time and date automatically from the NTP time server by default. The NTP time server address can be offered by the DHCP server or configured manually. NTP by DHCP Priority feature can configure the priority for the IP phone to use the NTP time server address offered by the DHCP server or configured manually.

### Time Zone

A time zone is a region on Earth that has a uniform standard time. It is convenient for areas in close commercial or other communication to keep the same time. When configuring the IP phone to obtain the time and date from the NTP time server, you must set the time zone.



## Procedure

NTP time server and time zone can be configured using the following methods.

Central Provisioning (Configuration File)	<MAC>.cfg	Configure NTP by DHCP priority feature and DHCP time feature. <b>Parameters:</b> local_time.manual_ntp_srv_prior local_time.dhcp_time
		Configure the NTP server, time zone. <b>Parameters:</b> local_time.ntp_server1 local_time.ntp_server2 local_time.interval local_time.time_zone local_time.time_zone_name
Web User Interface	Configure NTP by DHCP priority feature and DHCP time feature.  Configure the NTP server, time zone. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=settings-datetime&q=load	
Phone User Interface	Configure DHCP time feature.  Configure the NTP server, time zone.	

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
local_time.manual_ntp_srv_prior	0 or 1	0
<b>Description:</b> Configures the priority for the IP phone to use the NTP server address offered by the DHCP server.  <b>0</b> -High (use the NTP server address offered by the DHCP server preferentially) <b>1</b> -Low (use the NTP server address configured manually preferentially)  <b>Web User Interface:</b> Settings->Time & Date->NTP by DHCP Priority  <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>local_time.dhcp_time</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to update time with the offset time offered by the DHCP server.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It is only available to offset from Greenwich Mean Time (GMT).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Time &amp; Date-&gt;DHCP Time</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Time &amp; Date-&gt;DHCP Time-&gt;DHCP Time</p>		
<b>local_time.ntp_server1</b>	<b>IP address or domain name</b>	<b>cn.pool.ntp.org</b>
<p><b>Description:</b></p> <p>Configures the IP address or the domain name of the NTP server 1.</p> <p>The IP phone will obtain the current time and date from the NTP server 1.</p> <p><b>Example:</b></p> <p>local_time.ntp_server1 = 192.168.0.5</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Time &amp; Date-&gt;Primary Server</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Time &amp; Date-&gt;General-&gt;SNTP Settings-&gt;NTP Server1</p>		
<b>local_time.ntp_server2</b>	<b>IP address or domain name</b>	<b>pool.ntp.org</b>
<p><b>Description:</b></p> <p>Configures the IP address or the domain name of the NTP server 2.</p> <p>If the NTP server 1 is not configured (configured by the parameter "local_time.ntp_server1") or cannot be accessed, the IP phone will request the time and date from the NTP server 2.</p> <p><b>Example:</b></p> <p>local_time.ntp_server2 = 192.168.0.6</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Time &amp; Date-&gt;Secondary Server</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Time &amp; Date-&gt;General-&gt;SNTP Settings-&gt;NTP Server2</p>		

Parameters	Permitted Values	Default
<b>local_time.interval</b>	<b>Integer from 15 to 86400</b>	<b>1000</b>
<p><b>Description:</b> Configures the interval (in seconds) to update time and date from the NTP server.</p> <p><b>Example:</b> local_time.interval = 1000</p> <p><b>Web User Interface:</b> Settings-&gt;Time &amp; Date-&gt;Update Interval (15~86400s)</p> <p><b>Phone User Interface:</b> None</p>		
<b>local_time.time_zone</b>	<b>-11 to +14</b>	<b>+8</b>
<p><b>Description:</b> Configures the time zone. For more available time zones, refer to <a href="#">Appendix B: Time Zones</a> on page 873.</p> <p><b>Example:</b> local_time.time_zone = +8</p> <p><b>Web User Interface:</b> Settings-&gt;Time &amp; Date-&gt;Time Zone</p> <p><b>Phone User Interface:</b> Menu-&gt;Basic-&gt;Time &amp; Date-&gt;General-&gt;SNTP Settings-&gt;Time Zone</p>		
<b>local_time.time_zone_name</b>	<b>String within 32 characters</b>	<b>China(Beijing)</b>
<p><b>Description:</b> Configures the time zone name. The available time zone names depend on the time zone configured by the parameter "local_time.time_zone". For more information on the available time zone names for each time zone, refer to <a href="#">Appendix B: Time Zones</a> on page 873.</p> <p><b>Example:</b> local_time.time_zone_name = China(Beijing)</p> <p><b>Note:</b> It works only if the value of the parameter "local_time.summer_time" is set to 2 (Automatic) and the parameter "local_time.time_zone" should be configured in advance.</p> <p><b>Web User Interface:</b> Settings-&gt;Time &amp; Date-&gt;Location</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
Menu->Basic->Time & Date->General->SNTP Settings->Location		

To configure NTP by DHCP priority feature via web user interface:

1. Click on **Settings->Time & Date**.
2. Select the desired value from the pull-down list of **NTP by DHCP Priority**.

The screenshot shows the Yealink T466 web interface. The 'Settings' tab is selected, and the 'Time & Date' sub-tab is active. The 'NTP by DHCP Priority' dropdown menu is highlighted with a red rectangle and is currently set to 'High'. Other visible settings include 'DHCP Time' (Disabled), 'Manual Time' (Disabled), 'Time Zone' (+8 China, Singapore, Australia, Russia), 'Daylight Saving Time' (Automatic), 'Location' (China(Beijing)), 'Fixed Type' (DST by Date), 'Start Date', 'End Date', 'Offset (minutes)', 'Primary Server' (cn.pool.ntp.org), 'Secondary Server' (pool.ntp.org), 'Update Interval' (1000), 'Time Format' (Hour 24), and 'Date Format' (WWW MMM DD). A 'NOTE' section on the right provides information about Time and Date, Time Zone, NTP Server, and Daylight Saving Time.

3. Click **Confirm** to accept the change.

To configure the NTP server, time zone via web user interface:

1. Click on **Settings->Time & Date**.
2. Select **Disabled** from the pull-down list of **Manual Time**.
3. Select the desired time zone from the pull-down list of **Time Zone**.
4. Select the desired location from the pull-down list of **Location**.  
This field appears only if **Daylight Saving Time** is set to **Automatic**.
5. Enter the domain name or IP address in the **Primary Server** and **Secondary Server** field respectively.

- Enter the desired time interval in the **Update Interval (15~86400s)** field.

- Click **Confirm** to accept the change.

#### To configure the NTP server and time zone via phone user interface:

- Press **Menu->Basic->Time & Date->General->SNTP Settings**.
- Press **←** or **→**, or the **Switch** soft key to select the time zone that applies to your area from the **Time Zone** field.
- Enter the domain name or IP address in the **NTP Server1** and **NTP Server2** field respectively.
- Select **Automatic** from the **Daylight Saving** field.
- Press **←** or **→**, or the **Switch** soft key to select the desired value from the **Location** field.
- Press the **Save** soft key to accept the change.

## Time and Date Settings

You can set the time and date manually when IP phones cannot obtain the time and date from the NTP time server. The time and date display can use one of several different formats. You can customize date format as required.

You need to know the following rules when customizing date formats:

Format	Description
Y/YY	It represents a two-digit year. For example, 16, 17, 18...
Y is used more than twice (for example, YYY, YYYY)	It represents a four-digit year. For example, 2016, 2017, 2018...
M/MM	It represents a two-digit month. For example, 01, 02,..., 12
MMM	It represents the abbreviation of the month. For example, Jan, Feb,..., Dec
M is used more than three times (for example, MMMM)	It represents the long format of the month. For example, January, February,..., December
D is used more than once (for example, DD)	It represents a two-digit day. For example, 01, 02,..., 31
W/WW	It represents the abbreviation of the day of week. For example, Mon, Tue,..., Sun
W is used more than twice (for example, WWW, WWWW)	It represents the long format of the day of week. For example, Monday, Tuesday,..., Sunday

## Procedure

Time and date can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure the time and date manually. <b>Parameter:</b> local_time.manual_time_enable
		Configure the time and date formats. <b>Parameters:</b> local_time.time_format local_time.date_format
		Customize the date format. <b>Parameter:</b> lcl.datetime.date.format

<b>Web User Interface</b>	<p>Configure the time and date manually.</p> <p>Configure the time and date formats.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?parameters=settings-datetime&amp;q=load</p>
<b>Phone User Interface</b>	<p>Configure the time and date manually.</p> <p>Configure the time and date formats.</p>

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>local_time.manual_time_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to obtain time and date from manual settings.</p> <p><b>0</b>-Disabled (obtain time and date from NTP server)</p> <p><b>1</b>-Enabled (obtain time and date from manual settings)</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Time &amp; Date-&gt;Manual Time</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>local_time.time_format</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Configures the time format.</p> <p><b>0</b>-Hour 12, the time will be displayed in 12-hour format with AM or PM specified.</p> <p><b>1</b>-Hour 24, the time will be displayed in 24-hour format (for example, 2:00 PM displays as 14:00).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Time &amp; Date-&gt;Time Format</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Time &amp; Date-&gt;Time &amp; Date Format-&gt;Time Format</p>		
<b>local_time.date_format</b>	<b>0, 1, 2, 3, 4, 5 or 6</b>	<b>0</b>

Parameters	Permitted Values	Default
<p><b>Description:</b> Configures the date format.</p> <p><b>0</b>-WWW MMM DD  <b>1</b>-DD-MMM-YY  <b>2</b>-YYYY-MM-DD  <b>3</b>-DD/MM/YYYY  <b>4</b>-MM/DD/YY  <b>5</b>-DD MMM YYYY  <b>6</b>-WWW DD MMM</p> <p><b>Note:</b> "WWW" represents the abbreviation of the week, "DD" represents a two-digit day, "MMM" represents the first three letters of the month, "YYYY" represents a four-digit year, and "YY" represents a two-digit year. The value configured by the parameter "lcl.datetime.date.format" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b> Settings-&gt;Time &amp; Date-&gt;Date Format</p> <p><b>Phone User Interface:</b> Menu-&gt;Basic-&gt;Time &amp; Date-&gt;Time &amp; Date Format-&gt;Date Format</p>		
<b>lcl.datetime.date.format</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b> Configures the display format of date.</p> <p><b>Y</b> = year, <b>M</b> = month, <b>D</b> = day, <b>W</b> = day of week</p> <p><b>Value formats are:</b></p> <ul style="list-style-type: none"> <li>Any combination of W, M, D and the separator (for example, space, dash, slash).</li> </ul> <p><b>Example:</b> lcl.datetime.date.format = W,MD The IP phone displays the date in "W,MD" format (for example, Wed,0420).</p> <ul style="list-style-type: none"> <li>Any combination of Y, M, D, W and the separator (for example, space, dash, slash).</li> </ul> <p><b>Example:</b> lcl.datetime.date.format = YYYY-MMM-DDD-WWW The IP phone displays the date in "YYYY-MMM-DDD-WWW" format (for example, 2016-Apr-20-Wednesday).</p> <p><b>Note:</b> "Y"/"YY" represents a two-digit year, more than two "Y" letters (for example, YYYY) represent a four-digit year, "M"/"MM" represents a two-digit month, "MMM" represents the abbreviation of the month, three or more than three "M" letters (for example, MMM) represent the long format of the month, one or more than one "D" (for example, DDD)</p>		



Parameters	Permitted Values	Default
<p>represents a two-digit day, "W"/"WW" represents the abbreviation of the day of week, three or more three "W" letters (for example, WWW) represent the long format of the day of week.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure the time and date manually via web user interface:**

1. Click on **Settings->Time & Date**.
2. Select **Enabled** from the pull-down list of **Manual Time**.
3. Enter the time and date in the corresponding fields.

The screenshot shows the Yealink T46G web interface. The 'Settings' tab is selected, and the 'Time & Date' sub-tab is active. The 'Manual Time' option is selected from the 'DHCP Time' dropdown. The date is set to Year: 2016, Month: 12, Day: 7. The time is set to Hour: 8, Minute: 53, Second: 14. The 'Time Format' is set to 'Hour 24' and the 'Date Format' is set to 'WWW MMM DD'. A 'NOTE' section on the right explains the Time and Date settings.

4. Click **Confirm** to accept the change.

**To configure the time and date formats via web user interface:**

1. Click on **Settings->Time & Date**.
2. Select the desired value from the pull-down list of **Time Format**.

3. Select the desired value from the pull-down list of **Date Format**.

4. Click **Confirm** to accept the change.

#### To configure the time and date manually via phone user interface:

1. Press **Menu->Basic->Time & Date->General->Manual Settings**.
2. Enter the date and time in the **Date Time** field.
3. Press the **Save** soft key to accept the change.

#### To configure the time and date formats via phone user interface:

1. Press **Menu->Basic->Time & Date->Time & Date Format**.
2. Press **◀** or **▶**, or the **Switch** soft key to select the desired time format from the **Time Format** field.
3. Press **◀** or **▶**, or the **Switch** soft key to select the desired date format from the **Date Format** field.
4. Press the **Save** soft key to accept the change.

## Daylight Saving Time (DST)

Daylight Saving Time (DST) is the practice of temporary advancing clocks during the summer time so that evenings have more daylight and mornings have less. Typically, clocks are adjusted forward one hour at the start of spring and backward in autumn. Many countries have used the DST at various times, details vary by location. By default, the DST is set to Automatic, so it can be adjusted automatically from the current time zone configuration. You can configure DST for the desired area as required.

## Procedure

DST can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure DST. <b>Parameters:</b> local_time.summer_time local_time.dst_time_type local_time.start_time local_time.end_time local_time.offset_time
<b>Web User Interface</b>		Configure DST. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=setting-s-datetime&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=setting-s-datetime&amp;q=load</a>

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>local_time.summer_time</b>	<b>0, 1 or 2</b>	<b>2</b>
<b>Description:</b> Configures Daylight Saving Time (DST) feature. <b>0</b> -Disabled <b>1</b> -Enabled <b>2</b> -Automatic <b>Web User Interface:</b> Settings->Time & Date->Daylight Saving Time <b>Phone User Interface:</b> Menu->Basic->Time & Date->General->SNTP Settings->Daylight Saving		
<b>local_time.dst_time_type</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Configures the Daylight Saving Time (DST) type. <b>0</b> -DST by Date <b>1</b> -DST by Week <b>Note:</b> It works only if the value of the parameter "local_time.summer_time" is set to 1 (Enabled).		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Settings->Time & Date->Fixed Type <b>Phone User Interface:</b> None		
<b>local_time.start_time</b>	<b>Refer to the following content</b>	<b>1/1/0</b>
<b>Description:</b> Configures the starting time of the Daylight Saving Time (DST). <b>Value formats are:</b> <ul style="list-style-type: none"> <li>Month/Day/Hour (for DST by Date)</li> <li>Month/Week of Month/Day of Week/Hour of Day (for DST by Week)</li> </ul> If "local_time.dst_time_type" is set to 0 (DST by Date), use the mapping: <b>Month:</b> 1=January, 2=February,..., 12=December <b>Day:</b> 1=the first day in a month,..., 31= the last day in a month <b>Hour:</b> 0=0am, 1=1am,..., 23=11pm <b>Example:</b> local_time.start_time = 1/1/2 If "local_time.dst_time_type" is set to 1 (DST by Week), use the mapping: <b>Month:</b> 1=January, 2=February,..., 12=December <b>Week of Month:</b> 1=the first week in a month,..., 5=the last week in a month <b>Day of Week:</b> 1=Monday, 2=Tuesday,..., 7=Sunday <b>Hour of Day:</b> 0=0am, 1=1am,..., 23=11pm <b>Example:</b> local_time.start_time = 1/1/7/0 <b>Note:</b> It works only if the value of the parameter "local_time.summer_time" is set to 1 (Enabled). <b>Web User Interface:</b> Settings->Time & Date->Start Date <b>Phone User Interface:</b> None		
<b>local_time.end_time</b>	<b>Refer to the following content</b>	<b>12/31/23</b>
<b>Description:</b> Configures the ending time of the Daylight Saving Time (DST).		

Parameters	Permitted Values	Default
<p><b>Value formats are:</b></p> <ul style="list-style-type: none"> <li>Month/Day/Hour (for DST by Date)</li> <li>Month/Week of Month/Day of Week/Hour of Day (for DST by Week)</li> </ul> <p>If "local_time.dst_time_type" is set to 0 (DST by Date), use the mapping:</p> <p><b>Month:</b> 1=January, 2=February,..., 12=December</p> <p><b>Day:</b> 1=the first day in a month,..., 31= the last day in a month</p> <p><b>Hour:</b> 0=0am, 1=1am,..., 23=11pm</p> <p><b>Example:</b></p> <p>local_time.start_time = 12/12/22</p> <p>If "local_time.dst_time_type" is set to 1 (DST by Week), use the mapping:</p> <p><b>Month:</b> 1=January, 2=February,..., 12=December</p> <p><b>Week of Month:</b> 1=the first week in a month,..., 5=the last week in a month</p> <p><b>Day of Week:</b> 1=Monday, 2=Tuesday,..., 7=Sunday</p> <p><b>Hour of Day:</b> 0=0am, 1=1am,..., 23=11pm</p> <p><b>Example:</b></p> <p>local_time.start_time = 4/3/2/3</p> <p><b>Note:</b> It works only if the value of the parameter "local_time.summer_time" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Time &amp; Date-&gt;End Date</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
local_time.offset_time	Integer from -300 to 300	Blank
<p><b>Description:</b></p> <p>Configures the offset time (in minutes) of Daylight Saving Time (DST).</p> <p><b>Note:</b> It works only if the value of the parameter "local_time.summer_time" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Time &amp; Date-&gt;Offset(minutes)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure the DST via web user interface:**

1. Click on **Settings->Time & Date**.
2. Select **Disabled** from the pull-down list of **Manual Time**.

3. Select the desired time zone from the pull-down list of **Time Zone**.
4. Enter the domain name or IP address in the **Primary Server** and **Secondary Server** field respectively.
5. Enter the desired time interval in the **Update Interval (15~86400s)** field.
6. Mark the **Enabled** radio box in the **Daylight Saving Time** field.
  - Mark the **DST by Date** radio box in the **Fixed Type** field.

Enter the starting time in the **Start Date** field.

Enter the ending time in the **End Date** field.

- Mark the **DST by Week** radio box in the **Fixed Type** field.
- Select the desired values of DST Start Month, DST Start Week of Month, DST Start Day of Week, Start Hour of Day; DST Stop Month, DST Stop Week of Month, DST Stop Day of Week and End Hour of Day from the pull-down lists.

7. Enter the desired offset time in the **Offset(minutes)** field.
8. Click **Confirm** to accept the change.

## Customizing an AutoDST Template File

The time zone and corresponding DST pre-configurations exist in the AutoDST file. If the DST is

set to Automatic, the IP phone obtains the DST configuration from the AutoDST file. You can customize the AutoDST file if required. The AutoDST file allows you to add or modify time zone and DST settings for your area each year.

Before customizing, you need to obtain the AutoDST file. You can ask the distributor or Yealink FAE for DST template. You can also obtain the DST template online:

<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>. For more information on obtaining the template file, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

The following table lists description of each element in the template file:

Element	Type	Values	Description
<b>DSTData</b>	required	no	File root element
<b>DST</b>	required	no	Time Zone item's root element
<b>szTime</b>	required	[+/-][X]:[Y], X=0~14, Y=0~59	Time Zone
<b>szZone</b>	required	String (if the content is more than one city, it is the best to keep their daylight saving time the same)	Time Zone name
<b>iType</b>	optional	0/1 <b>0:</b> DST by Date <b>1:</b> DST by Week	DST time type (This item is needed if you want to configure DST.)
<b>szStart</b>	optional	<b>Month/Day/Hour</b> (for <b>iType=0</b> ) Month: 1~12 Day: 1~31 Hour: 0 (midnight)~23 <b>Month/Week of Month/Day of Week/Hour of Day</b> (for <b>iType=1</b> ) Month: 1~12 Week of Month: 1~5 (the last week) Day of Week: 1~7 Hour of Day: 0 (midnight)~23	Starting time of the DST
<b>szEnd</b>	optional	Same as szStart	Ending time of the DST
<b>szOffset</b>	optional	Integer from -300 to 300	The offset time (in minutes) of DST

**When customizing an AutoDST file, learn the following:**

- <DSTData> indicates the start of a template and </DSTData> indicates the end of a template.

- Add or modify time zone and DST settings between <DSTData> and </DSTData>.
- The display order of time zone is corresponding to the szTime order specified in the AutoDST.xml file.
- If the starting time of DST is greater than the ending time, the valid time of DST is from the starting time of this year to the ending time of the next year.

#### Customizing an AutoDST file:

1. Open the AutoDST file using an ASCII editor.
2. Add or modify time zone and DST settings as you want in the AutoDST file.

##### Example 1:

To modify the DST settings for the existing time zone "+5 Pakistan(Islamabad)" and add DST settings for the existing time zone "+5:30 India(Calcutta)".

```

AutoDST.xml x
0 10 20 30 40 50 60 70 80 90 100 110
<DST szTime="+3:30" szZone="Iran (Teheran)" iType="0" szStart="3/22/0" szEnd="9/22/0" szOffset="60"/>
<DST szTime="+4" szZone="Armenia (Yerevan)" iType="1" szStart="3/5/7/2" szEnd="10/5/7/3" szOffset="60"/>
<DST szTime="+4" szZone="Azerbaijan (Baku)" iType="1" szStart="3/5/7/4" szEnd="10/5/7/5" szOffset="60"/>
<DST szTime="+4" szZone="Georgia (Tbilisi)" />
<DST szTime="+4" szZone="Kazakhstan (Aktau)" />
<DST szTime="+4" szZone="Russia (Samara)" />
<DST szTime="+4:30" szZone="Afghanistan (Kabul)" />
<DST szTime="+5" szZone="Kazakhstan (Aqtobe)" />
<DST szTime="+5" szZone="Kyrgyzstan (Bishkek)" />
<DST szTime="+5" szZone="Pakistan (Islamabad)" iType="0" szStart="4/15/0" szEnd="11/1/0" szOffset="60"/>
<DST szTime="+5:30" szZone="India (Calcutta)" iType="1" szStart="9/5/7/3" szEnd="4/1/7/2" szOffset="60"/>
<DST szTime="+5:45" szZone="Nepal (Katmandu)" />
<DST szTime="+6" szZone="Kazakhstan (Astana, Almaty)" />
<DST szTime="+6" szZone="Russia (Novosibirsk, Omsk)" />
<DST szTime="+6:30" szZone="Myanmar (Naypyitaw)" />
<DST szTime="+7" szZone="Russia (Krasnoyarsk)" />
<DST szTime="+7" szZone="Thailand (Bangkok)" />
<DST szTime="+8" szZone="China (Beijing)" />
<DST szTime="+8" szZone="Singapore (Singapore)" />

```

##### Example 2:

Add a new time zone (+6 Paradise) with daylight saving time 30 minutes.

```

AutoDST.xml x
0 10 20 30 40 50 60 70 80 90
<DST szTime="+4:30" szZone="Afghanistan (Kabul)" />
<DST szTime="+5" szZone="Kazakhstan (Aqtobe)" />
<DST szTime="+5" szZone="Kyrgyzstan (Bishkek)" />
<DST szTime="+5" szZone="Pakistan (Islamabad)" iType="0" szStart="4/15/0" szEnd="11/1/0" />
<DST szTime="+5" szZone="Russia (Chelyabinsk)" />
<DST szTime="+5:30" szZone="India (Calcutta)" />
<DST szTime="+5:45" szZone="Nepal (Katmandu)" />
<DST szTime="+6" szZone="Paradise" iType="1" szStart="3/5/7/2" szEnd="10/5/7/3" szOffset="30"/>
<DST szTime="+6" szZone="Kazakhstan (Astana, Almaty)" />
<DST szTime="+6" szZone="Russia (Novosibirsk, Omsk)" />
<DST szTime="+6:30" szZone="Myanmar (Naypyitaw)" />
<DST szTime="+7" szZone="Russia (Krasnoyarsk)" />
<DST szTime="+7" szZone="Thailand (Bangkok)" />
<DST szTime="+8" szZone="China (Beijing)" />
<DST szTime="+8" szZone="Singapore (Singapore)" />
<DST szTime="+8" szZone="Australia (Perth)" iType="1" szStart="10/1/7/2" szEnd="3/5/7/3" />
<DST szTime="+8" szZone="Russia (Irkutsk, Ulan-Ude)" />
<DST szTime="+8:45" szZone="Eucla" />
<DST szTime="+9" szZone="Korea (Seoul)" />
<DST szTime="+9" szZone="Japan (Tokyo)" />
<DST szTime="+9" szZone="Russia (Yakutsk, Chita)" />
<DST szTime="+9:30" szZone="Australia (Adelaide)" iType="1" szStart="10/1/7/2" szEnd="4/1/7/3" />
<DST szTime="+9:30" szZone="Australia (Darwin)" />
<DST szTime="+10" szZone="Australia (Sydney, Melbourne, Canberra)" iType="1" szStart="10/1/7/2" />
<DST szTime="+10" szZone="Australia (Brisbane)" />

```

3. Save this file and place it to the provisioning server (for example, 192.168.1.100).
4. Specify the access URL of the AutoDST file in the configuration file.



## Procedure

The access URL of the AutoDST file can be specified using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Specify the access URL of the AutoDST file. <b>Parameter:</b> auto_dst.url
------------------------------------------------------	-----------	----------------------------------------------------------------------------------

## Details of Configuration Parameter:

Parameter	Permitted Values	Default
auto_dst.url	URL within 511 characters	Blank
<p><b>Description:</b> Configures the access URL of the AutoDST file (AutoDST.xml).</p> <p><b>Example:</b> auto_dst.url = tftp://192.168.1.100/AutoDST.xml</p> <p>During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.100", and downloads the AutoDST file "AutoDST.xml". After update, you will find a new time zone "Paradise" and updated DST of "Pakistan (Islamabad)" and "India (Calcutta)" via web user interface: <b>Settings-&gt;Time &amp; Date-&gt;Time Zone</b>.</p> <p><b>Note:</b> It works only if the value of the parameter "local_time.summer_time" is set to 2 (Automatic).</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		

## Language

IP phones support multiple languages. Languages used on the phone user interface and web user interface can be specified respectively as required.

The following table lists languages supported by the phone user interface and the web user interface.

Phone User Interface	Web User Interface
English	English
Chinese Simplified	Chinese Simplified
Chinese Traditional	Chinese Traditional

Phone User Interface	Web User Interface
French (Canada)	French
French (EU)	German
German	Italian
Italian	Polish
Polish	Portuguese
Portuguese (EU)	Spanish
Portuguese (Latin)	Turkish
Spanish (EU)	Russian
Spanish (Latin)	
Turkish	
Russian	

**Note**

For SIP-T42G/T41P/T40P/T40G/T23P/T23G/T21(P) E2/T19(P) E2 IP phones, they do not support Chinese by default. For more information, contact Yealink FAE.

For SIP-T48G/T46G/T42G/T41P/T40P IP phones, they do not support French (Canada), Portuguese (Latin) and Spanish (Latin).

## Loading Language Packs

Languages available for selection depend on language packs currently loaded to the IP phone. You can customize the translation of the existing language on the phone user interface or web user interface. You can also make new languages (not included in the available language list) available for use on the phone user interface and web user interface by loading language packs to the IP phone. Language packs can only be loaded using configuration file.

You can ask the distributor or Yealink FAE for language packs. You can also obtain the language packs online: <http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>.

For more information on obtaining the language packs, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

**Note**

To modify translation of an existing language, do not rename the language file.

The new added language must be supported by the font library on the IP phone. If the characters in the custom language file are not supported by the phone, the IP phone will display "?" instead.

## Customizing a Language for Phone User Interface

The following table lists the available languages and associated language packs for the phone user interface:

Available Language	Associated Language Pack
English	000.GUI.English.lang
Chinese Simplified	001.GUI.Chinese_S.lang
Chinese Traditional	002.GUI.Chinese_T.lang
French (Canada)	003.GUI.French_CA.lang
French (EU)	004.GUI.French.lang
German	005.GUI.German.lang
Italian	006.GUI.Italian.lang
Polish	007.GUI.Polish.lang
Portuguese (EU)	008.GUI.Portuguese.lang
Portuguese (Latin)	009.GUI.Portuguese_LA.lang
Spanish (EU)	010.GUI.Spanish.lang
Spanish (Latin)	011.GUI.Spanish_LA.lang
Turkish	012.GUI.Turkish.lang
Russian	013.GUI.Russian.lang

When adding a new language pack for the phone user interface, the language pack must be formatted as "X.GUI.name.lang" (X starts from 014, "name" is replaced with the language name). If the language name is the same as the existing one, the existing language pack will be overridden by the new uploaded one. We recommend that the filename of the new language pack should not be the same as the existing one.

### To customize a language file:

1. Open the desired language template file (for example, 000.GUI.English.lang) using an ASCII editor.
2. Modify the characters within the double quotation marks on the right of the equal sign. Don't modify the translation item on the left of the equal sign.

The following shows a portion of the language pack "000.GUI.English.lang" for the phone user interface (take SIP-T46G IP phones for example):

```

000.GUI.English.lang x
1 [ Lang ]
2
3 '*' or '#' as send="Key as send"
4 "(Empty)"="(Empty)"
5 "10min"="10min"
6 "12 Hour"="12 Hour"
7 "120s"="120s"
8 "15s"="15s"
9 "1min"="1min"
10 "24 Hour"="24 Hour"
11 "2min"="2min"
12 "30min"="30min"
13 "30s"="30s"
14 "5min"="5min"
15 "60s"="60s"
16 "802.1x Mode"="802.1x Mode"
17 "802.1x"="802.1x"
18 "ACD Login"="ACD Login"
19 "ACD State"="ACD State"
20 "ACD Trace"="Trace"
21 "ACD"="ACD"
22 "AES"="AES"
23 "ALERT"="ALERT"
24 "AP Mac Address"="AP Mac Address"
25 "Account ID"="Account ID"
26 "Account Status"="Account Status"

```

3. Save the language file and place it to the provisioning server (for example, 192.168.10.25).
4. Specify the access URL of the phone user interface language pack in the configuration file.

If you want to add a new custom language (for example, Guilan) to your IP phone (for example, SIP-T46G), prepare the language file named as "011.GUI.Guilan.lang" for downloading. After update, you will find a new language selection "Guilan" on the IP phone user interface:

**Menu->Basic->Language.**

## Procedure

Loading language pack can be performed using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify the access URL of the phone user interface language pack.  <b>Parameter:</b> gui_lang.url
		Delete custom LCD language packs of the phone user interface.  <b>Parameter:</b> gui_lang.delete

**Details of the Configuration Parameters:**

Parameters	Permitted Values	Default
<b>gui_lang.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the access URL of the custom LCD language pack for the phone user interface.</p> <p><b>Example:</b></p> <p>gui_lang.url = http://192.168.10.25/000.GUI.English.lang</p> <p>During the auto provisioning process, the IP phone connects to the HTTP provisioning server "192.168.10.25", and downloads the language pack "000.GUI.English.lang". The English language translation will be changed accordingly if you have modified the language template file.</p> <p>If you want to download multiple language packs to the phone simultaneously, you can configure as following:</p> <p>gui_lang.url = http://192.168.10.25/000.GUI.English.lang</p> <p>gui_lang.url = http://192.168.10.25/001.GUI.Chinese_S.lang</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>gui_lang.delete</b>	<b>http://localhost/all or http://localhost/Y.GUI.name.lang</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Deletes the specified or all custom LCD language packs of the phone user interface.</p> <p><b>Example:</b></p> <p>Delete all custom language packs of the phone user interface:</p> <p>gui_lang.delete = http://localhost/all</p> <p>Delete a custom language pack of the phone user interface (for example, 001.GUI.Chinese_S.lang):</p> <p>gui_lang.delete = http://localhost/001.GUI.Chinese_S.lang</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## Customizing a Language for Web User Interface

The following table lists available languages and associated language packs for the web user interface:

Available Language	Associated Language Pack	Associated Note Language Pack
English	1.English.js	1.English_note.xml
Chinese Simplified	2.Chinese_S.js	2.Chinese_S_note.xml
Chinese Traditional	3.Chinese_T.js	3.Chinese_T_note.xml
French	4.French.js	4.French_note.xml
German	5.German.js	5.German_note.xml
Italian	6.Italian.js	6.Italian_note.xml
Polish	7.Polish.js	7.Polish_note.xml
Portuguese	8.Portuguese.js	8.Portuguese_note.xml
Spanish	9.Spanish.js	9.Spanish_note.xml
Turkish	10.Turkish.js	10.Turkish_note.xml
Russian	11.Russian.js	11.Russian_note.xml

When adding a new language pack for the web user interface, the language pack must be formatted as "Y.name.js" (Y starts from 12, "name" is replaced with the language name). If the language name is the same as the existing one, the existing language file will be overridden by the new uploaded one. We recommend that the filename of the new language pack should not be the same as the existing one.

### To customize a language file:

1. Open the desired language template file (for example, 1.English.js) using an ASCII editor.
2. Modify the characters within the double quotation marks on the right of the colon. Don't modify the translation item on the left of the colon.


The following shows a portion of the language pack "1.English.js" for the web user interface (take SIP-T46G IP phones for example):

```

1 var _objTrans =
2 {
3   //login.htm
4   "The username can not be empty.":"The username can
5   "Failed to connect to the server. Please check net
6   "Login":"Login",
7   "Username":"Username",
8   "Password":"Password",
9   "Confirm":"Confirm",
10  "admin":"admin",
11  "user":"user",
12  "var":"var",
13
14  //header.htm
15  "Log_Out":"Log Out",
16  "Status":"Status",
17  "Network":"Network",
18  "Dsskey":"Dsskey",
19  "Features":"Features",
20  "Settings":"Settings",
21  "Directory":"Directory",
22  "Security":"Security",
23  "Applications":"Applications",

```

3. Save the language file and place it to the provisioning server (for example, 192.168.10.25).
4. Specify the access URL of the web user interface language pack in the configuration file.

You can also customize the translation of the note language pack. The note information is displayed in the icon  of the web user interface. The note language pack must be formatted as "Y.name\_note.xml" ("Y" and "name" are associated with web language pack).

#### To customize a note language file:

1. Open the desired note language template file (for example, 1.English\_note.xml) using an ASCII editor.
2. Modify the text of the note field. Don't modify the name of the note field.

The following shows a portion of the note language pack "1.English\_note.xml" for the web user interface (take SIP-T46G IP phones for example):

```

1 <?xml version="1.0" encoding="utf-8"?>
2 <notedata>
3
4 <status>
5   <note name = "version">
6     <head>Description:</head>
7     <text>It shows the current firmware version and hardware version of the device.</text>
8   </note>
9   <note name = "DeviceCertificate">
10    <head>Description:</head>
11    <text>It shows the Device Certificate of the device.</text>
12  </note>
13  <note name = "network">
14    <head>Description:</head>
15    <text>It shows the IP address mode of the device.</text>
16  </note>
17  <note name = "network-ipv4">
18    <head>Description:</head>
19    <text>It shows the basic IPv4 network configurations.</text>
20  </note>
21  <note name = "network-ipv6">
22    <head>Description:</head>
23    <text>It shows the basic IPv6 network configurations.</text>
24  </note>

```

3. Save the language file and place it to the provisioning server (for example, 192.168.10.25).
4. Specify the access URL of the note language pack of the web user interface.

If you want to add a new language (for example, Wuilan) to IP phones, prepare the language file named as "12.Wuilan.js" and "12.Wuilan\_note.xml" for downloading. After update, you will find a new language selection "Wuilan" at the top-right corner of the web user interface, and new note information is displayed in the icon when the new language is selected.

## Procedure

Loading language pack can be performed using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify the access URL of the custom language pack for web user interface. <b>Parameter:</b> wui_lang.url
		Specify the access URL of the custom note language pack for web user interface. <b>Parameter:</b> wui_lang_note.url
		Delete custom language packs and note language packs of the web user interface. <b>Parameter:</b> wui_lang.delete

## Details of the Configuration Parameters:

Parameters	Permitted Values	Default
<b>wui_lang.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the access URL of the custom language pack for the web user interface.  <b>Example:</b> wui_lang.url = http://192.168.10.25/1.English.js  During the auto provisioning process, the IP phone connects to the HTTP provisioning server "192.168.10.25", and downloads the language pack "1.English.js". The English language translation will be changed accordingly if you have modified the language template file.  If you want to download multiple language packs to the web user interface simultaneously, you can configure as following:  wui_lang.url = http://192.168.10.25/1.English.js		



Parameters	Permitted Values	Default
<p>wui_lang.url = http://192.168.10.25/11.Russian.js</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>wui_lang_note.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the access URL of the custom note language pack for web user interface.</p> <p><b>Example:</b></p> <p>wui_lang_note.url = http://192.168.10.25/1.English_note.xml</p> <p>During the auto provisioning process, the IP phone connects to the HTTP provisioning server "192.168.10.25", and downloads the note language pack "1.English_note.xml". The English language translation will be changed accordingly if you have modified the language template file.</p> <p>If you want to download multiple language packs to the phone simultaneously, you can configure as following:</p> <p>wui_lang.url = http://192.168.10.25/1.English_note.xml</p> <p>wui_lang.url = http://192.168.10.25/11.Russian_note.xml</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>wui_lang.delete</b>	<b>http://localhost/all or http://localhost/Y.name.js</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Delete the specified or all custom web language packs and note language packs of the web user interface.</p> <p><b>Example:</b></p> <p>Delete all custom language packs of the web user interface:</p> <p>wui_lang.delete = http://localhost/all</p> <p>Delete a custom language pack of the web user interface (for example, 11.Russian.js):</p> <p>wui_lang.delete = http://localhost/11.Russian.js</p> <p>The corresponding note language pack (for example, 11.Russian_note.xml) will also be deleted.</p>		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> None		
<b>Phone User Interface:</b> None		

## Specifying the Language to Use

The default language used on the phone user interface is English. If the language of your web browser is not supported by the IP phone, the web user interface will use English by default. You can specify the languages for the phone user interface and web user interface respectively.

### Procedure

Specify the language for the phone user interface or the web user interface using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify the languages for the phone user interface or the web user interface.  <b>Parameters:</b> static.lang.gui static.lang.wui
<b>Web User Interface</b>		Specify the language for the web user interface.
<b>Phone User Interface</b>		Specify the language for the phone user interface.

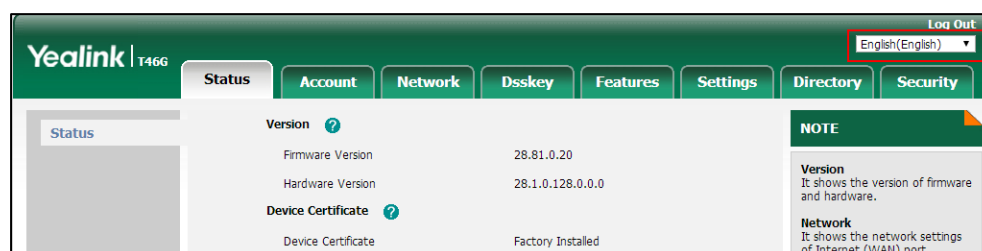
### Details of Configuration Parameters:



Parameters	Permitted Values	Default
<b>static.lang.gui</b>	<b>Refer to the following content</b>	<b>English</b>
<b>Description:</b> Configures the language used on the phone user interface.		
<b>Permitted Values:</b> For SIP-T54S/T52S/T48S/T46S/T42S/T41S/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2/T19(P) E2 IP phones: English, Chinese_S, Chinese_T, French_CA, French, German, Italian, Polish, Portuguese, Portuguese_LA, Spanish, Spanish_LA, Turkish, Russian or the custom language name.		

Parameters	Permitted Values	Default
<p>For SIP-T48G/T46G/T42G/T41P/T40P IP phones:</p> <p>English, Chinese_S, Chinese_T, French, German, Italian, Polish, Portuguese, Spanish, Turkish, Russian or the custom language name.</p> <p><b>Example:</b></p> <p>static.lang.gui = English</p> <p>If you want to use the custom language (for example, Guilan) for the IP phone, configure the parameter "static.lang.gui = Guilan".</p> <p><b>Note:</b> For SIP-T42G/T41P/T40P/T40G/T23P/T23G/T21(P) E2/T19(P) E2 IP phones, they do not support Chinese by default. For more information, contact Yealink FAE.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Language</p>		
<b>static.lang.wui</b>	<b>Refer to the following content</b>	<b>English</b>
<p><b>Description:</b></p> <p>Configures the language used on the web user interface.</p> <p><b>Permitted Values:</b></p> <p>English, Chinese_S, Chinese_T, French, German, Italian, Polish, Portuguese, Spanish, Turkish, Russian or the custom language name.</p> <p><b>Example:</b></p> <p>static.lang.wui = English</p> <p>If you want to use the custom language (for example, Wuilan) for the IP phone, configure the parameter "static.lang.wui = Wuilan".</p> <p><b>Note:</b> If the language of your browser is not supported by the IP phone, the web user interface will use English by default. For SIP-T42G/T41P/T40P/T40G/T23P/T23G/T21(P) E2/T19(P) E2 IP phones, they do not support Chinese by default. For more information, contact Yealink FAE.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Note:</b> You can change the language for the web user interface on each page via web user interface.</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To specify the language for the web user interface via web user interface:**

1. Select the desired language from the pull-down list at the top-right corner of the web user interface.

**To specify the language for the phone user interface via phone user interface:**

1. Press **Menu**->**Basic**->**Language**.
2. Press  or  to select the desired language.
3. Press the **Save** soft key to accept the change.

## Input Method

### Keypad Input Method Customization

Keypad input method customization allows users to customize the existing input method on IP phones. You can first customize the Yealink-supplied keypad input method file "ime.txt", "Russian\_ime.txt" or "Hebrew\_ime.txt", and then download it to the IP phone. The changes in the "Russian\_ime.txt" file becomes affective when the language is set to Russian. The changes in the "Hebrew\_ime.txt" file becomes affective when the language is set to Hebrew. The changes in the "ime.txt" file is affective for all the languages. IP phones support 6 input methods: 2aB, abc, Abc, 123, ABC and Hebrew. By default, Hebrew input method is hidden, the IP phone will automatically display the Hebrew input method when the language is set to Hebrew. If you want to set the language to Hebrew, you have to add the new custom language - Hebrew to your IP phone in advance. For more information, refer to [Customizing a Language for Phone User Interface](#) on page 233.

If you just want to customize the input method for a certain language, the filename must be formatted as "language name\_ime.txt" (for example, German\_ime.txt).

You can ask the distributor or Yealink FAE for keypad input method file. You can also obtain the keypad input method file online:

<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>. For more information on obtaining the keypad input method file, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

The following shows a portion of the keypad input method file "ime.txt":

```

ime.txt x
1 [2aB]
2 1 = "1"
3 2 = "2abcABC"
4 3 = "3defDEF"
5 4 = "4ghiGHI"
6 5 = "5jklJKL"
7 6 = "6mnoMNO"
8 7 = "7pqrsPQRS"
9 8 = "8tuvTUV"
10 9 = "9wxyzWXYZ"
11 0 = "0"
12 * = "*, '?!\\-()@/_:~;+&%=<>£$%[]{}~^;ç$#"|"
13 # = "#"
14
15 [abc]
16 1 = ""
17 2 = "abc2äæåääååç"
18 3 = "def3èéêëè"
19 4 = "ghi4íîïì"
20 5 = "jkl5ê"
21 6 = "mno6öøóóóñ"
22 7 = "pqrs7ßs"
23 8 = "tuv8ùúûü"
24 9 = "wxyz9ýÿ"
25 0 = " "
26 * = "*, '?!\\-()@/_:~;+&%=<>£$%[]{}~^;ç$#"|"
27 # = "#"

```

The following shows a portion of the keypad input method file "Hebrew\_ime.txt":

```

Hebrew ime.txt x
1 [123_PWD]
2 1 = "1"
3 2 = "2"
4 3 = "3"
5 4 = "4"
6 5 = "5"
7 6 = "6"
8 7 = "7"
9 8 = "8"
10 9 = "9"
11 0 = "0"
12 * = "*, !:@/"
13 # = "#"
14
15 [Hebrew]
16 1 = "1"
17 2 = "א ב ג ד ה ו ז ח ט י כ ל מ נ ס ע פ צ ק ר ש ת"
18 3 = "א ב ג"
19 4 = "ח ט נ ו"
20 5 = "כ ד ק י"
21 6 = "ט ח ט"
22 7 = "ר ש ת"
23 8 = "ק צ ק"
24 9 = "ט ע פ"
25 0 = "0"
26 * = "*, !:@+- $"
27 # = "#"

```

The following shows a portion of the keypad input method file "Russian\_ime.txt":

```

Russian_ime.txt x
0 1.0 2.0 3.0 4.0 5.0
[2aB]
1 = "1"
2 = "АБВГабвг2ABCabC"
3 = "ДЕЖЗдежз3DEFdef"
4 = "ИЙКЛийкл4GHIghi"
5 = "МНОПмноп5JKLjkl"
6 = "РСТУрсту6MNOmno"
7 = "ФХЦУфхц7PQRSpqrs"
8 = "ШЩЪЫыщъ8TUVtuv"
9 = "ЪЮЯъюя9WXYZwxyz"
0 = "0"
* = "*, '?! \- () @ / : _ ; + & % = < > £ $ ¥ ¤ [ ] { } ~ ^ ¡ ¢ § ¨"
# = "#"

[ABC]
1 = "*, '?! | @ ' - _ ( ) ; & / % * # + < = > " $ £ ¥ ¤ ¨ ¡ ¢"
2 = "АБВГABC"
3 = "ДЕЖЗDEF"
4 = "ИЙКЛIGHI"
5 = "МНОПJKL"
6 = "РСТУMNO"
7 = "ФХЦУPQRS"
8 = "ШЩЪTUV"
9 = "ЪЮЯWXYZ"
0 = "0"
* = "*, '?! \- () @ / : _ ; + & % = < > £ $ ¥ ¤ [ ] { } ~ ^ ¡ ¢ § ¨"
# = "#"

```

#### To customize a keypad input method file:

1. Open the desired keypad input method file (for example, ime.txt) using an ASCII editor.
2. Under the input method field (for example, [abc]), add new characters or adjust the characters order within the double quotation marks on the right of the equal sign.

Don't modify the item on the left of the equal sign.

```

ime.txt x
0 1.0 2.0 3.0 4.0
1 [2aB]
2 1 = "1"
3 2 = "2abcABC"
4 3 = "3defDEF"
5 4 = "4ghiGHI"
6 5 = "5jklJKL"
7 6 = "6mnoMNO"
8 7 = "7pqrsPQRS"
9 8 = "8tuvTUV"
10 9 = "9wxyzWXYZ"
11 0 = "0"
12 * = "*, '?! \- () @ / : _ ; + & % = < > £ $ ¥ ¤ [ ] { } ~ ^ ¡ ¢ § ¨"
13 # = "#"
14
15 [abc]
16 1 = ""
17 2 = "abc2âæãäåäâç"
18 3 = "def3ëèéèè"
19 4 = "ghi4îïîîî"
20 5 = "jkl5ê"
21 6 = "mno6ôøöôöðñ"
22 7 = "pqrs7ßs"
23 8 = "tuv8ùúúù"
24 9 = "wxyz9ýþ"
25 0 = ""
26 * = "*, '?! \- () @ / : _ ; + & % = < > £ $ ¥ ¤ [ ] { } ~ ^ ¡ ¢ § ¨"
27 # = "#"

```

Don't rename it.

Add new characters here or adjust the order of these characters.  
For example: abc2âæãäåäâç#@ or âæãäåäâçabc2.

Don't modify the items on the left of the equal sign.

3. Save the keypad input method file and place it to the provisioning server (for example, 192.168.10.25).
4. Specify the access URL of the custom keypad input method file in the configuration file.

**Note**

When adding new characters for the existing input method, ensure that the added characters are supported by IP phones.

The IP phones can only recognize the keypad input method files uploaded using Unicode encoding.

Do not rename the keypad input method filename.

**Procedure**

Specify the access URL of the custom keypad input method file using the configuration file.

<b>Configuration File</b>	<y0000000000xx>.cfg	Specify the access URL of the custom keypad input method file.  <b>Parameter:</b> gui_input_method.url
		Delete custom keypad input method file of the phone user interface.  <b>Parameter:</b> gui_input_method.delete

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
gui_input_method.url	URL within 511 characters	Blank
<b>Description:</b> Configures the access URL of the custom keypad input method file.  <b>Example:</b> gui_input_method.url = http://192.168.10.25/ime.txt  During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.25", and downloads the custom keypad input method file "ime.txt".  gui_input_method.url = http://192.168.10.25/Russian_ime.txt  During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.25", and downloads the custom keypad input method file "Russian_ime.txt" for Russian language.  <b>Note:</b> If you want to upload a custom keypad input method file for the desired language, you can name the file "language name_ime.txt".		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>gui_input_method.delete</b>	http://localhost/all or http://localhost/Name.txt	Blank
<b>Description:</b> Delete the specified or all custom keypad input method files of the phone user interface. <b>Example:</b> Delete all custom keypad input method files: gui_input_method.delete = http://localhost/all Delete a custom keypad input method file (for example, ime.txt) for the phone: gui_input_method.delete = http://localhost/ime.txt <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Specifying the Default Input Method

In addition to customizing the keypad input method file, you can also specify the default input method for the IP phone when editing or searching for contacts.

### Procedure

Specify the default input methods using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify the default input method when editing contacts. <b>Parameter:</b> directory.edit_default_input_method
		Specify the default input method when searching for contacts. <b>Parameter:</b> directory.search_default_input_method



**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>directory.edit_default_input_method</b>	<b>Abc, 2aB, 123, abc or ABC</b>	<b>Abc</b>
<p><b>Description:</b> Configures the default input method when the user edits contacts in the Local Directory, LDAP, Remote Phone Book or Blacklist.</p> <p><b>Example:</b> directory.edit_default_input_method = abc</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>directory.search_default_input_method</b>	<b>Abc, 2aB, 123, abc or ABC</b>	<b>Abc</b>
<p><b>Description:</b> Configures the default input method when the user searches for contacts in the Local Directory, LDAP, Remote Phone Book or Blacklist.</p> <p><b>Example:</b> directory.search_default_input_method = abc</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		

## Logo Customization

Logo customization allows unifying the IP phone appearance or displaying a custom image on the LCD screen such as a company logo, instead of the default system logo.

You may need to resize your logo file to fit the LCD screen. The image logo screen and the idle screen are displayed alternately. The logo may affect the invisibility of time and date, soft keys and the label of current account on the LCD screen.

In this example, a system logo (Yealink logo) is displayed on the image logo screen:



Logo is not applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.

The following table lists the supported logo file format and resolution for each phone model.

Phone Model	Logo File Format	Resolution
SIP-T42G/T42S/T41P/T41S	.dob	<b>&lt;=192*64</b> 2 gray scale
SIP-T27P/G	.dob	<b>&lt;=240*120</b> 2 gray scale
SIP-T40P/T40G/T23P/T23G/T21(P) E2/ T19(P) E2	.dob	<b>&lt;=132*64</b> 2 gray scale

#### Note

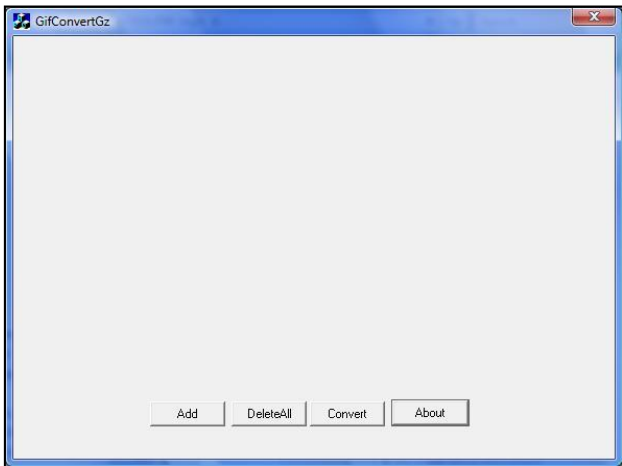
Before uploading your custom logo to IP phones, ensure your logo file is correctly formatted.

## Customizing a Logo Template File

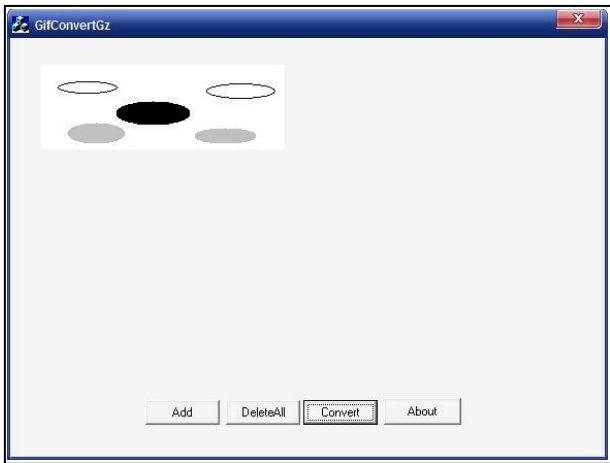
The common picture format can be \*.bmp. Yealink IP phones only support the \*.dob format logo files. Yealink provides PictureExDemo tool to convert \*.bmp format to \*.dob format. You can ask the distributor or Yealink FAE for the PictureExDemo tool.

**To customize a dob formatted logo file using the PictureExDemo tool:**

1. Double click the PictureExDemo.exe.



2. Click **Add** button to open a \*.bmp file.  
You can repeat the second step to add multiple original picture files.
3. Click the **Convert** button.



Then you can find the **DOB** logo files in the **adv** directory.

## Configuring the Logo Shown on the Idle Screen

### Procedure

The logo shown on the idle screen can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the logo shown on the LCD screen. <b>Parameter:</b> phone_setting.lcd_logo.mode
------------------------------------------------------	---------------------	-------------------------------------------------------------------------------------------------

		<p>Specify the access URL of the custom logo file.</p> <p><b>Parameter:</b></p> <p>lcd_logo.url</p>
		<p>Delete all custom logo files.</p> <p><b>Parameter:</b></p> <p>lcd_logo.delete</p>
<b>Web User Interface</b>		<p>Configure the logo shown on the idle screen.</p> <p>Upload the custom logo file.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?parameter=features-general&amp;q=load</p>

**Details of Configuration Parameters:**

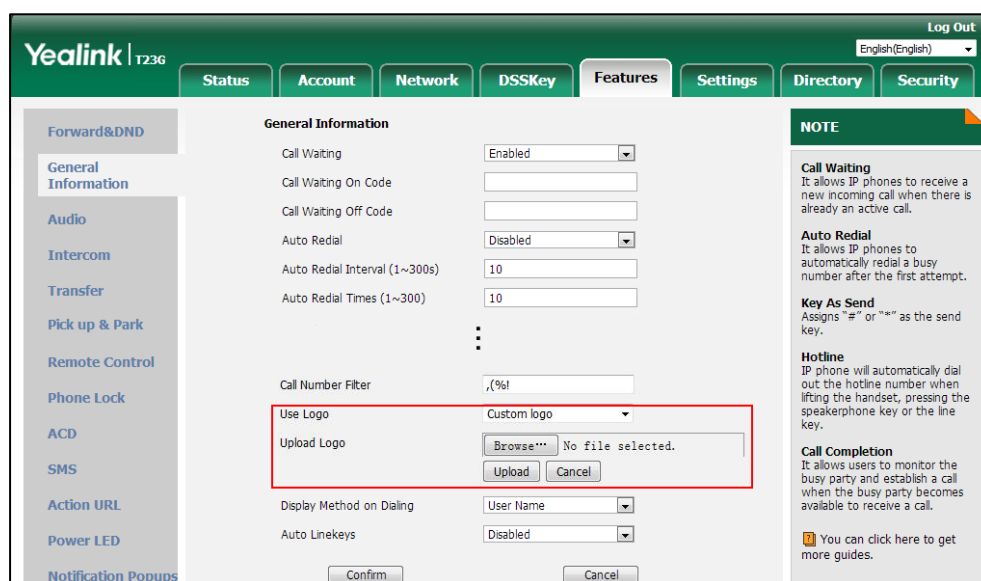
Parameters	Permitted Values	Default
<b>phone_setting.lcd_logo.mode</b>	<b>0, 1 or 2</b>	<b>0</b>
<b>Description:</b> Configures the logo shown on the LCD screen. <b>0</b> -Off, the IP phone is not allowed to display a logo. <b>1</b> -System Logo, the LCD screen will display the system logo. <b>2</b> -Custom Logo, the LCD screen will display the custom logo (you need to upload a custom logo file to the IP phone). <b>Note:</b> It is not applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones. <b>Web User Interface:</b> Features->General Information->Use Logo <b>Phone User Interface:</b> None		
<b>lcd_logo.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the access URL of the custom logo file. <b>Example:</b> lcd_logo.url = <a href="http://192.168.10.25/logo.dob">http://192.168.10.25/logo.dob</a> During the auto provisioning process, the IP phone connects to the provisioning server		

Parameters	Permitted Values	Default
<p>"192.168.10.25", and downloads the custom logo file "logo.dob".</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.lcd_logo.mode" is set to 2 (Custom Logo). It is not applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.</p> <p><b>Web User Interface:</b> Features-&gt;General Information-&gt;Upload Logo</p> <p><b>Phone User Interface:</b> None</p>		
lcd_logo.delete	http://localhost/all	Blank
<p><b>Description:</b> Deletes all custom logo files.</p> <p><b>Example:</b> lcd_logo.delete = http://localhost/all</p> <p><b>Note:</b> It is not applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		

**To configure an image logo via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Use Logo**.

- If you select **Custom logo**, you can click **Browse** to select the logo file from your local system, and then click **Upload** to upload the file.



- Click **Confirm** to accept the change.

## Softkey Layout

Softkey layout is used to customize the soft keys at the bottom of the LCD screen to best meet users' requirements. In addition to specifying which soft keys to display, you can determine their display order. It can be configured based on call states.

The following shows the softkeys displaying on the phone in the Dialing state:



You can configure the softkey layout using the softkey layout templates for different call states. For more information on how to configure a softkey layout template, refer to [Customizing Softkey Layout Template File](#) on page 253.

You are allowed to configure the EDK soft keys in different call states. For more information on EDK soft keys, refer to [Configuring EDK Soft Keys](#) on page 698.

## Customizing Softkey Layout Template File

The softkey layout template allows you to customize soft key layout for different call states. The call states include CallFailed, CallIn, Connecting, Dialing (not applicable to SIP-T48G/S), RingBack and Talking.

You can ask the distributor or Yealink FAE for softkey layout template. You can also obtain the softkey layout template online:

<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>. For more information on obtaining the softkey layout template, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

The following table lists soft keys available for IP phones in different call states.

Call State		Default Soft Keys	Optional Soft Keys
CallFailed (Call Failed)		NewCall Empty Empty Empty	Empty Switch End Call
CallIn (Incoming Call)		Answer Forward Silence Reject	Empty Switch Decline
Connecti ng	Connecting	Empty Empty Empty End Call	Empty Switch
	SemiAttendTrans (Transfer Connecting)	Transfer Empty Empty End Call	Empty Switch
Dialing (not applicable to SIP-T48G/S IP phones)		Send IME Delete End Call	Empty History Switch Line (not applicable to SIP-T19(P) E2 IP phones) Favorite (Directory) GPickup DPickup

Call State		Default Soft Keys	Optional Soft Keys
			Retrieve
RingBack	RingBack	Empty Empty Empty End Call	Empty Switch
	SemiAttendTrans Back (Transfer Ring Back)	Transfer Empty Empty End Call	Empty Switch
Talking	Talk (On Talk)	Transfer Hold Conference End Call	Empty Mute SWAP NewCall Switch Answer Reject PriHold Park GPark RTP Status
	Hold	Transfer Resume NewCall End Call	Empty Switch Answer Reject
	Held	Transfer Empty Conference End Call	Empty Switch Answer Reject NewCall Hold
	PreTrans (Transfer to, not applicable to SIP-T48G/S IP phones)	B Transfer Send Delete End Call	Empty IME Directory Switch



Call State		Default Soft Keys	Optional Soft Keys
	Conferenced	Empty Hold Split End Call	Empty Switch Answer Reject Mute Manage (not applicable to SIP-T48G/S IP phones) RTP Status

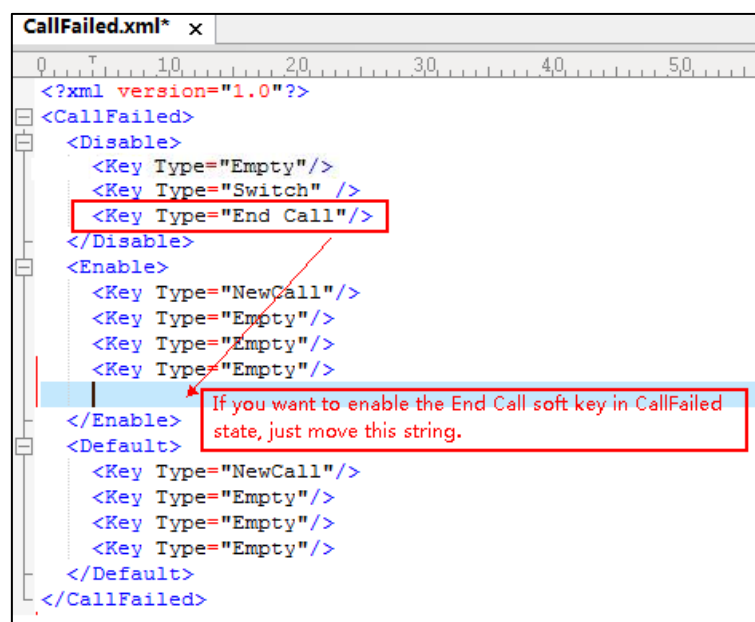
When editing a softkey layout template, learn the following:

- `<Call States>` indicates the start of a template and `</Call States>` indicates the end of a template. For example, `<CallFailed> </CallFailed>`.
- `<Disable>` indicates the start of the disabled soft key list and `</Disable>` indicates the end of the soft key list. The disabled soft keys are not displayed on the LCD screen.
- Create disabled soft keys between `<Disable>` and `</Disable>`.
- `<Enable>` indicates the start of the enabled soft key list and `</Enable>` indicates the end of the soft key list. The enabled soft keys are displayed on the LCD screen.
- Create enabled soft keys between `<Enable>` and `</Enable>`.
- `<Default>` indicates the start of the default soft key list and `</Default>` indicates the end of the default soft key list. The default soft keys are displayed on the LCD screen by default.

**To customize a softkey layout template:**

1. Open the template file using an ASCII editor.

- For each soft key that you want to enable, move the string from the disabled soft key list to enabled soft key list in the file.



For each soft key that you want disabled, just move the string from the enabled soft key list to disabled soft key list.



- Save the change and place this file to the provisioning server.
- Specify the access URL of the softkey layout template in the configuration file.

## Procedure

Specify the access URL of the softkey layout template using the configuration file.

Central Provisioning (Configuration File)	<y0000000000xx>.cfg	Configure the softkey layout. <b>Parameter:</b> phone_setting.custom_softkey_enable
		Specify the access URL of the softkey layout template. <b>Parameters:</b> custom_softkey_call_failed.url custom_softkey_call_in.url custom_softkey_connecting.url custom_softkey_dialing.url custom_softkey_ring_back.url custom_softkey_talking.url
Web User Interface		Configure the softkey layout. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?phone_setting.custom_softkey_enable=settings-softkey&q=load

## Details of Configuration Parameters:





Parameters	Permitted Values	Default
phone_setting.custom_softkey_enable	0 or 1	0
<b>Description:</b> Enables or disables custom soft keys layout feature. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Settings->Softkey Layout->Custom Softkey <b>Phone User Interface:</b> None		
custom_softkey_call_failed.url	URL within 511 characters	Blank
<b>Description:</b>		

Parameters	Permitted Values	Default
<p>Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Call Failed state.</p> <p><b>Example:</b></p> <p>custom_softkey_call_failed.url = http:// 192.168.1.20/XMLfiles/CallFailed.xml</p> <p>During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.20", and downloads the CallFailed.xml file from the "XMLfiles" directory.</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.custom_softkey_enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>custom_softkey_call_in.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the access URL of the custom file for the soft key presented on the LCD screen when in the CallIn state.</p> <p><b>Example:</b></p> <p>custom_softkey_call_in.url = http://192.168.1.20/XMLfiles/CallIn.xml</p> <p>During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.20", and downloads the CallIn.xml file from the "XMLfiles" directory.</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.custom_softkey_enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>custom_softkey_connecting.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Connecting (callout) state.</p> <p><b>Example:</b></p> <p>custom_softkey_connecting.url = http://192.168.1.20/XMLfiles/Connecting.xml</p> <p>During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.20", and downloads the Connecting.xml file from the "XMLfiles" directory.</p>		

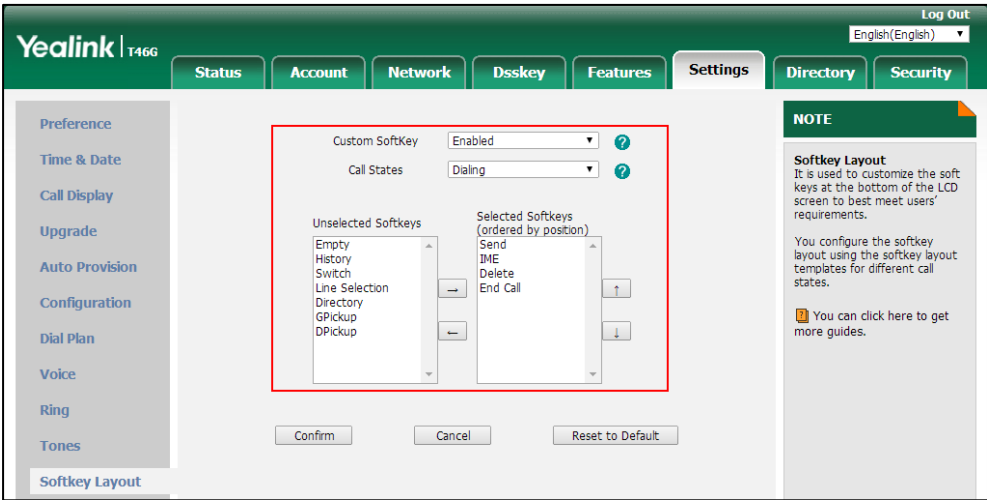
Parameters	Permitted Values	Default
<p><b>Note:</b> It works only if the value of the parameter "phone_setting.custom_softkey_enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>custom_softkey_dialing.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Dialing state.</p> <p><b>Example:</b> custom_softkey_dialing.url = http://192.168.1.20/XMLfiles/Dialing.xml</p> <p>During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.20", and downloads the Dialing.xml file from the "XMLfiles" directory.</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.custom_softkey_enable" is set to 1 (Enabled). It is not applicable to SIP-T48G/S IP phones.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>custom_softkey_ring_back.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of the custom file for the soft key presented on the LCD screen when in the RingBack state.</p> <p><b>Example:</b> custom_softkey_ring_back.url = http://192.168.1.20/XMLfiles/RingBack.xml</p> <p>During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.20", and downloads the RingBack.xml file from the "XMLfiles" directory.</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.custom_softkey_enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		

Parameters	Permitted Values	Default
<b>custom_softkey_talking.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Talking state.</p> <p><b>Example:</b></p> <p>custom_softkey_talking.url = http://192.168.1.20/XMLfiles/Talking.xml</p> <p>During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.20", and downloads the Talking.xml file from the "XMLfiles" directory.</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.custom_softkey_enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure softkey layout via web user interface:**

1. Click on **Settings->Softkey Layout**.
2. Select the desired value from the pull-down list of **Custom Softkey**.
3. Select the desired state from the pull-down list of **Call States**.
4. Select the desired soft key from the **Unselected Softkeys** column and then click  .  
The selected soft key appears in the **Selected Softkeys** column. If more than four soft keys are selected, a **More** soft key will appear on the LCD screen, and the selected soft keys are displayed in two pages.
5. Repeat the step 4 to add more soft keys to the **Selected Softkeys** column.
6. To remove the soft key from the **Selected Softkeys** column, select the desired soft key and then click  .
7. To adjust the display order of soft keys, select the desired soft key and then click  or  .

The LCD screen displays the soft keys in the adjusted order.



- 8. Click **Confirm** to accept the change.

## Key As Send

Key as send allows assigning the pound key ("#") or asterisk key ("\*") as the send key.

### Procedure

Key as send can be configured using the following methods.

Central Provisioning (Configuration File)	<y0000000000xx>.cfg	Configure a send key. <b>Parameter:</b> features.key_as_send
		Configure send pound key. <b>Parameter:</b> features.send_pound_key
Web User Interface		Configure a send key. Configure send pound key. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =features-general&q=load
Phone User Interface		Configure a send key.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>features.key_as_send</b>	<b>0, 1 or 2</b>	<b>1</b>
<p><b>Description:</b></p> <p>Configures the "#" or "*" key as the send key.</p> <p><b>0</b>-Disabled, neither "#" nor "*" can be used as the send key.</p> <p><b>1</b>-# key, the pound key is used as the send key.</p> <p><b>2</b>-* key, the asterisk key is used as the send key.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Key As Send</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Others-&gt;General-&gt;Key As Send</p>		
<b>features.send_pound_key</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to dial out "##" when the user only presses the # key on dialing screen or pre-dialing screen.</p> <p><b>0</b>-Disabled, the IP phone will dial out "#" when the user presses the # key for the second time.</p> <p><b>1</b>-Enabled, the IP phone will dial out "##" when the user presses the # key for the third time.</p> <p><b>Note:</b> It works only if the value of the parameter "features.key_as_send" is set to 1 (# key).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Send Pound Key</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure a send key via web user interface:**

1. Click on **Features->General Information**.



2. Select the desired value from the pull-down list of **Key As Send**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected. Under 'General Information', the 'Key As Send' field is highlighted with a red box, showing a pull-down menu with '#' and '\*' options. Other fields include 'Call Waiting' (Enabled), 'Call Waiting On Code', 'Call Waiting Off Code', 'Auto Redial' (Disabled), 'Auto Redial Interval (1~300s)' (10), 'Auto Redial Times (1~300)' (10), and 'Reserve # in User Name' (Enabled). A 'NOTE' section on the right provides details for 'Call Waiting', 'Auto Redial', 'Key As Send', and 'Hotline'.

3. Click **Confirm** to accept the change.

#### To configure a send key via phone user interface:

1. Press **Menu->Features->Others->General->Key As Send**.
2. Press or , or the **Switch** soft key to select # or \* from the **Key As Send** field, or select **Disabled** to disable this feature.
3. Press the **Save** soft key to accept the change.

## Send Tone

Send tone allows the IP phone to play a key tone when a user presses the send key. It works only if key tone is enabled. For more information on key tone, refer to [Key Tone](#) on page 264.

### Procedure

Send tone can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure a send tone. <b>Parameter:</b> features.send_key_tone
<b>Web User Interface</b>		Configure a send tone. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =features-audio&q=load

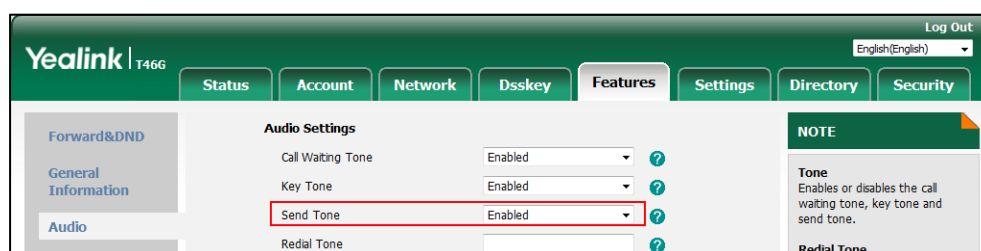
#### Details of Configuration Parameter:

Parameter	Permitted Values	Default
features.send_key_tone	0 or 1	1

Parameter	Permitted Values	Default
<p><b>Description:</b></p> <p>Enables or disables the IP phone to play a key tone when a user presses a send key.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "features.key_tone" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Audio-&gt;Send Tone</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure a send tone via web user interface:**

1. Click on **Features->Audio**.
2. Select the desired value from the pull-down list of **Send Tone**.



3. Click **Confirm** to accept the change.

## Key Tone

Key tone allows the IP phone to play a key tone when a user presses any key.

### Procedure

Key tone can be configured using the following methods.

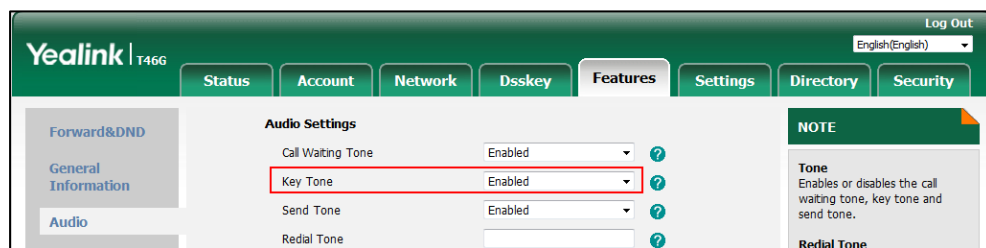
<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure a key tone. <b>Parameter:</b> features.key_tone
<b>Web User Interface</b>		Configure a key tone. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?path=features-audio&q=load
<b>Phone User Interface</b>		Configure a key tone.

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.key_tone</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to play a key tone when a user presses any key on your phone keypad.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Audio-&gt;Key Tone</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Sound-&gt;Key Tone</p>		

**To configure a key tone via web user interface:**

1. Click on **Features->Audio**.
2. Select the desired value from the pull-down list of **Key Tone**.



3. Click **Confirm** to accept the change.

**To configure a key tone via web user interface:**

1. Press **Menu->Basic->Sound->Key Tone**.
2. Press or , or the **Switch** soft key to select the desired value from the **Key Tone** field.
3. Press the **Save** soft key to accept the change.

## Dial Plan

Dial plan is a string of characters that governs the way for IP phones to process the inputs received from the IP phone's keypads. You can use regular expression to define dial plan. Regular expression, often called a pattern, is an expression that specifies a set of strings. A regular expression provides a concise and flexible means to "match" (specify and recognize) strings of text, such as particular characters, words, or patterns of characters.

Yealink IP phones support two methods to help creating a dial plan: [Dial Plan using XML Template Files](#) (old dial plan mechanism) and [Dial Plan using Digit Map String Rules](#) (new dial plan mechanism). Old dial plan method supports replace rule, dial now, area code and block out features, and each dial plan feature need its own matching rule. By contrast, new dial plan supports one or more matching rules in one digit map string. It is helpful for completing multiple dial plan features: replace, dial now, block out, and so on by one matching string.

If you enable new dial plan mechanism, old dial plan will be ignored.

## Dial Plan using XML Template Files

Yealink IP phones support the following dial plan features:

- [Replace Rule](#)
- [Dial Now](#)
- [Area Code](#)
- [Block Out](#)

You can configure these dial plan features via web user interface or using configuration file. You can select to add a replace rule/dial now rule one by one or using the replace rule/dial now template file to add multiple replace rules at a time.

You need to know the following basic regular expression syntax when creating old dial plan:

.	The dot "." can be used as a placeholder or multiple placeholders for any string. Example: "12." would match "123", "1234", "12345", "12abc", and so on.
x	The "x" can be used as a placeholder for any character. Example: "12x" would match "121", "122", "123", "12a", and so on.
-	The dash "-" can be used to match a range of characters within the brackets. Example: "[5-7]" would match the number "5", "6" or "7".
,	The comma "," can be used as a separator within the bracket. Example: "[2,5,8]" would match the number "2", "5" or "8".
[]	The square bracket "[]" can be used as a placeholder for a single character which matches any of a set of characters. Example: "91[5-7]1234" would match "9151234", "9161234", "9171234".
()	The parenthesis "()" can be used to group together patterns, for instance, to logically combine two or more patterns. Example: "([1-9])([2-7])3" would match "923", "153", "673", and so on.
\$	The "\$" followed by the sequence number of a parenthesis means

	<p>the characters placed in the parenthesis. The sequence number stands for the corresponding parenthesis. Example:</p> <p>A replace rule configuration, Prefix: "001(xxx)45(xx)", Replace: "9001\$145\$2". When you dial out "0012354599" on your phone, the IP phone will replace the number with "9001<b>235</b>45<b>99</b>". "\$1" means 3 digits in the first parenthesis, that is, "235". "\$2" means 2 digits in the second parenthesis, that is, "99".</p>
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## Replace Rule

Replace rule is an alternative string that replaces the numbers entered by the user. IP phones support up to 100 replace rules, which can be created either one by one or in batch using a replace rule template. For more information on how to customize a replace rule template, refer to [Customizing Replace Rule Template File](#) on page 269.

### Procedure

Replace rule can be created using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	<p>Create the replace rule for the IP phone.</p> <p><b>Parameters:</b></p> <p>dialplan.replace.prefix.X</p> <p>dialplan.replace.replace.X</p> <p>dialplan.replace.line_id.X</p>
<b>Web User Interface</b>		<p>Create the replace rule for the IP phone.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=settings-dialplan&amp;q=load</p>

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>dialplan.replace.prefix.X</b> (X ranges from 1 to 100)	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the entered number to be replaced.</p> <p><b>Example:</b></p> <p>dialplan.replace.prefix.1 = 1</p> <p><b>Note:</b> It works only if the values of the parameters "dialplan.digitmap.enable" and</p>		

Parameters	Permitted Values	Default
<p>"account.X.dialplan.digitmap.enable" are set to 0 (Disabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Dial Plan-&gt;Replace Rule-&gt;Prefix</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>dialplan.replace.replace.X</b></p> <p>(X ranges from 1 to 100)</p>	String within 32 characters	Blank
<p><b>Description:</b></p> <p>Configures the alternate number to replace the entered number.</p> <p><b>Example:</b></p> <p>dialplan.replace.prefix.1 =1 and dialplan.replace.replace.1 = 254245</p> <p>When you enter the number "1" and press the send key, the number "254245" will replace the entered number "1".</p> <p><b>Note:</b> It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Dial Plan-&gt;Replace Rule-&gt;Replace</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>dialplan.replace.line_id.X</b></p> <p>(X ranges from 1 to 100)</p>	Refer to the following content	Blank (for all lines)
<p><b>Description:</b></p> <p>Configures the desired line to apply the replace rule. The digit 0 stands for all lines. If it is left blank, the replace rule will apply to all lines on the IP phone.</p> <p><b>Permitted Values:</b></p> <p>0 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G)</p> <p>0 to 12 (for SIP-T52S/T42G/T42S)</p> <p>0 to 6 (for SIP-T41P/T41S/T27P/T27G)</p> <p>0 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>0 to 2 (for SIP-T21(P) E2)</p> <p><b>Example:</b></p> <p>dialplan.replace.line_id.1 = 1,2</p> <p><b>Note:</b> Multiple line IDs are separated by commas. It is not applicable to SIP-T19(P) E2 IP phones. It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).</p>		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Settings->Dial Plan->Replace Rule->Account  <b>Phone User Interface:</b> None		

#### To create a replace rule via web user interface:

1. Click on **Settings->Dial Plan->Replace Rule**.
2. Enter the string in the **Prefix** field.
3. Enter the string in the **Replace** field.
4. Enter the desired line ID in the **Account** field or leave it blank.

If you leave this field blank or enter 0, the replace rule will apply to all accounts on the IP phone.

**Yealink T46G** Log Out English(English)

Status Account Network Dsskey Features **Settings** Directory Security

Preference  
Time & Date  
Call Display  
Upgrade  
Auto Provision  
Configuration  
Dial Plan  
Voice  
Ring  
Tones  
Softkey Layout  
TR069  
Voice Monitoring

**Replace Rule** Dial Now Area Code Block Out

Index	Prefix	Replace	Account
1			
2			
3			
4			
5			
6			
7			
8			
9			
10			

Prefix: 1 Replace: 254245 Account: 1,2

Add Edit Del

**NOTE**  
**Replace Rule:** An alternative string that replaces the entered numbers.  
**Dial Now:** Automatically dial out the entered numbers.  
**Area Code:** Automatically add the area code before the numbers when dialing.  
**Block Out:** It prevents users from dialing out specific numbers.  
 " " represents any string.  
 "x" represents any character.  
 "[]" match a range of characters within the brackets.  
 " " a separator within the bracket.  
 "[]" a character matches any of character sets.  
 "()" combines two or more patterns.  
 "s": followed by the sequence number of a parenthesis means the characters placed in the parenthesis.  
 You can click here to get more guides.

5. Click **Add** to add the replace rule.

## Customizing Replace Rule Template File

The replace rule template helps with the creation of multiple replace rules.

You can ask the distributor or Yealink FAE for replace rule template. You can also obtain the replace rule template online:

<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>. For more information on obtaining the replace rule template, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

When editing a replace rule template file, learn the following:

- <DialRule> indicates the start of the template file and </DialRule> indicates the end of the

template file.

- When specifying the desired line(s) to apply the replace rule, the valid values are 0 and line ID. Multiple line IDs are separated by commas. It is not applicable to SIP-T19(P) E2 IP phones.

The following table lists valid values of line ID for each phone model.

Phone Model	Values	Description
SIP-T54S/T48G/T48S/T46G/T46S/T29G	0~16	0 stands for all lines 1~16 stand for line1~line16
SIP-T52S/T42G/T42S	0~12	0 stands for all lines 1~12 stand for line1~line12
SIP-T41P/T41S/T27P/T27G	0~6	0 stands for all lines 1~6 stand for line1~line6
SIP-T40P/T40G/T23P/T23G	0~3	0 stands for all lines 1~3 stand for line1~line3
SIP-T21(P) E2	0~2	0 stands for all lines 1~2 stand for line1~line2

- At most 100 replace rules can be added to the IP phone.

The expression syntax in the replace rule template is the same as that introduced in the section [Dial Plan using XML Template Files](#) on page 266.

#### To customize a replace rule template:

- Open the template file using an ASCII editor.
- Create replace rules between <DialRule> and </DialRule>.

#### For example:

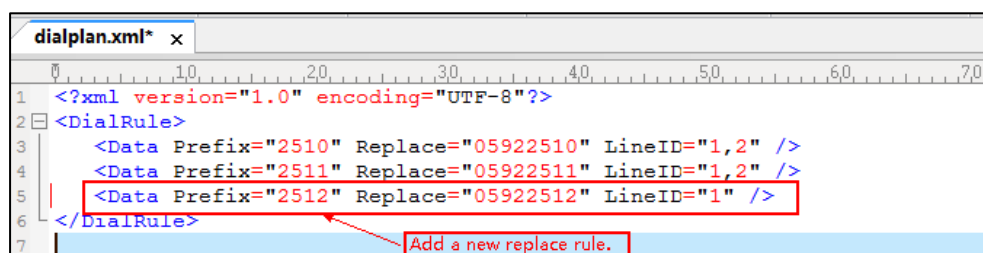
```
<Data Prefix="2512" Replace="05922512" LineID="1" />
```

Where:

Prefix="" specifies the numbers to be replaced.

Replace="" specifies the alternate string instead of what the user enters.

LineID="" specifies the desired line(s) for this rule. When you leave it blank or enter 0, this replace rule will apply to all lines.



If you want to change the replace rule, specify the values within double quotes.

- Save the change and place this file to the provisioning server.



- Specify the access URL of the replace rule template in the configuration file.

### Procedure

Specify the access URL of the replace rule template using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify the access URL of the replace rule template. <b>Parameter:</b> dialplan_replace_rule.url
------------------------------------------------------	---------------------	--------------------------------------------------------------------------------------------------------

### Details of Configuration Parameter:

Parameter	Permitted Values	Default
dialplan_replace_rule.url	URL within 511 characters	Blank
<p><b>Description:</b> Configures the access URL of the replace rule template file.</p> <p><b>Example:</b> dialplan_replace_rule.url = http://192.168.10.25/dialplan.xml</p> <p>During the auto provisioning process, the IP phone connects to the provisioning server "192.168.10.25", and downloads the replace rule file "dialplan.xml".</p> <p><b>Note:</b> It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		

## Dial Now

Dial now is a string used to match numbers entered by the user. When entered numbers match the predefined dial now rule, the IP phone will automatically dial out the numbers without pressing the send key. IP phones support up to 100 dial now rules, which can be created either one by one or in batch using a dial now rule template. For more information on how to customize a dial now template, refer to [Customizing Dial Now Template File](#) on page 275.

### Time Out for Dial Now Rule

The IP phone will automatically dial out the entered number, which matches the dial now rule, after a specified period of time.

## Procedure

Dial now rule can be created using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Create the dial now rule for the IP phone. <b>Parameters:</b> dialplan.dialnow.rule.X dialplan.dialnow.line_id.X
		Configure the delay time for the dial now rule. <b>Parameter:</b> phone_setting.dialnow_delay
<b>Web User Interface</b>		Create the dial now rule for the IP phone. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?path=settings-dialnow&q=load
		Configure the delay time for the dial now rule. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?path=features-general&q=load

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>dialplan.dialnow.rule.X</b> (X ranges from 1 to 100)	<b>String within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the dial now rule (the string used to match the numbers entered by the user). When entered numbers match the predefined dial now rule, the IP phone will automatically dial out the numbers without pressing the send key. <b>Example:</b> dialplan.dialnow.rule.1 = 123 <b>Note:</b> It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled). <b>Web User Interface:</b> Settings->Dial Plan->Dial Now->Rule		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>dialplan.dialnow.line_id.X</b> (X ranges from 1 to 100)	Refer to the following content	Blank (for all lines)
<b>Description:</b> Configures the desired line to apply the dial now rule. The digit 0 stands for all lines. If it is left blank, the dial now rule will apply to all lines on the IP phone.  <b>Permitted Values:</b> 0 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G) 0 to 12 (for SIP-T52S/T42G/T42S) 0 to 6 (for SIP-T41P/T41S/T27P/T27G) 0 to 3 (for SIP-T40P/T40G/T23P/T23G) 0 to 2 (for SIP-T21(P) E2)  <b>Example:</b> dialplan.dialnow.line_id.1 = 1,2  <b>Note:</b> Multiple line IDs are separated by commas. It is not applicable to SIP-T19(P) E2 IP phones. It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).  <b>Web User Interface:</b> Settings->Dial Plan->Dial Now->Account  <b>Phone User Interface:</b> None		
<b>phone_setting.dialnow_delay</b>	Integer from 0 to 14	1
<b>Description:</b> Configures the delay time (in seconds) for the dial now rule.  When entered numbers match the predefined dial now rule, the IP phone will automatically dial out the entered number after the designated delay time.  If it is set to 0, the IP phone will automatically dial out the entered number immediately.  <b>Note:</b> It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).  <b>Web User Interface:</b> Features->General Information->Time Out for Dial Now Rule  <b>Phone User Interface:</b> None		

**To create a dial now rule via web user interface:**

1. Click on **Settings->Dial Plan->Dial Now**.
2. Enter the desired value in the **Rule** field.
3. Enter the desired line ID in the **Account** field or leave it blank.

If you leave this field blank or enter 0, the dial now rule will apply to all accounts on the IP phone.

The screenshot shows the Yealink T466 web interface. The 'Settings' tab is active, and the 'Dial Now' sub-tab is selected. A table with 10 rows is displayed. The first row is highlighted. Below the table, there are three buttons: 'Add', 'Edit', and 'Del'. A red box highlights the 'Add' button. To the right of the table, there is a 'NOTE' section with a list of symbols and their meanings: '\*' for any string, 'x' for any character, '-' for a range of characters, '[' for a separator within the bracket, ']' for a character matching any of the character sets, '()' for combining two or more patterns, and '\$' for a sequence number of a parenthesis means the characters placed in the parenthesis. Below the note, there is a link: 'You can click here to get more guides.'

4. Click **Add** to add the dial now rule.

**To configure the time out for dial now rule via web user interface:**

1. Click on **Features->General Information**.
2. Enter the desired time within 0-14 (in seconds) in the **Time Out for Dial Now Rule** field.

The screenshot shows the Yealink T466 web interface. The 'Features' tab is active, and the 'General Information' sub-tab is selected. A list of features is displayed, each with a dropdown menu and a question mark icon. The 'Time Out for Dial Now Rule' field is highlighted with a red box, showing the value '1'. To the right of the list, there is a 'NOTE' section with a list of features and their descriptions: 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion'. Below the note, there is a link: 'You can click here to get more guides.'

3. Click **Confirm** to accept the change.

## Customizing Dial Now Template File

The dial now template helps with the creation of multiple dial now rules. After setup, place the dial now template to the provisioning server and specify the access URL in the configuration file.

You can ask the distributor or Yealink FAE for dial now template. You can also obtain the dial now template online:

<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>. For more information on obtaining the dial now template, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

When editing a dial now template, learn the following:

- `<DialNow>` indicates the start of a template and `</DialNow>` indicates the end of a template.
- When specifying the desired line(s) for the dial now rule, the valid values are 0 and line ID. Multiple line IDs are separated by commas. It is not applicable to SIP-T19(P) E2 IP phones.

The following table lists valid values of line ID for each phone model.

Phone Model	Values	Description
SIP-T54S/T48G/T48S/T46G/T46S/T29G	0~16	0 stands for all lines 1~16 stand for line1~line16
SIP-T52S/T42G/T42S	0~12	0 stands for all lines 1~12 stand for line1~line12
SIP-T41P/T41S/T27P/T27G	0~6	0 stands for all lines 1~6 stand for line1~line6
SIP-T40P/T40G/T23P/T23G	0~3	0 stands for all lines 1~3 stand for line1~line3
SIP-T21(P) E2	0~2	0 stands for all lines 1~2 stand for line1~line2

- At most 100 rules can be added to the IP phone.

The expression syntax in the dial now rule template is the same as that introduced in the section [Dial Plan using XML Template Files](#) on page 266.

### To customize a dial now template:

1. Open the template file using an ASCII editor.
2. Create dial now rules between `<DialNow>` and `</DialNow>`.

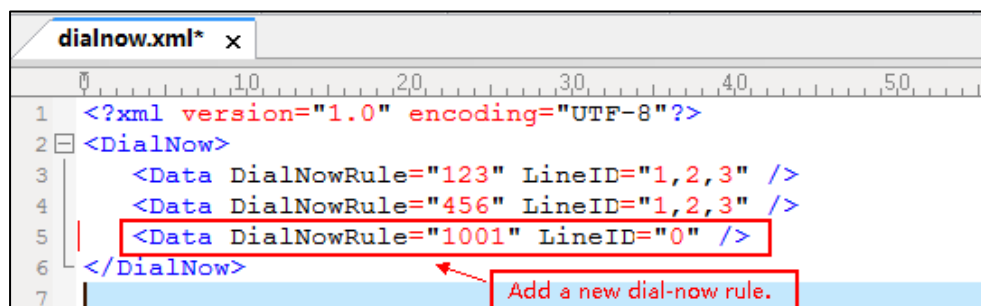
#### For example:

```
<Data DialNowRule="1001" LineID="0" />
```

Where:

DialNowRule="" specifies the dial now rule.

LineID="" specifies the desired line(s) for this rule. When you leave it blank or enter 0, this dial now rule will apply to all lines.



If you want to change the dial now rule, specify the values within double quotes.

3. Save the change and place this file to the provisioning server.
4. Specify the access URL of the dial now template.

## Procedure

Specify the access URL of the dial now template using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the access URL of the dial now template. <b>Parameter:</b> dialplan_dialnow.url
------------------------------------------------------	---------------------	-------------------------------------------------------------------------------------------------

## Details of Configuration Parameter:

Parameter	Permitted Values	Default
dialplan_dialnow.url	URL within 511 characters	Blank
<b>Description:</b> Configures the access URL of the dial now rule template file.		
<b>Example:</b> dialplan_dialnow.url = http://192.168.10.25/dialnow.xml		
During the auto provisioning process, the IP phone connects to the provisioning server "192.168.10.25", and downloads the dial now rule file "dialnow.xml".		
<b>Note:</b> It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).		
<b>Web User Interface:</b> None		
<b>Phone User Interface:</b> None		

## Area Code

Area codes are also known as Numbering Plan Areas (NPAs). They usually indicate geographical areas in one country. When entered numbers match the predefined area code rule, the IP phone will automatically add the area code before the numbers when dialing out them. IP phones only support one area code rule.

### Procedure

Area code rule can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Create the area code rule and specify the maximum and minimum lengths of entered numbers.  <b>Parameters:</b> dialplan.area_code.code dialplan.area_code.min_len dialplan.area_code.max_len dialplan.area_code.line_id
<b>Web User Interface</b>		Create the area code rule and specify the maximum and minimum lengths of entered numbers.  <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?parameters=settings-areacode&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?parameters=settings-areacode&amp;q=load</a>

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
dialplan.area_code.code	String within 16 characters	Blank
<b>Description:</b> Configures the area code to be added before the entered numbers when dialing out. <b>Example:</b> dialplan.area_code.code = 0592 <b>Note:</b> The length of the entered number must be between the minimum length configured by the parameter "dialplan.area_code.min_len" and the maximum length configured by the parameter "dialplan.area_code.max_len". It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled). <b>Web User Interface:</b> Settings->Dial Plan->Area Code->Code		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>dialplan.area_code.min_len</b>	<b>Integer from 1 to 15</b>	<b>1</b>
<b>Description:</b> Configures the minimum length of the entered numbers. <b>Note:</b> It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled). <b>Web User Interface:</b> Settings->Dial Plan->Area Code->Min Length (1-15) <b>Phone User Interface:</b> None		
<b>dialplan.area_code.max_len</b>	<b>Integer from 1 to 15</b>	<b>15</b>
<b>Description:</b> Configures the maximum length of the entered numbers. <b>Note:</b> The value must be larger than the minimum length. It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled). <b>Web User Interface:</b> Settings->Dial Plan->Area Code->Max Length (1-15) <b>Phone User Interface:</b> None		
<b>dialplan.area_code.line_id</b>	<b>Refer to the following content</b>	<b>Blank (for all lines)</b>
<b>Description:</b> Configures the desired line to apply the area code rule. The digit 0 stands for all lines. If it is left blank, the area code rule will apply to all lines on the IP phone. <b>Permitted Values:</b> 0 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G) 0 to 12 (for SIP-T52S/T42G/T42S) 0 to 6 (for SIP-T41P/T41S/T27P/T27G) 0 to 3 (for SIP-T40P/T40G/T23P/T23G) 0 to 2 (for SIP-T21(P) E2) <b>Example:</b>		



Parameters	Permitted Values	Default
<p>dialplan.area_code.line_id = 1</p> <p><b>Note:</b> Multiple line IDs are separated by commas. It is not applicable to SIP-T19(P) E2 IP phones. It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Dial Plan-&gt;Area Code-&gt;Account</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure an area code rule via web user interface:**

1. Click on **Settings->Dial Plan->Area Code**.
2. Enter the desired values in the **Code**, **Min Length (1-15)** and **Max Length (1-15)** fields.
3. Enter the desired line ID in the **Account** field or leave it blank.

If you leave this field blank or enter 0, the area code rule will apply to all accounts on the IP phone.

The screenshot shows the Yealink T46G web interface. The 'Settings' tab is selected, and the 'Area Code' sub-tab is active. The configuration fields are as follows:

Field	Value
Code	0592
Min Length (1-15)	1
Max Length (1-15)	15
Account	1

Buttons: Confirm, Cancel

**NOTE:**

- Replace Rule:** An alternative string that replaces the entered numbers.
- Dial-now:** Automatically dial out the entered numbers.
- Area Code:** Automatically add the area code before the numbers when dialing.
- Block Out:** It prevents users from dialing out specific numbers.

Legend:

- "\*": represents any string.
- "x": represents any character.
- "-": match a range of characters

4. Click **Confirm** to accept the change.

## Block Out

Block out rule prevents users from dialing out specific numbers. When entered numbers match the predefined block out rule, the LCD screen prompts "Forbidden Number". IP phones support up to 10 block out rules.

### Procedure

Block out rule can be created using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	<p>Create the block out rule for the IP phone.</p> <p><b>Parameters:</b></p> <p>dialplan.block_out.number.X</p>
------------------------------------------------------	---------------------	-----------------------------------------------------------------------------------------------------------------

	dialplan.block_out.line_id.X
<b>Web User Interface</b>	<p>Create the block out rule for the IP phone.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?parameter=settings-blackout&amp;q=load</p>

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>dialplan.block_out.number.X</b> (X ranges from 1 to 10)	String within 32 characters	Blank
<p><b>Description:</b></p> <p>Configures the block out numbers.</p> <p><b>Example:</b></p> <p>dialplan.block_out.number.1 = 4321</p> <p>When you dial the number "4321" on your phone, the dialing will fail and the LCD screen will prompt "Forbidden Number".</p> <p><b>Note:</b> It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Dial Plan-&gt;Block Out-&gt;BlockOut NumberX</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>dialplan.block_out.line_id.X</b> (X ranges from 1 to 10)	Refer to the following content	Blank (for all lines)
<p><b>Description:</b></p> <p>Configures the desired line to apply the block out rule. The digit 0 stands for all lines. If it is left blank, the block out rule will apply to all lines on the IP phone.</p> <p><b>Permitted Values:</b></p> <p>0 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G)</p> <p>0 to 12 (for SIP-T52S/T42G/T42S)</p> <p>0 to 6 (for SIP-T41P/T41S/T27P/T27G)</p> <p>0 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>0 to 2 (for SIP-T21(P) E2)</p> <p><b>Example:</b></p> <p>dialplan.block_out.line_id.1 = 1,2,3</p> <p><b>Note:</b> Multiple line IDs are separated by commas. It is not applicable to SIP-T19(P) E2 IP</p>		

Parameters	Permitted Values	Default
<p>phones. It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Dial Plan-&gt;Block Out-&gt;Account</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To create a block out rule via web user interface:**

1. Click on **Settings->Dial Plan->Block Out**.
2. Enter the desired value in the **BlockOut NumberX** field.
3. Enter the desired line ID in the **Account** field or leave it blank.

If you leave this field blank or enter 0, the block out rule will apply to all accounts on the IP phone.

The screenshot shows the Yealink T46G web interface. The 'Settings' tab is active, and the 'Block Out' sub-tab is selected. A table lists 'BlockOut Number' fields from 1 to 10. The first row, 'BlockOut Number1', is highlighted with a red border. Its 'Account' field contains the value '1,2,3'. Below the table are 'Confirm' and 'Cancel' buttons. On the right, a 'NOTE' section provides definitions for 'Replace Rule', 'Dial now', 'Area Code', and 'Block Out', along with a list of regular expression symbols and their meanings.

4. Click **Confirm** to add the block out rule.

## Dial Plan using Digit Map String Rules

Digit maps, described in [RFC 3435](#), are defined by a single string or a list of strings. If a number entered matches any string of a digit map, the call is automatically placed. If a number entered matches no string - an impossible match - you can specify the phone's behavior. You can specify the digit map timeout, the period of time before the entered number is dialed out.

You need to know the following basic regular expression syntax when creating new dial plan:

T	The timer letter "T" indicates a timer expiry. If "T" is used alone (for example, 123T), the default timeout value of 3 will be used. If "T" is not used alone (for example, 123<Tx>, x can be a digit from 0 to 99),
---	-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

	a complete match occurs when waiting x seconds after inputting 123.
x	<p>The "x" can be used as a placeholder for any digit from 0 to 9.</p> <p>Example:</p> <p>"12x" would match "121", "122", "123", and so on.</p>
[]	<p>The square bracket "[]" can be used as a placeholder for a single character which matches any of a set of characters. Example:</p> <p>"91[5-7]1234" would match "9151234", "9161234", "9171234".</p>
-	<p>The dash "-" can be used to match a range of digits within the brackets.</p> <p>Example:</p> <p>"[35-7]" would match the number "3", "5", "6" or "7".</p> <p><b>Note:</b> The digits must be concrete, for example, [3-x] is invalid.</p>
.	<p>The dot "." can be used as a placeholder or multiple placeholders, including zero, of occurrences of the preceding construct.</p> <p>Examples:</p> <p>"123.T" would match "123", "1233", "12333", "123333", and so on.</p> <p>"x.T" would match an arbitrary number.</p> <p>"[x*#+].T" would match an arbitrary character.</p> <p><b>Note:</b> If the string ends with a dot (for example, 123.), a match will occur immediately after inputting the characters before the dot (e., 123) since the dot allows for zero occurrences of the preceding construct. So we recommend you to add a letter "T" after the dot (for example, 123.T) for inputting more characters.</p>
R	<p>The letter "R" indicates that certain matched strings are replaced.</p> <p>Using a RRR syntax, you can replace the digits between the first two Rs with the digits between the last two Rs. Example:</p> <p>"R12R234R" would replace 12 with 234.</p>
<:>	<p>The letter ":" in the angle bracket indicates that certain matched strings are replaced. Using the &lt;:&gt; syntax, you can replace the digits before the colon with the digits after the colon.</p> <p>Example:</p> <p>"&lt;12:234&gt;" would replace 12 with 234. It is the same with R12R234R.</p>
!	<p>The exclamation mark "!" can be used to prevent users from dialing out specific numbers. It can only be put last in each string of the digit map.</p> <p>Example:</p> <p>"235x!" would match "2351", "2352", "2353", and so on. The number starting with 235 will be blocked to dial out.</p>

,	<p>The comma “,” can be used as a separator to generate secondary dial tone.</p> <p>Example:</p> <p>“&lt;9,:55&gt;xx”, after entering digit “9”, secondary dial tone plays and you can complete the remaining two-digit number.</p> <p><b>Note:</b> The secondary dial tone can be customized. For more information, refer to <a href="#">Tones</a> on page 724.</p>
A	<p>The letter “A” indicates the account that is applied to the digit map. You can use A alone or a combination Of A and account ID (for example, &lt;A1&gt;).</p> <p>Example:</p> <p>“123A”, the default account will be applied to the digit map.</p> <p>“123&lt;A2&gt;”, the second account will be applied to the digit map.</p> <p><b>Note:</b> It is not applicable to the digit map on a per-line basis.</p>

## Procedure

Digit map can be created using the configuration file.

Central Provisioning (Configuration Files)	<y0000000000xx>.cfg	<p>Configure digit map on a phone basis.</p> <p><b>Parameters:</b></p> <p>dialplan.digitmap.enable</p> <p>dialplan.digitmap.string</p> <p>dialplan.digitmap.interdigit_long_timer</p> <p>dialplan.digitmap.interdigit_short_timer</p> <p>dialplan.digitmap.no_match_action</p> <p>dialplan.digitmap.active.on_hook_dialing</p> <p>dialplan.digitmap.apply_to.on_hook_dial</p> <p>dialplan.digitmap.apply_to.directory_dial</p> <p>dialplan.digitmap.apply_to.history_dial</p> <p>dialplan.digitmap.apply_to.forward</p> <p>dialplan.digitmap.apply_to.press_send</p>
	<MAC>.cfg	<p>Configure digit map on a per-line basis.</p> <p><b>Parameters:</b></p> <p>account.X.dialplan.digitmap.enable</p> <p>account.X.dialplan.digitmap.string</p> <p>account.X.dialplan.digitmap.interdigit_long_timer</p> <p>account.X.dialplan.digitmap.interdigit_short_timer</p>

		mer account.X.dialplan.digitmap.no_match_action account.X.dialplan.digitmap.active.on_hook_d ialing account.X.dialplan.digitmap.apply_to.on_hook _dial account.X.dialplan.digitmap.apply_to.director y_dial account.X.dialplan.digitmap.apply_to.history_ dial account.X.dialplan.digitmap.apply_to.forward account.X.dialplan.digitmap.apply_to.press_se nd
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**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>dialplan.digitmap.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the digit map feature for the IP phone. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> The value configured by the parameter "account.X.dialplan.digitmap.enable" takes precedence over that configured by this parameter. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>dialplan.digitmap.string</b>	<b>String within 2048 characters</b>	[2-9]11   0T   011xxx.T   [0-1][2-9]xx xxxxxxx   [2-9]xx xxxxxxx   [2-9]xxxT   **x.T   +x.T   00x.T
<b>Description:</b>		

Parameters	Permitted Values	Default
<p>Configures digit map pattern used for the dial plan.</p> <p><b>Example:</b></p> <pre>dialplan.digitmap.string = &lt;[2-9]x:86&gt;3.T 0x.! 1xxx</pre> <p><b>Note:</b> The string must be compatible with the digit map feature of MGCP described in 2.1.5 of <a href="#">RFC 3435</a>. It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.string" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>dialplan.digitmap.interdigit_long_timer</b>	<b>Integer from 0 to 255</b>	<b>10</b>
<p><b>Description:</b></p> <p>Configures the time (in seconds) for the IP phone to wait before dialing an entered number if it matches part of any string of the digit map.</p> <p>If it is set to 0, the IP phone will not dial the entered number if only a partial match exists.</p> <p>The value of this parameter should be greater than that configured by the parameter "dialplan.digitmap.interdigit_short_timer".</p> <p><b>For example:</b></p> <pre>dialplan.digitmap.string = 1xxT xxxxx&lt;T1&gt;</pre> <pre>dialplan.digitmap.interdigit_long_timer = 10</pre> <pre>dialplan.digitmap.interdigit_short_timer = 5</pre> <p>When you enter 1, it matches part of two digit maps, the IP phone tries to wait 10 seconds and then dials out 1 if no numbers entered;</p> <p>When you enter 15, it also matches part of two digit maps, the IP phone tries to wait 10 seconds and then dials out 15 if no numbers entered;</p> <p>When you enter 153, it also matches part of two digit maps, the IP phone tries to wait 10 seconds. But after waiting for 5 seconds, it completely matches the first digit map and then immediately dials out 153.</p> <p><b>Note:</b> It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.interdigit_long_timer" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
None <b>Phone User Interface:</b> None		
<b>dialplan.digitmap.interdigit_short_timer</b>	<b>Refer to the following content</b>	<b>3</b>
<p><b>Description:</b></p> <p>Configures the timeout interval (in seconds) for any string of digit map.</p> <p>The IP phone will wait this many seconds before matching the entered digits to the dial plan and placing the call.</p> <p><b>Valid values are:</b></p> <ul style="list-style-type: none"> <li>Single configuration (configure a specific value for the timer letter "T" for all strings with "T" of the digit map)</li> </ul> <p><b>Example:</b></p> <p>dialplan.digitmap.interdigit_short_timer = 5</p> <p>If the value of the parameter "dialplan.digitmap.string" is set to &lt;[2-9]x:86&gt;3.T 0T, the IP phone will wait 5 seconds before matching the entered digits to this dial plan and placing the call.</p> <ul style="list-style-type: none"> <li>Distribution configuration (configure a string of positive integers separated by " " for each string of the digit map in the corresponding position)</li> </ul> <p>If there are more digit maps than timeout values, the last timeout is applied to the extra digit map. If there are more timeout values than digit maps, the extra timeout values are ignored.</p> <p><b>Example:</b></p> <p>dialplan.digitmap.interdigit_short_timer = 4 5 3 6 2 1</p> <p>If the value of the parameter "dialplan.digitmap.string" is set to &lt;[2-9]x:86&gt;3.T 2T 1xxT 0x.![2-9]11T, 4 is applied to the "&lt;[2-9]x:86&gt;3.T" digit map, 5 is applied to "2T" digit map, 3 is applied to "1xxT" digit map, 6 is applied to "0x.!" digit map, 2 is applied to the "[2-9]11T" digit map, the last digit 1 is ignored.</p> <p><b>Note:</b> It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.interdigit_short_timer" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		



Parameters	Permitted Values	Default
<b>dialplan.digitmap.no_match_action</b>	<b>0, 1 or 2</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the behavior when an impossible digit map match occurs.</p> <p><b>0</b>-prevent users from entering a number and immediately dial out the entered numbers</p> <p><b>1</b>-the dialing will fail and the LCD screen will prompt "Forbidden Number"</p> <p><b>2</b>-allow users to accumulate digits and dispatch call manually with the send key or automatically dial out the entered number after a certain period of time configured by the parameter "dialplan.digitmap.interdigit_long_timer"</p> <p><b>Note:</b> It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.no_match_action" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>dialplan.digitmap.active.on_hook_dialing</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the entered numbers to match the predefined string of the digit map in real time. It is only applicable to the on-hook dialing.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.active.on_hook_dialing" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>dialplan.digitmap.apply_to.on_hook_dial</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the entered number to match the predefined string of the digit map</p>		

Parameters	Permitted Values	Default
<p>after pressing a send key when dialing on-hook or pressing the DSS key (for example, speed dial, BLF or prefix DSS key).</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.apply_to.on_hook_dial" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>dialplan.digitmap.apply_to.directory_dial</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the digit map to be applied to the numbers dialed from the directory.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.apply_to.directory_dial" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>dialplan.digitmap.apply_to.history_dial</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the digit map to be applied to the numbers (received calls or missed calls) dialed from call log lists.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.apply_to.history_dial" takes precedence over that</p>		

Parameters	Permitted Values	Default
<p>configured by this parameter.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>dialplan.digitmap.apply_to.forward</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the digit map to be applied to the numbers that you want to forward to when performing call forward.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, the incoming calls will be forwarded to a desired destination number according to the string of the digit map.</p> <p><b>Note:</b> It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.apply_to.forward" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>dialplan.digitmap.apply_to.press_send</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the entered number to match the predefined string of the digit map after pressing a send key. It is only applicable to the off-hook dialing.</p> <p>The off-hook dialing includes: pick up the handset, press the Speakerphone key or press the line key when the phone is idle.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.apply_to.press_send" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b></p> <p>None</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b>		
None		

### Per-Line Parameters:

The parameters listed in the above table have a per-line equivalent that you can configure. All of the per-line parameters are listed in the following table. Note that the per-line parameters take precedence over the global parameters. For example, "account.X.dialplan.digitmap.enable" takes precedence over "dialplan.digitmap.enable".

X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

Per-Line Parameters	Global Parameters
account.X.dialplan.digitmap.enable	dialplan.digitmap.enable
account.X.dialplan.digitmap.string	dialplan.digitmap.string
account.X.dialplan.digitmap.interdigit_long_timer	dialplan.digitmap.interdigit_long_timer
account.X.dialplan.digitmap.interdigit_short_timer	dialplan.digitmap.interdigit_short_timer
account.X.dialplan.digitmap.no_match_action	dialplan.digitmap.no_match_action
account.X.dialplan.digitmap.active.on_hook_dialing	dialplan.digitmap.active.on_hook_dialing
account.X.dialplan.digitmap.apply_to.on_hook_dial	dialplan.digitmap.apply_to.on_hook_dial
account.X.dialplan.digitmap.apply_to.directory_dial	dialplan.digitmap.apply_to.directory_dial
account.X.dialplan.digitmap.apply_to.history_dial	dialplan.digitmap.apply_to.history_dial
account.X.dialplan.digitmap.apply_to.forward	dialplan.digitmap.apply_to.forward
account.X.dialplan.digitmap.apply_to.press_send	dialplan.digitmap.apply_to.press_send

## Emergency Dialplan

Yealink IP phones support dialing emergency telephone numbers when the phone is locked

(refer to [Phone Lock](#)). Due to the fact that the IP phone must have a registered account or a configured SIP server, it may not meet the need of dialing emergency telephone number at any time.

Emergency dialplan allows users to dial the emergency telephone number (emergency services number) at any time when the IP phone is powered on and has been connected to the network. It is available even if your phone keypad is locked or no SIP account is registered.

#### Note

Contact your local phone service provider for available emergency numbers in your area.

## Emergency Dial Plan

Users can configure the emergency dial plan for the phone (for example, emergency number, emergency routing). The phone determines if this is an emergency number by checking the emergency dial plan. When placing an emergency call, the call is directed to the configured emergency server. Multiple emergency servers may need to be configured for emergency routing, avoiding that emergency calls couldn't get through because of the server failure. If the phone is not locked, it checks against the regular dial plan (refer to [Dial Plan](#)). If the phone is locked, it checks against the emergency dial plan.

## Emergency Location Identification Number (ELIN)

The IP Phones support Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED). LLDP-MED allows the phone to use the location information, Emergency Location Identification Number (ELIN), sent by the switch, as a caller ID for making emergency calls. The outbound identity used in the P-Asserted-Identity (PAI) header of the SIP INVITE request is taken from the network using an LLDP-MED Emergency Location Identifier Number (ELIN). The administrator can customize the outbound identity. The custom outbound identity will be used if the phone fails to get the LLDP-MED ELIN value.

The following is an example of the PAI header:

P-asserted-identity: <sip: **1234567890**@abc.com > (where 1234567890 is the custom outbound identity.)

## P-Access-Network-Info (PANI)

When placing an emergency call, the MAC address of the phone/connected switch should be added in the P-Access-Network-Info (PANI) header of the INVITE message. It helps the aid agency to immediately identify the caller's location, improving rescue efficiency.

The following is an example of the PANI header:

P-Access-Network-Info: IEEE-802.3; eth-location="**00:15:65:74:b1:6e**" (where 00156574B16E is the phone's MAC address.)

## Procedure

Emergency dialplan can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the emergency dialplan. <b>Parameters:</b> dialplan.emergency.asserted_id_source dialplan.emergency.custom_asserted_id dialplan.emergency.server.X.address dialplan.emergency.server.X.port dialplan.emergency.server.X.transport_type dialplan.emergency.X.value dialplan.emergency.X.server_priority
--------------------------------------------------	---------------------	------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>dialplan.emergency.asserted_id_source</b>	<b>ELIN or CUSTOM</b>	<b>ELIN</b>
<b>Description:</b> Configures the precedence of source of emergency outbound identities when placing an emergency call.  If it is set to ELIN, the outbound identity used in the P-Asserted-Identity (PAI) header of the SIP INVITE request is taken from the network using an LLDP-MED Emergency Location Identifier Number (ELIN). The custom outbound identity configured by "dialplan.emergency.custom_asserted_id" will be used if the phone fails to get the LLDP-MED ELIN value.  If it is set to CUSTOM, the custom outbound identity configured by "dialplan.emergency.custom_asserted_id" will be used; if the value of the parameter "dialplan.emergency.custom_asserted_id" is left blank, the LLDP-MED ELIN value will be used.  <b>Note:</b> If the obtained LLDP-MED ELIN value is blank and no custom outbound identity, the PAI header will not be included in the SIP INVITE request.  <b>Web User Interface:</b> None  <b>Phone User Interface:</b> None		
<b>dialplan.emergency.custom_asserted_id</b>	<b>10-25 digits, SIP URI, or TEL URI</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the custom outbound identity when placing an emergency call.</p> <p>If using a TEL URI, for example, tel:+16045558000. The full URI is included in the P-Asserted-Identity (PAI) header (for example, &lt;tel:+16045558000&gt;).</p> <p>If using a SIP URI, for example, sip:1234567890123@abc.com. The full URI is included in the P-Asserted-Identity (PAI) header and the address will be replaced by the emergency server (for example, &lt;sip:1234567890123@emergency.com&gt;).</p> <p>If using a 10-25 digit number, for example, 1234567890. The SIP URI constructed from the number and SIP server (for example, abc.com) is included in the P-Asserted-Identity (PAI) header (for example, &lt;sip:1234567890@abc.com&gt;).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>dialplan.emergency.server.X.address</b></p> <p>(X ranges from 1 to 3)</p>	<p><b>IP address or domain name</b></p>	<p><b>Blank</b></p>
<p><b>Description:</b></p> <p>Configures the IP address or domain name of the emergency server X to be used for routing calls.</p> <p><b>Note:</b> If the account is registered successfully or failed (the account information has been configured), the emergency calls will be dialed using the following priority: SIP server&gt;emergency server; if the account is not registered, the emergency server will be used.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>dialplan.emergency.server.X.port</b></p> <p>(X ranges from 1 to 3)</p>	<p><b>Integer from 1 to 65535</b></p>	<p><b>5060</b></p>
<p><b>Description:</b></p> <p>Configures the port of emergency server X to be used for routing calls.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

Parameters	Permitted Values	Default
<b>dialplan.emergency.server.X.transport_type</b> (X ranges from 1 to 3)	0, 1, 2 or 3	0
<b>Description:</b> Configures the transport method the IP phone uses to communicate with the emergency server X. <b>0</b> -UDP <b>1</b> -TCP <b>2</b> -TLS <b>3</b> -DNS-NAPTR <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>dialplan.emergency.X.value</b> (X ranges from 1 to 255)	number or SIP URI	Refer to the following content
<b>Description:</b> Configures the emergency number to use on your IP phone so a caller can contact emergency services in the local area when required. <b>Default:</b> When X = 1, the default value is 911; When X = 2-255, the default value is Blank. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>dialplan.emergency.X.server_priority</b> (X ranges from 1 to 255)	a combination of digits 1, 2 and 3	1, 2, 3
<b>Description:</b> Configures the priority for the emergency servers to be used. The digits are separated by commas. The servers to be used in the order listed (left to right). The IP phone tries to send the INVITE request to the emergency server with higher priority. If the emergency server with higher priority does not respond correctly to the INVITE, then the phone tries to make the call using the emergency server with lower priority, and so forth.		



Parameters	Permitted Values	Default
<p>The IP phone tries to send the INVITE request to each emergency server for three times.</p> <p><b>Example:</b></p> <p>dialplan.emergency.1.server_priority = 2, 1, 3</p> <p>It means the IP phone sends the INVITE request to the emergency server 2 first. If the emergency server 2 does not respond correctly to the INVITE, then tries to make the call using the emergency server 1. If the emergency server 1 does not respond correctly to the INVITE, then tries to make the call using the emergency server 3. The IP phone tries to send the INVITE request to each emergency server for three times.</p> <p><b>Note:</b> If the IP address of the emergency server with higher priority has not been configured, the emergency server with lower priority will be used. If the account is registered successfully or failed (the account information has been configured), the emergency calls will be dialed using the following priority: SIP server&gt;emergency server; if the account is not registered, the emergency server will be used.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## Hotline

Hotline, sometimes referred to as hot dialing, is a point-to-point communication link in which a call is automatically directed to the preset hotline number. The IP phone automatically dials out the hotline number using the first available line after a specified time interval when you lift the handset, press the Speakerphone key or the line key. IP phones only support one hotline number.

It works only if the Off Hook Hot Line Dialing feature is disabled. For more information, refer to [Off Hook Hot Line Dialing](#) on page 297.

### Procedure

Hotline can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the hotline number. <b>Parameter:</b> features.hotline_number
		Specify the time the IP phone waits before automatically dialing out the hotline number. <b>Parameter:</b> features.hotline_delay

<b>Web User Interface</b>	<p>Configure the hotline number.</p> <p>Specify the time the IP phone waits before automatically dialing out the hotline number.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load</p>
<b>Phone User Interface</b>	<p>Configure the hotline number.</p> <p>Specify the time the IP phone waits before automatically dialing out the hotline number.</p>

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>features.hotline_number</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the hotline number that the IP phone automatically dials out when you lift the handset, press the Speakerphone or the line key.</p> <p>Leaving it blank disables hotline feature.</p> <p><b>Example:</b></p> <p>features.hotline_number = 1234</p> <p><b>Note:</b> Line key is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Hotline Number</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Others-&gt;Hot Line-&gt;Hotline Number</p>		
<b>features.hotline_delay</b>	<b>Integer from 0 to 10</b>	<b>4</b>
<p><b>Description:</b></p> <p>Configures the waiting time (in seconds) for the IP phone to automatically dial out the preset hotline number.</p> <p>If it is set to 0 (0s), the IP phone will immediately dial out the preset hotline number when you lift the handset, press the Speakerphone or press the line key.</p> <p>If it is set to a value greater than 0, the IP phone will wait the designated seconds before dialing out the preset hotline number when you lift the handset, press the Speakerphone key or press the line key.</p>		

Parameters	Permitted Values	Default
<p><b>Note:</b> Line key is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Hotline Delay(0~10s)</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Others-&gt;Hot Line-&gt;Hotline Delay</p>		

#### To configure hotline via web user interface:

1. Click on **Features->General Information**.
2. Enter the hotline number in the **Hotline Number** field.
3. Enter the delay time in the **Hotline Delay(0~10s)** field.

4. Click **Confirm** to accept the change.

#### To configure hotline via phone user interface:

1. Press **Menu->Features->Others->Hot Line**.
2. Enter the hotline number in the **Hotline Number** field.
3. Enter the waiting time (in seconds) in the **Hotline Delay** field.
4. Press the **Save** soft key to accept the change.

## Off Hook Hot Line Dialing

For security reasons, IP phones support off hook hot line dialing feature, which allows the phone to first dial out the pre-configured number when the user lifts the handset, presses the Speakerphone key or desired line key, dials out a call using the account with this feature enabled. The SIP server may then prompt the user to enter an activation code for call service. Only if the user enters a valid activation code, the IP phone will use this account to dial out a call successfully.

Off hook hot line dialing feature is configurable on a per-line basis and depends on support

from a SIP server.

### Note

Off hook hot line dialing feature limits the call-out permission of this account and disables the hotline feature. For example, when the phone goes off hook using the account with this feature enabled, the configured hotline number will not be dialed out automatically.

The server actions may vary from different servers.

It is also applicable to the IP call and intercom call.

## Procedure

Off hook hot line dialing can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure off hook hot line dialing feature. <b>Parameter:</b> account.X.auto_dial_enable
		Specify the number that the phone first dials out. <b>Parameter:</b> account.X.auto_dial_num

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>account.X.auto_dial_enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to first dial out a pre-configured number when a user lifts the handset, presses the Speakerphone or desired line key or dials out a call using account X. <b>0</b> -Disabled <b>1</b> -Enabled, the phone will first dial out the pre-configured number (configured by the parameter "account.X.auto_dial_num") when a user lifts the handset, presses the Speakerphone or desired line key, dials out a call using account X. <b>Note:</b> Line key is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>account.X.auto_dial_num<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the number that the IP phone first dials out when a user lifts the handset, presses the Speakerphone or desired line key, dials out a call using account X.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.auto_dial_enable" is set to 1 (Enabled). Line key is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## Directory List

Users can access frequently used directory lists by pressing the **Directory/Dir** soft key when the IP phone is idle. The lists can be Local Directory, History, Remote Phone Book and LDAP. The desired lists can be added to directory lists using a directory file (favorite\_setting.xml).

### Note

The LDAP list is hidden by default. To configure the LDAP list, you should enable LDAP feature first. For more information, refer to [Lightweight Directory Access Protocol \(LDAP\)](#) on page 519.

## Customizing a Directory Template File

You can ask the distributor or Yealink FAE for directory template. You can also obtain the directory template online:

<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>. For more information on obtaining the directory template, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

The following table lists meaning of each variable in the directory template file:

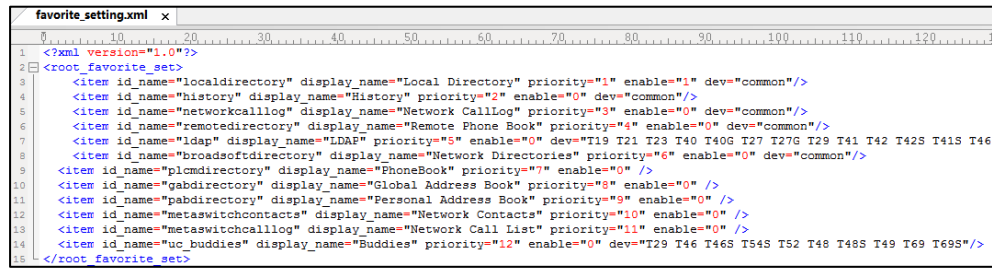
Element	Attribute	Description
<b>root_favorite_set</b>	no	File root element
<b>item</b>	no	Directory list's root element
<b>id_name</b>	localdirectory history networkcalllog	The existing directory list (For example, "localdirectory" for the local directory list).

Element	Attribute	Description
	remotedirectory ldap broadsoftdirectory plcmdirectory gabdirectory pabdirectory metaswitchcontacts metaswitchcallog uc_buddies	<b>Note:</b> Do not edit this field.
<b>display_name</b>	Local Directory History Network CallLog Remote Phone Book LDAP Network Directories PhoneBook Global Address Book Personal Address Book Network Contacts Network Call List Buddies	The display name of the directory list. <b>Note:</b> We recommend you do not edit this field. Network Directories and Network CallLog lists are hidden for IP phones in neutral firmware, which are designed for the BroadWorks environment.
<b>priority</b>	1 to 12. 1 is the highest priority, 12 is the lowest.	The display priority of the directory list.
<b>enable</b>	0/1, <b>0:</b> Disabled <b>1:</b> Enabled.	Directory list whether to display on the IP phone LCD screen.
<b>dev</b>	common; T19 T21 T23 T40 T27 T27G T29 T41 T42 T42S T41S T46 T46S T48 T48S T52 T54S.	The applicable phone models of the directory list. Common represents that the desired directory list is applicable to all IP phone models. <b>Note:</b> Do not edit this field.

**Customizing a directory template:**

1. Open the template file using an ASCII editor.
2. For each directory list that you want to configure, edit the corresponding string in the file. For example, configure the local directory list, edit the values within double quotes in the following strings:

```
<item id_name="localdirectory" display_name="Local Directory" priority="1" enable="1"
dev="common"/>
```



```
<?xml version="1.0"?>
<root favorite_set>
  <item id_name="localdirectory" display_name="Local Directory" priority="1" enable="1" dev="common"/>
  <item id_name="history" display_name="History" priority="2" enable="0" dev="common"/>
  <item id_name="networkcalllog" display_name="Network CallLog" priority="3" enable="0" dev="common"/>
  <item id_name="remotelocaldirectory" display_name="Remote Phone Book" priority="4" enable="0" dev="common"/>
  <item id_name="ldap" display_name="LDAP" priority="5" enable="0" dev="T19 T21 T23 T40 T40G T27 T27G T29 T41 T42 T42S T41S T46" />
  <item id_name="broadsoftdirectory" display_name="Network Directories" priority="6" enable="0" dev="common"/>
  <item id_name="plmcdirectory" display_name="PhoneBook" priority="7" enable="0" />
  <item id_name="gabdirectory" display_name="Global Address Book" priority="8" enable="0" />
  <item id_name="pabdirectory" display_name="Personal Address Book" priority="9" enable="0" />
  <item id_name="metaswitchcontacts" display_name="Network Contacts" priority="10" enable="0" />
  <item id_name="metaswitchcalllog" display_name="Network Call List" priority="11" enable="0" />
  <item id_name="uc_buddies" display_name="Buddies" priority="12" enable="0" dev="T29 T46 T46S T54S T52 T48 T48S T49 T69 T69S"/>
</root favorite_set>
```

3. Save the change and place this file to the provisioning server (for example, 192.168.1.20).
4. Specify the access URL of the custom directory template file in the configuration files (for example, static.directory\_setting.url = http://192.168.1.20/favorite\_setting.xml).

### Procedure

Directory can be configured using the following methods.





<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify the access URL of the directory template file. <b>Parameter:</b> static.directory_setting.url
<b>Web User Interface</b>		Configure the directory. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=contacts-favorite&q=load

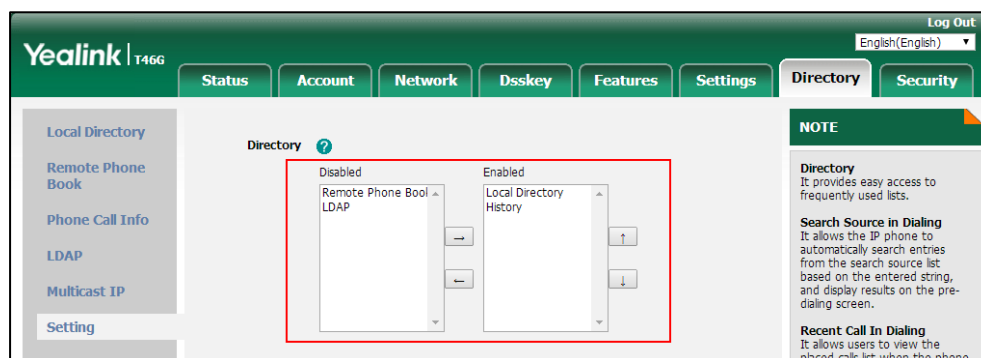
### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
static.directory_setting.url	URL within 511 characters	Blank
<b>Description:</b> Configures the access URL of the directory template file.  <b>Example:</b> static.directory_setting.url = http://192.168.1.20/favorite_setting.xml  During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.20", and downloads the directory file "favorite_setting.xml".  <b>Web User Interface:</b> Directory->Setting->Directory  <b>Phone User Interface:</b>		

Parameter	Permitted Values	Default
None		

To configure the directory via web user interface:

1. Click on **Directory->Setting**.
2. In the **Directory** block, select the desired list from the **Disabled** column and then click  .  
The selected list appears in the **Enabled** column.
3. Repeat the step 2 to add more lists to the **Enabled** column.
4. To remove a list from the **Enabled** column, select the desired list and then click  .
5. To adjust the display order of list, select the desired list and then click  or  .



6. Click **Confirm** to accept the change.

The IP phone LCD screen will display the enabled list(s) in the adjusted order.

## Search Source List In Dialing

Search source list in dialing allows the IP phone to automatically search entries from the search source list based on the entered string, and display results on the pre-dialing/dialing screen. The user can select the desired entry to dial out quickly.





The search source list can be Local Directory, History, Remote Phone Book and LDAP. The search source list can be configured using a supplied super search template file (super\_search.xml).

### Note

The LDAP list is hidden by default. To configure the LDAP list, you should enable LDAP feature first. For more information, refer to [Lightweight Directory Access Protocol \(LDAP\)](#) on page 519.

## Customizing a Super Search Template File

You can ask the distributor or Yealink FAE for super search template. You can also obtain the super search template online:

<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>. For more information on obtaining the super search template, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

The following table lists meaning of each variable in the super search template file:

Element	Attribute	Description
<b>root_super_search</b>	No	File root element
<b>Item</b>	No	Super search list's root element
<b>id_name</b>	local_directory_search calllog_search remote_directory_search ldap_search BroadSoft_directory_search BroadSoft_UC_search plcm_directory_search genband_directory_search MetaSwitch_directory_search MetaSwitch_calllog_search	The directory list (For example, "local_directory_search" for the local directory list). <b>Note:</b> Do not edit this field.
<b>display_name</b>	Local Contacts History Remote Phonebook LDAP Network Directories BroadSoft Buddies PhoneBook Personal Address Book Network Contacts Network Call List	The display name of the directory list. <b>Note:</b> We recommend you do not edit this field. Network Directories and BroadSoft Buddies lists are hidden for IP phones in neutral firmware, which are designed for the BroadWorks environment.
<b>priority</b>	1 to 10. 1 is the highest priority, 10 is the lowest.	The priority of the search results.

Element	Attribute	Description
<b>enable</b>	0/1, <b>0:</b> Disabled <b>1:</b> Enabled	Enable or disable the IP phone to search the desired directory list.
<b>dev</b>	T19 T21 T23 T40 T27 T27G T29 T41 T42 T42S T41S T46 T46S T48 T48S T54S T52; T29 T46 T46S T48 T48S T52 T54S.	The applicable phone models of the directory list. <b>Note:</b> Do not edit this field.

### Customizing a super search template:

1. Open the template file using an ASCII editor.
2. For each directory list that you want to configure, edit the corresponding string in the file.  
For example, configure the local directory list, edit the values within double quotes in the following strings:

```
<item id_name="local_directory_search" display_name="Local Contacts" priority="1"
enable="1"/>
```

```
super_search.xml x
1 <?xml version="1.0"?>
2 <root_super_search>
3   <item id_name="local_directory_search" display_name="Local Contacts" priority="1" enable="1" />
4   <item id_name="calllog_search" display_name="History" priority="2" enable="1" />
5   <item id_name="remote_directory_search" display_name="Remote Phonebook" priority="3" enable="0" />
6   <item id_name="ldap_search" display_name="LDAP" priority="4" enable="0" dev="T19 T21 T23 T40 T40G T27 T29 T41 T42 T42S T41S T46 T46S T48 T48S T54S T52" />
7   <item id_name="BroadSoft_UC_search" display_name="BroadSoft Buddies" priority="6" enable="0" dev="T29 T46 T46S T48 T48S T52 T54S" />
8   <item id_name="plcm_directory_search" display_name="PhoneBook" priority="7" enable="0" />
9   <item id_name="genband_directory_search" display_name="Personal Address Book" priority="8" enable="1" />
10  <item id_name="MetaSwitch_directory_search" display_name="Network Contacts" priority="9" enable="0" />
11  <item id_name="MetaSwitch_calllog_search" display_name="Network Call List" priority="10" enable="0" />
12 </root_super_search>
```

3. Save the change and place this file to the provisioning server (for example, 192.168.1.20).
4. Specify the access URL of the custom super search template file in the configuration files (for example, super\_search.url = http://192.168.1.20/super\_search.xml).

### Procedure





Search source list in dialing can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify the access URL of the super search template file. <b>Parameter:</b> super_search.url
<b>Web User Interface</b>		Configure the search source list in dialing. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p= =contacts-favorite&q=load

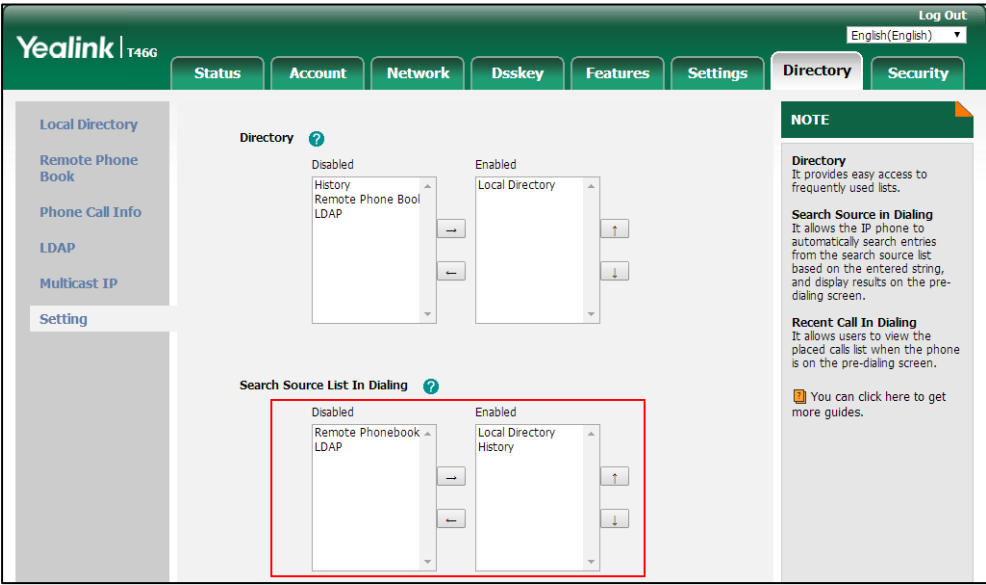
**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>super_search.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of the super search template file.</p> <p><b>Example:</b> super_search.url = http://192.168.1.20/super_search.xml</p> <p>During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.20", and downloads the super search template file "super_search.xml".</p> <p><b>Web User Interface:</b> Directory-&gt;Setting-&gt;Search Source List In Dialing</p> <p><b>Phone User Interface:</b> None</p>		

**To configure search source list in dialing via web user interface:**

1. Click on **Directory->Setting**.
2. In the **Search Source List In Dialing** block, select the desired list from the **Disabled** column and then click  .  
The selected list appears in the **Enabled** column.
3. Repeat the step 2 to add more lists to the **Enabled** column.
4. To remove a list from the **Enabled** column, select the desired list and then click  .
5. To adjust the display order of search results, select the desired list and then click  or  .

The LCD screen displays the search results in the adjusted order.



6. Click **Confirm** to accept the change.

## Call Log

### Save Call Log

IP phones record and maintain phone events to a call log, also known as a call list. The call log contains call information such as remote party identification, time and date of the call, and call duration. It can be used to redial previous outgoing calls, return incoming calls, and save contact information from call log lists to the contact directory.

IP phones maintain a local call log. Call log consists of four lists: Missed Calls, Placed Calls, Received Calls, and Forwarded Calls. Each call log list supports up to 100 entries. To store call information, you must enable save call log feature in advance. You can access the call history information via web user interface: **Directory->Phone Call Info**.

**Note** You can identify call types by the icons from a combined call log list (e.g., All Calls). For more information on icons, refer to [Appendix G: Reading Icons](#) on page 899.

### Procedure

Call log can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure call log feature. <b>Parameter:</b> features.save_call_history
------------------------------------------------------	---------------------	--------------------------------------------------------------------------------

<b>Web User Interface</b>	Configure call log feature. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =features-general&q=load
<b>Phone User Interface</b>	Configure call log feature.

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.save_call_history</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the IP phone to save the call log.</p> <p><b>0</b>-Disabled, the IP phone cannot log the missed calls, placed calls, received calls and forwarded calls in the call log lists.</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> To log the missed calls, the value of the parameter "account.X.missed_calllog" should be set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Features-&gt;General Information-&gt;Save Call Log</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;Others-&gt;General-&gt;History Record</p>		

**To configure call log feature via web user interface:**

1. Click on **Features->General Information**.

2. Select the desired value from the pull-down list of **Save Call Log**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected. Under 'General Information', the 'Save Call Log' option is highlighted with a red box, showing it is set to 'Enabled'. Other features like Call Waiting, Auto Redial, and DTMF settings are also visible. A 'NOTE' section on the right provides additional information about the 'Call Waiting' and 'Key As Send' features.

3. Click **Confirm** to accept the change.

#### To configure call log feature via phone user interface:

1. Press **Menu->Features->Others->General->History Record**.
2. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **History Record** field.
3. Press the **Save** soft key to accept the change.

## Backing up the Call Logs

Yealink IP phones support storing all call logs to a call log file named <MAC>-calllog.xml. You can back up this file to the server, avoiding data loss. Once the call logs update, the IP phone will automatically upload this file to the provisioning server or a specific server. If a call log file exists on the server, the file will be overridden. The IP phone will request to download the <MAC>-calllog.xml file according to its MAC address from the server during auto provisioning. The call log file is named after the MAC address of the IP phone. MAC address, a unique 12-digit serial number assigned to each phone, can be obtained from the bar code on the back of the IP phone. For example, if the MAC address of an IP phone is 00156574B150, the name of the call log file is 00156574b150-calllog.xml (lowercase).

## Procedure

Backing up the call log file can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	<p>Configure the IP phone to back up the call log.</p> <p><b>Parameters:</b></p> <p>static.auto_provision.local_calllog.backup.enable</p> <p>static.auto_provision.local_calllog.backup.path</p> <p>static.auto_provision.local_calllog.write_delay.terminated</p> <p>static.auto_provision.custom.upload_method</p>
------------------------------------------------------	---------------------	----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

### Details of the Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.auto_provision.local_calllog.backup.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to upload the &lt;MAC&gt;-calllog.xml file to the server each time the call logs update and download the &lt;MAC&gt;-calllog.xml file from the server during auto provisioning.</p> <p><b>0</b>-Disabled, the IP phone does not upload/download the call log file "&lt;MAC&gt;-calllog.xml" to the server.</p> <p><b>1</b>-Enabled, the IP phone uploads the call log file "&lt;MAC&gt;-calllog.xml" to the specific path configured by the parameter "static.auto_provision.local_calllog.backup.path" each time the call logs update; and downloads the call logs in the "&lt;MAC&gt;-calllog.xml" according to its MAC address from the specific path during auto provisioning.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.local_calllog.backup.path</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures a path or URL for the IP phone to upload/download the &lt;MAC&gt;-calllog.xml file. If it is left blank, the IP phone connects to the provisioning server URL, and</p>		

Parameters	Permitted Values	Default
<p>uploads/downloads the contact file "&lt;MAC&gt;-calllog.xml".</p> <p><b>Example:</b></p> <p>static.auto_provision.local_calllog.backup.path = http://192.168.1.20/calllog</p> <p>Once the call logs update, the IP phone will upload the call log file to the specified path "http://192.168.1.20/calllog".</p> <p>During the auto provisioning process, the IP phone downloads the call log file "&lt;MAC&gt;-calllog.xml" from the specified path "http://192.168.1.20/calllog".</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.local_calllog.backup.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.local_calllog.write_delay.terminated</b>	<b>Integer from 10 to 600</b>	<b>60</b>
<p><b>Description:</b></p> <p>Configures the delay time (in seconds) for the IP phone to upload the &lt;MAC&gt;-calllog.xml file each time the call logs update.</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.local_calllog.backup.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.custom.upload_method</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the way the IP phone uploads the &lt;MAC&gt;-calllog.xml file to the provisioning server (for HTTP/HTTPS server only).</p> <p><b>0</b>-PUT</p> <p><b>1</b>-POST</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.local_calllog.backup.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p>		



Parameters	Permitted Values	Default
<b>Phone User Interface:</b>		
None		

## Call List Show Number

Call list show number allows the IP phone to show the phone number instead of the name in the call log list. To use this feature, make sure the save call log feature is enabled. For more information on save call log, refer to [Save Call Log](#) on page 306.

### Procedure

Call list show number can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure call list show number. <b>Parameter:</b> features.call_log_show_num
<b>Web User Interface</b>		Configure call list show number. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =features-general&q=load

### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.call_log_show_num	0 or 1	0
<b>Description:</b> Enables or disables the IP phone to show the other party's phone number instead of the name in the call log lists. <b>0</b> -Disabled, the IP phone will show the other party's name in the call log lists. <b>1</b> -Enabled, the IP phone will show the other party's phone number in the call log lists. <b>Note:</b> It works only if the value of the parameter "features.save_call_history" is set to 1 (Enabled). <b>Web User Interface:</b> Features->General Information->Call List Show Number <b>Phone User Interface:</b> None		

**To configure call list show number via web user interface:**

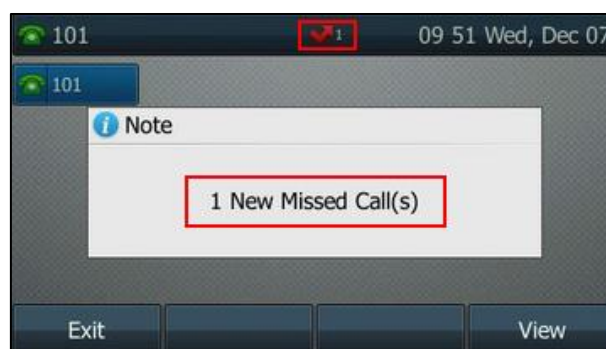
1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Call List Show Number**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'Call List Show Number' option is highlighted with a red box and is set to 'Enabled'. Other options include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Call Waiting Off Code' (empty), 'Auto Redial' (Disabled), 'Auto Redial Interval (1~300s)' (10), 'Auto Redial Times (1~300)' (10), 'Accept SIP Trust Server Only' (Enabled), 'Allow IP Call' (Enabled), 'IP Direct Auto Answer' (Enabled), 'Voice Mail Tone' (Enabled), and 'Auto Linekeys' (Disabled). A 'NOTE' sidebar on the right provides details for various features: 'Call Waiting' (allows IP phones to receive a new incoming call when there is already an active call), 'Auto Redial' (allows IP phones to automatically redial a busy number after the first attempt), 'Key As Send' (Assigns '#' or '\*' as the send key), 'Hotline' (IP phone will automatically dial out the hotline number when lifting the handset, pressing the speakerphone key or the line key), and 'Call Completion' (allows users to monitor the busy party and establish a call when the busy party becomes available to receive a call). There is also a link to get more guides.

3. Click **Confirm** to accept the change.

## Missed Call Log

Missed call log allows the IP phone to display the prompt message and an indicator icon on the idle screen, and to log missed calls in the Missed Calls list when the IP phone misses calls. It is configurable on a per-line basis. Once the user accesses the Missed Calls list, the prompt message and indicator icon on the idle screen disappear.



You can configure whether to display a prompt message when missing calls. For more information, refer to [Notification Popups](#) on page 138.

## Procedure

Missed call log can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure missed call log feature for the SIP account. <b>Parameter:</b> account.X.missed_calllog
<b>Web User Interface</b>		Configure missed call log feature. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-basic&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-basic&amp;q=load&amp;acc=0</a>

## Details of the Configuration Parameters:

Parameters	Permitted Values	Default
<b>account.X.missed_calllog<sup>[1]</sup></b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to indicate and record missed calls for SIP account X.</p> <p><b>0</b>-Disabled, the IP phone does not display a prompt message and an indicator icon on the idle screen and log the missed call in the Missed Calls list when it misses calls.</p> <p><b>1</b>-Enabled, the IP phone displays a prompt message and an indicator icon on the idle screen and logs the missed call in the Missed Calls list when it misses calls.</p> <p><b>Note:</b> It works only if the value of the parameter "features.save_call_history" is set to 1 (Enabled). The prompt message displays only if the value of the parameter "features.missed_call_popup.enable" is set to 1 (Enabled). For SIP-T29G/T46G/T46S/T48G/T48S/T52S/T54S IP phones, the indicator icon has a counter indicating the number of missed calls.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Basic-&gt;Missed Call Log</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

### To configure missed call log via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.

3. Select the desired value from the pull-down list of **Missed Call Log**.

The screenshot shows the Yealink T466 web interface. The 'Account' tab is selected. The 'Missed Call Log' option is highlighted with a red box, showing it is set to 'Enabled'. Other settings include Proxy Require, Local Anonymous, Local Anonymous Rejection, Send Anonymous Code, On Code, Off Code, Send Anonymous Rejection Code, Auto Answer, and Ring Type. A 'NOTE' section on the right explains 'Anonymous Call' and 'Anonymous Call Rejection'.

4. Click **Confirm** to accept the change.

## Local Directory

IP phones maintain a local directory. The local directory can store up to 1000 contacts and 48 groups. When adding a contact to the local directory, in addition to name and phone numbers, you can also specify the account, ring tone and group for the contact. Contacts and groups can be added either one by one or in batch using a local contact file. Yealink IP phones support both \*.xml and \*.csv format contact files, but you can only customize the \*.xml format contact file.

Contacts can be configured to have a backup on the provisioning server or a specific server. The contact file format is \*.xml.

For SIP-T29G/T46G/T46S/T48G/T48S/T52S/T54S IP phones, you can customize the contact avatar. You can also customize a contact icon for SIP-T48G/T48S IP phones.

The contact avatar format must meet the following:

Phone Model	Format	Resolution	Single File Size
SIP-T48S/T46S	.jpg/.png/.bmp/.jpeg	<=110*110	<=5MB
SIP-T54S/T52S			
SIP-T48G/T46G/T29G		<=110*110	<=5MB

The contact icon format must meet the following:

Phone Model	Format	Resolution
SIP-T48G/S	.jpg/.png/.bmp/.jpeg	<=41*41

## Customizing a Local Contact File

You can add contacts one by one on the IP phone directly. You can also add multiple contacts at a time and/or share contacts between IP phones using the local contact template file. After setup, place the template file to the provisioning server and specify the access URL of the template file in the configuration file. The existing local contacts on the IP phones will be overridden by the downloaded local contacts.

You can ask the distributor or Yealink FAE for local contact template. You can also obtain the local contact template online:

<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>. For more information on obtaining the local contact file, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

The following table lists meaning of each variable in the local contact template file:

Element	Values	Description
<b>root_group</b>	no	Group list's root element.
<b>group</b>	no	Group's root element.
<b>display_name</b>	All Contacts Blacklist	An element of group. Group name.
<b>ring</b>	Format of the value: System ring tone: <b>Auto</b> <b>Silent.wav</b> <b>Splash.wav</b> <b>RingN.wav</b> (integer N ranges from 1 to 8) Custom ring tone: <b>Name.wav</b>	An element of group. Group ring tone.
<b>root_contact</b>	no	Contact list's root element.
<b>contact</b>	no	Contact's root element.
<b>display_name</b>	String	An element of contact. Contact name. <b>Note:</b> This value cannot be left blank or duplicated.
<b>office_number</b>	String	Office number of the contact.
<b>mobile_number</b>	String	Mobile number of the contact.
<b>other_number</b>	String	Other number of the contact.
<b>line</b>	-1~15; Multiple line IDs are separated by commas. For more information on the valid values of	The desired line you want to add the contact to. <b>Note:</b> This is not applicable

Element	Values	Description
	this element for each phone model, refer to the following table.	to SIP-T19(P) E2 IP phones.
<b>ring</b>	Format of the value: System ring tone: <b>Auto</b> <b>Silent.wav</b> <b>Splash.wav</b> <b>RingN.wav</b> (integer N ranges from 1 to 8) Custom ring tone: <b>Name.wav</b>	An element of contact. Contact ring tone.
<b>group_id_name</b>	Valid Value: built-in: All Contacts, Blacklist custom: XXX (for example, Friend)	Group name of a contact.
<b>default_photo</b>	Format of the value: <b>Resource:avatar name</b> (the built-in avatar) <b>Config:avatar name</b> (the custom avatar)	Contact avatar. <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.

The following table lists valid values of line for each phone model.

Phone Model	Values	Description
SIP-T54S/T48G/T48S/T46G/T46S/T29G	-1~15	-1 stands for Auto (the first registered line) 0~15 stand for line1~line16
SIP-T52S/T42G/T42S	-1~11	-1 stands for Auto (the first registered line) 0~11 stand for line1~line12
SIP-T41P/T41S/T27P/T27G	-1~5	-1 stands for Auto (the first registered line) 0~5 stand for line1~line6
SIP-T40P/T40G/T23P/T23G	-1~2	-1 stands for Auto (the first registered line) 0~2 stand for line1~line3
SIP-T21(P) E2	-1~1	-1 stands for Auto (the first registered line) 0~1 stand for line1~line2

## Configuration Parameters

The following table lists the configuration parameters used when customizing a local contact file:

Parameters	Permitted Values	Default
<b>local_contact.data.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of the local contact file (*.xml).</p> <p><b>Example:</b> local_contact.data.url = http://192.168.10.25/contact.xml</p> <p><b>Note:</b> If the value of the parameter "static.auto_provision.local_contact.backup.enable" is set to 1 (Enabled), the contacts in the local contact file "contact.xml" downloaded from the provisioning server will not take effect.</p> <p><b>Web User Interface:</b> Directory-&gt;Local Directory-&gt;Import Local Directory File</p> <p><b>Phone User Interface:</b> None</p>		
<b>local_contact.photo.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of a contact avatar file.</p> <p>The format of the contact avatar must be *.png, *.jpg, *.bmp or *.jpeg.</p> <p>The contact avatar file should be uploaded to the provisioning server in advance.</p> <p><b>Example:</b> local_contact.photo.url = tftp://192.168.10.25/Photo.jpg</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>local_contact.icon_image.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of a contact icon file.</p> <p>The format of the contact icon must be *.png, *.jpg, *.bmp or *.jpeg.</p>		

Parameters	Permitted Values	Default
<p>The contact icon file should be uploaded to the provisioning server in advance.</p> <p><b>Example:</b></p> <p>local_contact.icon_image.url = tftp://192.168.10.25/Photo.jpg</p> <p><b>Note:</b> It is only applicable to SIP-T48G/S IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>local_contact.image.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the access URL of a TAR contact avatar file.</p> <p>The format of the contact avatar must be *.png, *.jpg, *.bmp or *.jpeg.</p> <p>The contact avatar file should be compressed as a TAR file in advance and then place it to the provisioning server.</p> <p><b>Example:</b></p> <p>local_contact.image.url = tftp://192.168.10.25/photo.tar</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>local_contact.data_photo_tar.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the access URL of the compressed TAR file consisting of the avatars TAR file and contact XML file.</p> <p>All avatars needed for contacts should be compressed as a TAR file in advance.</p> <p><b>Example:</b></p> <p>local_contact.data_photo_tar.url = tftp://192.168.10.25/Contact.tar</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones. The name of the avatars TAR file must be photo.tar (case-sensitive), and the name of the contact XML file must be ContactData.xml (case-sensitive).</p> <p><b>Web User Interface:</b></p> <p>None</p>		



Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>local_contact.icon.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the access URL of a TAR contact icon file. The format of the contact icon must be *.png, *.jpg, *.bmp or *.jpeg. The contact icon file should be compressed as a TAR file in advance and then place it to the provisioning server. <b>Example:</b> local_contact.icon.url = tftp://192.168.10.25/photo2.tar <b>Note:</b> It is only applicable to SIP-T48G/S IP phones. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

**To add a group to the local directory via web user interface:**

1. Click on **Directory->Local Directory**.
2. In the **Group Setting** block, enter the desired group name in the **Group** field.

3. Select the desired ring tone from the pull-down list of **Ring**.

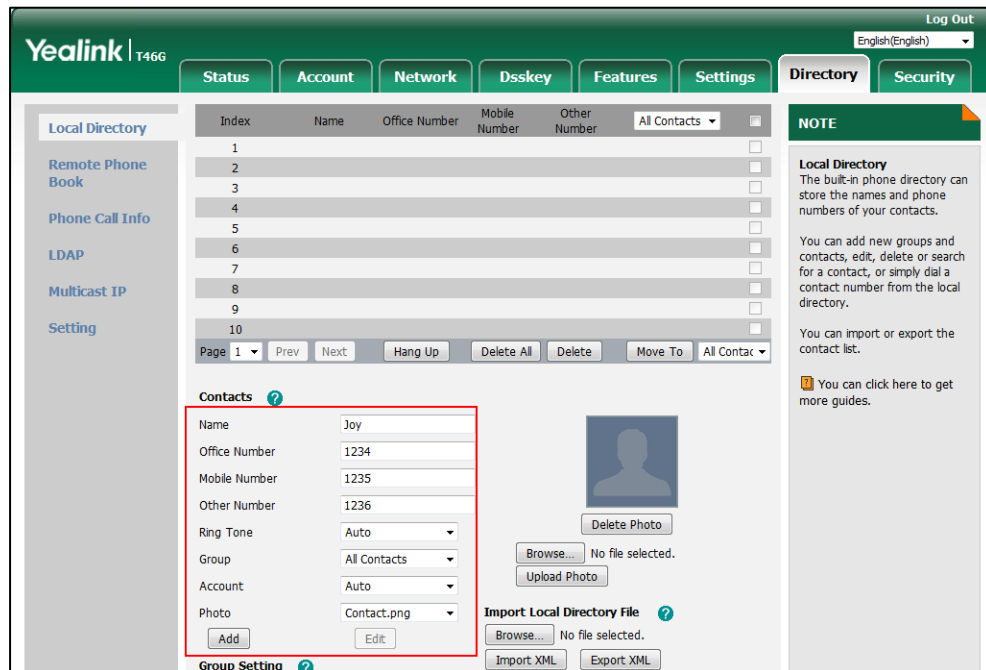
The screenshot shows the Yealink T466 web interface. The top navigation bar includes tabs for Status, Account, Network, Dsskey, Features, Settings, Directory, and Security. The left sidebar lists options: Local Directory, Remote Phone Book, Phone Call Info, LDAP, Multicast IP, and Setting. The main content area is divided into a table for 'All Contacts' and a 'Contacts' form. The 'Contacts' form includes fields for Name, Office Number, Mobile Number, Other Number, Ring Tone (set to Auto), Group (set to All Contacts), Account (set to Auto), and Photo (set to Contact.png). Below the form is the 'Group Setting' section, which contains a 'Group' dropdown (set to 'test') and a 'Ring' dropdown (set to 'Auto'). The 'Ring' dropdown is highlighted with a red box. To the right of the form is an 'Import Local Directory File' section with buttons for Import XML, Export XML, Import CSV, and Export CSV. A 'NOTE' box on the right explains the Local Directory functionality.

4. Click **Add** to add the group.

#### To add a contact to the local directory via web user interface:

1. Click on **Directory->Local Directory**.
2. In the **Contacts** block, enter the name and the office, mobile or other numbers in the corresponding fields.
3. Select the desired ring tone from the pull-down list of **Ring Tone**.
4. Select the desired group from the pull-down list of **Group**.
5. Select the desired account from the pull-down list of **Account**.

If **Auto** is selected, the IP phone will use the default account when placing calls to the contact from the local directory.



6. Click **Add** to add the contact.

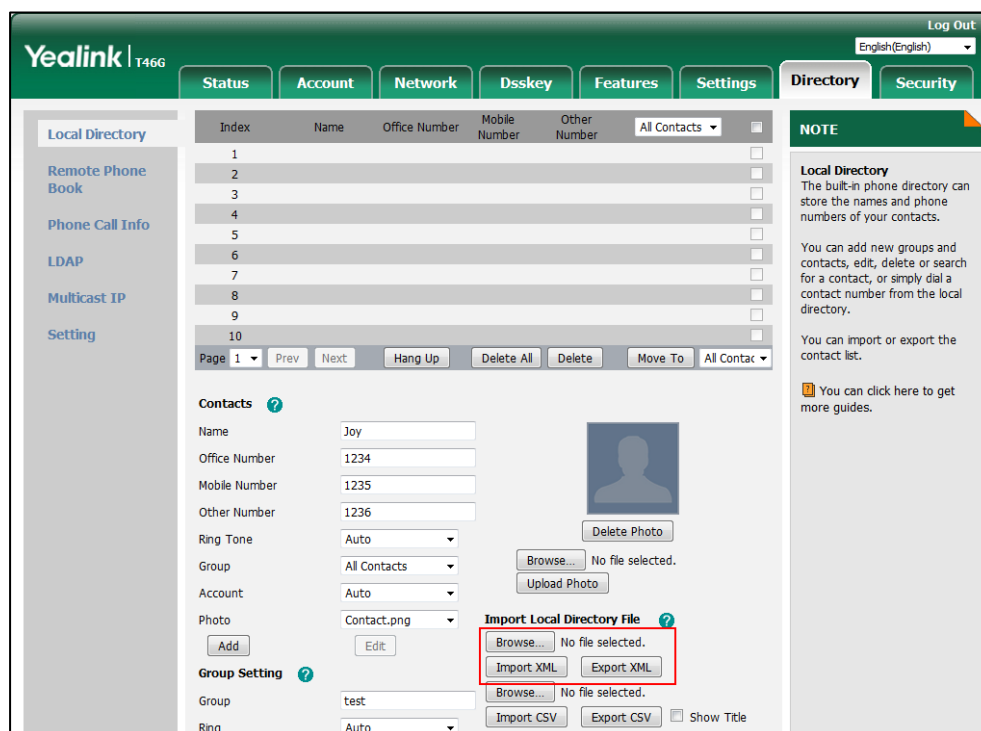
#### To add a group to the local directory via phone user interface:

1. Press **Menu**->**Directory**->**Local Directory**.
2. Press the **AddGroup** soft key.
3. Enter the desired group name in the **Name** field.
4. Press **◀** or **▶** , or the **Switch** soft key to select the desired group ring tone from the **Ring** field.
5. Press the **Add** soft key to accept the change.

#### To import an XML contact list file via web user interface:

1. Click on **Directory**->**Local Directory**.

- Click **Browse** to locate a contact list file (the file format must be \*.xml) from your local system.

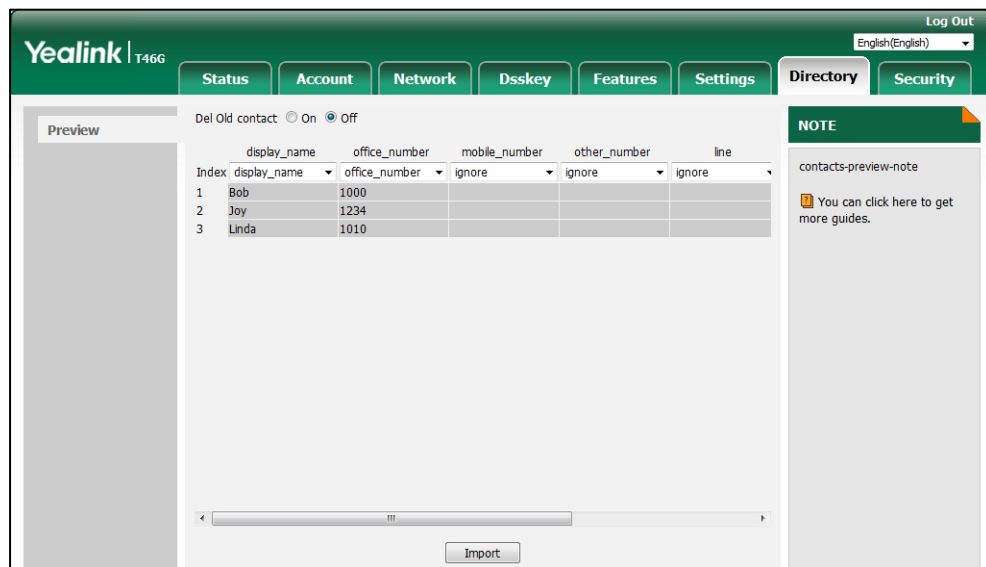


- Click **Import XML** to import the contact list.  
The web user interface prompts "The original contact will be covered, continue?".
- Click **OK** to complete importing the contact list.

#### To import a CSV contact list file via web user interface:

- Click on **Directory->Local Directory**.
- Click **Browse** to locate a contact list file (the file format must be \*.csv) from your local system.
- (Optional.) Check the **Show Title** checkbox.  
It will prevent importing the title of the contact information which is located in the first line of the CSV file.
- Click **Import CSV** to import the contact list.
- (Optional.) Mark the **On** radio box in the **Delete Old Contacts** field.  
It will delete all existing contacts while importing the contact list.
- Select the contact information you want to import into the local directory from the pull-down list of **Index**.

At least one item should be selected to be imported into the local directory.



7. Click **Import** to complete importing the contact list.

#### To export a contact list via web user interface:

1. Click on **Directory->Local Directory**.
2. Click **Export XML** (or **Export CSV**).
3. Click **Save** to save the contact list to your local system.

#### To add a contact to the local directory via phone user interface:

1. Press **Menu->Directory->Local Directory**.
2. Select the desired contact group and then press the **Enter** soft key.
3. Press the **Add** soft key.
4. Enter the name and the office, mobile or other numbers in the corresponding fields.
5. Press **◀** or **▶**, or the **Switch** soft key to select the desired account from the **Account** field.

If **Auto** is selected, the IP phone will use the default account when placing calls to the contact from the local directory.

6. Press **◀** or **▶**, or the **Switch** soft key to select the desired ring tone from the **Ring** field.
7. Press the **Save** soft key to accept the change.

## Customizing a Local Contact File (Black-and-white Screen Phones)

The following shows the procedure of customizing a local contact file for SIP-T42G/T42S/T41P/T41S/T40P/T40G/T27P/T27G/T23P/T23G/T21(P) E2/T19(P) E2 IP phones:

#### To customize a local contact file:

1. Open the template file using an ASCII editor.

- For each group that you want to add, add the following string to the file. Each starts on a separate line:

```
<group display_name="" ring=""/>
```

- For each contact that you want to add, add the following string to the file. Each starts on a separate line:

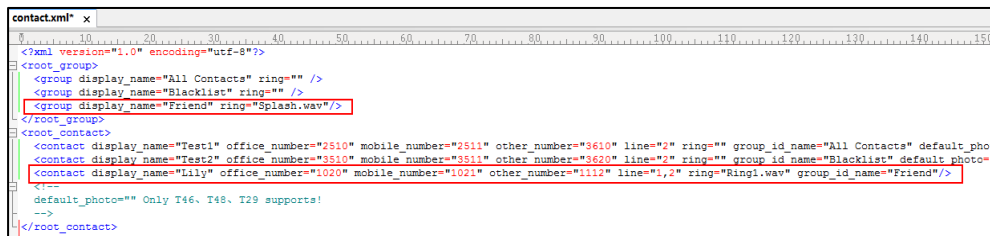
```
<contact display_name="" office_number="" mobile_number="" other_number="" line=""
ring="" group_id_name=""/>
```

- Specify the values within double quotes.

For example:

```
<group display_name="Friend" ring="Splash.wav"/>
```

```
<contact display_name="Lily" office_number="1020" mobile_number="1021"
other_number="1112" line="1,2" ring="Ring1.wav" group_id_name="Friend"/>
```



- Save the change and place this file to the provisioning server.
- Specify the access URL of the custom local contact template in the configuration file.

For example:

```
local_contact.data.url = tftp://192.168.10.25/contact.xml
```

During the auto provisioning process, the IP phone connects to the provisioning server "192.168.10.25", and downloads the local contact file "contact.xml".

#### Note

If you have configured to back up the local contacts to the server, the IP phone will download the contact file "<MAC>-contact.xml" from the server backup path and display the downloaded contacts in the local directory list. The contacts in the local contact file "contact.xml" downloaded from the provisioning server don't take effect. For more information on how to back up the local contacts, refer to [Backing up the Local Contacts](#) on page 332.

## Customizing a Local Contact File (Color Screen Phones)

The following shows the procedure of customizing a local contact file for SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones:

### Scenario A - Using the Built-in Avatar for Contact

This scenario is applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.

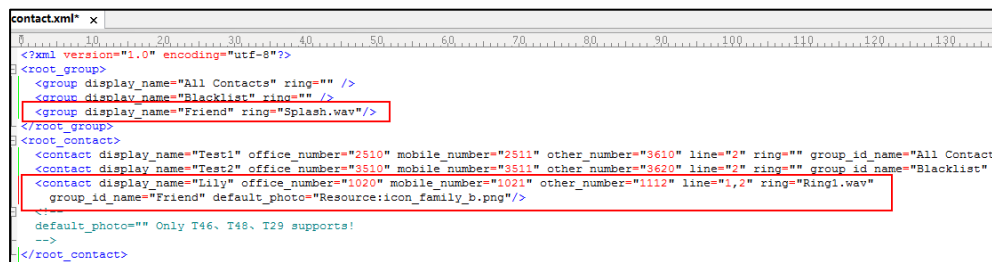
**To customize a local contact file:**

1. Open the template file using an ASCII editor.
2. For each group that you want to add, add the following string to the file. Each starts on a separate line:  
`<group display_name="" ring=""/>`
3. For each contact that you want to add, add the following string to the file. Each starts on a separate line:  
`<contact display_name="" office_number="" mobile_number="" other_number="" line="" ring="" group_id_name="" default_photo=""/>`
4. Specify the values within double quotes.

For example:

```
<group display_name="Friend" ring="Splash.wav"/>

<contact display_name="Lily" office_number="1020" mobile_number="1021"
other_number="1112" line="1,2" ring="Ring1.wav" group_id_name="Friend"
default_photo="Resource:icon_family_b.png"/>
```



5. Save the change and place this file to the provisioning server.
6. Specify the access URL of the custom local contact template in the configuration file.

For example:

```
local_contact.data.url = tftp://192.168.10.25/contact.xml
```

During the auto provisioning process, the IP phone connects to the provisioning server "192.168.10.25", and downloads the local contact file "contact.xml".

**Note**

If you have configured to back up the local contacts to the server, the IP phone will download the contact file "<MAC>-contact.xml" from the backup path from the server and display the downloaded contacts in the local directory list. The contacts in the local contact file "contact.xml" downloaded from the provisioning server don't take effect. For more information on how to back up the local contacts, refer to [Backing up the Local Contacts](#) on page 332.

**Scenario B - Using the Custom Avatar for Contact**

This scenario is applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.

To specify custom avatars for contacts, you need to upload the custom avatars to the

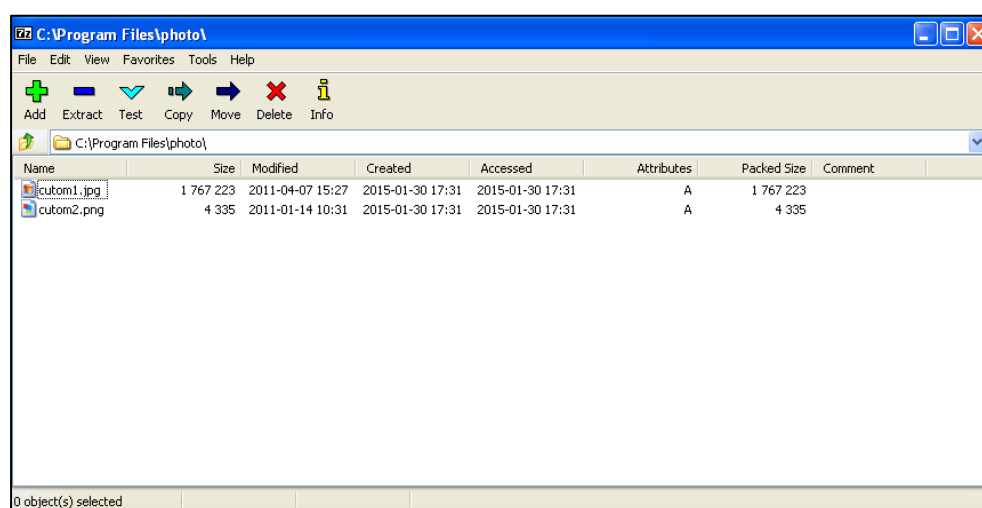
provisioning server in advance. In addition, you can also compress all the avatars as a tar formatted file, and then upload the tar formatted file to the provisioning server.

## Preparing the Tar Formatted File

You can package the tar formatted file using the tool 7-Zip or GnuWin32. You can download 7-Zip online: <http://www.7-zip.org/> and GnuWin32 online: <http://gnuwin32.sourceforge.net/packages/gtar.htm>. This section provides you on how to package the tar file using 7-Zip.

### To package a tar formatted file using the tool 7-Zip on the Windows platform:

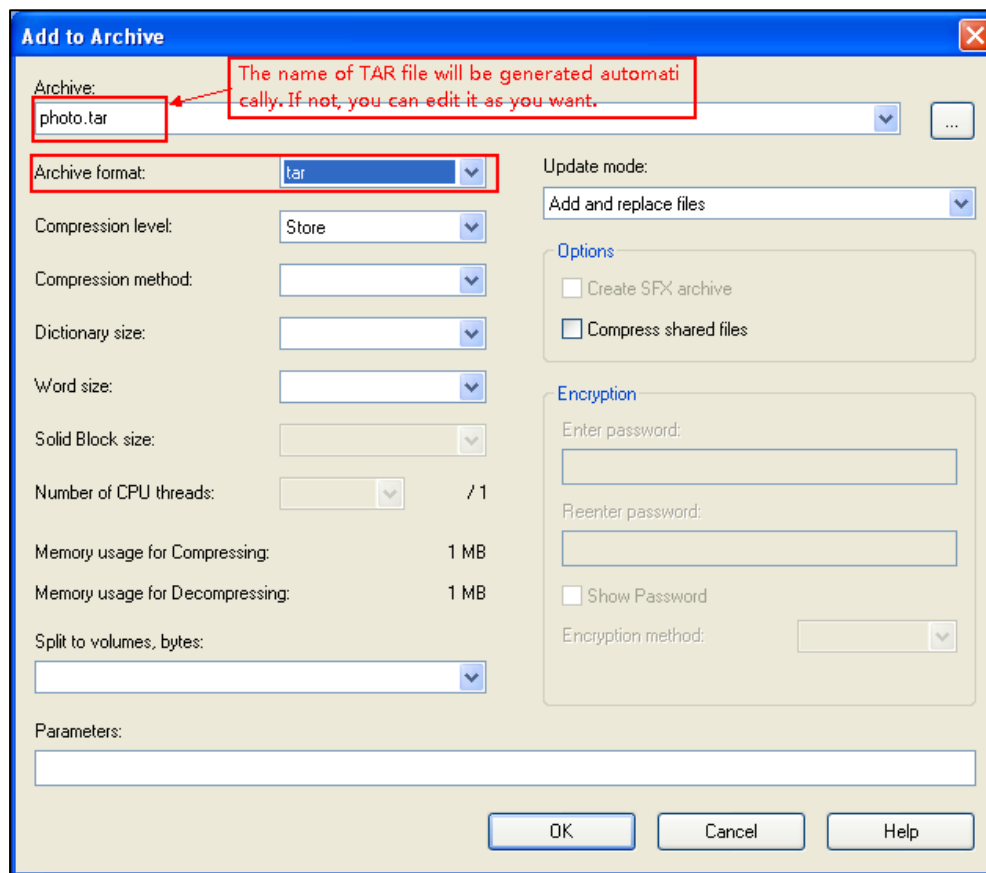
1. Download and install 7-Zip on the local system.
2. Create a folder (for example, photo) on the local system (for example, C:\Program Files) and place the file that will be compressed (for example, cutom1.jpg, cutom2.png) to this folder.
3. Start the 7-Zip file manager application (7zFM.exe).
4. Locate the photo folder from the local system (C:\Program Files\photo\).



5. Select the desired photos that will be compressed.
6. Click the **Add** button.



7. Select **tar** from the pull-down list of **Archive format**.



8. Click the **OK** button.  
A photo.tar file is generated in the directory C:\Program Files\photo.
9. Place this file to the provisioning server (for example, 192.168.10.25).

## Customizing a Local Contact File

### To customize a local contact file:

1. Open the template file using an ASCII editor.
2. For each group that you want to add, add the following string to the file. Each starts on a separate line:  
`<group display_name="" ring=""/>`
3. For each contact that you want to add, add the following string to the file. Each starts on a separate line:  
`<contact display_name="" office_number="" mobile_number="" other_number="" line="" ring="" group_id_name="" default_photo=""/>`
4. Specify the values within double quotes.

For example:

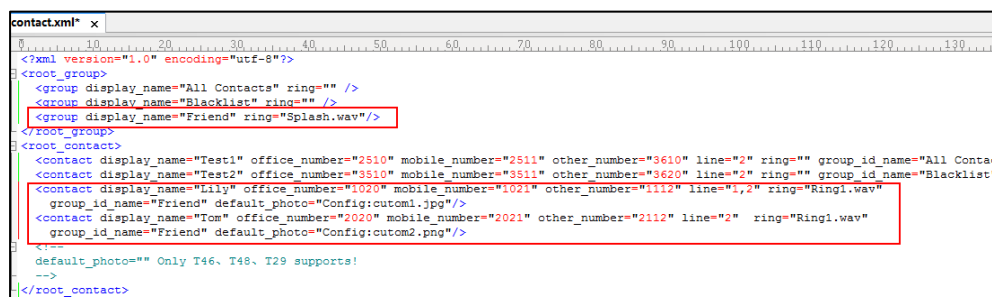
```
<group display_name="Friend" ring="Splash.wav"/>
```

```
<contact display_name="Lily" office_number="1020" mobile_number="1021"
```

```

other_number="1112" line="1,2" ring="Ring1.wav" group_id_name="Friend"
default_photo="Config:cutom1.jpg"/>
<contact display_name="Tom" office_number="2020" mobile_number="2021"
other_number="2112" line="2" ring="Ring1.wav" group_id_name="Friend"
default_photo="Config:cutom2.png"/>

```



5. Save the change and place this file to the provisioning server.
6. Specify the access URL of the custom local contact template in the configuration file.

There are three methods to specify custom avatar for contacts:

#### Method 1:

local\_contact.data.url = tftp://192.168.10.25/contact.xml

local\_contact.photo.url = tftp://192.168.10.25/cutom1.jpg

local\_contact.photo.url = tftp://192.168.10.25/cutom2.png

During the auto provisioning process, the IP phone connects to the provisioning server "192.168.10.25", and downloads the local contact file "contact.xml" and avatar pictures ("cutom1.jpg" and "cutom2.png").

#### Method 2:

local\_contact.data.url = tftp://192.168.10.25/contact.xml

local\_contact.image.url = tftp://192.168.10.25/photo.tar

For more information on generating a contact avatar file "photo.tar", refer to [Preparing the Tar Formatted File](#) on page 326.

During the auto provisioning process, the IP phone connects to the provisioning server "192.168.10.25", and downloads the local contact file "contact.xml" and avatar file "photo.tar".

#### Method 3:

If the local contact file (ContactData.xml) and custom avatars (photo.tar) are compressed as a tar formatted file (for example, Contact.tar), you can only configure the following parameter to upload contacts and avatars:

local\_contact.data\_photo\_tar.url = tftp://192.168.10.25/Contact.tar

For more information on generating "photo.tar" and "Contact.tar", refer to [Preparing the Tar Formatted File](#) on page 326.

During the auto provisioning process, the IP phone connects to the provisioning server "192.168.10.25", and downloads the file "Contact.tar".

#### Note

Note that if you are using method 3 to specify custom avatar for contacts, the name of the avatars TAR file must be photo.tar (case-sensitive), and the name of the contact XML file must be ContactData.xml (case-sensitive).

If you have configured to back up the local contacts to the server, the IP phone will download the contact file "<MAC>-contact.xml" from the backup path from the server and display the downloaded contacts in the local directory list. The contacts in the local contact file "contact.xml" downloaded from the provisioning server don't take effect. But the avatars downloaded take effect. For more information on how to back up the local contacts, refer to [Backing up the Local Contacts](#) on page 332.

## Scenario C - Using the Custom Avatar and Custom Icon for Contact

This scenario is only applicable to SIP-T48G/S IP phones.

#### Scenario Conditions I:

- Provisioning server URL: tftp://192.168.10.25.
- The custom avatars and icons: "cutom1.jpg" and "cutom2.png". They are all uploaded to the provisioning server in advance.

#### Scenario Conditions II:

- Provisioning server URL: tftp://192.168.10.25.
- The custom avatars: "cutom1.jpg" and "cutom2.png". They are compressed as a tar formatted file (photo1.tar).
- The custom icons: "cutom1.jpg" and "cutom2.png". They are compressed as a tar formatted file (photo2.tar).

For more information on generating a tar formatted file, refer to [Preparing the Tar Formatted File](#) on page 326.

#### Note

The custom avatar and icon can be different, but make sure the icon name is the same as avatar name.

#### To customize a local contact file:

1. Open the template file using an ASCII editor.
2. For each group that you want to add, add the following string to the file. Each starts on a separate line:  

```
<group display_name="" ring=""/>
```
3. For each contact that you want to add, add the following string to the file. Each starts on a separate line:

```
<contact display_name="" office_number="" mobile_number="" other_number="" line=""
ring="" group_id_name="" default_photo=""/>
```

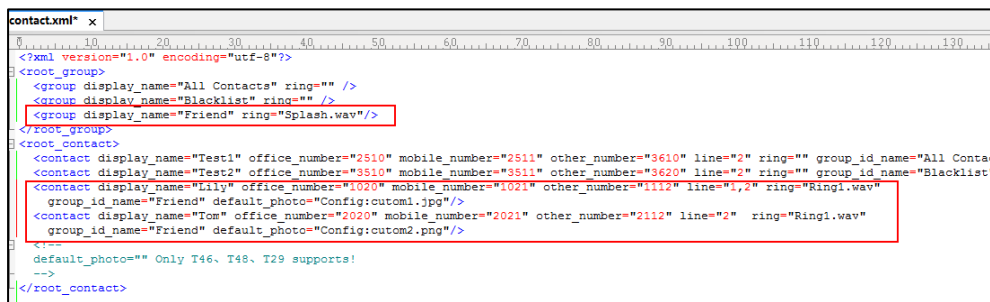
4. Specify the values within double quotes.

For example:

```
<group display_name="Friend" ring="Splash.wav"/>

<contact display_name="Lily" office_number="1020" mobile_number="1021"
other_number="1112" line="1,2" ring="Ring1.wav" group_id_name="Friend"
default_photo="Config:cutom1.jpg"/>

<contact display_name="Tom" office_number="2020" mobile_number="2021"
other_number="2112" line="2" ring="Ring1.wav" group_id_name="Friend"
default_photo="Config:cutom2.png"/>
```



5. Save the change and place this file to the provisioning server.
6. Specify the access URL of the custom local contact template file in the configuration file.

There are two methods to specify custom avatar and icon for contacts:

**Method 1:**

```
local_contact.photo.url = tftp://192.168.10.25/cutom1.jpg
local_contact.photo.url = tftp://192.168.10.25/cutom2.png
local_contact.data.url = tftp://192.168.10.25/contact.xml
local_contact.icon_image.url = tftp://192.168.10.25/cutom1.jpg
local_contact.icon_image.url = tftp://192.168.10.25/cutom2.png
```

During the auto provisioning process, the IP phone connects to the provisioning server "192.168.10.25", and downloads the avatars (cutom1.jpg and cutom2.png), icons (cutom1.jpg and cutom2.png) and the contact file "contact.xml".

**Method 2:**

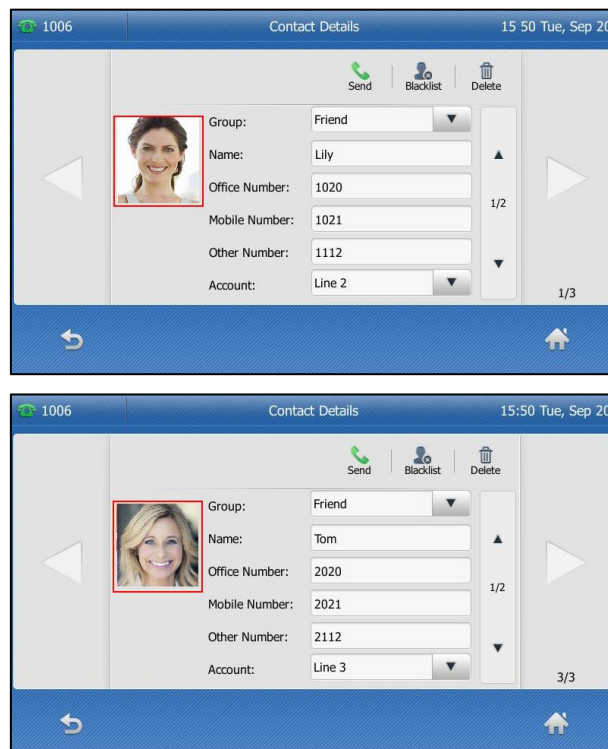
```
local_contact.image.url = tftp://192.168.10.25/photo1.tar
local_contact.data.url = tftp://192.168.10.25/contact.xml
local_contact.icon.url = tftp://192.168.10.25/photo2.tar
```

During the auto provisioning process, the IP phone connects to the provisioning server "192.168.10.25", and downloads the avatar file "photo1.tar", icon file "photo2.tar" and the local contact file "contact.xml".

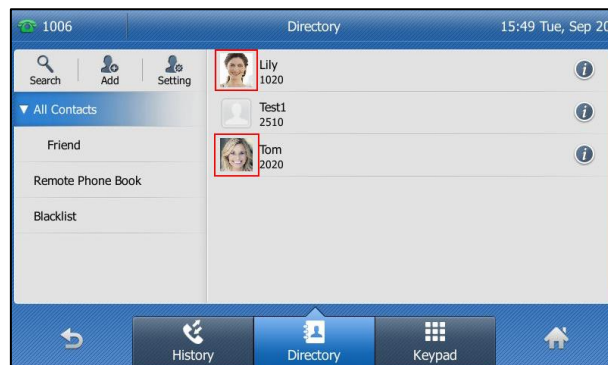
### Note

If you have configured to back up the local contacts to the server, the IP phone will download the contact file "<MAC>-contact.xml" from the backup path from the server and display the downloaded contacts in the local directory list. The contacts in the local contact file "contact.xml" downloaded from the provisioning server don't take effect. But the avatars and icons downloaded take effect. For more information on how to back up the local contacts, refer to [Backing up the Local Contacts](#) on page 332.

The following shows the custom avatars downloaded from the provisioning server:



The following shows the custom icons downloaded from the provisioning server:



## Backing up the Local Contacts

Yealink IP phones support storing all local contacts to a contact file named <MAC>-contact.xml. You can back up this file to the server, avoiding data loss. Once the contacts update, the IP phone will automatically upload this file to the provisioning server or a specific server. If a contact file exists on the server, the file will be overridden. The IP phone will request to download the <MAC>-contact.xml file according to its MAC address from the server during auto provisioning.

The contact file is named after the MAC address of the IP phone. MAC address, a unique 12-digit serial number assigned to each phone, can be obtained from the bar code on the back of the IP phone. For example, if the MAC address of an IP phone is 00156574B150, the name of the contact file is 00156574b150-contact.xml (lowercase).

### Procedure

Backing up the local contact file can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the IP phone to back up the local contacts.  <b>Parameters:</b> static.auto_provision.local_contact.backup.enable static.auto_provision.local_contact.backup.path static.auto_provision.custom.upload_method
------------------------------------------------------	---------------------	---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

### Details of the Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.auto_provision.local_contact.backup.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to upload the <MAC>-contact.xml file to the server each time the contacts update and download the <MAC>-contact.xml file from the server during auto provisioning.  <b>0</b> -Disabled, the IP phone will not upload the contact file "<MAC>-contact.xml" to the server, so the IP phone downloads the contacts in the "contact.xml" from the access URL configured by the parameter "local_contact.data.url" or "local_contact.data_photo_tar.url" during auto provisioning.  <b>1</b> -Enabled, the IP phone uploads the contact file "<MAC>-contact.xml" to the specific path configured by the parameter "static.auto_provision.local_contact.backup.path" each time the		

Parameters	Permitted Values	Default
<p>contacts update; and downloads the contacts in the "&lt;MAC&gt;-contact.xml" according to its MAC address from the specific path during auto provisioning.</p> <p><b>Note:</b> It does not affect the downloading of the contact avatar/icon files.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.local_contact.backup.path</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures a path or URL for the IP phone to upload/download the &lt;MAC&gt;-contact.xml file. If it is left blank, the IP phone connects to the provisioning server URL, and uploads/downloads the contact file "&lt;MAC&gt;-contact.xml".</p> <p><b>Example:</b></p> <p>static.auto_provision.local_contact.backup.path = http://192.168.1.20/contact</p> <p>Once the contacts update, the IP phone will upload the contact file to the specified path "http://192.168.1.20/contact".</p> <p>During the auto provisioning process, the IP phone downloads the contact file "&lt;MAC&gt;-contact.xml" from the specified path "http://192.168.1.20/contact".</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.local_contact.backup.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.custom.upload_method</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the way the IP phone uploads the &lt;MAC&gt;-contact.xml file to the provisioning server (for HTTP/HTTPS server only).</p> <p><b>0</b>-PUT</p> <p><b>1</b>-POST</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.local_contact.backup.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
None		
<b>Phone User Interface:</b>		
None		

## Live Dialpad

Live dialpad allows IP phones to automatically dial out the entered phone number without pressing the send key after a designated period of time.

### Procedure

Live dialpad can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure live dialpad. <b>Parameters:</b> phone_setting.predial_autodial phone_setting.inter_digit_time
<b>Web User Interface</b>		Configure live dialpad. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =settings-preference&q=load

### Details of Configuration Parameters:

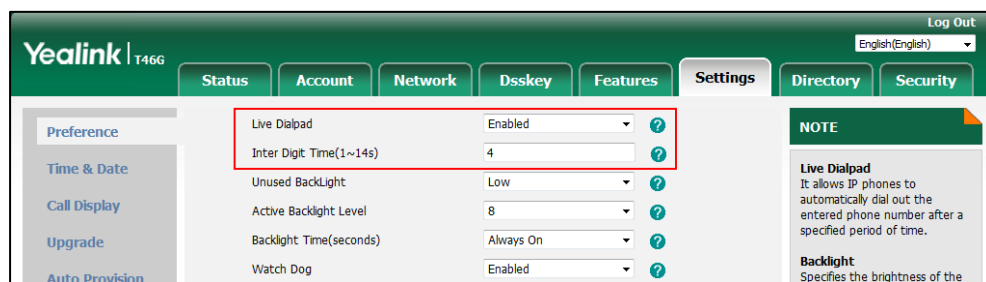
Parameters	Permitted Values	Default
<b>phone_setting.predial_autodial</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the live dialpad feature. <b>0</b> -Disabled <b>1</b> -Enabled, the IP phone will automatically dial out the entered phone number on the dialing screen without pressing a send key. <b>Web User Interface:</b> Settings->Preference->Live Dialpad <b>Phone User Interface:</b> None		
<b>phone_setting.inter_digit_time</b>	<b>Integer from 1 to 14</b>	<b>4</b>



Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the delay time (in seconds) for the IP phone to automatically dial out the entered phone number without pressing a send key.</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.predial_autodial" is set to 1 (Enabled) and the value of the parameter "dialplan.digitmap.enable" is set to 0 (Disabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Preference-&gt;Inter Digit Time(1~14s)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure live dialpad via web user interface:**

1. Click on **Settings->Preference**.
2. Select the desired value from the pull-down list of **Live Dialpad**.
3. Enter the desired delay time in the **Inter Digit Time(1~14s)** field.



4. Click **Confirm** to accept the change.

## Speed Dial

Speed dial allows users to speed up dialing the contacts on the phone's idle screen using dedicated DSS keys.

### Procedure

Speed dial DSS key can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	<p>Assign a speed dial DSS key.</p> <p><b>Parameters:</b></p> <p>linekey.X.type/  programablekey.X.type/  expansion_module.X.key.Y.type</p> <p>linekey.X.line/  programablekey.X.line/</p>
------------------------------------------------------	---------------------	--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

		expansion_module.X.key.Y.line linekey.X.value/ programablekey.X.value/ expansion_module.X.key.Y.value linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Assign a speed dial DSS key.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =dsskey&q=load&model=0
<b>Phone User Interface</b>		Assign a speed dial DSS key.

## Speed Dial DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type</b> <sup>[1]</sup> <b>programablekey.X.type</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.type</b> <sup>[3]</sup>	<b>13</b>
<b>Description:</b> Configures a DSS key as a speed dial DSS key on the IP phone.  <b>Example:</b> linekey.1.type = 13  <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.  <b>Web User Interface:</b> Dsskey->Line Key/Programable Key/Ext Key->Type  <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.line</b> <sup>[1]</sup> <b>programablekey.X.line</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.line</b> <sup>[3]</sup>	<b>Refer to the following content</b>

Parameters	Permitted Values
<p><b>Description:</b></p> <p>Configures the desired line to apply the speed dial DSS key.</p> <p><b>Permitted Values:</b></p> <p>1 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G)</p> <p>1 to 12 (for SIP-T52S/T42G/T42S)</p> <p>1 to 6 (for SIP-T41P/T41S/T27P/T27G)</p> <p>1 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>1 to 2 (for SIP-T21(P) E2)</p> <p>1-Line 1</p> <p>2-Line 2</p> <p>...</p> <p>16-Line 16</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Example:</b></p> <p>linekey.1.line = 1</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key/Programable Key/Ext Key-&gt;Line</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Account ID</p>	
<b>linekey.X.value<sup>[1]</sup></b> <b>programablekey.X.value<sup>[2]</sup></b> <b>expansion_module.X.key.Y.value<sup>[3]</sup></b>	String within 99 characters
<p><b>Description:</b></p> <p>Configures the extension you want to dial out.</p> <p><b>Example:</b></p> <p>linekey.1.value = 1008</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key/Programable Key/Ext Key-&gt;Value</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Value</p>	
<b>linekey.X.label<sup>[1]</sup></b> <b>programablekey.X.label<sup>[2]</sup></b>	String within 99 characters

Parameters	Permitted Values
<b>expansion_module.X.key.Y.label<sup>[3]</sup></b>	
<p><b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key.</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key/Programable Key/Ext Key-&gt;Label</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Label</p>	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	<b>String within 99 characters</b>
<p><b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key.</p> <p><b>Note:</b> It is only applicable to SIP-T52S IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key-&gt;Short</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Short Label</p>	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

**To configure a speed dial DSS key via web user interface:**

1. Click on **Dsskey->Line Key** (or **Programable Key**).
2. In the desired DSS key field, select **Speed Dial** from the pull-down list of **Type**.
3. Enter the phone number or extension you want to dial out in the **Value** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

5. Select the desired line from the pull-down list of **Line**.

6. Click **Confirm** to accept the change.

**To configure a speed dial DSS key via phone user interface:**

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press or , or the **Switch** soft key to select **Speed Dial** from the **Type** field.
4. Press or , or the **Switch** soft key to select the desired line from the **Account ID** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the phone number or extension you want to dial out in the **Value** field.
7. Press the **Save** soft key to accept the change.

# Call Waiting

Call waiting allows IP phones to receive a new incoming call when there is already an active call. The new incoming call is presented to the user visually on the LCD screen.

Call waiting tone allows the IP phone to play a short tone, to remind the user audibly of a new incoming call during conversation. Call waiting tone works only if call waiting is enabled. You can customize call waiting tone or select specialized tone sets (vary from country to country) for your IP phone. For more information, refer to [Tones](#) on page 724.

The call waiting on code and call waiting off code configured on IP phones are used to activate/deactivate the server-side call waiting feature. They may vary on different servers.

## Procedure

Call waiting and call waiting tone can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure call waiting and call waiting tone.
------------------------------------------------------	---------------------	-----------------------------------------------

		<b>Parameters:</b> call_waiting.enable call_waiting.tone call_waiting.on_code call_waiting.off_code
<b>Web User Interface</b>		Configure call waiting. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?parameters=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?parameters=features-general&amp;q=load</a>
		Configure call waiting tone. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?parameters=features-audio&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?parameters=features-audio&amp;q=load</a>
<b>Phone User Interface</b>		Configure call waiting and call waiting tone.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>call_waiting.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables call waiting feature. <b>0</b> -Disabled, a new incoming call is automatically rejected by the IP phone with a busy signal (configured by the parameter "features.normal_refuse_code") during a call. <b>1</b> -Enabled, the LCD screen will present a new incoming call during a call. In both cases, users can put an active call on hold to make outgoing calls. <b>Note:</b> It works only if the value of the parameter "features.ignore_incoming_call.enable" is set to 0 (Disabled). <b>Web User Interface:</b> Features->General Information->Call Waiting <b>Phone User Interface:</b> Menu->Features->Call Waiting->Call Waiting		
<b>call_waiting.tone</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to play the call waiting tone when the IP phone receives an		

Parameters	Permitted Values	Default
<p>incoming call during a call.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "call_waiting.enable" is set to 1 (Enabled). It works only if the value of the parameter "features.ignore_incoming_call.enable" is set to 0 (Disabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Audio-&gt;Call Waiting Tone</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Waiting-&gt;Play Tone</p>		
<b>call_waiting.on_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the call waiting on code.</p> <p>The IP phone will send the code to activate call waiting on server-side when you activate it on the IP phone.</p> <p><b>Example:</b></p> <p>call_waiting.on_code = *71</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Call Waiting On Code</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Waiting-&gt;On Code</p>		
<b>call_waiting.off_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the call waiting off code.</p> <p>The IP phone will send the code to deactivate call waiting on server-side when you deactivate it on the IP phone.</p> <p><b>Example:</b></p> <p>call_waiting.off_code = *72</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Call Waiting Off Code</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Waiting-&gt;Off Code</p>		

**To configure call waiting via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Call Waiting**.
3. (Optional.) Enter the call waiting on code in the **Call Waiting On Code** field.
4. (Optional.) Enter the call waiting off code in the **Call Waiting Off Code** field.

The screenshot shows the 'General Information' section of the Yealink T466 web interface. The 'Call Waiting' dropdown is set to 'Enabled'. The 'Call Waiting On Code' field contains '\*71' and the 'Call Waiting Off Code' field contains '\*72'. The 'Auto Redial' dropdown is set to 'Disabled' and the 'Auto Redial Interval' is set to '10'. A red box highlights the 'Call Waiting', 'Call Waiting On Code', and 'Call Waiting Off Code' fields. A 'NOTE' box on the right explains 'Call Waiting' and 'Auto Redial'.

5. Click **Confirm** to accept the change.

**To configure call waiting tone via web user interface:**

1. Click on **Features->Audio**.
2. Select the desired value from the pull-down list of **Call Waiting Tone**.

The screenshot shows the 'Audio Settings' section of the Yealink T466 web interface. The 'Call Waiting Tone' dropdown is set to 'Enabled'. The 'Key Tone' dropdown is set to 'Enabled', the 'Send Tone' dropdown is set to 'Enabled', and the 'Redial Tone' field is empty. A red box highlights the 'Call Waiting Tone' dropdown. A 'NOTE' box on the right explains 'Tone' and 'Redial Tone'.

3. Click **Confirm** to accept the change.

**To configure call waiting and call waiting tone via phone user interface:**

1. Press **Menu->Features->Call Waiting**.
2. Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select the desired value from the **Call Waiting** field.
3. Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select the desired value from the **Play Tone** field.
4. (Optional.) Enter the call waiting on code in the **On Code** field.
5. (Optional.) Enter the call waiting off code in the **Off Code** field.
6. Press the **Save** soft key to accept the change.

## Auto Redial

Auto redial allows IP phones to automatically redial the last dialed number when the callee is temporarily unavailable. Both the number of attempts and waiting time between redials are configurable.



## Procedure

Auto redial can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure auto redial feature. <b>Parameters:</b> auto_redial.enable auto_redial.interval auto_redial.times
<b>Web User Interface</b>		Configure auto redial feature. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =features-general&q=load
<b>Phone User Interface</b>		Configure auto redial feature.

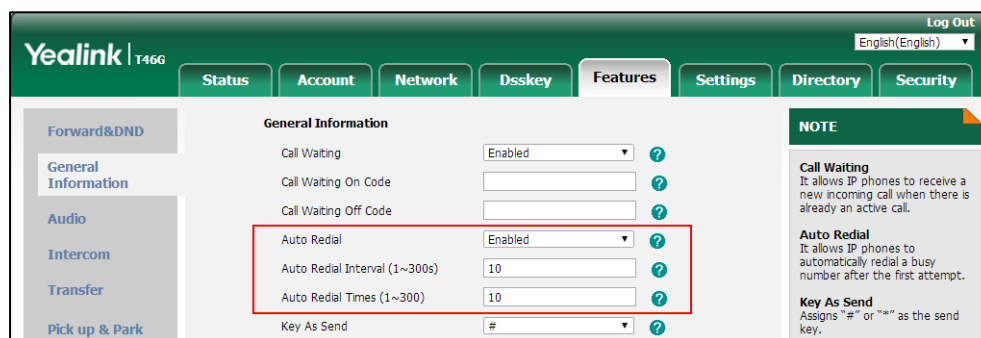
## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>auto_redial.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to automatically redial the last dialed number when the callee is temporarily unavailable. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Features->General Information->Auto Redial <b>Phone User Interface:</b> Menu->Features->Others->Auto Redial->Auto Redial		
<b>auto_redial.interval</b>	<b>Integer from 1 to 300</b>	<b>10</b>
<b>Description:</b> Configures the interval (in seconds) for the IP phone to wait between redials. The IP phone redials the last dialed number at regular intervals till the callee answers the call. <b>Note:</b> It works only if the value of the parameter "auto_redial.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Features->General Information->Auto Redial Interval (1~300s) <b>Phone User Interface:</b>		

Parameters	Permitted Values	Default
Menu->Features->Others->Auto Redial->Redial Interval		
<b>auto_redial.times</b>	<b>Integer from 1 to 300</b>	<b>10</b>
<p><b>Description:</b></p> <p>Configures the auto redial times when the callee is temporarily unavailable.</p> <p>The IP phone tries to redial the dialed number as many times as configured till the callee answers the call.</p> <p><b>Note:</b> It works only if the value of the parameter "auto_redial.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Auto Redial Times (1~300)</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Others-&gt;Auto Redial-&gt;Redial Times</p>		

**To configure auto redial via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Auto Redial**.
3. Enter the waiting time in the **Auto Redial Interval (1~300s)** field.
4. Enter the desired times in the **Auto Redial Times (1~300)** field.



5. Click **Confirm** to accept the change.

**To configure auto redial via phone user interface:**

1. Press **Menu->Features->Others->Auto Redial**.
2. Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select the desired value from the **Auto Redial** field.
3. Enter the waiting time (in seconds) in the **Redial Interval** field.
4. Enter the desired times in the **Redial Times** field.
5. Press the **Save** soft key to accept the change.

## Auto Answer

Auto answer allows IP phones to automatically answer an incoming call. By default, the IP phones will not automatically answer the incoming call during a call or while dialing even if auto answer is enabled; and the incoming call will not be automatically answered after you end the current call or cancel the dialing. You can configure the IP phones to automatically answer the incoming call after ending all calls or canceling the dialing. Auto answer is configurable on a per-line basis. Auto-Answer delay defines a period of delay time before the IP phone automatically answers incoming calls.

### Auto Answer Tone

Auto answer tone allows the IP phones to play a tone when an incoming call is automatically answered. You can customize the auto answer tone or select specialized tone sets (vary from country to country) for your IP phone. For more information, refer to [Tones](#) on page 724.

### Procedure

Auto answer can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<MAC>.cfg	Configure auto answer and auto answer mute for SIP account X. <b>Parameter:</b> account.X.auto_answer
	<y0000000000xx>.cfg	Specify a period of delay time for auto answer. <b>Parameter:</b> features.auto_answer_delay
		Configure auto answer tone. <b>Parameter:</b> features.auto_answer_tone.enable
		Configure auto answer mute. <b>Parameter:</b> features.mute.autoanswer_mute.enable
		Configure the IP phone whether to handle the incoming call arriving when the phone is in another call or is dialing like a normal call. <b>Parameter:</b> features.auto_answer.first_call_only

<b>Web User Interface</b>	Configure auto answer and auto answer mute.  <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-basic&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-basic&amp;q=load&amp;acc=0</a>
	Specify a period of delay time for auto answer.  Configure auto answer tone.  <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;servlet?p=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;servlet?p=features-general&amp;q=load</a>
<b>Phone User Interface</b>	Configure auto answer and auto answer mute.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>account.X.auto_answer<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables auto answer feature for a specific account. <b>0</b> -Disabled <b>1</b> -Enabled, the IP phone can automatically answer an incoming call. <b>Note:</b> The IP phone cannot automatically answer the incoming call during a call or while dialing even if auto answer is enabled. <b>Web User Interface:</b> Account->Basic->Auto Answer <b>Phone User Interface:</b> Menu->Features->Auto Answer->Line X->Auto Answer		
<b>features.auto_answer.first_call_only</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to only automatically answer the incoming call arrived when the phone is idle. <b>0</b> -Disabled. If there is an incoming call arrived when the IP phone is in a call or dialing, the IP phone will automatically answer after you end the current call or cancel the dialing. <b>1</b> -Enabled. The IP phone can only automatically answer the incoming call arrived when the phone is idle. If there is an incoming call arrived when the IP phone is in a call or dialing, you		

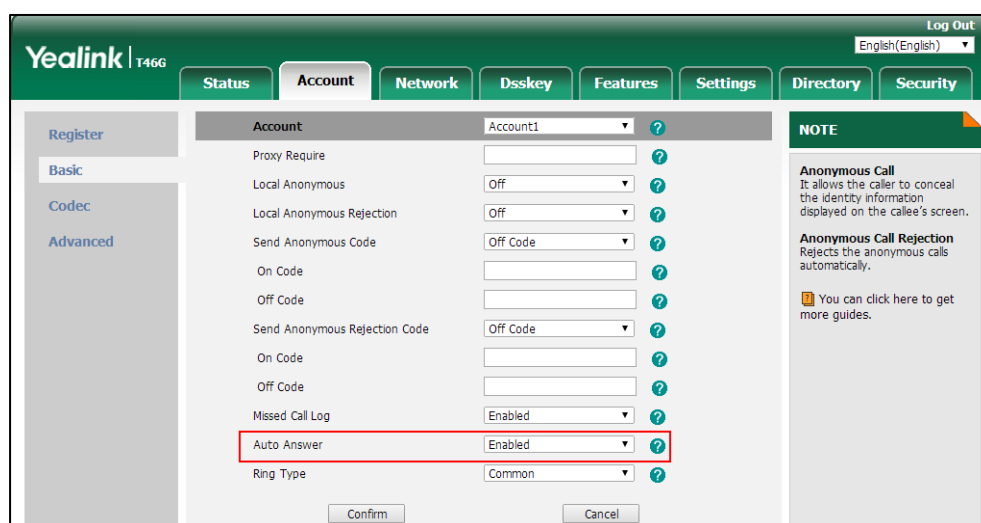
Parameters	Permitted Values	Default
<p>have to manually pick up the incoming call.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.auto_answer" or "features.ip_call.auto_answer.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.auto_answer_delay</b>	<b>Integer from 1 to 4</b>	<b>1</b>
<p><b>Description:</b></p> <p>Configures the delay time (in seconds) before the IP phone automatically answers an incoming call.</p> <p><b>Note:</b> For the call coming from a SIP account, it works only if the value of the parameter "account.X.auto_answer" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Auto Answer Delay(1~4s)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.auto_answer_tone.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the phone to play a warning tone when an incoming call is automatically answered.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> For the call coming from a SIP account, it works only if the value of the parameter "account.X.auto_answer" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Enable Auto Answer Tone</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.mute.autoanswer_mute.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the auto answer mute feature for the IP phone.</p>		

Parameters	Permitted Values	Default
<b>0-Disabled</b>  <b>1-Enabled</b> , the IP phone will mute the microphone when an incoming call is automatically answered, and then the other party cannot hear you.  <b>Note:</b> It works only if the values of parameters "account.X.auto_answer" and "features.allow_mute" are set to 1 (Enabled). It is not available to the intercom call.  <b>Web User Interface:</b>  None  <b>Phone User Interface:</b>  None		

<sup>[1]</sup> X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

#### To configure auto answer via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Auto Answer**.



4. Click **Confirm** to accept the change.

#### To configure a period of delay time for auto answer via web user interface:

1. Click on **Features->General Information**.

- Enter the desired time in the **Auto Answer Delay(1~4s)** field.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' section is active. The 'Auto Answer Delay(1~4s)' field is highlighted with a red box and contains the value '1'. Other settings in the 'General Information' section include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Allow Mute' (Enabled), 'Dual Headset' (Enabled), 'Enable Auto Answer Tone' (Enabled), 'Headset Prior' (Enabled), 'DTMF Replace Tran' (Disabled), 'Accept SIP Trust Server Only' (Disabled), 'Display Method on Dialing' (User Name), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right provides details about 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion'.

- Click **Confirm** to accept the change.



#### To configure auto answer tone via web user interface:

- Click on **Features->General Information**.
- Select the desired value from the pull-down list of **Enable Auto Answer Tone**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' section is active. The 'Enable Auto Answer Tone' dropdown menu is highlighted with a red box and is set to 'Enabled'. Other settings in the 'General Information' section include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Allow Mute' (Enabled), 'Dual Headset' (Enabled), 'Auto Answer Delay(1~4s)' (1), 'Headset Prior' (Enabled), 'DTMF Replace Tran' (Disabled), 'Accept SIP Trust Server Only' (Disabled), 'Display Method on Dialing' (User Name), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right provides details about 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion'.

- Click **Confirm** to accept the change.

#### To configure auto answer via phone user interface:

- Press **Menu->Features->Auto Answer**.
- Select the desired line.
- Press  or  , or the **Switch** soft key to select the desired value from the **Auto Answer** field.

4. Press the **Save** soft key to accept the change.

## IP Direct Auto Answer

IP direct auto answer allows IP phones to automatically answer an IP call. IP direct auto answer works only if allow IP call is enabled. For more information on allow IP call, refer to [Allow IP Call](#) on page 351.

### Procedure

IP direct auto answer can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure IP direct auto answer feature. <b>Parameter:</b> features.ip_call.auto_answer.enable
<b>Web User Interface</b>		Configure IP direct auto answer feature. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-general&q=load

### Details of Configuration Parameter:

Parameter	Permitted Values	Default
features.ip_call.auto_answer.enable	0 or 1	0
<b>Description:</b> Enables or disables the auto answer feature for IP call. <b>0</b> -Disabled <b>1</b> -Enabled, the IP phone can automatically answer an IP call. <b>Note:</b> It works only if the value of the parameter "features.direct_ip_call_enable" is set to 1 (Enabled). The IP phone cannot automatically answer the incoming IP call when the IP phone is in a call or dialing even if IP call auto answer is enabled. <b>Web User Interface:</b> Features->General Information->IP Direct Auto Answer <b>Phone User Interface:</b> None		

### To configure IP direct auto answer via web user interface:

1. Click on **Features->General Information**.



- Select the desired value from the pull-down list of **IP Direct Auto Answer**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected. Under 'General Information', the 'IP Direct Auto Answer' feature is set to 'Enabled' and is highlighted with a red box. Other features include Call Waiting (Enabled), Auto Redial (Disabled), and Call Completion (Enabled). A 'NOTE' section on the right provides details for Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

- Click **Confirm** to accept the change.

## Allow IP Call

Allow IP Call feature allows IP phones to receive or place an IP call. You can neither receive nor place an IP call if allow IP call feature is disabled.

### Procedure

Allow IP call can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure allow IP call. <b>Parameter:</b> features.direct_ip_call_enable
<b>Web User Interface</b>		Configure allow IP call. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-general&q=load

### Details of Configuration Parameter:

Parameter	Permitted Values	Default
features.direct_ip_call_enable	0 or 1	1
<b>Description:</b> Enables or disables allow IP call.		

Parameter	Permitted Values	Default
<b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> If you want to receive an IP call, make sure the value of the parameter "sip.trust_ctrl" is set to 0 (Disabled). <b>Web User Interface:</b> Features->General Information->Allow IP Call <b>Phone User Interface:</b> None		

To configure allow IP call feature via web user interface:

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Allow IP Call**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' section is active. The 'Allow IP Call' option is highlighted with a red box and is set to 'Enabled'. Other options include 'Call Waiting' (Enabled), 'Auto Redial' (Disabled), 'Accept SIP Trust Server Only' (Enabled), 'IP Direct Auto Answer' (Enabled), 'Call List Show Number' (Enabled), 'Voice Mail Tone' (Enabled), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right provides additional information about these features.

3. Click **Confirm** to accept the change.

## Accept SIP Trust Server Only

Accept SIP trust server only enables the IP phones to only accept the SIP message from your SIP server and outbound proxy server. It can prevent the phone receiving ghost calls from random numbers like 100, 1000, and so on. To stop this from happening, you also need to disable allow IP call feature. For more information on allow IP call, refer to [Allow IP Call](#) on page 351.

### Procedure

Accept SIP trust server only can be configured using the following methods.

<b>Central Provisioning</b>	<y0000000000xx>.cfg	Configure accept SIP trust server only.
-----------------------------	---------------------	-----------------------------------------

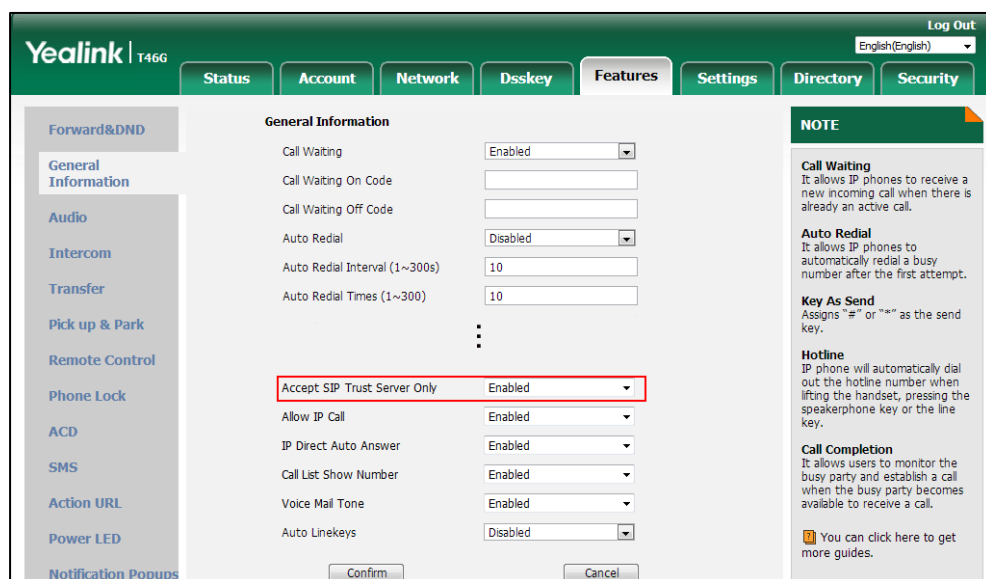
<b>(Configuration File)</b>		<b>Parameter:</b> sip.trust_ctrl
<b>Web User Interface</b>		Configure accept SIP trust server only. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-general&q=load

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
sip.trust_ctrl	0 or 1	0
<p><b>Description:</b></p> <p>Enables or disables the IP phone to only accept the SIP message from the SIP server and outbound proxy server.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Accept SIP Trust Server Only</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure accept SIP trust server only feature via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Accept SIP Trust Server Only**.



3. Click **Confirm** to accept the change.

## Call Completion

Call completion allows users to monitor the busy party and establish a call when the busy party becomes available to receive a call. Two factors commonly prevent a call from connecting successfully:

- Callee does not answer
- Callee actively rejects the incoming call before answering

IP phones support call completion using the SUBSCRIBE/NOTIFY method, which is specified in draft-poetzl-sipping-call-completion-00, to subscribe to the busy party and receive notifications of their status changes.

The caller subscribes for update notifications of the dialog event from the busy party. Example of a SUBSCRIBE message:

```
SUBSCRIBE sip:1000@10.10.20.34:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.20.32:5060;branch=z9hG4bK2880274891
From: "10111" <sip:10111@10.2.1.48:5060>;tag=8643512
To: <sip:1000@10.2.1.48:5060>;tag=4025601441
Call-ID: 4_2103527761@10.10.20.32
CSeq: 2 SUBSCRIBE
Contact: <sip:10111@10.10.20.32:5060>
Accept: application/dialog-info+xml
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Expires: 60
Event: dialog
Content-Length: 0
```

Example of a NOTIFY message (The subscription (SUBSCRIBE message) of the dialog event "Call Completion" is confirmed by the busy party):

```
NOTIFY sip:10111@10.10.20.32:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.20.31:5060;branch=z9hG4bK1830418099
From: <sip:1000@10.2.1.48:5060>;tag=1032948194
To: "10111" <sip:10111@10.2.1.48:5060>;tag=722495580
Call-ID: 0_160090766@10.10.20.32
CSeq: 2 NOTIFY
Contact: <sip:1000@10.10.20.31:5060>
Content-Type: application/dialog-info+xml
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Subscription-State: active;expires=60
Event: dialog
```

```

Content-Length: 584

<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="1" state="full"
entity="sip:1000@10.2.1.48:5060">
<dialog id="65626" call-id="0_3138198645@10.10.20.31" local-tag="2331766736" remote-tag="1786911541"
direction="initiator">
<state>confirmed</state>
<local>
<identity>sip:1000@10.2.1.48:5060</identity>
<target uri="sip:1000@10.2.1.48:5060"/>
</local>
<remote>
<identity>sip:1@10.2.1.48:5060</identity>
<target uri="sip:1@10.2.1.48:5060"/>
</remote>
</dialog>
<dialog id="65622">
<state>terminated</state>
</dialog>
</dialog-info>

```

Example of a NOTIFY message (The busy party has finished the call and is available again. A new notification update from the busy party is received by the caller):

```

NOTIFY sip:10111@10.10.20.32:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.20.31:5060;branch=z9hG4bK3431394016
From: <sip:1000@10.2.1.48:5060>;tag=1558968605
To: "10111" <sip:10111@10.2.1.48:5060>;tag=140677866
Call-ID: 0_2584152566@10.10.20.32
CSeq: 5 NOTIFY
Contact: <sip:1000@10.10.20.31:5060>
Content-Type: application/dialog-info+xml
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Subscription-State: active;expires=48
Event: dialog
Content-Length: 217

<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="4" state="partial"
entity="sip:1000@10.2.1.48:5060">
<dialog id="65644">
<state>terminated</state>
</dialog>

```

</dialog-info>

## Procedure

Call completion can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure call completion. <b>Parameter:</b> features.call_completion_enable
<b>Web User Interface</b>		Configure call completion. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =features-general&q=load
<b>Phone User Interface</b>		Configure call completion.

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
<b>features.call_completion_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables call completion feature. If a user places a call and the callee is temporarily unavailable to answer the call, call completion feature allows notifying the user when the callee becomes available to receive a call.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, the caller is notified when the callee becomes available to receive a call.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Call Completion</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Others-&gt;Call Completion-&gt;Call Completion</p>		

## To configure call completion via web user interface:



1. Click on **Features->General Information**.

2. Select the desired value from the pull-down list of **Call Completion**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected. In the 'General Information' section, the 'Call Completion' feature is highlighted with a red box, showing it is set to 'Enabled'. The interface includes a sidebar with navigation options like Forward&DND, General Information, Audio, Intercom, Transfer, Pick up & Park, Remote Control, Phone Lock, ACD, SMS, Action URL, and Bluetooth. The main area displays various settings for Call Waiting, Auto Redial, Key As Send, Reserve # in User Name, Hotline Number, Busy Tone Delay, Return Code When Refuse, Return Code When DND, Call Completion, and Feature Key Synchronization. A 'NOTE' section on the right provides details for Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

3. Click **Confirm** to accept the change.

#### To configure call completion via phone user interface:

1. Press **Menu->Features->Others->Call Completion**.
2. Press  or , or the **Switch** soft key to select the desired value from the **Call Completion** field.
3. Press the **Save** soft key to accept the change.

## Anonymous Call

Anonymous call allows the caller to conceal the identity information displayed on the callee's screen. The callee's phone LCD screen prompts an incoming call from anonymity. Anonymous call is configurable on a per-line basis.

Example of anonymous SIP header when the anonymous call feature is performed on local:

```
Via: SIP/2.0/UDP 10.3.20.14:5060;branch=z9hG4bK3074920774
From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=131654239
To: <sip:1006@10.2.1.48:5060>
Call-ID: 0_288363101@10.3.20.14
CSeq: 1 INVITE
Contact: <sip:1009@10.3.20.14:5060>
Content-Type: application/sdp
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH,
UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Allow-Events: talk,hold,conference,refer,check-sync
```

```
P-Preferred-Identity: <sip:1009@10.2.1.48>
Privacy: id
Content-Length: 302
```

The anonymous call on code and anonymous call off code configured on IP phones are used to activate/deactivate the server-side anonymous call feature. They may vary on different servers.

Yealink IP phones running firmware version 82 or later support a new Anonymous Call mechanism. The following only describe how to configure the new Anonymous Call mechanism. For more information on old Anonymous Call mechanism, refer to [Yealink SIP-T2 Series T19\(P\) E2\\_T4 Series T5 Series CP860 IP Phones Administrator Guide V81](#).

## Procedure

Anonymous call can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure anonymous call. <b>Parameters:</b> account.X.anonymous_call account.X.anonymous_call.server_base_only account.X.anonymous_call_oncode account.X.anonymous_call_offcode
<b>Web User Interface</b>		Configure anonymous call. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-basic&q=load&acc=0
<b>Phone User Interface</b>		Configure anonymous call.

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>account.X.anonymous_call<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Triggers the anonymous call feature to on or off for a specific account. <b>0-Off</b> <b>1-On</b> , the IP phone will block its identity from showing up to the callee when placing a call. The callee's phone LCD screen presents "Anonymous" instead of the caller's identity. <b>Web User Interface:</b> Account->Basic->Anonymous (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank) <b>Phone User Interface:</b>		



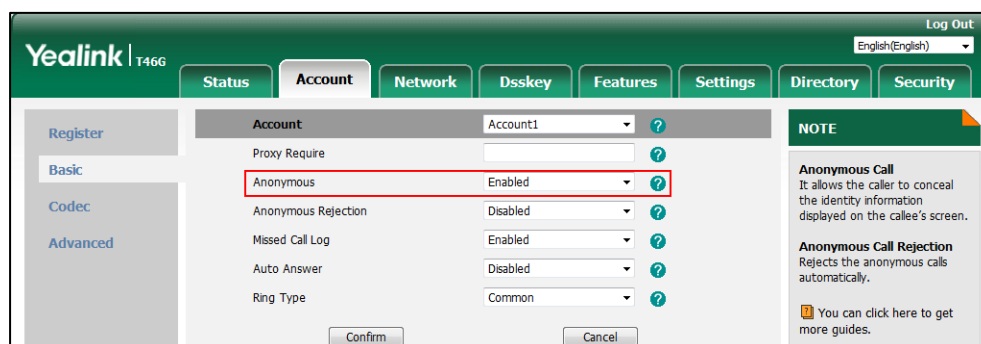
Parameters	Permitted Values	Default
Menu->Features->Anonymous->Line X->Anonymous (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)		
<b>account.X.anonymous_call.server_base_only<sup>[1]</sup></b>	<b>0 or 1</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to perform the anonymous call feature on server-side only.</p> <p><b>0</b>-Disabled, the IP phone will perform the anonymous call feature on server-side and local. If the anonymous call feature is enabled on the IP phone, the IP phone will carry the Anonymous attribute in the From header of the INVITE message.</p> <p><b>1</b>-Enabled, the IP phone will perform the anonymous call feature on server-side only. The IP phone will not carry the Anonymous attribute in the From header of the INVITE message even if the anonymous call feature is enabled on the IP phone.</p> <p><b>Note:</b> You need to configure parameters "account.X.anonymous_call_oncode" and "account.X.anonymous_call_offcode" to activate or deactivate the server-side anonymous call feature.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.anonymous_call_oncode<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the anonymous call on code for a specific account.</p> <p>The IP phone will send the code to activate anonymous call feature on server-side when you activate it on the IP phone.</p> <p><b>Example:</b></p> <p>account.1.anonymous_call_oncode = *72</p> <p><b>Web User Interface:</b></p> <p>None (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)</p> <p><b>Phone User Interface:</b></p> <p>None (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)</p>		
<b>account.X.anonymous_call_offcode<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p>		

Parameters	Permitted Values	Default
<p>Configures the anonymous call off code for a specific account.</p> <p>The IP phone will send the code to deactivate anonymous call feature on server-side when you deactivate it on the IP phone.</p> <p><b>Example:</b></p> <p>account.1.anonymous_call_offcode = *73</p> <p><b>Web User Interface:</b></p> <p>None (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)</p> <p><b>Phone User Interface:</b></p> <p>None (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure anonymous call via web user interface (only for new Anonymous Call mechanism):**

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Anonymous**.



4. Click **Confirm** to accept the change.

**To configure the anonymous call via phone user interface (only for new anonymous call mechanism):**

1. Press **Menu->Features->Anonymous**.
2. Select the desired line and then press the **Enter** soft key.
3. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Anonymous** field.
4. Press the **Save** soft key to accept the change.

## Anonymous Call Rejection

Anonymous call rejection allows IP phones to automatically reject incoming calls from callers whose identity has been deliberately concealed. The anonymous caller's phone LCD screen presents "Anonymity Disallowed". Anonymous call rejection is configurable on a per-line basis.

Example of anonymous call rejection SIP header:

```
SIP/2.0 433 Anonymity Disallowed
Via: SIP/2.0/UDP 10.10.20.32:5060;branch=z9hG4bK2816884590
From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=2625078618
To: <sip:1058@10.2.1.48:5060>;tag=2781829106
Call-ID: 4_510565349@10.10.20.32
CSeq: 1 INVITE
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH,
UPDATE, MESSAGE
User-Agent: Yealink SIP-T46G 28.82.0.20
Allow-Events: talk, hold, conference, refer, check-sync
Content-Length: 0
```

The anonymous call rejection on code and anonymous call rejection off code configured on IP phones are used to activate/deactivate the server-side anonymous call rejection feature. They may vary on different servers.

Yealink IP phones running firmware version 82 or later support a new Anonymous Call Rejection mechanism. The following only describe how to configure the new Anonymous Call Rejection mechanism. For more information on old Anonymous Call Rejection, refer to

[Yealink SIP-T2\\_Series\\_T19\(P\)](#)

[E2\\_T4\\_Series\\_T5\\_Series\\_CP860\\_IP\\_Phones\\_Administrator\\_Guide\\_V81](#).

### Procedure

Anonymous call rejection can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure anonymous call rejection. <b>Parameters:</b> account.X.reject_anonymous_call account.X.anonymous_reject_oncode account.X.anonymous_reject_offcode
<b>Web User Interface</b>		Configure anonymous call rejection. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-basic&q=load&acc=0
<b>Phone User Interface</b>		Configure anonymous call rejection.

**Details of Configuration Parameters:**

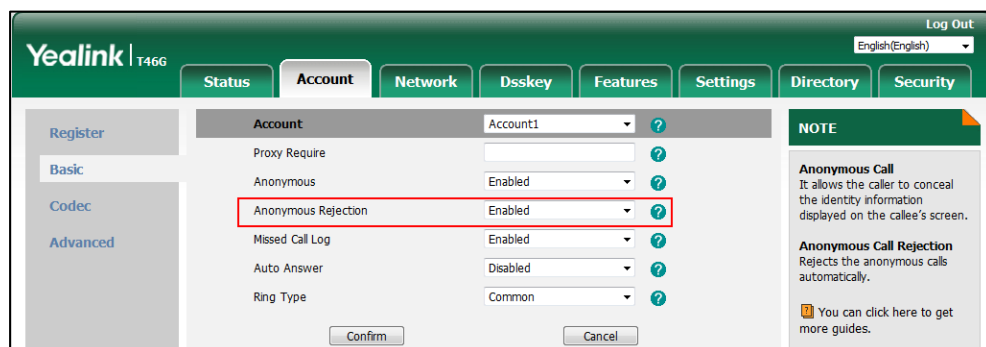
Parameters	Permitted Values	Default
<b>account.X.reject_anonymous_call<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Triggers the anonymous call rejection feature to on or off for a specific account.</p> <p><b>0</b>-Off</p> <p><b>1</b>-On, the IP phone will automatically reject incoming calls from users enabled anonymous call feature. The anonymous user's phone LCD screen presents "Anonymity Disallowed".</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Basic-&gt;Anonymous Rejection (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Anonymous-&gt;Line X-&gt;Anonymous Rejection (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)</p>		
<b>account.X.anonymous_reject_oncode<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the anonymous call rejection on code for a specific account.</p> <p>The IP phone will send the code to activate anonymous call rejection feature on server-side when you activate it on the IP phone.</p> <p><b>Example:</b></p> <p>account.1.anonymous_reject_oncode = *74</p> <p><b>Web User Interface:</b></p> <p>None (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)</p> <p><b>Phone User Interface:</b></p> <p>None (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)</p>		
<b>account.X.anonymous_reject_offcode<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the anonymous call rejection off code for a specific account.</p> <p>The IP phone will send the code to deactivate anonymous call rejection feature on server-side when you deactivate it on the IP phone.</p> <p><b>Example:</b></p>		

Parameters	Permitted Values	Default
<p>account.1.anonymous_reject_offcode = *75</p> <p><b>Web User Interface:</b></p> <p>None (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)</p> <p><b>Phone User Interface:</b></p> <p>None (the value of the parameter "account.X.anonymous_call.server_base_only" must not be left blank)</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure anonymous call rejection via web user interface (only for new Anonymous Call Rejection mechanism):**

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Anonymous Rejection**.



4. Click **Confirm** to accept the change.

**To configure anonymous call rejection via phone user interface (only for new Anonymous Call Rejection mechanism):**

1. Press **Menu->Features->Anonymous**.
2. Select the desired line and then press the **Enter** soft key.
3. Press **◀** or **▶** to select **Enabled** from the **Anonymous Rejection** field.
4. Press the **Save** soft key to accept the change.

## Do Not Disturb (DND)

DND allows IP phones to ignore incoming calls. Incoming calls received while DND is activated are logged in the Missed Calls list. DND feature can be configured on a phone or a per-line basis

depending on the DND mode.

There are two DND modes:

- **Phone** (default): DND feature is effective for the phone system. You can apply DND to all line registrations on the phone.
- **Custom**: DND feature can be configured for each or all accounts. You can specify individual or all line registrations on a phone that has multiply registered lines.

The DND feature is enabled by default, and you have the option of disabling it. Users have no permission to configure DND on the phone if DND feature is disabled. When DND is enabled, it can be activated locally through the phone or through a server. A user can activate or deactivate DND using the DND DSS key or DND soft key on the phone. The server-side DND feature deactivates the local DND and Call Forward settings. If the server-side DND feature is activated on any of the IP phone's line registrations, the other line registrations are not affected. For more information on Call Forward, refer to [Call Forward](#) on page 400.

The DND on code and DND off code configured on IP phones are used to activate/deactivate the server-side DND feature. They may vary on different servers.

## Return Message When DND

This feature defines the return code and the reason of the SIP response message for the rejected incoming call when DND is activated on the IP phone. The caller's phone LCD screen displays the received return code.

## DND Emergency

This feature allows users to receive the incoming calls from some authorized numbers even if the DND feature is activated. This feature is disabled by default.

## DND Large Icon

This feature allows the IP phones to display a large DND icon on the idle screen. It helps users to clearly view that DND is activated.



## DND Feature Synchronization

DND Feature Synchronization feature provides the capability to synchronize the status of the DND features between the IP phone and the server.

If the DND is activated in phone mode, the DND status changed on local will be synchronized to all registered accounts on server; but if the DND status of specific account is changed on server, the DND status on local will be changed.

## Procedure

DND can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<MAC>.cfg	Configure DND in the custom mode. <b>Parameters:</b> account.X.dnd.enable account.X.dnd.on_code account.X.dnd.off_code
		Configure DND Feature Synchronization on a per-line basis. <b>Parameter:</b> account.X.features.dnd.feature_key_sync.local_processing.enable
	<y0000000000xx>.cfg	Enable or disable the DND feature. <b>Parameter:</b> features.dnd.allow
		Configure the DND mode. <b>Parameter:</b> features.dnd_mode
		Configure DND in the phone mode. <b>Parameters:</b> features.dnd.enable features.dnd.on_code features.dnd.off_code
		Specify the authorized numbers when DND is activated. <b>Parameters:</b> features.dnd.emergency_enable features.dnd.emergency_authorized_number

		Specify the return code and the reason of the SIP response message when DND is activated.  <b>Parameter:</b> features.dnd_refuse_code
		Configure large DND icon feature.  <b>Parameter:</b> features.dnd.large_icon.enable
		Assign a DND DSS key.  <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type  linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label  linekey.X.shortlabel
		Configure DND Feature Synchronization.  <b>Parameters:</b> features.feature_key_sync.enable features.dnd.feature_key_sync.enable features.dnd.feature_key_sync.local_processing.enable
<b>Web User Interface</b>		Configure DND.  <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-forward&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-forward&amp;q=load</a>
		Specify the authorized numbers when DND is activated.  Specify the return code and the reason of the SIP response message when DND is activated.  <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load</a>



	Assign a DND DSS key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0">http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0</a>
<b>Phone User Interface</b>	Configure DND. Assign a DND DSS key.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>features.dnd.allow</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the DND feature. <b>0</b> -Disabled, DND cannot be activated and users are not allowed to configure DND on the phone. <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.feature_key_sync.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to send a SUBSCRIBE message with event as-feature-event. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.dnd.feature_key_sync.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the DND feature synchronization. <b>0</b> -Disabled <b>1</b> -Enabled, a user changes the status of DND feature on the server, and then the server		

<p>notifies the phone of synchronizing the status. Conversely, if the user changes the DND feature status on the phone, the IP phone notifies the server of synchronizing the status.</p> <p><b>Note:</b> It works only if the value of the parameter "features.feature_key_sync.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.dnd.feature_key_sync.local_processing.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the local DND when DND is activated.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameters "features.feature_key_sync.enable" and "features.dnd.feature_key_sync.enable" are set to 1 (Enabled). The value configured by the parameter "account.X.features.dnd.feature_key_sync.local_processing.enable" takes precedence over that configured by this parameter.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.dnd_mode</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the DND mode for the IP phone.</p> <p><b>0</b>-Phone, DND feature is effective for the phone system.</p> <p><b>1</b>-Custom, you can configure DND feature for each or all accounts.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;DND-&gt;Mode</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.dnd.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p>		

<p>Triggers DND feature to on or off.</p> <p><b>0-Off</b></p> <p><b>1-On</b>, the IP phone will reject incoming calls on all accounts.</p> <p><b>Note:</b> It works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled) and value of the parameter "features.dnd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;DND-&gt;DND Status</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DND-&gt;DND Status</p>		
<b>features.dnd.on_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the DND on code to activate the server-side DND feature.</p> <p>The IP phone will send the DND on code to the server when you activate DND feature on the IP phone.</p> <p><b>Example:</b></p> <p>features.dnd.on_code = *71</p> <p><b>Note:</b> It works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled) and value of the parameter "features.dnd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;DND-&gt;On Code</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DND-&gt;On Code</p>		
<b>features.dnd.off_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the DND off code to deactivate the server-side DND feature.</p> <p>The IP phone will send the DND off code to the server when you deactivate DND feature on the IP phone.</p> <p><b>Example:</b></p> <p>features.dnd.off_code = *72</p> <p><b>Note:</b> It works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled) and value of the parameter "features.dnd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones).</p> <p><b>Web User Interface:</b></p>		

Features->Forward&DND->DND->Off Code <b>Phone User Interface:</b> Menu->Features->DND->Off Code		
<b>account.X.dnd.enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Triggers DND feature to on or off for a specific account. <b>0</b> -Off <b>1</b> -On, the IP phone will reject incoming calls on account X. <b>Note:</b> It works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled) and value of the parameter "features.dnd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> Features->Forward&DND->DND->AccountX->DND Status <b>Phone User Interface:</b> Menu->Features->DND->AccountX->DND Status		
<b>account.X.features.dnd.feature_key_sync.local_processing.enable</b>	<b>0 or 1</b>	<b>Blank</b>
<b>Description:</b> Enables or disables the local DND for a specific account when DND is activated. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameters "features.feature_key_sync.enable" and "features.dnd.feature_key_sync.enable" are set to 1 (Enabled). It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>account.X.dnd.on_code<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the DND on code to activate the server-side DND feature for a specific account. The IP phone will send the DND on code to the server when you activate DND feature for a specific account on the IP phone. <b>Example:</b> account.1.dnd.on_code = *73		

**Note:** It works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled) and value of the parameter "features.dnd\_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones.

**Web User Interface:**

Features->Forward&DND->AccountX->On Code

**Phone User Interface:**

Menu->Features->DND->AccountX->On Code

**account.X.dnd.off\_code<sup>[1]</sup>**

**String within 32  
characters**

**Blank**

**Description:**

Configures the DND off code to deactivate the server-side DND feature for a specific account.

The IP phone will send the DND off code to the server when you deactivate DND feature for a specific account on the IP phone.

**Example:**

account.1.dnd.off\_code = \*74

**Note:** It works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled) and value of the parameter "features.dnd\_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones.

**Web User Interface:**

Features->Forward&DND->AccountX->Off Code

**Phone User Interface:**

Menu->Features->DND->AccountX->Off Code

**features.dnd.emergency\_enable**

**0 or 1**

**0**

**Description:**

Enables or disables the IP phone to receive incoming calls from authorized numbers when DND feature is enabled.

**0**-Disabled

**1**-Enabled

**Note:** The authorized numbers are configured by the parameter "features.dnd.emergency\_authorized\_number". It works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled).

**Web User Interface:**

Features->Forward&DND->DND->DND Emergency

**Phone User Interface:**

None

<b>features.dnd.emergency_authorized_number</b>	<b>String within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the authorized numbers the IP phone can receive incoming calls from even if DND feature is activated.</p> <p>Multiple numbers are separated by commas.</p> <p><b>Example:</b></p> <p>features.dnd.emergency_authorized_number = 123,124</p> <p><b>Note:</b> It works only if the values of the parameters "features.dnd.allow" and "features.dnd.emergency_enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;DND-&gt;DND Authorized Numbers</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.dnd_refuse_code</b>	<b>404, 480, 486 or 603</b>	<b>480</b>
<p><b>Description:</b></p> <p>Configures a return code and reason of SIP response messages when rejecting an incoming call by DND. A specific reason is displayed on the caller's phone LCD screen.</p> <p><b>404</b>-Not Found</p> <p><b>480</b>-Temporarily Unavailable</p> <p><b>486</b>-Busy Here, the caller's phone LCD screen will display the reason "Busy Here" when the callee enables DND.</p> <p><b>603</b>-Decline</p> <p><b>Note:</b> For Yealink IP phones, it works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Return Code When DND</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.dnd.large_icon.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to display a large DND icon on the idle screen.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> For Yealink IP phones, it works only if the value of the parameter "features.dnd.allow"</p>		

is set to 1 (Enabled).

**Web User Interface:**

None

**Phone User Interface:**

None

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## DND DSS Key

DND DSS key can be configured only if the DND feature is enabled. For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>programablekey.X.type<sup>[2]</sup></b> <b>expansion_module.X.key.Y.type<sup>[3]</sup></b>	5
<p><b>Description:</b></p> <p>Configures a DSS key as a DND DSS key on the IP phone.</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key/Programable Key-&gt;Type</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Type</p>	
<b>linekey.X.label<sup>[1]</sup></b> <b>programablekey.X.label<sup>[2]</sup></b> <b>expansion_module.X.key.Y.label<sup>[3]</sup></b>	String within 99 characters
<p><b>Description:</b></p> <p>(Optional.) Configures the label displayed on the LCD screen for each DSS key.</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key/Programable Key-&gt;Label</p> <p><b>Phone User Interface:</b></p>	

Parameters	Permitted Values
Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	<b>String within 99 characters</b>
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Short <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Short Label	

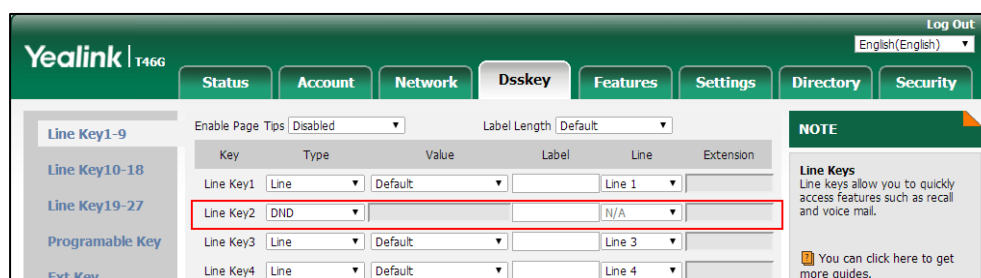
<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programmable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure a DND DSS key via web user interface:

1. Click on **Dsskey->Line Key** (or **Programmable Key**).
2. In the desired DSS key field, select **DND** from the pull-down list of **Type**.
3. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.



4. Click **Confirm** to accept the change.

#### To configure DND feature via web user interface:

1. Click on **Features->Forward&DND**.
2. In the **DND** block, mark the desired radio box in the **Mode** field.
  - a) If you mark the **Phone** radio box:
    - 1) Mark the desired radio box in the **DND Status** field.



- 2) (Optional.) Enter the DND on code in the **On Code** field.
- 3) (Optional.) Enter the DND off code in the **Off Code** field.

The screenshot displays the Yealink T46G web interface for configuring Forward and DND features. The interface includes a sidebar with navigation links and a main content area with tabs for Status, Account, Network, Dsskey, Features, Settings, Directory, and Security. The 'Features' tab is selected, and the 'Forward&DND' section is active. The 'Forward' section contains fields for Forward Emergency (Disabled), Forward Authorized Numbers, Mode (Phone/Custom), Account (101), Always Forward (On/Off), Busy Forward (On/Off), and No Answer Forward (On/Off). The 'DND' section includes DND Emergency (Disabled), DND Authorized Numbers, Mode (Phone/Custom), Account (101), DND Status (On/Off), On Code (\*71), and Off Code (\*72). A red box highlights the DND Status, On Code, and Off Code fields. A 'NOTE' section on the right provides information about Call Forward, Do Not Disturb (DND), and DND Mode.

- b) If you mark the **Custom** radio box:
- 1) Select the desired account from the pull-down list of **Account**.
  - 2) Mark the desired radio box in the **DND Status** field.
  - 3) (Optional.) Enter the DND on code in the **On Code** field.

- 4) (Optional.) Enter the DND off code in the **Off Code** field.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'Forward' section is expanded. The 'DND' (Do Not Disturb) section is highlighted with a red box. The 'DND Status' is set to 'On', and the 'On Code' is \*73 and the 'Off Code' is \*74. The 'NOTE' section on the right provides additional information about Call Forward and DND features.

**NOTE**

**Call Forward**  
It allows users to redirect an incoming call to a third party.

**Call Forward Mode**  
**Phone:** Call forward feature is effective for the IP phone.  
**Custom:** Call forward feature can be configured for each or all accounts.

**Do Not Disturb (DND)**  
It allows IP phones to ignore incoming calls.

**DND Mode**  
**Phone:** DND feature is effective for the IP phone.  
**Custom:** DND feature can be configured for each or all accounts.

You can click here to get more guides.

3. Click **Confirm** to accept the change.

**To specify the authorized numbers when DND is activated via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **DND Emergency**.
3. Enter the desired value in the **DND Authorized Numbers** field.

Multiple numbers are separated by commas.

The screenshot shows the Yealink T46G web interface with the 'Features' tab selected. The 'Forward' section is active, and the 'DND' (Do Not Disturb) settings are highlighted with a red box. The 'DND Emergency' is set to 'Enabled', and 'DND Authorized Numbers' is set to '123,124'. Other settings like 'Forward Emergency', 'Forward Authorized Numbers', 'Mode', 'Account', 'Always Forward', 'Busy Forward', and 'No Answer Forward' are also visible.

4. Click **Confirm** to accept the change.

To specify the return code and the reason when DND is activated via web user interface:





1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Return Code When DND**.

The screenshot shows the Yealink T46G web interface with the 'Features' tab selected. The 'General Information' section is active, and the 'Return Code When DND' is highlighted with a red box. It is set to '480 (Temporarily Unavailable)'. Other settings like 'Call Waiting', 'Auto Redial', 'Key As Send', and 'Hotline' are also visible.

3. Click **Confirm** to accept the change.

To configure a DND DSS key via phone user interface:



1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.

3. Press  or  , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press  or  , or the **Switch** soft key to select **DND** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Press the **Save** soft key to accept the change.

**To configure DND in the phone mode via phone user interface:**

1. Press the **DND** soft key or the DND DSS key when the IP phone is idle.

**To configure DND in the custom mode for a specific account via phone user interface:**

1. Press the **DND** soft key or the DND DSS key when the IP phone is idle.  
The LCD screen displays a list of accounts registered on the IP phone.
2. Press  or  to select the desired account and then press the **Enter** soft key.
3. Press the **Switch** soft key to activate DND.
4. (Optional.) Enter the DND on code or off code respectively in the **On Code** or **Off Code** field.
5. Press the **Save** soft key to accept the change.

You can configure DND in the custom mode for all accounts by pressing the **All On** soft key.

## Busy Tone Delay

Busy tone is audible to indicate that the call is released by the other party. You can define the amount of time to play the busy tone.

### Procedure

Busy tone delay can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure busy tone delay. <b>Parameter:</b> features.busy_tone_delay
<b>Web User Interface</b>		Configure busy tone delay. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?parameters=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?parameters=features-general&amp;q=load</a>

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.busy_tone_delay</b>	<b>0, 3 or 5</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the duration time (in seconds) to play the busy tone when the call is released by the other party.</p> <p><b>0-0s</b>, the IP phone will not play a busy tone.</p> <p><b>3-3s</b>, a busy tone is audible for 3 seconds on the IP phone.</p> <p><b>5-5s</b></p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Busy Tone Delay (Seconds)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure busy tone delay via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Busy Tone Delay (Seconds)**.

The screenshot displays the Yealink T46G web management interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'Busy Tone Delay (Seconds)' parameter is set to 0 and is highlighted with a red rectangular box. Other parameters visible include Call Waiting (Enabled), Auto Redial (Disabled), and Hotline Delay (4). A 'NOTE' panel on the right side of the interface provides explanatory text for several features: Call Waiting, Auto Redial, Key As Send, and Hotline.

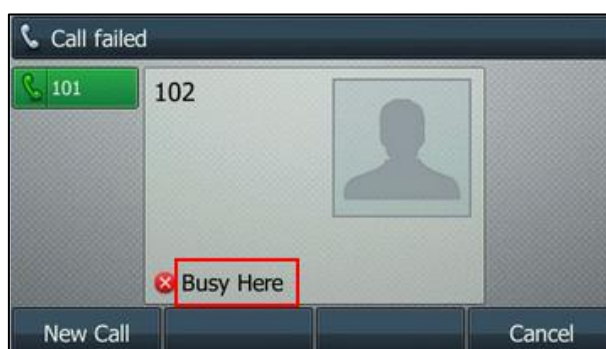
3. Click **Confirm** to accept the change.

## Return Code When Refuse

Return code when refuse defines the return code and reason of the SIP response message for the refused call. The caller's phone LCD screen displays the reason according to the received return code. Available return codes and reasons are:

- 404 (Not Found)
- 480 (Temporarily Unavailable)
- 486 (Busy Here)
- 603 (Decline)

The following shows an example of the returned reason for the refused call:



### Procedure

Return code for refused call can be configured using the following methods.

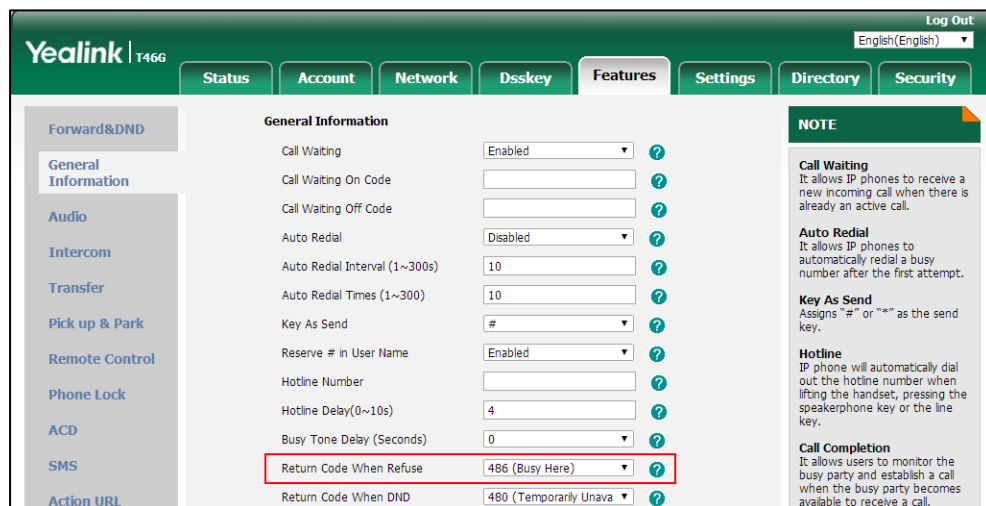
<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify the return code and the reason of the SIP response message when refusing a call.  <b>Parameter:</b> features.normal_refuse_code
<b>Web User Interface</b>		Specify the return code and the reason of the SIP response message when refusing a call.  <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?parameters=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?parameters=features-general&amp;q=load</a>

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.normal_refuse_code</b>	<b>404, 480, 486 or 603</b>	<b>486</b>
<p><b>Description:</b></p> <p>Configures a return code and reason of SIP response messages when the IP phone rejects an incoming call. A specific reason is displayed on the caller's phone LCD screen.</p> <p><b>404</b>-Not Found</p> <p><b>480</b>-Temporarily Unavailable</p> <p><b>486</b>-Busy Here, the caller's phone LCD screen will display the message "Busy Here" when the callee rejects the incoming call.</p> <p><b>603</b>-Decline</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Return Code When Refuse</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

To specify the return code and the reason when refusing a call via web user interface:

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Return Code When Refuse**.



3. Click **Confirm** to accept the change.

## Early Media

Early media refers to media (for example, audio and video) played to the caller before a SIP call

is actually established. Current implementation supports early media through the 183 message. When the caller receives a 183 message with SDP before the call is established, a media channel is established. This channel is used to provide the early media stream for the caller.

You can configure the IP phone to play the local ringback tone when receiving the 183 message but do not receive the RTP stream for the early media.

## Procedure

Early media can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configures the time to wait for the IP phone to play the local ringback tone when the early media cannot be played.  <b>Parameter:</b> phone_setting.early_media.rtp_sniffer.timeout
------------------------------------------------------	---------------------	-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
phone_setting.early_media.rtp_sniffer.timeout	-1, 0 or integer from 1 to 100	1
<b>Description:</b> Configures the time to wait for the IP phone to play the local ringback tone when the early media cannot be played. <b>-1</b> -Do not play the local ringback tone <b>0</b> -Play the local ringback tone immediately <b>1-100</b> -Wait for specified time to play the local ringback tone <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## 180 Ring Workaround

180 ring workaround defines whether to deal with the 180 message received after the 183 message. When the caller receives a 183 message, it suppresses any local ringback tone and begins to play the media received. 180 ring workaround allows IP phones to resume and play the local ringback tone upon a subsequent 180 message received.



## Procedure

180 ring workaround can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure 180 ring workaround. <b>Parameter:</b> phone_setting.is_deal180
<b>Web User Interface</b>		Configure 180 ring workaround. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =features-general&q=load

### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
<b>phone_setting.is_deal180</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to deal with the 180 SIP message received after the 183 SIP message. <b>0</b> -Disabled <b>1</b> -Enabled, the IP phone will resume and play the local ringback tone upon a subsequent 180 message received. <b>Web User Interface:</b> Features->General Information->180 Ring Workaround <b>Phone User Interface:</b> None		

**To configure 180 ring workaround via web user interface:**

1. Click on **Features->General Information**.

2. Select the desired value from the pull-down list of **180 Ring Workaround**.

The screenshot shows the Yealink T466 web interface. The 'Features' tab is selected. In the 'General Information' section, the '180 Ring Workaround' is set to 'Enabled' and is highlighted with a red box. Other settings include 'Call Waiting' (Enabled), 'Auto Redial' (Disabled), and 'Auto Redial Interval' (10). A 'NOTE' section on the right explains features like 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion'. At the bottom, there are 'Confirm' and 'Cancel' buttons.

3. Click **Confirm** to accept the change.

## Use Outbound Proxy in Dialog

An outbound proxy server can receive all initiating request messages and route them to the designated destination. If the IP phone is configured to use an outbound proxy server within a dialog, all SIP request messages from the IP phone will be sent to the outbound proxy server forcibly.

**Note** To use this feature, make sure the outbound server has been correctly configured on the IP phone. For more information on how to configure outbound server, refer to [Account Registration](#) on page 192.

### Procedure

Use outbound proxy in dialog can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify whether to use outbound proxy in a dialog. <b>Parameter:</b> sip.use_out_bound_in_dialog
<b>Web User Interface</b>		Specify whether to use outbound proxy in a dialog. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-general&q=load

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>sip.use_out_bound_in_dialog</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to send all SIP requests to the outbound proxy server forcibly in a dialog.</p> <p><b>0</b>-Disabled, only the new SIP request messages from the IP phone will be sent to the outbound proxy server in a dialog.</p> <p><b>1</b>-Enabled, all the SIP request messages from the IP phone will be forced to send to the outbound proxy server in a dialog.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.outbound_proxy_enable" is set to 1 (Enabled) and the outbound server address has been correctly configured on the phone.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Use Outbound Proxy In Dialog</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure use outbound proxy in dialog via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Use Outbound Proxy In Dialog**.

The screenshot shows the Yealink 1466 web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'Use Outbound Proxy In Dialog' option is highlighted with a red rectangle and is currently set to 'Enabled'. Other visible options include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Call Waiting Off Code' (empty), 'Auto Redial' (Disabled), 'Auto Redial Interval (1~300s)' (10), 'Auto Redial Times (1~300)' (10), '180 Ring Workaround' (Enabled), 'Logon Wizard' (Enabled), 'PswPrefix' (12), 'PswLength' (3), and 'Auto Linekeys' (Disabled). A 'NOTE' panel on the right provides details for 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion'. At the bottom, there are 'Confirm' and 'Cancel' buttons.

3. Click **Confirm** to accept the change.

## SIP Session Timer

SIP session timers T1, T2 and T4 are SIP transaction layer timers defined in [RFC 3261](#). These session timers are configurable on IP phones.

### Timer T1

Timer T1 is an estimate of the Round Trip Time (RTT) of transactions between a SIP client and SIP server.

### Timer T2

Timer T2 represents the maximum retransmitting time of any SIP request message. The re-transmitting and doubling of T1 will continue until the retransmitting time reaches the T2 value.

#### Example:

The user registers a SIP account for the IP phone and then set the value of Timer T1, Timer T2 respectively (Timer T1: 0.5, Timer T2: 4). The SIP registration request message will be re-transmitted between the IP phone and SIP server. The re-transmitting and doubling of Timer T1 (0.5) will continue until the retransmitting time reaches the Timer T2 (4). The total registration request retry time will be less than 64 times of T1 ( $64 * 0.5 = 32$ ). The re-transmitting interval in sequence is: 0.5s, 1s, 2s, 4s, 4s, 4s, 4s, 4s, 4s and 4s.

### Timer T4

Timer T4 represents that the network will take to clear messages between the SIP client and server.

### Procedure

SIP session timer can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure SIP session timer. <b>Parameters:</b> sip.timer_t1 sip.timer_t2 sip.timer_t4
<b>Web User Interface</b>		Configure SIP session timer. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =settings-sip&q=load

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>sip.timer_t1</b>	<b>Float from 0.5 to 10</b>	<b>0.5</b>
<p><b>Description:</b> Configures the SIP session timer T1 (in seconds). T1 is an estimate of the Round Trip Time (RTT) of transactions between a SIP client and SIP server.</p> <p><b>Web User Interface:</b> Settings-&gt;SIP-&gt;SIP Session Timer T1 (0.5~10s)</p> <p><b>Phone User Interface:</b> None</p>		
<b>sip.timer_t2</b>	<b>Float from 2 to 40</b>	<b>4</b>
<p><b>Description:</b> Configures the SIP session timer T2 (in seconds). Timer T2 represents the maximum retransmitting time of any SIP request message.</p> <p><b>Web User Interface:</b> Settings-&gt;SIP-&gt;SIP Session Timer T2 (2~40s)</p> <p><b>Phone User Interface:</b> None</p>		
<b>sip.timer_t4</b>	<b>Float from 2.5 to 60</b>	<b>5</b>
<p><b>Description:</b> Configures the SIP session timer of T4 (in seconds). T4 represents the maximum duration a message will remain in the network.</p> <p><b>Web User Interface:</b> Settings-&gt;SIP-&gt;SIP Session Timer T4 (2.5~60s)</p> <p><b>Phone User Interface:</b> None</p>		

**To configure session timer via web user interface:**

1. Click on **Settings->SIP**.
2. Enter the desired value in the **SIP Session Timer T1 (0.5~10s)** field.
3. Enter the desired value in the **SIP Session Timer T2 (2~40s)** field.

4. Enter the desired value in the **SIP Session Timer T4 (2.5~60s)** field.

The screenshot shows the Yealink T46G web interface for SIP configuration. The 'SIP Config' section is highlighted with a red box. It contains the following fields:

- SIP Session Timer T1 (0.5~10s): 0.5
- SIP Session Timer T2 (2~40s): 4
- SIP Session Timer T4 (2.5~60s): 5
- Local SIP Port: 5060
- TLS SIP Port: 5061

Buttons for 'Confirm' and 'Cancel' are at the bottom of the config section. A 'NOTE' section on the right provides details about the timers:

**NOTE**

**SIP Session Timers**  
SIP session timers T1, T2 and T4 are SIP transaction layer timers defined in RFC 3261.

Timer T1 is an estimate of the Round Trip Time (RTT) of transactions between a SIP client and SIP server.

Timer T2 represents the maximum retransmitting time of any SIP request message.

Timer T4 represents the time the network will take to clear messages between the SIP client and server.

You can click here to get more guides.

5. Click **Confirm** to accept the change.

## Session Timer

Session timer allows a periodic refresh of SIP sessions through an UPDATE request, to determine whether a SIP session is still active. Session timer is specified in [RFC 4028](#). IP phones support two refresher modes: UAC and UAS. Whether the endpoint functions as a UAC or a UAS depends on the UA that initiates the SIP request. If the initiator is configured as UAC, the other client or the SIP server will function as a UAS. If the initiator is configured as UAS, the other client or the SIP server will function as a UAC. The session expiration is negotiated via the Session-Expires header in the INVITE message. The negotiated refresher is always the UAC and it will send an UPDATE request at the negotiated session expiration. The value "refresher=uac" included in the UPDATE message means that the UAC performs the refresh.

Example of UPDATE message (UAC mode):

```
UPDATE sip:1058@10.10.20.34:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.20.32:5060;branch=z9hG4bK2104991394
From: "10111" <sip:10111@10.2.1.48:5060>;tag=2170397024
To: <sip:1058@10.2.1.48:5060>;tag=200382096
Call-ID: 4_1556494084@10.10.20.32
CSeq: 2 UPDATE
Contact: <sip:10111@10.10.20.32:5060>
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Session-Expires: 90;refresher=uac
Supported: timer
Content-Length: 0
```

## Procedure

Session timer can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure session timer. <b>Parameters:</b> account.X.session_timer.enable account.X.session_timer.expires account.X.session_timer.refresher
<b>Web User Interface</b>		Configure session timer. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>account.X.session_timer.enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the session timer for a specific account. <b>0</b> -Disabled <b>1</b> -Enabled, IP phone will send periodic UPDATE requests to refresh the session during a call. <b>Web User Interface:</b> Account->Advanced->Session Timer <b>Phone User Interface:</b> None		
<b>account.X.session_timer.expires<sup>[1]</sup></b>	<b>Integer from 30 to 7200</b>	<b>1800</b>
<b>Description:</b> Configures the interval (in seconds) for refreshing the SIP session during a call for a specific account. For example, an UPDATE will be sent after 50% of its value has elapsed. If it is set to 1800 (1800s), the IP phone will refresh the session during a call before 900 seconds. <b>Example:</b> account.1.session_timer.expires = 1800 <b>Note:</b> It works only if the value of the parameter "account.X.session_timer.enable" is set to 1 (Enabled).		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Account->Advanced->Session Expires(30~7200s) <b>Phone User Interface:</b> None		
<b>account.X.session_timer.refresher<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Configures the function of the endpoint who initiates the SIP request for a specific account. <b>0</b> -UAC <b>1</b> -UAS <b>Note:</b> It works only if the value of the parameter "account.X.session_timer.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Account->Advanced->Session Refresher <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure session timer via web user interface:**

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Session Timer**.
4. Enter the desired time interval in the **Session Expires(30~7200s)** field.



5. Select the desired refresher from the pull-down list of **Session Refresher**.

The screenshot shows the Yealink T46G web interface with the 'Account' tab selected. The 'Session Refresher' is set to 'UAC'. A red box highlights the 'Session Timer', 'Session Expires(30~7200s)', and 'Session Refresher' settings.

Account	Account1
Keep Alive Type	Default
Keep Alive Interval(Seconds)	30
RPort	Disabled
Subscribe Period(Seconds)	1800
DTMF Type	RFC2833
DTMF Info Type	DTMF-Relay
DTMF Payload Type(96~127)	101
Retransmission	Disabled
Subscribe Register	Disabled
Subscribe for MWI	Disabled
MWI Subscription Period(Seconds)	3600
Subscribe MWI To Voice Mail	Disabled
Voice Mail	
Voice Mail Display	Enabled
Caller ID Source	FROM
Session Timer	Enabled
Session Expires(30~7200s)	1800
Session Refresher	UAC

**NOTE**

**DTMF**  
It is the signal sent from the IP phone to the network, which is generated when pressing the IP phone's keypad during a call.

**Session Timer**  
It allows a periodic refresh of SIP sessions through a re-INVITE request, to determine whether a SIP session is still active.

**Busy Lamp Field/BLF List**  
Monitors a specific extension/a list of extensions for status changes on IP phones.

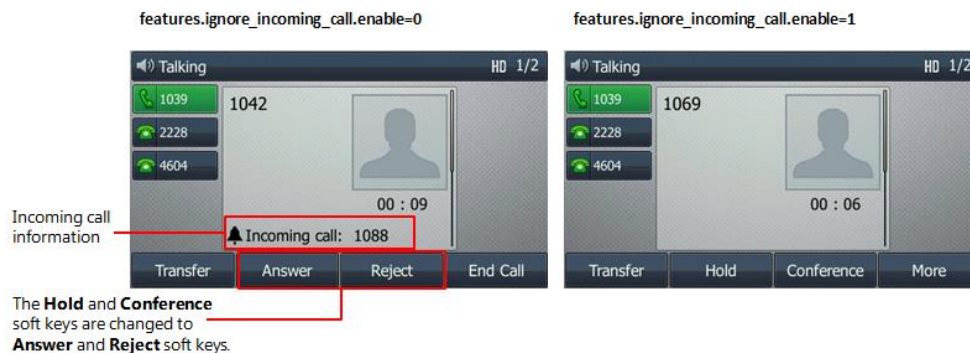
**Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA)**  
It allows users to share a SIP line on several IP phones. Any IP phone can be used to originate or receive calls on the shared line.

**Network Conference**  
It allows multiple participants (more than three) to join in a call.

6. Click **Confirm** to accept the change.

## Ignoring Incoming Calls

The phone behavior for the incoming call arrived when the IP phone is in a call is depended on the parameter "features.ignore\_incoming\_call.enable".



When you set the value of the parameter "features.ignore\_incoming\_call.enable" to 1, and if the IP phone receives an incoming call in the idle state, the user can only press the corresponding line key, the **Answer** soft key or the OK key to answer the call. And the IP phone will enter the dialing screen if the user picking up the handset, pressing the Speakerphone key or pressing the HEADSET key.

## Procedure

Ignore Incoming Call can be configured using the configuration file.

<b>Central Provisioning (Configuration Files)</b>	<y0000000000xx>.cfg <MAC>.cfg	Configure the Ignore Incoming Call. <b>Parameter:</b> features.ignore_incoming_call.enable
-------------------------------------------------------	----------------------------------	--------------------------------------------------------------------------------------------------

## Details of Configuration Parameter:

Parameter	Permitted Values	Default
<b>features.ignore_incoming_call.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the Ignore Incoming Call feature. <b>0</b> -Disabled <b>1</b> -Enabled, the users cannot answer the incoming call during the call. The IP phone will not prompt the incoming call information until the current call is end. And if there is an incoming call arrives when the IP phone is idle, the user can only press the corresponding line key, the <b>Answer</b> soft key or the OK key to answer the call. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Call Hold

Call hold provides a service of placing an active call on hold. The purpose of call hold is to pause activity on the existing call so that you can use the phone for another task (for example, to place or receive another call).

When a call is placed on hold, the IP phones send an INVITE request with HOLD SDP to request remote parties to stop sending media and to inform them that they are being held. IP phones support two call hold methods, one is [RFC 3264](#), which sets the "a" (media attribute) in the SDP to sendonly, recvonly or inactive (for example, a=sendonly). The other is [RFC 2543](#), which sets the "c" (connection addresses for the media streams) in the SDP to zero (for example, c=0.0.0.0).

When you place the call on hold, the IP phone is in the call hold state, and the other phone is in the call held state.

Call Hold Tone/Call Held Tone feature allows IP phone to play a call hold/call held tone at specified intervals when there is a call on hold/held. The call hold/call held tone is played through the speakerphone.

## Procedure

Call hold can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<y0000000000xx>.cfg	Configure the Call Hold Tone. <b>Parameters:</b> features.play_hold_tone.enable features.play_hold_tone.delay features.play_hold_tone.interval
		Configure the Call Held Tone. <b>Parameters:</b> features.play_held_tone.enable features.play_held_tone.delay features.play_held_tone.interval
		Specify whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used. <b>Parameter:</b> sip.rfc2543_hold
	<MAC>.cfg	Specify whether to use inactive hold signaling. <b>Parameter:</b> account.X.hold_use_inactive
<b>Web User Interface</b>		Configure the call hold tone and call hold tone delay.  Specify whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used.  <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load</a>

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
features.play_hold_tone.enable	0 or 1	1
<b>Description:</b> Enables or disables the IP phone to play the call hold tone when you place a call on hold.		

Parameters	Permitted Values	Default
<b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Features->General Information->Play Hold Tone <b>Phone User Interface:</b> None		
<b>features.play_hold_tone.delay</b>	<b>Integer from 3 to 3600</b>	<b>30</b>
<b>Description:</b> Configures the time (in seconds) to wait for the IP phone to play the initial call hold tone. If it is set to 30 (30s), the IP phone will wait 30 seconds to play the initial call hold tone after you place a call on hold. <b>Note:</b> It works only if the value of the parameter "features.play_hold_tone.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Features->General Information->Play Hold Tone Delay <b>Phone User Interface:</b> None		
<b>features.play_hold_tone.interval</b>	<b>Integer from 3 to 3600</b>	<b>30</b>
<b>Description:</b> Configures the time (in seconds) between subsequent call hold tones. If it is set to 3 (3s) and the value of the parameter "features.play_hold_tone.delay" is set to 30 (30s), the IP phone will begin to play a hold tone after you place a call on hold for 30 seconds, and repeat the call hold tone every 3 seconds. <b>Note:</b> It works only if the value of the parameter "features.play_hold_tone.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Features->General Information->Hold Tone Interval(second) <b>Phone User Interface:</b> None		
<b>features.play_held_tone.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to play the call held tone when a call is held by the other		

Parameters	Permitted Values	Default
party. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Features->General Information->Play Held Tone <b>Phone User Interface:</b> None		
<b>features.play_held_tone.delay</b>	<b>Integer from 3 to 3600</b>	<b>30</b>
<b>Description:</b> Configures the time (in seconds) to wait for the IP phone to play the initial call held tone. If it is set to 30 (30s), the IP phone will wait 30 seconds to play the initial call held tone after you are held by the other party. <b>Note:</b> It works only if the Music on Hold feature is disabled and the value of the parameter "features.play_held_tone.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Features->General Information->Play Held Tone Delay <b>Phone User Interface:</b> None		
<b>features.play_held_tone.interval</b>	<b>Integer from 3 to 3600</b>	<b>60</b>
<b>Description:</b> Configures the time (in seconds) between subsequent call held tones. If it is set to 3 (3s) and the value of the parameter "features.play_held_tone.delay" is set to 30 (30s), the IP phone will begin to play a held tone after a call is held by the other party for 30 seconds, and repeat the call held tone every 3 seconds. <b>Note:</b> It works only if the Music on Hold feature is disabled and the value of the parameter "features.play_held_tone.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Features->General Information->Held Tone Interval(second) <b>Phone User Interface:</b> None		
<b>sip.rfc2543_hold</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b>		

Parameters	Permitted Values	Default
<p>Enables or disables the IP phone to use RFC 2543 (c=0.0.0.0) outgoing hold signaling.</p> <p><b>0</b>-Disabled, SDP media direction attributes (such as a=sendonly) per RFC 3264 is used when placing a call on hold.</p> <p><b>1</b>-Enabled, SDP media connection address c=0.0.0.0 per RFC 2543 is used when placing a call on hold.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;RFC 2543 Hold</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.hold_use_inactive<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to use inactive outgoing hold signaling.</p> <p><b>0</b>-Disabled, SDP media direction attribute "a=sendonly" is used when placing a call on hold.</p> <p><b>1</b>-Enabled, SDP media direction attribute "a=inactive" is used when placing a call on hold. RTP packets will not be sent or received.</p> <p><b>Note:</b> It works only if the value of the parameter "sip.rfc2543_hold" is set to 0 (Disabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure call hold tone and call hold tone delay via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Play Hold Tone**.

- Enter the desired time in the **Play Hold Tone Delay** field.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' section is expanded. The 'Play Hold Tone' is set to 'Enabled' and 'Play Hold Tone Delay' is set to '30'. A red box highlights these two fields. The interface includes a sidebar with navigation options like Forward&DND, General Information, Audio, Intercom, Transfer, Pick up & Park, Remote Control, Phone Lock, ACD, SMS, Action URL, Power LED, and Notification Popups. The top navigation bar includes Status, Account, Network, Dsskey, Features, Settings, Directory, and Security. A 'NOTE' section on the right provides details about Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

- Click **Confirm** to accept the change.

**To configure call hold method via web user interface:**

- Click on **Features->General Information**.
- Select the desired value from the pull-down list of **RFC 2543 Hold**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' section is expanded. The 'RFC 2543 Hold' is set to 'Enabled'. A red box highlights this field. The interface includes a sidebar with navigation options like Forward&DND, General Information, Audio, Intercom, Transfer, Call Pickup, Remote Control, Phone Lock, ACD, SMS, Action URL, Power LED, and Notification Popups. The top navigation bar includes Status, Account, Network, Dsskey, Features, Settings, Directory, and Security. A 'NOTE' section on the right provides details about Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

- Click **Confirm** to accept the change.

## Music on Hold (MoH)

Music on Hold (MoH) is the business practice of playing recorded music to fill the silence that would be heard by the party who has been placed on hold. To use this feature, specify a SIP URI pointing to a MoH server account. When a call is placed on hold, the IP phone will send an INVITE message to the specified MoH server account according to the SIP URI. The MoH server

account automatically responds to the INVITE message and immediately plays audio from some source located anywhere (LAN, Internet) to the held party. For more information, refer to draft RFC [draft-worley-service-example](#).

**Note**

Music on Hold is not available on all servers. It is no need to specify the SIP URI if the MoH feature is enabled by default on the server and the server can play audio to the held party. For more information, contact your server administrator.

**Procedure**

Music on hold can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure music on hold on a per-line basis. <b>Parameter:</b> account.X.music_server_uri
		Configure the way on how the IP phone processes music on hold when placing an active call on hold. <b>Parameter:</b> account.X.music_on_hold_type
<b>Web User Interface</b>		Configure music on hold on a per-line basis. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>account.X.music_server_uri<sup>[1]</sup></b>	<b>SIP URI within 256 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the SIP URI that provides the audio to play for the held party for a specific account. The URI points to a Music On Hold (MoH) server.  Examples for valid values: <10.1.3.165>, 10.1.3.165, sip:moh@sip.com, <sip:moh@sip.com>, <yealink.com> or yealink.com.  <b>Example:</b> account.1.music_server_uri = sip:moh@sip.com  <b>Note:</b> The DNS query in this parameter only supports A query.		



Parameters	Permitted Values	Default
<b>Web User Interface:</b> Account->Advanced->Music Server URI <b>Phone User Interface:</b> None		
<b>account.X.music_on_hold_type<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Configures the way to process Music On Hold when placing an active call on hold for a specific account. <b>0</b> -Calling the Music On Hold server before holding the call <b>1</b> -Calling the Music On Hold server after holding the call <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure MoH via web user interface:**

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.

- Enter the SIP URI (for example, sip:moh@sip.com) in the **Music Server URI** field.

The screenshot shows the Yealink T46G web interface with the 'Account' tab selected. The 'Music Server URI' field is highlighted with a red box and contains the value 'sip:moh@sip.com'. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'Early Media' (Disabled), 'SIP Server Type' (Default), 'Directed Call Pickup Code' (\*97), 'Group Call Pickup Code' (\*98), 'Distinctive Ring Tones' (Enabled), 'Unregister When Reboot' (Enabled), 'Out Dialog BLF' (Enabled), 'VQ RTPC-XR Collector Name' (collector), 'VQ RTPC-XR Collector Address' (10.2.1.98), and 'VQ RTPC-XR Collector Port' (5060). A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

- Click **Confirm** to accept the change.

## Call Forward

Call forward allows users to redirect an incoming call to a third party. IP phones redirect an incoming INVITE message by responding with a 302 Moved Temporarily message, which contains a Contact header with a new URI that should be tried. Three types of call forward:

- Always Forward** -- Forward the incoming call immediately.
- Busy Forward** -- Forward the incoming call when the IP phone or the specified account is busy.
- No Answer Forward** -- Forward the incoming call after a period of ring time.

Call forward can be configured on a phone or a per-line basis depending on the call forward mode.

There are two call forward modes:

- Phone (default):** Call forward feature is effective for the phone system. You can apply call forward to all line registrations on the phone.
- Custom:** Call forward feature can be configured for each or all accounts. You can specify individual or all line registrations on a phone that has multiply registered lines.

The call forward feature is enabled by default, and you have the option of disabling it. Users have no permission to configure call forward on the phone if call forward feature is disabled. The server-side Call Forward settings deactivate the local Call Forward settings. If the server-side Call Forward feature is activated on any of the IP phone's registrations, the other registrations

are not affected. DND activated on the IP phone deactivates the local No Answer Forward settings.

You can configure a call forward DSS key with an assigned value. When you press the call forward DSS key, the IP phone will activate the Always Forward feature and forward all incoming calls to the number specified in the value field. For more information, refer to [Yealink phone-specific user guide](#).

The call forward on code and call forward off code configured on IP phones are used to activate/deactivate the server-side Call Forward feature. They may vary on different servers.

## Diversion/History-Info

IP phones support the redirected call information sent by the SIP server with Diversion header, per draft-levy-sip-diversion-08, or History-info header, per [RFC 4244](#). The Diversion/History-info header is used to inform the IP phone of a call's history. For example, when a phone has been set to enable call forward, the Diversion/History-info header allows the receiving phone to indicate who the call was from, and from which phone number it was forwarded.

## Forward International

Forward international allows users to forward an incoming call to an international telephone number (the prefix is 00). This feature is enabled by default.

## Forward Emergency

Forward emergency allows the incoming calls from some authorized numbers not to be forwarded when the call forward feature is enabled. The incoming call will not be logged in the Forwarded Calls list. This feature is disabled by default.

## Call Forward Feature Synchronization

Call Forward Feature Synchronization feature provides the capability to synchronize the status of the Call Forward features between the IP phone and the server.

If the call forward is activated in phone mode, the forward status changes on local will be synchronized to all registered accounts on server; but if the forward status of specific account is changed on server, the forward status on local will be changed.

## Procedure

Call forward can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<MAC>.cfg	Configure call forward in custom mode.  <b>Parameters:</b> account.X.always_fwd.enable account.X.always_fwd.target account.X.always_fwd.on_code
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		account.X.always_fwd.off_code account.X.busy_fwd.enable account.X.busy_fwd.target account.X.busy_fwd.on_code account.X.busy_fwd.off_code account.X.timeout_fwd.enable account.X.timeout_fwd.target account.X.timeout_fwd.timeout account.X.timeout_fwd.on_code account.X.timeout_fwd.off_code
		Configure Call Forward Feature Synchronization on a per-line basis.  <b>Parameter:</b> account.X.features.forward.feature_key_sync.local_processing.enable
	<y0000000000xx>.cfg	Enable or disable the call forward feature.  <b>Parameter:</b> features.fwd.allow
		Specify the authorized numbers when call forward is activated.  <b>Parameters:</b> features.forward.emergency.enable features.forward.emergency.authorized_number
		Configure the call forward mode.  <b>Parameter:</b> features.fwd_mode

		<p>Configure call forward in phone mode.</p> <p><b>Parameters:</b></p> <p>forward.always.enable</p> <p>forward.always.target</p> <p>forward.always.on_code</p> <p>forward.always.off_code</p> <p>forward.busy.enable</p> <p>forward.busy.target</p> <p>forward.busy.on_code</p> <p>forward.busy.off_code</p> <p>forward.no_answer.enable</p> <p>forward.no_answer.target</p> <p>forward.no_answer.timeout</p> <p>forward.no_answer.on_code</p> <p>forward.no_answer.off_code</p>
		<p>Configure diversion/history-info feature.</p> <p><b>Parameter:</b></p> <p>features.fwd_diversion_enable</p>
		<p>Configure forward international.</p> <p><b>Parameter:</b></p> <p>forward.international.enable</p>
		<p>Configure Call Forward Feature Synchronization.</p> <p><b>Parameters:</b></p> <p>features.feature_key_sync.enable</p> <p>features.forward.feature_key_sync.enable</p> <p>features.forward.feature_key_sync.local_processing.enable</p>
<b>Web User Interface</b>		<p>Specify the authorized numbers when call forward is activated.</p> <p>Configure call forward.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?parameters=features-forward&amp;q=load</p>

	Configure diversion/history-info feature. Configure forward international. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?parameters=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?parameters=features-general&amp;q=load</a>
<b>Phone User Interface</b>	Configure call forward. Configure forward international.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>features.fwd.allow</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the call forward feature. <b>0</b> -Disabled, call forward feature is not available to the users. <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.feature_key_sync.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to send a SUBSCRIBE message with event as-feature-event. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.forward.feature_key_sync.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the forward feature synchronization.		

Parameters	Permitted Values	Default
<b>0</b> -Disabled <b>1</b> -Enabled, a user changes the forward status on server, the server notifies the phone of synchronizing the status. Conversely, if the user changes forward status on the phone, the IP phone notifies the server of synchronizing the status. <b>Note:</b> It works only if the value of the parameter "features.feature_key_sync.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.forward.feature_key_sync.local_processing.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the local forward when forward is activated. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameters "features.feature_key_sync.enable" and "features.forward.feature_key_sync.enable" are set to 1 (Enabled). The value configured by the parameter "account.X.features.forward.feature_key_sync.local_processing.enable" takes precedence over that configured by this parameter. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>account.X.features.forward.feature_key_sync.local_processing.enable</b>	<b>0 or 1</b>	<b>Blank</b>
<b>Description:</b> Enables or disables the local forward for a specific account when forward is activated. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameters "features.feature_key_sync.enable" and "features.forward.feature_key_sync.enable" are set to 1 (Enabled). It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>features.fwd_mode</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the call forward mode for the phone system.</p> <p><b>0</b>-Phone, call forward feature is effective for the phone system.</p> <p><b>1</b>-Custom, you can configure call forward feature for each or all accounts.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;Mode</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.always_fwd.enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Triggers always forward feature to on or off for a specific account.</p> <p><b>0</b>-Off</p> <p><b>1</b>-On, incoming calls to the account X are forwarded to the destination number (configured by the parameter "account.X.always_fwd.target") immediately.</p> <p><b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;AccountX-&gt;Always Forward-&gt;On/Off</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Forward-&gt;AccountX-&gt;Always Forward-&gt;Always Forward</p>		
<b>account.X.always_fwd.target<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the destination number of the always forward for a specific account.</p> <p><b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Example:</b></p> <p>account.1.always_fwd.target = 1003</p>		



Parameters	Permitted Values	Default
<b>Web User Interface:</b> Features->Forward&DND->Forward->AccountX->Always Forward->Target  <b>Phone User Interface:</b> Menu->Features->Call Forward->AccountX->Always Forward->Forward to		
<b>account.X.always_fwd.on_code<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the always forward on code to activate the server-side always forward feature for a specific account.  The IP phone will send the always forward on code and the pre-configured destination number (configured by the parameter "account.X.always_fwd.target") to the server when you activate always forward feature for a specific account on the IP phone.  <b>Example:</b> account.1.always_fwd.on_code = *72  <b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones.  <b>Web User Interface:</b> Features->Forward&DND->Forward->AccountX->Always Forward->On Code  <b>Phone User Interface:</b> Menu->Features->Call Forward->AccountX->Always Forward->On Code		
<b>account.X.always_fwd.off_code<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the always forward off code to deactivate the server-side always forward feature for a specific account.  The IP phone will send the always forward off code to the server when you deactivate always forward feature for a specific account on the IP phone.  <b>Example:</b> account.1.always_fwd.off_code = *73  <b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones.  <b>Web User Interface:</b> Features->Forward&DND->Forward->AccountX->Always Forward->Off Code		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> Menu->Features->Call Forward->AccountX->Always Forward->Off Code		
<b>account.X.busy_fwd.enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Triggers busy forward feature to on or off for a specific account. <b>0</b> -Off <b>1</b> -On, incoming calls to the account X are forwarded to the destination number (configured by the parameter "account.X.busy_fwd.target") when the callee is busy. <b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> Features->Forward&DND->Forward->AccountX->Busy Forward->On/Off <b>Phone User Interface:</b> Menu->Features->Call Forward->AccountX->Busy Forward->Busy Forward		
<b>account.X.busy_fwd.target<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the destination number of the busy forward for a specific account. <b>Example:</b> account.1.busy_fwd.target = 3602 <b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> Features->Forward&DND->Forward->AccountX->Busy Forward->Target <b>Phone User Interface:</b> Menu->Features->Call Forward->AccountX->Busy Forward->Forward to		
<b>account.X.busy_fwd.on_code<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the busy forward on code to activate the server-side busy forward feature for a specific account. The IP phone will send the busy forward on code and the pre-configured destination		

Parameters	Permitted Values	Default
<p>number (configured by the parameter "account.X.busy_fwd.target") to the server when you activate busy forward feature for a specific account on the IP phone.</p> <p><b>Example:</b></p> <p>account.1.busy_fwd.on_code = *74</p> <p><b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;AccountX-&gt;Busy Forward-&gt;On Code</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Forward-&gt;AccountX-&gt;Busy Forward-&gt;On Code</p>		
<b>account.X.busy_fwd.off_code<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the busy forward off code to deactivate the server-side busy forward feature for a specific account.</p> <p>The IP phone will send the busy forward off code to the server when you deactivate busy forward feature for a specific account on the IP phone.</p> <p><b>Example:</b></p> <p>account.1.busy_fwd.off_code = *75</p> <p><b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;AccountX-&gt;Busy Forward-&gt;Off Code</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Forward-&gt;AccountX-&gt;Busy Forward-&gt;Off Code</p>		
<b>account.X.timeout_fwd.enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Triggers no answer forward feature to on or off for a specific account.</p> <p><b>0</b>-Off</p> <p><b>1</b>-On, incoming calls to the account X are forwarded to the destination number (configured by the parameter "account.X.timeout_fwd.target") after a period of ring time.</p> <p><b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to</p>		

Parameters	Permitted Values	Default
SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> Features->Forward&DND->Forward->AccountX->No Answer Forward->On/Off <b>Phone User Interface:</b> Menu->Features->Call Forward->AccountX->No Answer Forward->No Answer Forward		
<b>account.X.timeout_fwd.target<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the destination number of the no answer forward for a specific account. <b>Example:</b> account.1.timeout_fwd.target = 3603 <b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> Features->Forward&DND->Forward->AccountX->No Answer Forward->Target <b>Phone User Interface:</b> Menu->Features->Call Forward->AccountX->No Answer Forward->Forward to		
<b>account.X.timeout_fwd.timeout<sup>[1]</sup></b>	<b>Integer from 0 to 20</b>	<b>2</b>
<b>Description:</b> Configures ring times (N) to wait before forwarding incoming calls for a specific account. Incoming calls will be forwarded when not answered after N*6 seconds. <b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> Features->Forward&DND->Forward->AccountX->No Answer Forward->After Ring Time(0~120s) <b>Phone User Interface:</b> Menu->Features->Call Forward->AccountX->No Answer Forward->After Ring Time		
<b>account.X.timeout_fwd.on_code<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the no answer forward on code to activate the server-side no answer forward feature for a specific account.</p> <p>The IP phone will send the no answer forward on code and the pre-configured destination number (configured by the parameter "account.X.timeout_fwd.target") to the server when you activate no answer forward feature for a specific account on the IP phone.</p> <p><b>Example:</b></p> <p>account.1.timeout_fwd.on_code = *76</p> <p><b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;AccountX-&gt;No Answer Forward-&gt;On Code</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Forward-&gt;AccountX-&gt;No Answer Forward-&gt;On Code</p>		
<b>account.X.timeout_fwd.off_code<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the no answer forward off code to deactivate the server-side no answer forward feature for a specific account.</p> <p>The IP phone will send the no answer forward off code to the server when you deactivate no answer forward feature for a specific account on the IP phone.</p> <p><b>Example:</b></p> <p>account.1.timeout_fwd.off_code = *77</p> <p><b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 1 (Custom). It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;AccountX-&gt;No Answer Forward -&gt;Off Code</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Forward-&gt;AccountX-&gt;No Answer Forward-&gt;Off Code</p>		
<b>features.forward.emergency.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to receive the call from authorized numbers when the call forward feature is activated.</p>		

Parameters	Permitted Values	Default
<p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, the call from authorized numbers will not be forwarded when the call forward feature is activated.</p> <p><b>Note:</b> For Yealink IP phones, it works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;Forward Emergency</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.forward.emergency.authorized_number</b>	<b>String within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the authorized numbers.</p> <p>The call from the authorized numbers cannot be forwarded even if the call forward feature is activated.</p> <p>Multiple numbers are separated by commas.</p> <p><b>Example:</b></p> <p>features.forward.emergency.authorized_number = 1014,1015</p> <p><b>Note:</b> It works only if the values of the parameters "features.fwd.allow" and "features.forward.emergency.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;Forward Authorized Numbers</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>forward.always.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Triggers the always forward feature to on or off for the phone system.</p> <p><b>0</b>-Off</p> <p><b>1</b>-On, incoming calls are forwarded to the destination number (configured by the parameter "forward.always.target") immediately.</p> <p><b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;Always Forward-&gt;On/Off</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> Menu->Features->Call Forward->Always Forward->Always Forward		
<b>forward.always.target</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the destination number of the always forward for the phone system. <b>Example:</b> forward.always.target = 4608 <b>Note:</b> It works only if the values of parameters "features.fwd.allow" and "forward.always.enable" are set to 1 and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones). <b>Web User Interface:</b> Features->Forward&DND->Forward->Always Forward->Target <b>Phone User Interface:</b> Menu->Features->Call Forward->Always Forward->Forward to		
<b>forward.always.on_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the always forward on code to activate the server-side always forward feature. The IP phone will send the always forward on code and the pre-configured destination number (configured by the parameter "forward.always.target") to the server when you activate always forward feature on the phone system. <b>Example:</b> forward.always.on_code = *72 <b>Note:</b> It works only if the values of parameters "features.fwd.allow" and "forward.always.enable" are set to 1 and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones). <b>Web User Interface:</b> Features->Forward&DND->Forward->Always Forward->On Code <b>Phone User Interface:</b> Menu->Features->Call Forward->Always Forward->On Code		
<b>forward.always.off_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b>		

Parameters	Permitted Values	Default
<p>Configures the always forward off code to deactivate the server-side always forward feature. The IP phone will send the always forward off code to the server when you deactivate always forward feature on the phone system.</p> <p><b>Example:</b></p> <p>forward.always.off_code = *73</p> <p><b>Note:</b> It works only if the values of parameters "features.fwd.allow" and "forward.always.enable" are set to 1 and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;Always Forward-&gt;Off Code</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Forward-&gt;Always Forward-&gt;Off Code</p>		
<b>forward.busy.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Triggers the busy forward feature to on or off for the phone system.</p> <p><b>0</b>-Off</p> <p><b>1</b>-On, incoming calls are forwarded to the destination number (configured by the parameter "forward.busy.target") when the callee is busy.</p> <p><b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;Busy Forward-&gt;On/Off</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Forward-&gt;Busy Forward-&gt;Busy Forward</p>		
<b>forward.busy.target</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the destination number of the busy forward for the phone system.</p> <p><b>Example:</b></p> <p>forward.busy.target = 3602</p> <p><b>Note:</b> It works only if the values of parameters "features.fwd.allow" and "forward.busy.enable" are set to 1 and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones).</p> <p><b>Web User Interface:</b></p>		



Parameters	Permitted Values	Default
Features->Forward&DND->Forward->Busy Forward->Target <b>Phone User Interface:</b> Menu->Features->Call Forward-> Busy Forward->Forward to		
<b>forward.busy.on_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the busy forward on code to activate the server-side busy forward feature. The IP phone will send the busy forward on code and the pre-configured destination number (configured by the parameter "forward.busy.target") to the server when you activate busy forward feature on the phone system. <b>Example:</b> forward.busy.on_code = *74 <b>Note:</b> It works only if the values of parameters "features.fwd.allow" and "forward.busy.enable" are set to 1 and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones). <b>Web User Interface:</b> Features->Forward&DND->Forward->Busy Forward->On Code <b>Phone User Interface:</b> Menu->Features->Call Forward->Busy Forward->On Code		
<b>forward.busy.off_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the busy forward off code to deactivate the server-side busy forward feature. The IP phone will send the busy forward off code to the server when you deactivate busy forward feature on the phone system. <b>Example:</b> forward.busy.off_code = *75 <b>Note:</b> It works only if the values of parameters "features.fwd.allow" and "forward.busy.enable" are set to 1 and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones). <b>Web User Interface:</b> Features->Forward&DND->Forward->Busy Forward->Off Code <b>Phone User Interface:</b> Menu->Features->Call Forward->Busy Forward->Off Code		

Parameters	Permitted Values	Default
<b>forward.no_answer.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Triggers the no answer forward feature to on or off for the phone system.</p> <p><b>0</b>-Off</p> <p><b>1</b>-On, incoming calls are forwarded to the destination number (configured by the parameter "forward.no_answer.target") after a period of ring time.</p> <p><b>Note:</b> It works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;No Answer Forward-&gt;On/Off</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Forward-&gt;No Answer Forward-&gt;No Answer Forward</p>		
<b>forward.no_answer.target</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the destination number of the no answer forward for the phone system.</p> <p><b>Example:</b></p> <p>forward.no_answer.target = 3603</p> <p><b>Note:</b> It works only if the values of parameters "features.fwd.allow" and "forward.no_answer.enable" are set to 1 and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Forward&amp;DND-&gt;Forward-&gt;No Answer Forward-&gt;Target</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;Call Forward-&gt;No Answer Forward-&gt;Forward to</p>		
<b>forward.no_answer.timeout</b>	<b>Integer from 0 to 20</b>	<b>2</b>
<p><b>Description:</b></p> <p>Configures ring times (N) to wait before forwarding incoming calls.</p> <p>Incoming calls will be forwarded when not answered after N*6 seconds.</p> <p><b>Note:</b> It works only if the values of parameters "features.fwd.allow" and "forward.no_answer.enable" are set to 1 and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones).</p>		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Features->Forward&DND->Forward->No Answer Forward->After Ring Time (0~120s) <b>Phone User Interface:</b> Menu->Features->Call Forward->No Answer Forward->After Ring Time		
<b>forward.no_answer.on_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the no answer forward on code to activate the server-side no answer forward feature. The IP phone will send the no answer forward on code and the pre-configured destination number (configured by the parameter "forward.no_answer.target") to the server when you activate no answer forward feature on the phone system. <b>Example:</b> forward.no_answer.on_code = *76 <b>Note:</b> It works only if the values of parameters "features.fwd.allow" and "forward.no_answer.enable" are set to 1 and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones). <b>Web User Interface:</b> Features->Forward&DND->Forward->No Answer Forward->On Code <b>Phone User Interface:</b> Menu->Features->Call Forward->No Answer Forward->On Code		
<b>forward.no_answer.off_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the no answer forward off code to deactivate the server-side no answer forward feature. The IP phone will send the no answer forward off code to the server when you deactivate no answer forward feature on the phone system. <b>Example:</b> forward.no_answer.off_code = *77 <b>Note:</b> It works only if the values of parameters "features.fwd.allow" and "forward.no_answer.enable" are set to 1 and value of the parameter "features.fwd_mode" is set to 0 (Phone) (except SIP-T19(P) E2 IP phones). <b>Web User Interface:</b> Features->Forward&DND->Forward->No Answer Forward->Off Code		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> Menu->Features->Call Forward->No Answer Forward->Off Code		
<b>features.fwd_diversion_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to present the diversion information when an incoming call is forwarded to your IP phone. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Features->General Information->Diversion/History-Info <b>Phone User Interface:</b> None		
<b>forward.international.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to forward incoming calls to international numbers (the prefix is 00). <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> For Yealink IP phones, it works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled). <b>Web User Interface:</b> Features->General Information->Fwd International <b>Phone User Interface:</b> Menu->Advanced (default password: admin)->FWD International->FWD International		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2.

**To specify the authorized numbers when call forward is enabled via web user interface:**

1. Click on **Features->Forward&DND**.
2. Select the desired value from the pull-down list of **Forward Emergency**.
3. Enter the desired value in the **Forward Authorized Numbers** field.

Multiple numbers are separated by commas.

The screenshot shows the 'Forward' configuration page in the Yealink T46G web interface. The 'Forward Emergency' dropdown is set to 'Enabled' and 'Forward Authorized Numbers' is set to '1014,1015'. The 'Mode' field has 'Phone' selected. A red box highlights the 'Forward Emergency' and 'Forward Authorized Numbers' fields.

4. Click **Confirm** to accept the change.

**To configure call forward via web user interface:**

1. Click on **Features->Forward&DND**.
2. In the **Forward** block, mark the desired radio box in the **Mode** field.
  - a) If you mark the **Phone** radio box:
    - 1) Mark the desired radio box in the **Always/Busy/No Answer Forward** field.
    - 2) Enter the destination number you want to forward in the **Target** field.
    - 3) (Optional.) Enter the on code and off code in the **On Code** and **Off Code** fields.
    - 4) Select the ring time to wait before forwarding from the pull-down list of **After Ring Time(0~120s)** (only for the no answer forward).

The screenshot shows the 'Forward' configuration page in the Yealink T46G web interface. The 'Mode' field has 'Phone' selected. The 'Always Forward' section is highlighted with a red box, showing 'On' selected, 'Target' set to '4608', 'On Code' set to '\*72', and 'Off Code' set to '\*73'.

- b) If you mark the **Custom** radio box:
  - 1) Select the desired account from the pull-down list of **Account**.
  - 2) Mark the desired radio box in the **Always/Busy/No Answer Forward** field.
  - 3) Enter the destination number you want to forward in the **Target** field.
  - 4) Enter the on code and off code in the **On Code** and **Off Code** fields.

- 5) Select the ring time to wait before forwarding from the pull-down list of **After Ring Time(0~120s)** (only for the no answer forward).

The screenshot shows the 'Forward' configuration page in the Yealink T466 web interface. The 'Forward' section is highlighted with a red box. It includes the following fields:

- Forward Emergency: Disabled
- Forward Authorized Numbers:
- Mode: Phone (selected), Custom
- Account: 101
- Always Forward: On (selected), Off
- Target: 102
- On Code: \*72
- Off Code: \*73
- Busy Forward: On (selected), Off

A 'NOTE' section on the right provides information about Call Forward, Call Forward Mode (Phone and Custom), Do Not Disturb (DND), and DND Mode (Phone).

3. Click **Confirm** to accept the change.

**To configure Diversion/History-Info feature via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Diversion/History-Info**.

The screenshot shows the 'General Information' configuration page in the Yealink T466 web interface. The 'Diversion/History-Info' field is highlighted with a red box and set to 'Enabled'. Other fields include:

- Call Waiting: Enabled
- Call Waiting On Code:
- Call Waiting Off Code:
- Auto Redial: Disabled
- Auto Redial Interval (1~300s): 10
- Auto Redial Times (1~300): 10
- Diversion/History-Info: Enabled
- Allow Trans Exist Call: Enabled
- BLF LED Mode: 0
- Auto-Logout Time(1~1000min): 5
- Call Number Filter: ,
- Auto Linekeys: Disabled

A 'NOTE' section on the right provides information about Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

3. Click **Confirm** to accept the change.

**To configure forward international via web user interface:**

1. Click on **Features->General Information**.

2. Select the desired value from the pull-down list of **Fwd International**.



The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected. Under 'General Information', the 'Fwd International' dropdown is highlighted with a red box and set to 'Enabled'. Other settings include 'Call Waiting' (Enabled), 'Auto Redial' (Disabled), 'Auto Redial Interval' (10), 'Auto Redial Times' (10), 'Allow Trans Exist Call' (Enabled), 'BLF LED Mode' (0), 'Auto-Logout Time' (5), 'Call Number Filter' (.), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right explains features like Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

3. Click **Confirm** to accept the change.

**To configure call forward in phone mode via phone user interface:**

1. Press **Menu->Features->Call Forward**.
2. Press  $\uparrow$  or  $\downarrow$  to select the desired forwarding type, and then press the **Enter** soft key.
3. Depending on your selection:
  - a) If you select **Always Forward**:
    - 1) Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select the desired value from the **Always Forward** field.
    - 2) Enter the destination number you want to forward all incoming calls to in the **Forward to** field.
    - 3) (Optional.) Enter the always forward on code and off code respectively in the **On Code** and **Off Code** field.
  - b) If you select **Busy Forward**:
    - 1) Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select the desired value from the **Busy Forward** field.
    - 2) Enter the destination number you want to forward all incoming calls to when the IP phone is busy in the **Forward to** field.
    - 3) (Optional.) Enter the busy forward on code and off code respectively in the **On Code** and **Off Code** field.
  - c) If you select **No Answer Forward**:
    - 1) Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select the desired value from the **No Answer Forward** field.
    - 2) Enter the destination number you want to forward all unanswered incoming calls to in the **Forward to** field.
    - 3) (Optional.) Enter the no answer forward on code and off code respectively in the





**On Code** and **Off Code** field.

- 4) Press  or , or the **Switch** soft key to select the ring time to wait before forwarding from the **After Ring Time** field.



The default ring time is 12 seconds.

4. Press the **Save** soft key to accept the change.



**To configure call forward in custom mode via phone user interface:**

1. Press **Menu->Features->Call Forward**.
2. Press  or  to select the desired account, and then press the **Enter** soft key.
3. Press  or  to select the desired forwarding type, and then press the **Enter** soft key.
4. Depending on your selection:

- a) If you select **Always Forward**, you can configure it for a specific account.



- 1) Press  or , or the **Switch** soft key to select the desired value from the **Always Forward** field.
- 2) Enter the destination number you want to forward all incoming calls to in the **Forward to** field.
- 3) (Optional.) Enter the always forward on code and off code respectively in the **On Code** and **Off Code** field.

You can also configure the always forward for all accounts. After the always forward was configured for a specific account, do the following:



- 1) Press  or  to highlight the **Always Forward** field.
- 2) Press the **All Lines** soft key.

The LCD screen prompts "Copy to all lines?".
- 3) Press the **OK** soft key to accept the change.

- b) If you select **Busy Forward**, you can configure it for a specific account.

- 1) Press  or , or the **Switch** soft key to select the desired value from the **Busy Forward** field.
- 2) Enter the destination number you want to forward all incoming calls to when the IP phone is busy in the **Forward to** field.
- 3) (Optional.) Enter the busy forward on code and off code respectively in the **On Code** and **Off Code** field.

You can also configure the busy forward for all accounts. After the busy forward was configured for a specific account, do the following:

- 1) Press  or  to highlight the **Busy Forward** field.
- 2) Press the **All Lines** soft key.



The LCD screen prompts "Copy to all lines?".
- 3) Press the **OK** soft key to accept the change.

- c) If you select **No Answer Forward**, you can configure it for a specific account.

- 1) Press  or , or the **Switch** soft key to select the desired value from the





**No Answer Forward** field.

- 2) Enter the destination number you want to forward all unanswered incoming calls to in the **Forward to** field.
- 3) Press  or  , or the **Switch** soft key to select the ring time to wait before forwarding from the **After Ring Time** field.

The default ring time is 12 seconds.



- 4) (Optional.) Enter the no answer forward on code and off code respectively in the **On Code** and **Off Code** field.

You can also configure the no answer forward for all accounts. After the no answer forward was configured for a specific account, do the following:

- 1) Press  or  to highlight the **No Answer Forward** field.
- 2) Press the **All Lines** soft key.  
The LCD screen prompts "Copy to all lines?".
- 3) Press the **OK** soft key to accept the change.

5. Press the **Save** soft key to accept the change.

**To configure forward international via phone user interface:**

1. Press **Menu->Advanced** (default password: admin) -> **FWD International**.
2. Press  or  , or the **Switch** soft key to select the desired value from the **FWD International** field.
3. Press the **Save** soft key to accept the change.

## Call Transfer

Call transfer enables IP phones to transfer an existing call to a third party. For example, if party A is in an active call with party B, party A can transfer this call to party C (the third party). Then, party B will begin a new call with party C, and party A will disconnect.

IP phones support call transfer using the REFER method specified in [RFC 3515](#) and offer three types of transfer:

- **Blind Transfer** -- Transfer a call directly to another party without consulting. Blind transfer is implemented by a simple REFER method without Replaces in the Refer-To header.
- **Semi-attended Transfer** -- Transfer a call after hearing the ringback tone. Semi-attended transfer is implemented by a REFER method with Replaces in the Refer-To header.
- **Attended Transfer (Consultative Transfer)** -- Transfer a call with prior consulting. Attended transfer is implemented by a REFER method with Replaces in the Refer-To header.

Normally, call transfer is completed by pressing the transfer key. Blind transfer on hook and attended transfer on hook features allow the IP phone to complete the transfer through on-hook.

You can configure which transfer type to perform for the IP phone when the entered transferee

numbers match the dial plan.

## Procedure

Call transfer can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	<p>Specify whether to complete the transfer through on-hook.</p> <p><b>Parameters:</b></p> <p>transfer.blind_tran_on_hook_enable</p> <p>transfer.on_hook_trans_enable</p>
		<p>Configure semi-attended transfer feature.</p> <p><b>Parameter:</b></p> <p>transfer.semi_attend_tran_enable</p>
		<p>Configures the transfer type when the entered transferee numbers match the dial plan.</p> <p><b>Parameter:</b></p> <p>dialplan.transfer.mode</p>
<b>Web User Interface</b>	<p>Specify whether to complete the transfer through on-hook.</p> <p>Configure semi-attended transfer feature.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=features-transfer&amp;q=load</p>	

## Details of Configuration Parameters:

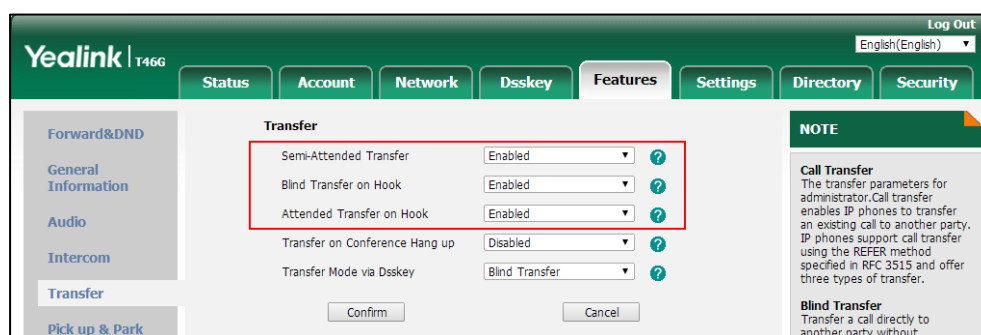
Parameters	Permitted Values	Default
<b>transfer.blind_tran_on_hook_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to complete the blind transfer through on-hook besides pressing the <b>Trans/Transfer</b> soft key or TRAN/TRANSFER key. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> Blind transfer means transfer a call directly to another party without consulting. <b>Web User Interface:</b>		

Parameters	Permitted Values	Default
Features->Transfer->Blind Transfer On Hook <b>Phone User Interface:</b> None		
<b>transfer.on_hook_trans_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to complete the semi-attended/attended transfer through on-hook besides pressing the <b>Trans/Transfer</b> soft key or TRAN/TRANSFER key. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> Semi-attended transfer means transfer a call after hearing the ringback tone; Attended transfer means transfer a call with prior consulting. <b>Web User Interface:</b> Features->Transfer->Attended Transfer On Hook <b>Phone User Interface:</b> None		
<b>transfer.semi_attend_tran_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the semi-attended transfer. <b>0</b> -Disabled. When the user presses the <b>Trans/Transfer</b> soft key after hearing the ringback tone, the IP phone will blind transfer the call. <b>1</b> -Enabled. When the user presses the <b>Trans/Transfer</b> soft key after hearing the ringback tone, the IP phone will transfer the call after the transferee answers the call. <b>Web User Interface:</b> Features->Transfer->Semi-Attended Transfer <b>Phone User Interface:</b> None		
<b>dialplan.transfer.mode</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Configures the transfer type the IP phone will perform when the entered transferee numbers match the Dial Now rule of dial plan. <b>0</b> -Semi-attended Transfer/Attended Transfer. <b>1</b> -Blind Transfer.		

Parameters	Permitted Values	Default
<p><b>Note:</b> It is only applicable to old dial plan mechanism.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure call transfer via web user interface:**

1. Click on **Features->Transfer**.
2. Select the desired values from the pull-down lists of **Semi-Attended Transfer**, **Blind Transfer on Hook** and **Attended Transfer on Hook**.



3. Click **Confirm** to accept the change.

## Conference

You can use the IP phones to set up a conference with other two parties using the phone's local conference or set up a conference with multiple participants which is called network conference.

### Procedure

Conference type can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure conference type. <b>Parameter:</b> account.X.conf_type
<b>Web User Interface</b>		Configure conference type. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =account-adv&q=load&acc=0

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>account.X.conf_type<sup>[1]</sup></b>	<b>0 or 2</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the conference type for a specific account.</p> <p><b>0</b>-Local Conference, conference is set up with other two parties via the IP phone when the value of parameter "features.conference.local.enable" is set to 1 (Enabled).</p> <p><b>2</b>-Network Conference, conference is set up with multiple parties via the server.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;Conference Type</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure the conference type via web user interface:**

1. Click on **Account**->**Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Conference Type**.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected, and the 'Advanced' sub-tab is active. The 'Account' dropdown is set to 'Account 1'. The 'Conference Type' dropdown is highlighted with a red box and set to 'Network Conference'. Other settings include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), and 'Subscribe Period(Seconds)' (1800). The 'NOTE' section on the right contains information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

4. Click **Confirm** to accept the change.

## Local Conference

Local conference requires a host phone to process the audio of all parties. Yealink IP phones support up to 3 parties (including yourself) in a local conference call.

### Note

You can configure the softkey layout feature to display the **RTP Status** soft key during the conference. Then you can press the **RTP Status** soft key to check the voice quality data with different parties.

### Procedure

Local conference can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure local conference. <b>Parameters:</b> features.conference.local.enable features.conference.with_previous_call.enable features.local_conf.combine_with_one_press.enable
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### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>features.conference.local.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the local conference for the IP phone. <b>0</b> -Disabled, the <b>Conf/Conference</b> soft key will disappear during a call and the CONF key/conference DSS key will not work. Local conference cannot be set up even though the value of the parameter "account.X.conf_type" is set to 0 (Local Conference). <b>1</b> -Enabled, you can set up a local conference with other two parties. It works only when the value of the parameter "account.X.conf_type" is set to 0 (Local Conference). <b>Note:</b> The CONF key is only applicable to SIP-T29G/T27P/T27G IP phones. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.conference.with_previous_call.enable</b>	<b>0 or 1</b>	<b>0</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Enables or disables the IP phone to merge two calls into a conference directly by pressing the <b>Conf/Conference</b> soft key, CONF key or conference DSS key when there are two calls on the phone.</p> <p><b>0</b>-Disabled, you can select to set up a conference with the held party or a new party when pressing the <b>Conf/Conference</b> soft key, CONF key or conference DSS key during multiple calls.</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> The CONF key is only applicable to SIP-T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.local_conf.combine_with_one_press.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to set up a three-way conference directly after the second party answers the call.</p> <p><b>0</b>-Disabled, the first call is placed on hold. The user needs to press the <b>Conf/Conference</b> soft key, CONF key or conference DSS key again to set up a conference after the second party answers the call.</p> <p><b>1</b>-Enabled. The second party joins a conference with the first party after answering the call, both phones play a warning tone.</p> <p><b>Note:</b> The CONF key is only applicable to SIP-T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## Network Conference

Network conference, also known as centralized conference, provides users with flexibility of call with multiple participants (more than three). IP phones implement network conference using the REFER method specified in [RFC 4579](#). This feature depends on support from a SIP server.

## Procedure

Network conference can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure network conference URI. <b>Parameter:</b> account.X.conf_uri
<b>Web User Interface</b>		Configure network conference URI. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =account-adv&q=load&acc=0

## Details of Configuration Parameter:

Parameter	Permitted Values	Default
<b>account.X.conf_uri<sup>[1]</sup></b>	<b>SIP URI within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the network conference URI for a specific account.</p> <p><b>Example:</b> account.1.conf_uri = conference@example.com</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.conf_type" is set to 2 (Network Conference).</p> <p><b>Web User Interface:</b> Account-&gt;Advanced-&gt;Conference URI</p> <p><b>Phone User Interface:</b> None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## To configure the network conference URI via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select **Network Conference** from the pull-down list of **Conference Type**.



4. Enter the conference URI in the **Conference URI** field.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected. The 'Conference Type' is set to 'Network Conference' and the 'Conference URI' is 'conference@example.com'. A red box highlights these two fields. The interface includes tabs for Status, Account, Network, Dsskey, Features, Settings, Directory, and Security. A sidebar on the left shows options for Register, Basic, Codec, and Advanced. A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

5. Click **Confirm** to accept the change.

## Transfer on Conference Hang Up

For a local three-way conference, if the conference initiator leaves the conference, all parties are disconnected and the conference call ends. Transfer on Conference Hang Up feature allows the other two parties to remain connected when the conference initiator drops the conference call. For network conference, if any party leaves the conference, the remaining parties are still connected. For more information on network conference, refer to [Network Conference](#) on page 429.

### Procedure

Transfer on conference hang up can be configured using the following methods.

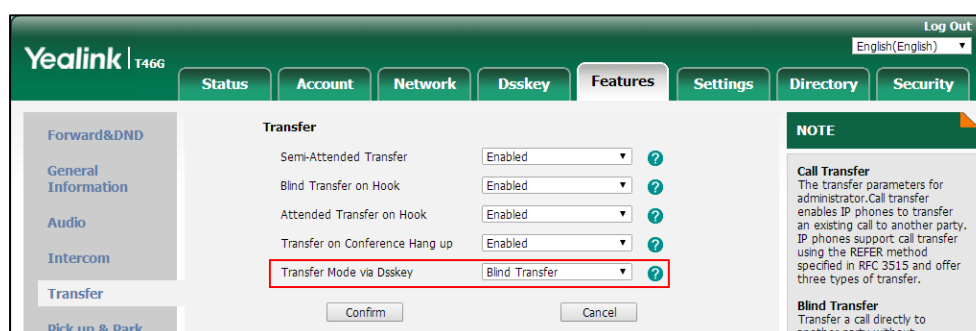
<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the transfer on conference hang up. <b>Parameter:</b> transfer.tran_others_after_conf_enable
<b>Web User Interface</b>		Configure the transfer on conference hang up. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=f eatures-transfer&q=load

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>transfer.tran_others_after_conf_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to transfer the local conference call to the other two parties after the conference initiator drops the local conference call.</p> <p><b>0</b>-Disabled, all parties are disconnected when the conference initiator drops the conference call.</p> <p><b>1</b>-Enabled, the other two parties remain connected when the conference initiator drops the conference call.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.conf_type" is set to 0 (Local Conference) and the value of the parameter "features.conference.local.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Transfer-&gt;Transfer on Conference Hang up</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure transfer on conference hang up via web user interface:**

1. Click on **Features->Transfer**.
2. Select the desired value from the pull-down list of **Transfer on Conference Hang up**.



3. Click **Confirm** to accept the change.

## Transfer Mode via Dsskey

Transfer Mode via Dsskey feature configures the transfer mode for the IP phone when transferring the current call via the specified DSS key.

IP phones support three transfer modes: New Call, Blind Transfer and Attended Transfer. For more information on Blind Transfer and Attended Transfer, refer to [Call Transfer](#) on page 423.

The transfer mode via dsskey feature is available to the following DSS keys:

- Speed Dial DSS key
- Transfer DSS key with an assigned value
- BLF/BLF List DSS key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

#### Note

It is not applicable to SIP-T19(P) E2 IP phones.

### Procedure

Transfer mode via dsskey can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the transfer mode via dsskey. <b>Parameter:</b> transfer.dsskey_deal_type
<b>Web User Interface</b>		Configure the transfer mode via dsskey. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-transfer&q=load

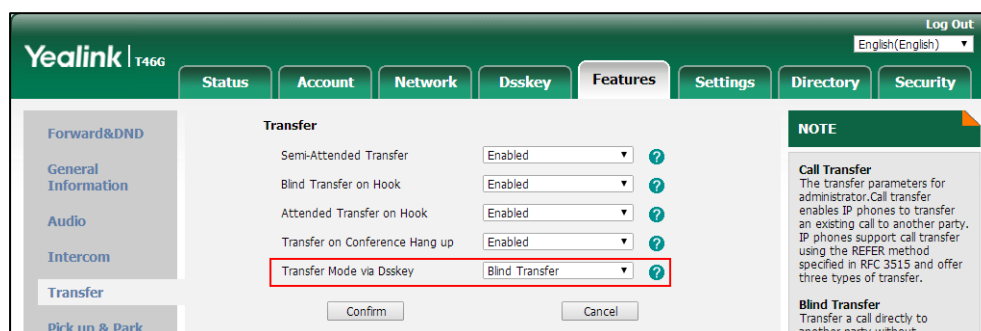
### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
transfer.dsskey_deal_type	0, 1 or 2	2
<b>Description:</b> Configures the transfer mode for DSS key. When the user presses the DSS Key during a call, the DSS Key behavior depends on the transfer mode. <b>0</b> -New Call <b>1</b> -Attended Transfer <b>2</b> -Blind Transfer <b>Note:</b> This feature is only applicable to the Speed Dial DSS key, BLF/BLF List DSS key or Transfer DSS key with an assigned value. It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> Features->Transfer->Transfer Mode via Dsskey <b>Phone User Interface:</b>		

Parameter	Permitted Values	Default
None		

To configure transfer mode via dsskey via web user interface:

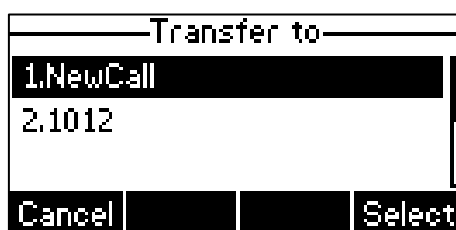
1. Click on **Features**->**Transfer**.
2. Select the desired value from the pull-down list of **Transfer Mode via Dsskey**.



3. Click **Confirm** to accept the change.

## Allow Trans Exist Call

Allow Trans Exist Call feature allows users to transfer the current call to a new call or another existing call. For example, party A (1001) has two calls on line 1, one with party B (1011) and the other with party C (1012). If party A transfers the call with party B by pressing the **Trans/Transfer** soft key, a "Transfer to" screen will display on the phone. In this case, party A can select to transfer the call to a new number or party C (1012).



It is not applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.

### Procedure

Allow trans exist call can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>. cfg	Configure allow trans exist call. <b>Parameter:</b> transfer.multi_call_trans_enable
<b>Web User Interface</b>		Configure allow trans exist call. <b>Navigate to:</b>

	http://<phoneIPAddress>/servlet?p=features-general&q=load
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**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>transfer.multi_call_trans_enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to transfer the current call to a new call or another existing call during multiple calls when user presses the <b>Trans/Transfer</b> soft key or TRAN/TRANSFER key.</p> <p><b>0</b>-Disabled, the user cannot transfer the current call to another existing call.</p> <p><b>1</b>-Enabled, the user can transfer the current call to a new call or another existing call.</p> <p><b>Note:</b> It is not applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Allow Trans Exist Call</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure allow trans exist call via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Allow Trans Exist Call**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'Allow Trans Exist Call' option is set to 'Enabled' and is highlighted with a red box. Other settings include 'Call Waiting' (Enabled), 'Auto Redial' (Disabled), 'Diversion/History-Info' (Enabled), 'BLF LED Mode' (0), 'Auto-Logout Time' (5), 'Call Number Filter' (.), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right provides additional information about various features like Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

3. Click **Confirm** to accept the change.

## Directed Call Pickup

Directed call pickup is used for picking up an incoming call on a specific extension. A user can pick up the incoming call using a directed pickup DSS key or the **DPickup** soft key. This feature depends on support from a SIP server. For many SIP servers, directed call pickup requires a directed pickup code, which can be configured on a phone or a per-line basis.

When you enable directed call pickup, the LCD screen will display a **DPickup** soft key when you pick up the handset, press the Speakerphone key or press the line key (You may need to press the **More** soft key to see the **DPickup** soft key). As shown below:



### Note

It is recommended not to configure the directed call pickup DSS key and the **DPickup** soft key simultaneously. If you do, the directed pickup DSS key will not be used correctly.

## Procedure

Directed call pickup can be configured using the following methods.

Central Provisioning (Configuration Files)	<MAC>.cfg	Configure the directed call pickup code on a per-line basis. <b>Parameter:</b> account.X.direct_pickup_code
		Configure directed call pickup features on a phone basis. <b>Parameters:</b> features.pickup.direct_pickup_enable features.pickup.direct_pickup_code
	<y0000000000xx>.cfg	Assign a directed call pickup DSS key. <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type linekey.X.line/ programablekey.X.line/

		expansion_module.X.key.Y.line linekey.X.value/ programablekey.X.value/ expansion_module.X.key.Y.value linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Assign a directed call pickup DSS key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0">http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0</a>
		Configure directed call pickup code on a per-line basis. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>
		Configure directed call pickup feature on a phone basis. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-callpickup&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-callpickup&amp;q=load</a>
<b>Phone User Interface</b>		Assign a directed call pickup DSS key.

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>account.X.direct_pickup_code<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the directed call pickup code for a specific account. <b>Example:</b> account.1.direct_pickup_code = *97 <b>Note:</b> The directed call pickup code configured on a per-line basis (configured by the parameter "account.X.direct_pickup_code") takes precedence over that configured on a phone basis (configured by the parameter "features.pickup.direct_pickup_code"). <b>Web User Interface:</b> Account->Advanced->Directed Call Pickup Code		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>features.pickup.direct_pickup_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to display the <b>DPickup</b> soft key when the IP phone is on the dialing screen. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Features->Pick up & Park->Directed Call Pickup <b>Phone User Interface:</b> None		
<b>features.pickup.direct_pickup_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the directed call pickup code on a phone basis. <b>Example:</b> features.pickup.direct_pickup_code = *97 <b>Note:</b> The directed call pickup code configured on a per-line basis (configured by the parameter "account.X.direct_pickup_code") takes precedence over that configured on a phone basis (configured by the parameter "features.pickup.direct_pickup_code"). <b>Web User Interface:</b> Features->Pick up & Park->Directed Call Pickup Code <b>Phone User Interface:-</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## Directed Call Pickup DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.



**Details of Configuration Parameters:**

Parameters	Permitted Values
<b>linekey.X.type</b> <sup>[1]</sup> <b>programmablekey.X.type</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.type</b> <sup>[3]</sup>	9
<p><b>Description:</b>  Configures a DSS key as a directed call pickup DSS key on the IP phone.</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b>  Dsskey-&gt;Line Key/Programable Key-&gt;Type</p> <p><b>Phone User Interface:</b>  Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Type</p>	
<b>linekey.X.line</b> <sup>[1]</sup> <b>programmablekey.X.line</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.line</b> <sup>[3]</sup>	Refer to the following content
<p><b>Description:</b>  Configures the desired line to apply the directed call pickup DSS key.</p> <p><b>Permitted Values:</b>  1 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G)  1 to 12 (for SIP-T52S/T42G/T42S)  1 to 6 (for SIP-T41P/T41S/T27P/T27G)  1 to 3 (for SIP-T40P/T40G/T23P/T23G)  1 to 2 (for SIP-T21(P) E2)  1-Line 1  2-Line 2  ...  16-Line 16</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Example:</b>  linekey.3.line = 1</p> <p><b>Web User Interface:</b>  Dsskey-&gt;Line Key/Programable Key-&gt;Line</p> <p><b>Phone User Interface:</b></p>	

Parameters	Permitted Values
Menu->Features->DSS Keys->Line Key X->Account ID	
<b>linekey.X.value</b> <sup>[1]</sup> <b>programablekey.X.value</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.value</b> <sup>[3]</sup>	String within 99 characters
<p><b>Description:</b></p> <p>Configures the directed call pickup feature code followed by the monitored extension.</p> <p>If you have configured the directed call pickup code on a phone basis (configured by the parameter "features.pickup.direct_pickup_code"), you can configure the monitored extension only.</p> <p><b>Example:</b></p> <p>linekey.3.value = *971008</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key/Programable Key-&gt;Value</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Value</p>	
<b>linekey.X.label</b> <sup>[1]</sup> <b>programablekey.X.label</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.label</b> <sup>[3]</sup>	String within 99 characters
<p><b>Description:</b></p> <p>(Optional.) Configures the label displayed on the LCD screen for each DSS key.</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key/Programable Key-&gt;Label</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Label</p>	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<p><b>Description:</b></p> <p>(Optional.) Configures the short label displayed on the LCD screen for line key.</p> <p><b>Note:</b> It is only applicable to SIP-T52S IP phones.</p>	

Parameters	Permitted Values
<b>Web User Interface:</b> Dsskey->Line Key->Short	
<b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programmable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure a directed call pickup DSS key via web user interface:

1. Click on **Dsskey->Line Key** (or **Programable Key**).
2. In the desired DSS key field, select **Direct Pickup** from the pull-down list of **Type**.
3. Enter the directed call pickup code followed by the specific extension in the **Value** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
5. Select the desired line from the pull-down list of **Line**.

6. Click **Confirm** to accept the change.

#### To configure the directed call pickup code on a per-line basis via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.

- Enter the directed call pickup code in the **Directed Call Pickup Code** field.

The screenshot shows the Yealink T46G web interface with the 'Account' tab selected. The 'Account' section is expanded, showing various settings. The 'Directed Call Pickup Code' field is highlighted with a red box and contains the value '\*97'. Other visible fields include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'Early Media' (Disabled), 'SIP Server Type' (Default), 'Music Server URI' (sip:moh@sip.com), 'Group Call Pickup Code' (\*98), 'Distinctive Ring Tones' (Enabled), 'Unregister When Reboot' (Enabled), 'Out Dialog BLF' (Enabled), 'VQ RTPC-XR Collector Name' (collector), 'VQ RTPC-XR Collector Address' (10.2.1.98), and 'VQ RTPC-XR Collector Port' (5060). A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

- Click **Confirm** to accept the change.

**To configure directed call pickup feature on a phone basis via web user interface:**

- Click on **Features->Pick up & Park**.
- Select the desired value from the pull-down list of **Directed Call Pickup**.
- Enter the directed call pickup code in the **Directed Call Pickup Code** field.

The screenshot shows the Yealink T46G web interface with the 'Features' tab selected. The 'Pick up & Park' section is expanded, showing various settings. The 'Directed Call Pickup' field is highlighted with a red box and contains the value 'Enabled'. Other visible fields include 'Directed Call Pickup Code' (\*97), 'Group Call Pickup' (Disabled), 'Group Call Pickup Code', 'Visual Alert for BLF Pickup' (Disabled), 'Audio Alert for BLF Pickup' (Disabled), and 'Ring Type for BLF Pickup' (Splash.wav). A 'NOTE' section on the right provides information about Directed Call Pickup and Visual Alert for BLF Pickup.

- Click **Confirm** to accept the change.

**To configure a directed pickup DSS key via phone user interface:**

- Press **Menu->Features->DSS Keys**.
- Select the desired DSS key.
- Press **◀** or **▶**, or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select **DPickup** from the **Key Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired line from the **Account ID** field.

- 6. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- 7. Enter the directed call pickup code followed by the specific extension in the **Value** field.
- 8. Press the **Save** soft key to accept the change.

Group Call Pickup

Group call pickup is used for picking up incoming calls within a pre-defined group. If the group receives many incoming calls at once, the user will pick up the first incoming call, using a group pickup DSS key or the **GPickup** soft key. This feature depends on support from a SIP server. For many SIP servers, group call pickup requires a group pickup code, which can be configured on a phone or a per-line basis.

When you enable group call pickup, the LCD screen will display a **GPickup** soft key when you pick up the handset, press the Speakerphone key or press the line key. As shown below:



Procedure

Group call pickup can be configured using the following methods.

Central Provisioning (Configuration Files)		Configure the group call pickup code on a per-line basis. <b>Parameter:</b> account.X.group_pickup_code
	<MAC>.cfg	Configure group call pickup features on a phone basis. <b>Parameters:</b> features.pickup.group_pickup_enable features.pickup.group_pickup_code
	<y0000000000xx>.cfg	Assign a group call pickup DSS key. <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type linekey.X.line/ programablekey.X.line/

		expansion_module.X.key.Y.line linekey.X.value/ programablekey.X.value/ expansion_module.X.key.Y.value linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Assign a group call pickup DSS key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0">http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0</a>
		Configure the group call pickup code on a per-line basis. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>
		Configure group call pickup feature on a phone basis. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-callpickup&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-callpickup&amp;q=load</a>
<b>Phone User Interface</b>		Assign a group call pickup DSS key.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>features.pickup.group_pickup_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to display the <b>GPickup</b> soft key when the IP phone is on the dialing screen. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Features->Pick up & Park->Group Call Pickup <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>account.X.group_pickup_code<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the group pickup code for a specific account.</p> <p><b>Example:</b> account.1.group_pickup_code = *98</p> <p><b>Note:</b> The group call pickup code configured on a per-line basis (configured by the parameter "account.X.group_pickup_code") takes precedence over that configured on a phone basis (configured by the parameter "features.pickup.group_pickup_code").</p> <p><b>Web User Interface:</b> Account-&gt;Advanced-&gt;Group Call Pickup Code</p> <p><b>Phone User Interface:</b> None</p>		
<b>features.pickup.group_pickup_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the group call pickup code on a phone basis.</p> <p><b>Example:</b> features.pickup.group_pickup_code = *98</p> <p><b>Note:</b> The group call pickup code configured on a per-line basis (configured by the parameter "account.X.group_pickup_code") takes precedence over that configured on a phone basis (configured by the parameter "features.pickup.group_pickup_code").</p> <p><b>Web User Interface:</b> Features-&gt;Pick up &amp; Park-&gt;Group Call Pickup Code</p> <p><b>Phone User Interface:</b> None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## Group Call Pickup DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

**Details of Configuration Parameters:**

Parameters	Permitted Values
<b>linekey.X.type</b> <sup>[1]</sup> <b>programablekey.X.type</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.type</b> <sup>[3]</sup>	23
<b>Description:</b> Configures a DSS key as a group call pickup DSS key on the IP phone. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.line</b> <sup>[1]</sup> <b>programablekey.X.line</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.line</b> <sup>[3]</sup>	Refer to the following content
<b>Description:</b> Configures the desired line to apply the group call pickup DSS key. <b>Permitted Values:</b> 1 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G) 1 to 12 (for SIP-T52S/T42G/T42S) 1 to 6 (for SIP-T41P/T41S/T27P/T27G) 1 to 3 (for SIP-T40P/T40G/T23P/T23G) 1 to 2 (for SIP-T21(P) E2) 1-Line 1 2-Line 2 ... 16-Line 16 <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Example:</b> linekey.2.line = 1 <b>Web User Interface:</b> Dsskey->Line Key/ Programable Key->Line <b>Phone User Interface:</b>	



Parameters	Permitted Values
Menu->Features->DSS Keys->Line Key X->Account ID	
<b>linekey.X.value<sup>[1]</sup></b> <b>programmablekey.X.value<sup>[2]</sup></b> <b>expansion_module.X.key.Y.value<sup>[3]</sup></b>	String within 99 characters
<p><b>Description:</b> Configures the group call pickup feature code.</p> <p><b>Example:</b> linekey.2.value = *98</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key/Programable Key-&gt;Value</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Value</p>	
<b>linekey.X.label<sup>[1]</sup></b> <b>programmablekey.X.label<sup>[2]</sup></b> <b>expansion_module.X.key.Y.label<sup>[3]</sup></b>	String within 99 characters
<p><b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key.</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key/Programable Key-&gt;Label</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Label</p>	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<p><b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key.</p> <p><b>Note:</b> It is only applicable to SIP-T52S IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key-&gt;Short</p> <p><b>Phone User Interface:</b></p>	

Parameters	Permitted Values
Menu->Features->DSS Keys->Line Key X->Short Label	

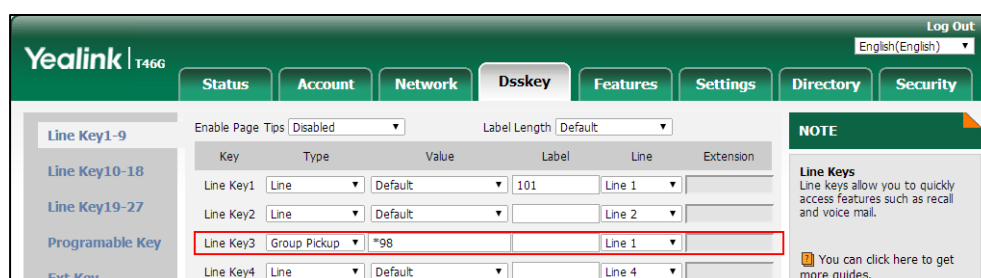
<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programmable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure a group call pickup DSS key via web user interface:

1. Click on **Dsskey->Line Key (or Programmable Key)**.
2. In the desired DSS key field, select **Group Pickup** from the pull-down list of **Type**.
3. Enter the group call pickup code in the **Value** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
5. Select the desired line from the pull-down list of **Line**.



6. Click **Confirm** to accept the change.

#### To configure the group call pickup code on a per-line basis via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.

- Enter the group call pickup code in the **Group Call Pickup Code** field.

The screenshot shows the Yealink T46G web interface with the 'Account' tab selected. The 'Group Call Pickup Code' field is highlighted with a red box and contains the value '\*98'. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'Early Media' (Disabled), 'SIP Server Type' (Default), 'Music Server URI' (sip:moh@sp.com), 'Directed Call Pickup Code' (\*97), 'Distinctive Ring Tones' (Enabled), 'Unregister When Reboot' (Enabled), 'Out Dialog BLF' (Enabled), 'VQ RTPC-XR Collector Name' (collector), 'VQ RTPC-XR Collector Address' (10.2.1.98), and 'VQ RTPC-XR Collector Port' (5060). A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

- Click **Confirm** to accept the change.

**To configure group call pickup feature on a phone basis via web user interface:**

- Click on **Features->Pick up & Park**.
- Select the desired value from the pull-down list of **Group Call Pickup**.
- Enter the group call pickup code in the **Group Call Pickup Code** field.

The screenshot shows the Yealink T46G web interface with the 'Features' tab selected. The 'Group Call Pickup' field is highlighted with a red box and contains the value 'Enabled'. Other fields include 'Directed Call Pickup' (Disabled), 'Directed Call Pickup Code' (empty), 'Visual Alert for BLF Pickup' (Disabled), 'Audio Alert for BLF Pickup' (Disabled), and 'Ring Type for BLF Pickup' (Splash.wav). A 'NOTE' section on the right provides information about Directed Call Pickup and Visual Alert for BLF Pickup.

- Click **Confirm** to accept the change.

**To configure a group pickup DSS key via phone user interface:**

- Press **Menu->Features->DSS Keys**.
- Select the desired DSS key.
- Press **◀** or **▶**, or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select **GPickup** from the **Key Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired line from the **Account ID**

field.

6. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
7. Enter the group call pickup code in the **Value** field.
8. Press the **Save** soft key to accept the change.

## Dialog Info Call Pickup

Call pickup is implemented through SIP signals on some specific servers. When this feature is enabled, IP phones support picking up incoming calls via the INVITE message which includes a Replaces header in the message body. The value of Replaces is derived from a NOTIFY message with dialog-info event. A user can pick up an incoming call by pressing the DSS key used to monitor a specific extension (such as the BLF DSS key). For more information on BLF, refer to [Busy Lamp Field \(BLF\)](#) on page 532.

If the visual alert for blf pickup feature is enabled, a user can also pick up an incoming call by pressing the **Pick Up** soft key. For more information on visual alert for blf pickup, refer to [Visual Alert and Audio Alert for BLF Pickup](#) on page 537.

It is not applicable to SIP-T19(P) E2 IP phones.

### Note

In this way, you do not need to configure the directed call pickup code.

Example of the dialog-info carried in NOTIFY message:

```
<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="6" state="partial"
entity="sip:1011@10.2.1.48:5060">
  <dialog id="65603" call-id="0_1756536024@10.10.20.34" local-tag="3408640225" remote-tag="3779921438"
  direction="recipient">
    <state>early</state>
    <local>
      <identity>sip:1011@10.2.1.48:5060</identity>
      <target uri="sip:1011@10.2.1.48:5060"/>
    </local>
    <remote>
      <identity>sip:1058@10.2.1.48:5060</identity>
      <target uri="sip:1058@10.2.1.48:5060"/>
    </remote>
  </dialog>
</dialog-info>
```

Example of the Replaces carried in INVITE message:

```
Via: SIP/2.0/UDP 10.10.20.18:5060;branch=z9hG4bK2026058891
From: "1010" <sip:1010@10.2.1.48:5060>;tag=826048502
```

```

To: <sip:1058@10.2.1.48:5060>
Call-ID: 0_572446084@10.10.20.18
CSeq: 1 INVITE
Contact: <sip:1010@10.10.20.18:5060>
Content-Type: application/sdp
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH,
UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Replaces: 0_1756536024@10.10.20.34;to-tag=3779921438;from-tag=3408640225
Allow-Events: talk,hold,conference,refer,check-sync
Supported: replaces
Content-Length: 304

```

## Procedure

Dialog info call pickup can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure dialog info call pickup. <b>Parameter:</b> account.X.dialoginfo_callpickup
<b>Web User Interface</b>		Configure dialog info call pickup. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
account.X.dialoginfo_callpickup <sup>[1]</sup>	0 or 1	0
<b>Description:</b>  Enables or disables the IP phone to pick up a call according to the Replaces header of the INVITE message for a specific account.  <b>0</b> -Disabled <b>1</b> -Enabled, call pickup is implemented through SIP signals.  <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.  <b>Web User Interface:</b> Account->Advanced->Dialog Info Call Pickup  <b>Phone User Interface:</b>  None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2.

### To configure dialog info call pickup via web user interface:

1. Click on **Account**->**Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Dialog Info Call Pickup**.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected, and the 'Advanced' sub-tab is active. The 'Account' dropdown is set to 'Account 1'. The 'Dialog Info Call Pickup' option is highlighted with a red box and set to 'Enabled'. Other settings include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'RTP Encryption(SRTP)' (Optional), 'PTime(ms)' (20), 'BLF List URI' (4609@pbx.yealink.com), 'BLF List Pickup Code' (\*97), 'BLF List Barge In Code' (\*33), 'BLF List Retrieve Call Parked Code' (\*88), 'Shared Line' (Shared Call Appearance), 'Call Pull Feature Access Code' (\*11), 'VQ RTPC-XR Collector name' (collector), 'VQ RTPC-XR Collector address' (10.2.1.98), and 'VQ RTPC-XR Collector port' (5060). A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

4. Click **Confirm** to accept the change.

## Recent Call In Dialing

Recent call in dialing feature allows users to view the placed calls list when the phone is on the dialing screen (lifts the handset, presses the Speakerphone key or desired line key). Users can select to place a call from the placed calls list. For some phones, you may need to press up/down navigation key to browse all the placed call number.



## Procedure

Recent call in dialing can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure recent call in dialing feature. <b>Parameter:</b> super_search.recent_call
<b>Web User Interface</b>		Configure recent call in dialing feature. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=contacts-favorite&q=load

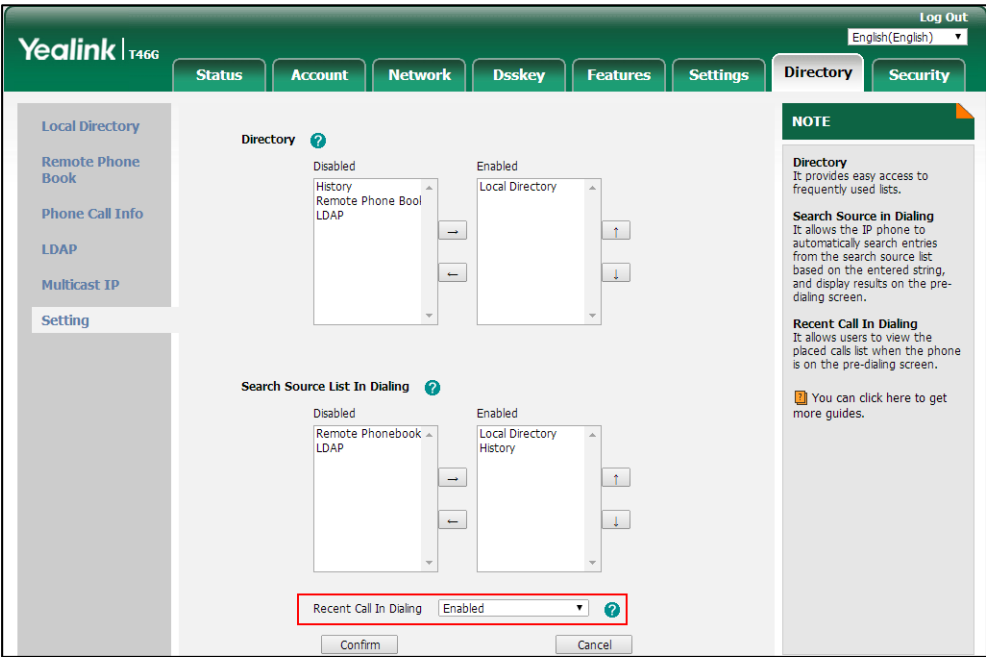
### Details of Configuration Parameter:

Parameter	Permitted Values	Default
super_search.recent_call	0 or 1	1
<b>Description:</b> Enables or disables Recent Call in Dialing feature. <b>0</b> -Disabled <b>1</b> -Enabled, users can view the placed calls list when the phone is on the dialing screen. <b>Web User Interface:</b> Directory->Setting->Recent Call In Dialing <b>Phone User Interface:</b> None		

**To configure recent call in dialing via web user interface:**

1. Click on **Directory->Setting**.

2. Select the desired value from the pull-down list of **Recent Call In Dialing**.



3. Click **Confirm** to accept the change.

## ReCall

ReCall, also known as last call return, allows users to place a call back to the last caller. Recall is implemented on IP phones using a recall DSS key. When you press the recall DSS key, you will place a call to the phone number that last called you.

### Procedure

Recall DSS key can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Assign a recall DSS key. <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Assign a recall DSS key. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =dsskey&q=load&model=0



**Phone User Interface**

Assign a recall DSS key.

**ReCall DSS Key**

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

**Details of Configuration Parameters:**

Parameters	Permitted Values
<b>linekey.X.type</b> <sup>[1]</sup> <b>programablekey.X.type</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.type</b> <sup>[3]</sup>	7
<b>Description:</b> Configures a DSS key as a recall DSS key on the IP phone. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.label</b> <sup>[1]</sup> <b>programablekey.X.label</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.label</b> <sup>[3]</sup>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones.	

Parameters	Permitted Values
<b>Web User Interface:</b> Dsskey->Line Key->Short	
<b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programmable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure a recall DSS key via web user interface:

1. Click on **Dsskey->Line Key** (or **Programmable Key**).
2. In the desired DSS key field, select **ReCall** from the pull-down list of **Type**.
3. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

4. Click **Confirm** to accept the change.

#### To configure a recall DSS key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press or , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press or , or the **Switch** soft key to select **ReCall** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Press the **Save** soft key to accept the change.

## Call Number Filter

Call number filter feature allows IP phone to automatically filter designated characters when dialing.

## Procedure

Call number filter can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the characters the IP phone filters when dialing. <b>Parameter:</b> features.call_num_filter
<b>Web User Interface</b>		Configure the characters the IP phone filters when dialing. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-general&q=load

## Details of Configuration Parameter:

Parameter	Permitted Values	Default
features.call_num_filter	String within 99 characters	, -()
<p><b>Description:</b></p> <p>Configures the characters the IP phone filters when dialing.</p> <p>If the dialed number contains configured characters, the IP phone will automatically filter these characters when dialing.</p> <p><b>Example:</b></p> <p>features.call_num_filter = -</p> <p>If you dial 3-61, the IP phone will filter the character - and then dial out 361.</p> <p><b>Note:</b> If it is left blank, the IP phone will not automatically filter any characters when dialing.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Call Number Filter</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure the characters the IP phone will filter via web user interface:**

1. Click on **Features->General Information**.

2. Enter the desired characters in the **Call Number Filter** field.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected. In the 'General Information' section, the 'Call Number Filter' field is highlighted with a red box. The field contains the characters '-'. Other settings visible include 'Call Waiting' (Enabled), 'Auto Redial' (Disabled), 'Auto Redial Interval' (10), 'Auto Redial Times' (10), 'Diversion/History-Info' (Enabled), 'Allow Trans Exist Call' (Enabled), 'BLF LED Mode' (0), 'Auto-Logout Time' (5), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right provides additional information about various call features.

3. Click **Confirm** to accept the change.

## Call Park

Call park allows users to park a call on a special extension and then retrieve it from another phone (for example, a phone in another office or conference room). This feature depends on support from a SIP server.

Call park feature supports the following two modes:

- **FAC mode:** Call park feature via FAC mode allows users to park an active call to a desired extension or local extension through dialing the call park code.
- **Transfer mode:** Call park feature via Transfer mode allows users to park an active call to the shared parking lot through performing a blind transfer to a call park shared number (call park code). For some servers, the system will return a specific call park retrieve number (park retrieve code) from which the call can be retrieved after parking successfully.

Users can park calls on the extension, known as call park orbit, by pressing the **Park** soft key or a call park DSS key. You need to configure the call park code for the **Park** soft key or the call park DSS key. Call park code configured for the **Park** soft key will also apply to the call park DSS key. If the call is parked successfully, users will hear a voice prompt confirming that the call was parked. The current call is placed on hold and can be retrieved on another IP phone. To retrieve a parked call, dial the park retrieve code or press the **Retrieve** soft key or press the retrieve park DSS key. If the parked call is not retrieved within a period of time assigned by the system, the phone performing call park will receive call back.

### Note

If the call park code or park retrieve code has been configured for the **Park** soft key or the **Retrieve** soft key, you don't need to configure the call park code or the park retrieve code for the call park DSS key or the retrieve park DSS key.

## Procedure

Call park can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	<p>Configure call park feature.</p> <p><b>Parameters:</b></p> <p>features.call_park.park_mode</p> <p>features.call_park.enable</p> <p>features.call_park.park_code</p> <p>features.call_park.park_retrieve_code</p> <p>features.call_park.direct_send.enable</p> <p>features.call_park.line_restriction.enable</p>
		<p>Assign a call park DSS key.</p> <p>Assign a retrieve park DSS key.</p> <p><b>Parameters:</b></p> <p>linekey.X.type/ expansion_module.X.key.Y.type</p> <p>linekey.X.line/ expansion_module.X.key.Y.line</p> <p>linekey.X.value/ expansion_module.X.key.Y.value</p> <p>linekey.X.label/ expansion_module.X.key.Y.label</p> <p>linekey.X.shortlabel</p>
<b>Web User Interface</b>		<p>Configure call park feature.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?parameters=features-callpickup&amp;q=load</p>
		<p>Assign a call park DSS key.</p> <p>Assign a retrieve park DSS key.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?parameters=dsskey&amp;q=load&amp;model=0</p>
<b>Phone User Interface</b>		<p>Assign a call park DSS key.</p> <p>Assign a retrieve park DSS key.</p>

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>features.call_park.park_mode</b>	<b>1 or 2</b>	<b>2</b>
<b>Description:</b> Configures the call park mode. <b>1</b> -FAC <b>2</b> -Transfer <b>Web User Interface:</b> Features->Pick up & Park->Call Park Mode <b>Phone User Interface:</b> None		
<b>features.call_park.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to display the <b>Park</b> soft key during a call. <b>0</b> -Disabled <b>1</b> -Enabled, the <b>Retrieve</b> soft key will also be displayed on the dialing screen. <b>Web User Interface:</b> Features->Pick up & Park->Call Park <b>Phone User Interface:</b> None		
<b>features.call_park.park_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the call park code for the <b>Park</b> soft key. This call park code will also apply to the call park DSS key. <b>Example:</b> features.call_park.park_code = *68 <b>Web User Interface:</b> Features->Pick up & Park->Call Park Code <b>Phone User Interface:</b> None		
<b>features.call_park.park_retrieve_code</b>	<b>String within 32 characters</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the park retrieve code for the <b>Retrieve</b> soft key.</p> <p>This park retrieve code will also apply to the retrieve park DSS key.</p> <p><b>Example:</b></p> <p>features.call_park.park_retrieve_code = *88</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Pick up &amp; Park-&gt;Park Retrieve Code</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.call_park.direct_send.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to dial out the call park code/park retrieve code directly when pressing the <b>Park/Retrieve</b> soft key.</p> <p><b>0</b>-Disabled, the IP phone will enter the pre-dialing screen when pressing the <b>Park/Retrieve</b> soft key. And you can dial the specific extension manually or press the BLF/BLF List DSS key to park the call to the specific user or retrieve the call parked from the specific user.</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "features.call_park.park_mode" is set to 1 (FAC) and you have configured the call park code/park retrieve code.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.call_park.line_restriction.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to park/retrieve a call using the line specified by the parameter "<b>linekey.X.line/expansion_module.X.key.Y.line</b>". It is only applicable to the scenario that the user uses the call park/retrieve park DSS key to park/retrieve a call.</p> <p><b>0</b>-Disabled, the IP phone will park/retrieve the call to the server where the current account is registered.</p> <p><b>1</b>-Enabled, the phone will place a new call if the specified line is idle.</p> <p><b>Note:</b> It works only if the value of the parameter "features.call_park.park_mode" is set to 2 (Transfer). It is not applicable to SIP-T19(P) E2 IP phones.</p>		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Call Park DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>expansion_module.X.key.Y.type<sup>[2]</sup></b>	<b>10</b>
<b>Description:</b> Configures a DSS key as a call park DSS key on the IP phone. The digit <b>10</b> stands for the key type <b>Call Park</b> . <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.line<sup>[1]</sup></b> <b>expansion_module.X.key.Y.line<sup>[2]</sup></b>	<b>Refer to the following content</b>
<b>Description:</b> Configures the desired line to apply the call park DSS key. <b>Permitted Values:</b> 1 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G) 1 to 12 (for SIP-T52S/T42G/T42S) 1 to 6 (for SIP-T41P/T41S/T27P/T27G) 1 to 3 (for SIP-T40P/T40G/T23P/T23G) 1 to 2 (for SIP-T21(P) E2) 1-Line 1 2-Line 2	



Parameters	Permitted Values
<p>...</p> <p>16-Line 16</p> <p><b>Example:</b></p> <p>linekey.2.line = 1</p> <p><b>Note:</b> This parameter is only required if the value of the parameter "features.call_park.line_restriction.enable" is set to 1 (Enabled). It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line key-&gt;Line</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Account ID</p>	
<p><b>linekey.X.value<sup>[1]</sup></b></p> <p><b>expansion_module.X.key.Y.value<sup>[2]</sup></b></p>	<p><b>String within 99 characters</b></p>
<p><b>Description:</b></p> <p>Configures the call park code.</p> <p><b>Example:</b></p> <p>linekey.2.value = *68</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line key-&gt;Value</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Value</p>	
<p><b>linekey.X.label<sup>[1]</sup></b></p> <p><b>expansion_module.X.key.Y.label<sup>[2]</sup></b></p>	<p><b>String within 99 characters</b></p>
<p><b>Description:</b></p> <p>(Optional.) Configures the label displayed on the LCD screen for each DSS key.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key-&gt;Label</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Label</p>	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for

SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

## Retrieve Park DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>expansion_module.X.key.Y.type<sup>[2]</sup></b>	56
<p><b>Description:</b></p> <p>Configures a DSS key as a retrieve park DSS key on the IP phone.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line key-&gt;Type</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Type</p>	
<b>linekey.X.line<sup>[1]</sup></b> <b>expansion_module.X.key.Y.line<sup>[2]</sup></b>	Refer to the following content
<p><b>Description:</b></p> <p>Configures the desired line to apply the retrieve park DSS key.</p> <p><b>Permitted Values:</b></p> <p>1 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G)</p> <p>1 to 12 (for SIP-T52S/T42G/T42S)</p> <p>1 to 6 (for SIP-T41P/T41S/T27P/T27G)</p> <p>1 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>1 to 2 (for SIP-T21(P) E2)</p> <p>1-Line 1</p> <p>2-Line 2</p> <p>...</p> <p>16-Line 16</p> <p><b>Example:</b></p>	

Parameters	Permitted Values
<p>linekey.2.line = 1</p> <p><b>Note:</b> This parameter is only required if the value of the parameter "features.call_park.line_restriction.enable" is set to 1 (Enabled). It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line key-&gt;Line</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Account ID</p>	
<p><b>linekey.X.value<sup>[1]</sup></b></p> <p><b>expansion_module.X.key.Y.value<sup>[2]</sup></b></p>	<p><b>String within 99 characters</b></p>
<p><b>Description:</b></p> <p>Configures the park retrieve code.</p> <p><b>Example:</b></p> <p>linekey.2.value = *88</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line key-&gt;Value</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Value</p>	
<p><b>linekey.X.label<sup>[1]</sup></b></p> <p><b>expansion_module.X.key.Y.label<sup>[2]</sup></b></p>	<p><b>String within 99 characters</b></p>
<p><b>Description:</b></p> <p>(Optional.) Configures the label displayed on the LCD screen for each DSS key.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key-&gt;Label</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Label</p>	
<p><b>linekey.X.shortlabel</b></p> <p>(X ranges from 1 to 21)</p>	<p><b>String within 99 characters</b></p>
<p><b>Description:</b></p>	

Parameters	Permitted Values
(Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Short <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure call park feature via web user interface:

1. Click on **Features->Pick up & Park**.
2. Select the desired call park mode from the pull-down list of **Call Park Mode**.
3. Select the desired value from the pull-down list of **Call Park**.
4. (Optional.) Enter the call park code in the **Call Park Code** field.
5. (Optional.) Enter the park retrieve code in the **Park Retrieve Code** field.

The screenshot shows the Yealink T466 web interface. The 'Features' tab is selected, and the 'Pick up & Park' sub-tab is active. The 'Call Park' section is highlighted with a red box, indicating the configuration for call parking. The 'Call Park Mode' is set to 'FAC', 'Call Park' is set to 'Enabled', 'Call Park Code' is set to '\*68', and 'Park Retrieve Code' is set to '\*88'. The 'Call Pickup' section is also visible, showing settings for 'Directed Call Pickup' (Disabled), 'Group Call Pickup' (Enabled), 'Visual Alert for BLF Pickup' (Disabled), 'Audio Alert for BLF Pickup' (Disabled), and 'Ring Type for BLF Pickup' (Splash.wav). A 'NOTE' section on the right provides additional information about the features.

6. Click **Confirm** to accept the change.

#### To configure a call park DSS key via web user interface:

1. Click on **Dsskey->Line Key**.
2. In the desired DSS key field, select **Call Park** from the pull-down list of **Type**.
3. Enter the call park code in the **Value** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

5. Select the desired line from the pull-down list of **Line**.

6. Click **Confirm** to accept the change.

**To configure a retrieve park DSS key via web user interface:**

1. Click on **Dsskey->Line Key**.
2. In the desired DSS key field, select **Retrieve Park** from the pull-down list of **Type**.
3. Enter the park retrieve code in the **Value** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
5. Select the desired line from the pull-down list of **Line**.

6. Click **Confirm** to accept the change.

**To configure a call park DSS key via phone user interface:**

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select **Call Park** from the **Key Type** field.
5. Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select the desired line from the **Account ID** field.
6. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
7. Enter the call park code in the **Value** field.
8. Press the **Save** soft key to accept the change.

**To configure a retrieve park DSS key via phone user interface:**

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select **Retrieve Park** from the **Type** field.

4. Press ◀ or ▶ , or the **Switch** soft key to select the desired account from the **Account ID** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the retrieve park code in the **Value** field.
7. Press the **Save** soft key to accept the change.

## Calling Line Identification Presentation (CLIP)

Calling Line Identification Presentation (CLIP) allows IP phones to display the caller identity, derived from a SIP header contained in the INVITE message when receiving an incoming call. IP phones support deriving caller identity from three types of SIP header: From, P-Asserted-Identity (PAI) and Remote-Party-ID (RPID). Identity presentation is based on the identity in the relevant SIP header.

### Note

If the caller already exists in the local directory, the local contact name assigned to the caller should be preferentially displayed and stored in the call log.

The following sessions show the enhancements of calling line identification presentation according to the calling line identification source configured on the IP phones.

### Caller ID source = FROM

- 1) The IP phone checks Privacy: id header preferentially, if there is a Privacy: id in the INVITE request, the calling line identification information will be hidden and the IP phone LCD screen presents anonymous.
- 2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.
- 3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone presents the caller identification derived from the FROM header.

### Caller ID source = PAI

- 1) The IP phone checks Privacy: id header preferentially, if there is a Privacy: id in the INVITE request, the caller identification information will be hidden and the IP phone LCD screen presents anonymous.
- 2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.
- 3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the P-Asserted-Identity header.

### Caller ID source = PAI-FROM

- 1) The IP phone checks Privacy: id header preferentially, if there is a Privacy: id in the INVITE request, the caller identification information will be hidden and the IP phone LCD screen presents anonymous.

- 2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.
- 3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the P-Asserted-Identity header.
- 4) If there is not P-Asserted-Identity header in the INVITE request, the IP phone presents the caller identification derived from the FROM header.

### **Caller ID source = RPID-FROM**

- 1) The IP phone checks Privacy: id header preferentially, if there is a Privacy: id in the INVITE request, the caller identification information will be hidden and the IP phone LCD screen presents anonymous.
- 2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.
- 3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the Remote-Party-ID header.
- 4) If there is not Remote-Party-ID header in the INVITE request, the IP phone presents the caller identification derived from the FROM header.

### **Caller ID source = PAI-RPID-FROM**

- 1) The IP phone checks Privacy: id header preferentially, if there is a Privacy: id in the INVITE request, the caller identification information will be hidden and the IP phone LCD screen presents anonymous.
- 2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.
- 3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the P-Asserted-Identity header.
- 4) If there is not P-Asserted-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the Remote-Party-ID header.
- 5) If there is not Remote-Party-ID header in the INVITE request, the IP phone presents the caller identification derived from the FROM header.

### **Caller ID source = RPID-PAI-FROM**

- 1) The IP phone checks Privacy: id header preferentially, if there is a Privacy: id in the INVITE request, the caller identification information will be hidden and the IP phone LCD screen presents anonymous.
- 2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.
- 3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the Remote-Party-ID header.
- 4) If there is not Remote-Party-ID header in the INVITE request, the IP phone checks and presents the caller identification from the P-Asserted-Identity header.
- 5) If there is not P-Asserted-Identity in the INVITE request, the IP phone presents the caller

identification derived from the FROM header.

For more information on calling line identification presentation, refer to [Calling and Connected Line Identification Presentation on Yealink IP Phones](#).

## Procedure

CLIP can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<MAC>.cfg	Configure the presentation of the caller identity. <b>Parameter:</b> account.X.cid_source
		Specify whether to process Privacy header field. <b>Parameter:</b> account.X.cid_source_privacy
		Specify whether to process the P-Preferred-Identity (PPI) header for caller identity presentation. <b>Parameter:</b> account.X.cid_source_ppi
	<y0000000000x x>.cfg	Specify the priority order for the sources of caller identity information. <b>Parameter:</b> sip.cid_source.preference
<b>Web User Interface</b>		Configure the presentation of the caller identity. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>

## Details of the Configuration Parameters:

Parameters	Permitted Values	Default
account.X.cid_source <sup>[1]</sup>	0, 1, 2, 3, 4, 5 or 6	0
<b>Description:</b> Configures the presentation of the caller identity when receiving an incoming call on a specific account.		



Parameters	Permitted Values	Default
<b>0</b> -FROM <b>1</b> -PAI <b>2</b> -PAI-FROM <b>3</b> -RPID-PAI-FROM <b>4</b> -PAI-RPID-FROM <b>5</b> -RPID-FROM <b>6</b> -PREFERENCE, the IP phone uses the custom priority order for the sources of caller identity information (configured by the parameter "sip.cid_source.preference"). <b>Web User Interface:</b> Account->Advanced->Caller ID Source <b>Phone User Interface:</b> None		
<b>account.X.cid_source_privacy<sup>[1]</sup></b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to process Privacy header field in the SIP message for a specific account. <b>0</b> -Disabled, the IP phone doesn't process Privacy header. <b>1</b> -Enabled, the caller identification information will be hidden and the IP phone LCD screen presents anonymity if there is a Privacy: id in the INVITE request. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>account.X.cid_source_ppi<sup>[1]</sup></b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to process the P-Preferred-Identity (PPI) header for caller identity presentation when receiving an incoming call on a specific account. <b>0</b> -Disabled, the IP phone doesn't process P-Preferred-Identity (PPI) header. <b>1</b> -Enabled, the IP phone presents the caller identification from the P-Preferred-Identity (PPI) header. <b>Web User Interface:</b> None <b>Phone User Interface:</b>		

Parameters	Permitted Values	Default
None		
<b>sip.cid_source.preference</b>	<b>String</b>	<b>P-Preferred-Identity, P-Asserted-Identity, Remote-Party-ID, From</b>
<p><b>Description:</b></p> <p>Configures the priority order for the sources of caller identity information. The headers can be in any order.</p> <p><b>Note:</b> Yealink IP phones support deriving caller identity from the following SIP headers: From, P-Asserted-Identity (PAI), P-Preferred-Identity and Remote-Party-ID (RPID). It works only if the value of the parameter "account.X.cid_source" is set to 6 (PREFERENCE).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

#### To configure the presentation of the caller identity via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Caller ID Source**.

The screenshot shows the Yealink T46G web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'Dsskey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Account' tab is active. On the left, there's a sidebar with 'Register', 'Basic', 'Codec', and 'Advanced' (selected). The main content area shows settings for 'Account1'. The 'Caller ID Source' is set to 'FROM' and is highlighted with a red box. Other settings include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'DTMF Type' (RFC2833), 'DTMF Info Type' (DTMF-Relay), 'DTMF Payload Type(96~127)' (101), 'Retransmission' (Disabled), 'Subscribe Register' (Disabled), 'Subscribe for MWI' (Disabled), 'MWI Subscription Period(Seconds)' (3600), 'Subscribe MWI To Voice Mail' (Disabled), 'Voice Mail' (empty), and 'Voice Mail Display' (Enabled). A 'NOTE' section on the right contains information about DTMF, Session Timer, Busy Lamp Field/BLF List, and Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA).

4. Click **Confirm** to accept the change.

## Connected Line Identification Presentation (COLP)

Connected Line Identification Presentation (COLP) allows IP phones to display the identity of the connected party specified for outgoing calls. IP phones can display the Dialed Digits, or the identity in a SIP header (Remote-Party-ID or P-Asserted-Identity) received, or the identity in the From header carried in the UPDATE message sent by the callee as described in [RFC 4916](#). Connected line identification presentation is also known as Called line identification presentation. In some cases, the remote party will be different from the called line identification presentation due to call diversion.

### Note

If the callee already exists in the local directory, the local contact name assigned to the callee should be preferentially displayed.

The following sessions show the enhancements of connected line identification according to the connected line identification source configured on the IP phones.

### Connected Line Identification source = PAI-RPID

- 1) The IP phone checks Privacy: id header preferentially, if there is a Privacy: id in the 18X or 200OK response, the connected line identification information will be hidden and the IP phone LCD screen presents anonymous.
- 2) If there is not any Privacy: id header in the 18X or 200OK response, the IP phone checks and presents the connected line identification from the P-Asserted-Identity header.
- 3) If there is not P-Asserted-Identity header in the I8X or 200OK response, the IP phone presents the connected line identification from the Remote-Party-ID header. If no, the IP phone presents the connected line identification according to the dialed digits.

### Connected Line Identification source = Dialed digits

Yealink IP phones present the connected line identification according to the dialed digits.

### Connected Line Identification source = RFC4916

Yealink IP phones support to present the connected line identification from UPDATE message following the [RFC 4916](#).

- 1) The IP phone receives an UPDATE message during a call, the connected line identification on the LCD screen should be refreshed according the FROM SIP carried in the UPDATE message.

For more information on connected line identification presentation, refer to [Calling and Connected Line Identification Presentation on Yealink IP Phones](#).

## Procedure

COLP can be configured only using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure the presentation of the callee's identity.  <b>Parameter:</b> account.X.cp_source
		Specify whether to process Privacy header field.  <b>Parameter:</b> account.X.cid_source_privacy

## Details of the Configuration Parameters:

Parameters	Permitted Values	Default
<b>account.X.cp_source<sup>[1]</sup></b>	<b>0, 1 or 2</b>	<b>0</b>
<b>Description:</b> Configures the presentation of the callee's identity for a specific account. <b>0</b> -PAI-RPID <b>1</b> -Dialed Digits <b>2</b> -RFC 4916 When the RFC 4916 is enabled on the IP phone, the caller sends the SIP request message which contains the from-change tag in the Supported header. The caller then receives an UPDATE message from the callee, and displays the identity in the "From" header. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>account.X.cid_source_privacy<sup>[1]</sup></b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to process Privacy header field in the SIP message for a specific account. <b>0</b> -Disabled, the IP phone doesn't process Privacy header. <b>1</b> -Enabled, the caller identification information will be hidden and the IP phone LCD screen presents anonymous if there is a Privacy: id in the INVITE request. <b>Web User Interface:</b>		

Parameters	Permitted Values	Default
None		
<b>Phone User Interface:</b>		
None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## Mute

Yealink IP phones support muting the microphone of the active audio device (handset, headset or speakerphone) during an active call or while dialing. You can activate the mute feature by pressing the MUTE key. Normally, mute feature is automatically deactivated when the active call ends. You can enable keep mute feature to keep the mute state persist across the calls.

## Allow Mute

### Mute during the call

You can mute the microphone of the active audio device during an active call, and then the other party cannot hear you.

### Mute while dialing/ringing

You can mute the microphone of the active audio device when the phone is on the pre-dialing/dialing/calling/ringing screen. The call is automatically muted when setting up successfully. This feature is available for SIP-T23P/T23G/T21(P) E2 IP phones only when the keep mute feature is enabled. For more information on keep mute, refer to [Keep Mute](#) on page 477. Mute while dialing feature helps prevent the other party from hearing your informal discussion.

### Procedure

Allow mute can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure allow mute feature. <b>Parameter:</b> features.allow_mute
<b>Web User Interface</b>		Configure allow mute feature. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-general&q=load

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.allow_mute</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the allow mute feature for the IP phone.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, you are allowed to mute an active call, or activate the mute feature on the pre-dialing/dialing/calling/ringing screen (Mute the microphone on the calling screen is only applicable to the SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G IP phones).</p> <p><b>Note:</b> For SIP-T23P/T23G/T21(P) E2 IP phones, you are allowed to activate the mute feature on the pre-dialing/dialing/calling/ringing screen only if the value of the parameter "features.keep_mute.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Allow Mute</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure allow mute via web user interface:**


1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Allow Mute**.

The screenshot displays the Yealink T466 web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'Allow Mute' dropdown menu is highlighted with a red rectangle and is currently set to 'Enabled'. Other visible settings include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Call Waiting Off Code' (empty), 'Auto Redial' (Disabled), 'Auto Redial Interval (1~300s)' (10), 'Auto Redial Times (1~300)' (10), 'Play Hold Tone' (Enabled), 'Play Hold Tone Delay' (30), 'Dual-Headset' (Disabled), 'Auto-Answer Delay(1~4s)' (1), and 'Auto Linekeys' (Disabled). A 'NOTE' panel on the right side of the screen provides detailed descriptions for 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion' features.

3. Click **Confirm** to accept the change.

## Keep Mute

Keep mute, also known as persistent mute, allows you to keep the mute state of your phone persist.

By default, the mute feature is automatically deactivated when the active call ends. When you enable keep mute feature and activate the mute feature, the phone stays in the mute state until you press the MUTE key again or the phone restarts. You can activate the mute feature in any state (for example, idle state) by pressing the MUTE key (for SIP-T23P/T23G/T21(P) E2 IP phones, you can only activate the mute feature by pressing the Cancel key (  ) on idle/pre-dialing/dialing/ringing/calling/talking screen).

### Note

For SIP-T23P/T23G/T21(P) E2 IP phones, the Cancel key can be used to cancel actions, reject incoming calls, mute/un-mutes calls or activate the mute feature. If keep mute feature is enabled, the Cancel key can only be used as a mute key.

### Procedure

Keep mute can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure keep mute feature. <b>Parameter:</b> features.keep_mute.enable
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### Details of Configuration Parameter:

Parameter	Permitted Values	Default
features.keep_mute.enable	0 or 1	0
<b>Description:</b> Enables or disables the keep mute feature for the IP phone. <b>0</b> -Disabled, mute feature is automatically deactivated when the active call ends. <b>1</b> -Enabled, the mute state is kept until you change the mute state manually or the phone restarts. <b>Note:</b> If it is set to 1 (Enabled), you cannot customize the Cancel key for SIP-T23P/T23G/T21(P) E2 IP phones, or the Mute key for SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T19(P) E2 IP phones. It works only if the value of the parameter "features.allow_mute" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b>		

Parameter	Permitted Values	Default
None		

## Intercom

Intercom allows establishing an audio conversation directly. The IP phone can answer intercom calls automatically. This feature depends on support from a SIP server.

## Outgoing Intercom Calls

Intercom is a useful feature in office environments to quickly connect with an operator or secretary. Users can press an intercom DSS key to automatically initiate an outgoing intercom call with a remote extension.

Intercom can be also used to monitor a specific user for status changes on IP phones. For example, you can configure an intercom DSS key on a supervisor's phone to monitor the IP phone user status (busy or idle). When the monitored user places a call, a busy indicator on the supervisor's phone indicates that the user's phone is in use. When the monitored user is idle, the supervisor can press the intercom DSS key to automatically connect with a preconfigured target extension for outgoing intercom calls. When the monitored user receives an incoming call, the supervisor can press the intercom DSS key to pick up the call directly. To pick up the call, you have to configure the directed call pickup code in advance. You can configure the directed call pickup code when configuring an intercom DSS key. IP phones support this feature using a SUBSCRIBE/NOTIFY mechanism as specified in [RFC 3265](#). This feature depends on support from a SIP server.

**Line key/Expansion Module Key LED** (configured as an intercom DSS key)

LED Status	Description
Solid green	The monitored user is idle.
Fast-flashing red (200ms)	The monitored user receives an incoming call.
Solid red	The monitored user is dialing. The monitored user is talking.
Off	The monitored user does not exist.



## Procedure

Intercom can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>. cfg	Configure the intercom subscription. <b>Parameters:</b> features.intercom.led.enable features.intercom.subscribe.enable
		Configure the intercom mode. <b>Parameter:</b> features.intercom.mode features.intercom.feature_access_code
		Assign an intercom DSS key. <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type linekey.X.line/ programablekey.X.line/ expansion_module.X.key.Y.line linekey.X.value/ programablekey.X.value/ expansion_module.X.key.Y.value linekey.X.pickup_value/ programablekey.X.pickup_value/ expansion_module.X.key.Y.pickup_value linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Assign an intercom DSS key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0">http://&lt;phoneIPAddress&gt;/servlet?p=dsskey &amp;q=load&amp;model=0</a>
<b>Phone User Interface</b>		Assign an intercom DSS key.

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
features.intercom.led.enable	0 or 1	0
<b>Description:</b> Enables or disables the IP phone to display different intercom DSS key LED status/icons		

Parameters	Permitted Values	Default
<p>when the status of monitored user changes.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "features.intercom.subscribe.enable" is set to 1 (Enabled). It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.intercom.subscribe.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables intercom subscription for the IP phone.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.intercom.mode</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the intercom mode.</p> <p><b>0</b>-SIP</p> <p><b>1</b>-FAC, the feature access code is configured by the parameter "features.intercom.feature_access_code".</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.intercom.feature_access_code</b>	<b>0 or 1</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<b>Description:</b> Configures the intercom feature access code. <b>Note:</b> It works only if the value of parameter "features.intercom.mode" is set to 1 (FAC). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Intercom DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type</b> <sup>[1]</sup> <b>programablekey.X.type</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.type</b> <sup>[3]</sup>	14
<b>Description:</b> Configures a DSS key as an intercom DSS key. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line key/Programable Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.line</b> <sup>[1]</sup> <b>programablekey.X.line</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.line</b> <sup>[3]</sup>	Refer to the following content
<b>Description:</b> Configures the desired line to apply the intercom DSS key. <b>Permitted Values:</b> 1 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G) 1 to 12 (for SIP-T52S/T42G/T42S) 1 to 6 (for SIP-T41P/T41S/T27P/T27G)	

Parameters	Permitted Values
1 to 3 (for SIP-T40P/T40G/T23P/T23G) 1 to 2 (for SIP-T21(P) E2) 1-Line 1 2-Line 2 ... 16-Line 16 <b>Example:</b> linekey.2.line = 1 <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line key/Programable Key->Line <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Account ID	
<b>linekey.X.value</b> <sup>[1]</sup> <b>programablekey.X.value</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.value</b> <sup>[3]</sup>	String within 99 characters
<b>Description:</b> Configures the intercom number. <b>Example:</b> linekey.2.value = 1008 <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line key/Programable Key->Value <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Value	
<b>linekey.X.pickup_value</b> <sup>[1]</sup> <b>programablekey.X.pickup_value</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.pickup_value</b> <sup>[3]</sup>	String within 256 characters
<b>Description:</b> (Optional.) Configures the pickup code for intercom feature. <b>Example:</b> linekey.2.pickup_value = *20*	

Parameters	Permitted Values
<p><b>Note:</b> This parameter only applies to intercom and BLF features. It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key/Programable Key-&gt;Extension</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Extension</p>	
<b>linekey.X.label<sup>[1]</sup></b> <b>programablekey.X.label<sup>[2]</sup></b> <b>expansion_module.X.key.Y.label<sup>[3]</sup></b>	String within 99 characters
<p><b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key.</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key/Programable Key-&gt;Label</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Label</p>	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<p><b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key.</p> <p><b>Note:</b> It is only applicable to SIP-T52S IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key-&gt;Short</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Short Label</p>	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

### To configure an intercom DSS key via web user interface:

1. Click on **Dsskey->Line Key** (or **Programable Key**).
2. In the desired DSS key field, select **Intercom** from the pull-down list of **Type**.
3. Enter the remote extension number in the **Value** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
5. Select the desired line from the pull-down list of **Line**.
6. (Optional.) Enter the directed call pickup code in the **Extension** field.

Key	Type	Value	Label	Line	Extension
Line Key1	Line	Default	101	Line 1	
Line Key2	Intercom	1008		Line 1	
Line Key3	Line	Default		Line 3	
Line Key4	Line	Default		Line 4	

7. Click **Confirm** to accept the change.

### To configure an intercom DSS key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press or , or the **Switch** soft key to select **Intercom** from the **Type** field.
4. Select the desired line from the **Account ID** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the remote extension number in the **Value** field.
7. (Optional.) Enter the directed call pickup code in the **Extension** field.  
This field appears only if intercom subscription feature is enabled.
8. Press the **Save** soft key to accept the change.

## Incoming Intercom Calls

The IP phone can process incoming calls differently depending on settings. There are four configuration options for incoming intercom calls:

### Intercom Allow

Intercom Allow allows the IP phone to answer an incoming intercom call.

If you disable this feature, the IP phone will handle an incoming intercom call like a normal incoming call.

### Intercom Mute

Intercom Mute allows the IP phone to mute the microphone for incoming intercom calls.

### Intercom Tone

Intercom Tone allows the IP phone to play a warning tone before answering an intercom call.

### Intercom Barge

Intercom Barge allows the IP phone to automatically answer an incoming intercom call while an active call is in progress. The active call will be placed on hold.

If you disable this feature, the IP phone will handle an incoming intercom call like a normal incoming call while there is already an active call on the IP phone.

### Procedure

Incoming intercom calls can be configured using the following methods.

Central Provisioning (Configuration File)	<y0000000000xx>.cfg	Configure incoming intercom call feature.  <b>Parameters:</b> features.intercom.allow features.intercom.mute features.intercom.tone features.intercom.barge
		Specify the channel mode to use when receiving an incoming intercom call.  <b>Parameter:</b> features.intercom.headset_prior.enable
Web User Interface		Configure incoming intercom call feature.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-intercom&q=load
Phone User Interface		Configure incoming intercom call feature.

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>features.intercom.allow</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to answer an incoming intercom call. <b>0</b> -Disabled, the IP phone will handle an incoming intercom call like a normal incoming call.		

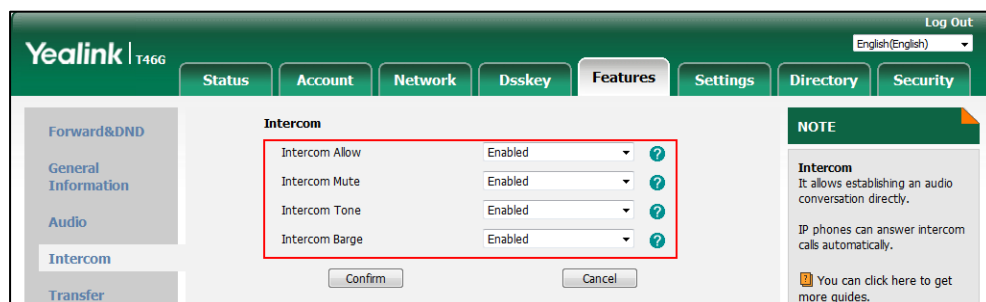
Parameters	Permitted Values	Default
<p><b>1</b>-Enabled, the IP phone will automatically answer an incoming intercom call.</p> <p><b>Web User Interface:</b> Features-&gt;Intercom-&gt;Intercom Allow</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;Intercom-&gt;Intercom Allow</p>		
<b>features.intercom.mute</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the IP phone to mute the microphone when answering an intercom call.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, the microphone is muted for intercom calls, and then the other party cannot hear you.</p> <p><b>Note:</b> It works only if the values of the parameters "features.intercom.allow" and "features.allow_mute" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Features-&gt;Intercom-&gt;Intercom Mute</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;Intercom-&gt;Intercom Mute</p>		
<b>features.intercom.tone</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the IP phone to play a warning tone when answering an intercom call.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "features.intercom.allow" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Features-&gt;Intercom-&gt;Intercom Tone</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;Intercom-&gt;Intercom Tone</p>		
<b>features.intercom.barge</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the IP phone to answer an incoming intercom call while there is already an active call on the IP phone.</p>		



Parameters	Permitted Values	Default
<p><b>0</b>-Disabled, the IP phone will handle an incoming intercom call like a normal incoming call while there is already an active call on the IP phone.</p> <p><b>1</b>-Enabled, the IP phone will automatically answer the intercom call while there is already an active call on the IP phone and place the active call on hold.</p> <p><b>Note:</b> It works only if the values of parameters "features.intercom.allow" and "call_waiting.enable" are set to 1 (Enabled) and the value of the parameters "phone_setting.call_appearance.calls_per_linekey" is greater than 1.</p> <p><b>Web User Interface:</b> Features-&gt;Intercom-&gt;Intercom Barge</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;Intercom-&gt;Intercom Barge</p>		
<b>features.intercom.headset_prior.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Configures the channel mode to use when receiving an incoming intercom call.</p> <p><b>0</b>-Speaker Mode</p> <p><b>1</b>-Headset Mode, it works only if you connect the headset to the IP phone and the headset mode is activated for use.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		



**To configure intercom via web user interface:**

1. Click on **Features->Intercom**.
2. Select the desired values from the pull-down lists of **Intercom Allow**, **Intercom Mute**, **Intercom Tone** and **Intercom Barge**.



3. Click **Confirm** to accept the change.

**To configure intercom via phone user interface:**

1. Press **Menu**->**Features**->**Intercom**.
2. Press  or , or the **Switch** soft key to select the desired values from the **Intercom Allow**, **Intercom Mute**, **Intercom Tone** and **Intercom Barge** fields.
3. Press the **Save** soft key to accept the change.

## Call Timeout

Call timeout defines a specific period of time within which the IP phone will cancel the dialing if the call is not answered.

### Procedure

Call timeout can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the duration time in the ringback state.  <b>Parameter:</b> phone_setting.ringback_timeout
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### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
phone_setting.ringback_timeout	Integer from 0 to 3600	180
<b>Description:</b>  Configures the duration time (in seconds) in the ringback state.  If it is set to 180, the phone will cancel the dialing if the call is not answered within 180 seconds.  <b>Web User Interface:</b>  None  <b>Phone User Interface:</b>  None		

## Ringing Timeout

Ringing timeout defines a specific period of time within which the IP phone will stop ringing if the call is not answered.

## Procedure

Ringing timeout can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the duration time in the ringing state. <b>Parameter:</b> phone_setting.ringing_timeout
------------------------------------------------------	---------------------	---------------------------------------------------------------------------------------------------------

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
phone_setting.ringing_timeout	Integer from 0 to 3600	180
<b>Description:</b> Configures the duration time (in seconds) in the ringing state. If it is set to 180, the phone will stop ringing if the call is not answered within 180 seconds. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Send user=phone

When placing a call, the IP phone will send an INVITE request to the proxy server. Send user=phone feature allows adding user=phone to the SIP header of the INVITE message.

Example of a SIP INVITE message:

```
INVITE sip:101@10.2.1.48:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.3.20.6:5060;branch=z9hG4bK2475812834
From: "1010" <sip:1010@10.2.1.48:5060>;tag=3747068208
To: <sip:101@10.2.1.48:5060;user=phone>
Call-ID: 0_4008470062@10.3.20.6
CSeq: 1 INVITE
Contact: <sip:1010@10.3.20.6:5060>
Content-Type: application/sdp
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH,
UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Allow-Events: talk,hold,conference,refer,check-sync
```

Content-Length: 300

## Procedure

Send user=phone can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure send user=phone feature on a per-line basis. <b>Parameter:</b> account.X.enable_user_equal_phone
<b>Web User Interface</b>		Configure send user=phone feature on a per-line basis. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
<b>account.X.enable_user_equal_phone<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to add "user=phone" to the SIP header of the INVITE message for a specific account. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Account->Advanced->Send user=phone <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

### To configure send user=phone feature via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.

3. Select the desired value from the pull-down list of **Send user=phone**.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected. The 'Send user=phone' option is highlighted with a red box. The interface includes a sidebar with 'Register', 'Basic', 'Codec', and 'Advanced' sections. The main area displays various account settings with pull-down menus and text input fields. A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), Network Conference, and VQ-RTCPXR.

Account	Account1
Keep Alive Type	Default
Keep Alive Interval(Seconds)	30
RPort	Disabled
Subscribe Period(Seconds)	1800
DTMF Type	RFC2833
DTMF Info Type	DTMF-Relay
DTMF Payload Type(96~127)	101
Retransmission	Disabled
Subscribe Register	Disabled
Subscribe for MWI	Disabled
MWI Subscription Period(Seconds)	3600
Subscribe MWI To Voice Mail	Disabled
Voice Mail	
Voice Mail Display	Enabled
Caller ID Source	FROM
Session Timer	Disabled
Session Expires(30~7200s)	1800
Session Refresher	UAC
Send user=phone	Enabled

**NOTE**

**DTMF**  
It is the signal sent from the IP phone to the network, which is generated when pressing the IP phone's keypad during a call.

**Session Timer**  
It allows a periodic refresh of SIP sessions through a re-INVITE request, to determine whether a SIP session is still active.

**Busy Lamp Field/BLF List**  
Monitors a specific extension/a list of extensions for status changes on IP phones.

**Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA)**  
It allows users to share a SIP line on several IP phones. Any IP phone can be used to originate or receive calls on the shared line.

**Network Conference**  
It allows multiple participants (more than three) to join in a call.

**VQ-RTCPXR**  
The VQ-RTCPXR mechanism,

4. Click **Confirm** to accept the change.

## SIP Send MAC

The IP phone can send the MAC address in the REGISTER message. SIP send MAC allow adding "Mac: <PhoneMACAddress>" (for example, Mac: 00:15:65:74:b1:50) to the SIP header of the REGISTER message.

Example of a SIP REGISTER message:

```
REGISTER sip:10.2.1.48:5060 SIP/2.0
Via: SIP/2.0/UDP 10.3.20.14:5060;branch=z9hG4bK3593117201
From: "11" <sip:11@10.2.1.48:5060>;tag=2788360609
To: "11" <sip:11@10.2.1.48:5060>
Call-ID: 1_1863786852@10.3.20.14
CSeq: 2 REGISTER
Contact: <sip:11@10.3.20.14:5060;line=cc75882e976e208>
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH, UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Expires: 0
Allow-Events: talk,hold,conference,refer,check-sync
Mac: 00:15:65:74:b1:50
Content-Length: 0
```

## Procedure

SIP send MAC can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure SIP send MAC on a per-line basis. <b>Parameter:</b> account.X.register_mac
<b>Web User Interface</b>		Configure SIP send MAC on a per-line basis. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
<b>account.X.register_mac<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to add MAC address to the SIP header of the REGISTER message for a specific account. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Account->Advanced->SIP Send MAC <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## To configure SIP send MAC feature via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.

3. Select the desired value from the pull-down list of **SIP Send MAC**.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected. Under the 'Account' section, the 'SIP Send MAC' dropdown menu is highlighted with a red box, showing 'Enabled' as the selected value. Other settings include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), and 'Subscribe Period(Seconds)' (1800). A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, and Shared Call Appearance (SCA) / Bridge Line Appearance (BLA).

4. Click **Confirm** to accept the change.

## SIP Send Line

The IP phone can send the line number in the REGISTER message. SIP send line allow adding "Line:<linenumber>" (for example, Line: 1) to the SIP header of the REGISTER message. The line number is a number between 0 and 15.

The following table lists line number values for each phone model.

Phone Model	Line Number	Description
SIP-T54S/T48G/T48S/T46G/T46S/T29G	0~15	0~15 stand for line1~line16
SIP-T52S/T42G/T42S	0~11	0~11 stand for line1~line12
SIP-T41P/T41S/T27P/T27G	0~5	0~5 stand for line1~line6
SIP-T40P/T40G/T23P/T23G	0~2	0~2 stand for line1~line3
SIP-T21(P) E2	0~1	0~1 stand for line1~line2
SIP-T19(P) E2	0	0 stand for line1

Example of a SIP REGISTER message:

```
REGISTER sip:10.2.1.48:5060 SIP/2.0
Via: SIP/2.0/UDP 10.3.20.14:5060;branch=z9hG4bK3990593443
From: "11" <sip:11@10.2.1.48:5060>;tag=255071842
To: "11" <sip:11@10.2.1.48:5060>
Call-ID: 1_2369214377@10.3.20.14
CSeq: 2 REGISTER
```

Contact: <sip:11@10.3.20.14:5060;line=1da6aa8d7254654>  
 Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH, UPDATE, MESSAGE  
 Max-Forwards: 70  
 User-Agent: Yealink SIP-T46G 28.82.0.20  
 Expires: 0  
 Allow-Events: talk,hold,conference,refer,check-sync  
 Line: 1  
 Content-Length: 0

## Procedure

SIP send line can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure SIP send line on a per-line basis. <b>Parameter:</b> account.X.register_line
<b>Web User Interface</b>		Configure SIP send line on a per-line basis. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
<b>account.X.register_line<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to add line number to the SIP header of the REGISTER message for a specific account. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Account->Advanced->SIP Send Line <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.



To configure SIP send Line feature via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **SIP Send Line**.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected, and the 'Advanced' sub-tab is active. The 'Account' dropdown is set to 'Account 1'. The 'SIP Send Line' dropdown is highlighted with a red box and set to 'Enabled'. Other settings include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), and 'Subscribe Period(Seconds)' (1800). A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, and Shared Call Appearance (SCA)/Bridge Line Appearance (BLA).

4. Click **Confirm** to accept the change.

## Reserve # in User Name

Reserve # in User Name feature allows IP phones to reserve “#” in user name. When Reserve # in User Name feature is disabled, “#” will be converted into “%23”. For example, the user registers an account (user name: 1010#) on the phone, the phone will send 1010%23 instead of 1010# in the REGISTER message or INVITE message to SIP server.

Example of a SIP REGISTER message:

```
INVITE sip:2@10.2.1.48:5060 SIP/2.0
Via: SIP/2.0/UDP 10.3.20.6:5060;branch=z9hG4bK1867789050
From: "1010" <sip:1010%23@10.2.1.48:5060>;tag=1945988802
To: <sip:2@10.2.1.48:5060>
Call-ID: 0_2336101648@10.3.20.6
CSeq: 1 INVITE
Contact: <sip:1010%23@10.3.20.6:5060>
Content-Type: application/sdp
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH,
UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Allow-Events: talk,hold,conference,refer,check-sync
Content-Length: 300
```

## Procedure

Reserve # in User Name can be configured using the following methods.

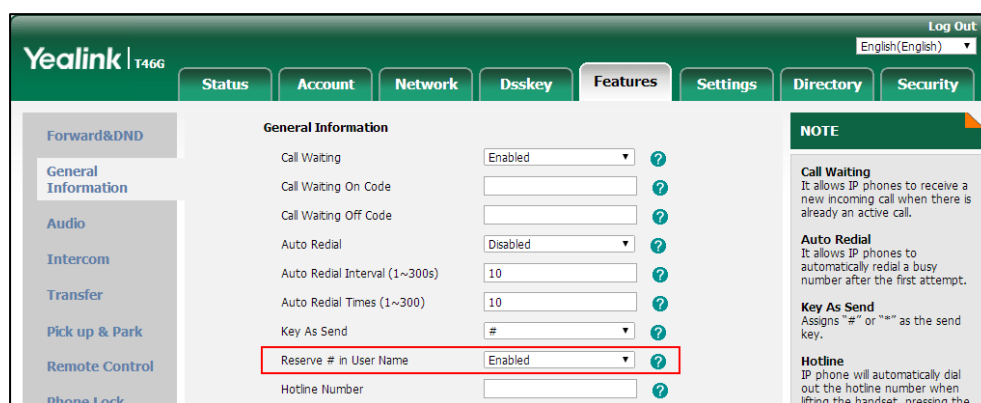
<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure reserve # in user name. <b>Parameter:</b> sip.use_23_as_pound
<b>Web User Interface</b>		Configure reserve # in user name. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =features-general&q=load

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
<b>sip.use_23_as_pound</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to reserve the pound sign (#) in the user name.</p> <p><b>0</b>-Disabled (convert the pound sign into "%23")</p> <p><b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Reserve # in User Name</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure reserve # in user name feature via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Reserve # in User Name**.



3. Click **Confirm** to accept the change.

## Password Dial

Password dial feature allows the callee number to be partly displayed on the IP phone when placing a call. The hidden digits are displayed as asterisks on the LCD screen. The number in placed call list is also partly displayed on the IP phone. This feature is especially useful for users who are always placing important and confidential calls.

### Procedure

Password dial feature can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure password dial feature. <b>Parameters:</b> features.password_dial.enable features.password_dial.prefix features.password_dial.length
<b>Web User Interface</b>		Configure password dial feature. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?parameters=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?parameters=features-general&amp;q=load</a>

### Details of the Configuration Parameters:

Parameters	Permitted Values	Default
<b>features.password_dial.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to partly display the callee number when placing a call. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Features->General Information->PswDial <b>Phone User Interface:</b> None		
<b>features.password_dial.prefix</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the prefix of the number that needs to be partly displayed. <b>Example:</b>		

Parameters	Permitted Values	Default
<p>features.password_dial.prefix = 12</p> <p><b>Note:</b> It works only if the value of the parameter "features.password_dial.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;PswPrefix</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.password_dial.length</b>	<b>Integer from 0 to 99</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configure how many digits to be displayed as asterisks.</p> <p><b>Example:</b></p> <p>features.password_dial.length = 3</p> <p>If you set the prefix to 12 (configured by the parameter "features.password_dial.prefix") and the length to 3, when you want to dial the number 123456, the entered number is displayed as 12***6 on the LCD screen.</p> <p><b>Note:</b> It works only if the value of the parameter "features.password_dial.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;PswLength</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure password dial feature via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **PswDial**.
3. Enter the prefix of password dial in the **PswPrefix** field.

4. Enter the desired number of hidden digits in the **PswLength** field.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' section is expanded. The 'PswLength' field is highlighted with a red box and contains the value '3'. Other fields in the 'General Information' section include 'Call Waiting' (Enabled), 'Call Waiting On Code', 'Call Waiting Off Code', 'Auto Redial' (Disabled), 'Auto Redial Interval (1~300s)' (10), 'Auto Redial Times (1~300)' (10), '180 Ring Workaround' (Enabled), 'Logon Wizard' (Enabled), 'PswPrefix' (12), 'PswDial' (Enabled), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right provides details for 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion'.

5. Click **Confirm** to accept the change.

## Unregister When Reboot

Unregister when reboot feature allows IP phones to unregister first before re-registering the account when finishing a reboot.

### Procedure

Unregister when reboot can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure unregister when reboot. <b>Parameter:</b> account.X.unregister_on_reboot
<b>Web User Interface</b>		Configure unregister when reboot. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
account.X.unregister_on_reboot <sup>[1]</sup>	0 or 1	0
<b>Description:</b> Enables or disables the IP phone to unregister first before re-registering account X when		

Parameter	Permitted Values	Default
finishing a reboot. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Account->Advanced->Unregister When Reboot <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

#### To configure unregister when reboot via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Unregister When Reboot**.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected, and 'Account 1' is chosen from the dropdown. The 'Advanced' sub-tab is active. The 'Unregister When Reboot' setting is highlighted with a red rectangle and is set to 'Enabled'. Other settings are visible, including 'Keep Alive Type', 'Keep Alive Interval', 'RPort', 'Subscribe Period', 'Early Media', 'SIP Server Type', 'Music Server URI', 'Directed Call Pickup Code', 'Group Call Pickup Code', 'Distinctive Ring Tones', 'Out Dialog BLF', 'VQ RTPC-XR Collector Name', 'VQ RTPC-XR Collector Address', and 'VQ RTPC-XR Collector Port'. A 'NOTE' section on the right provides additional information about various features.

4. Click **Confirm** to accept the change.

## 100 Reliable Retransmission

As described in [RFC 3262](#), 100rel tag is for reliability of provisional responses. When present in a Supported header, it indicates that the IP phone can send or receive reliable provisional responses. When present in a Require header in a reliable provisional response, it indicates that

the response is to be sent reliably.

Example of a SIP INVITE message:

```
INVITE sip:1024@pbx.test.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.3.6.197:5060;branch=z9hG4bK1708689023
From: "1025" <sip:1025@pbx.test.com:5060>;tag=1622206783
To: <sip:1024@pbx.test.com:5060>
Call-ID: 0_537569052@10.3.6.197
CSeq: 2 INVITE
Contact: <sip:1025@10.3.6.197:5060>
Authorization: Digest username="1025", realm="pbx.test.com", nonce="BroadWorksXi5stub71Ts2nb05BW",
uri="sip:1024@pbx.test.com:5060", response="f7e9d35c55af45b3f89beae95e913171", algorithm=MD5,
cnonce="0a4f113b", qop=auth, nc=00000001
Content-Type: application/sdp
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH,
UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Supported: 100rel
Allow-Events: talk,hold,conference,refer,check-sync
Content-Length: 302
```

Procedure

100 Reliable Retransmission can be configured using the following methods.

Central Provisioning (Configuration File)	<MAC>.cfg	Configure the 100 reliable retransmission feature. <b>Parameter:</b> account.X.100rel_enable
Web User Interface		Configure the 100 reliable retransmission feature. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

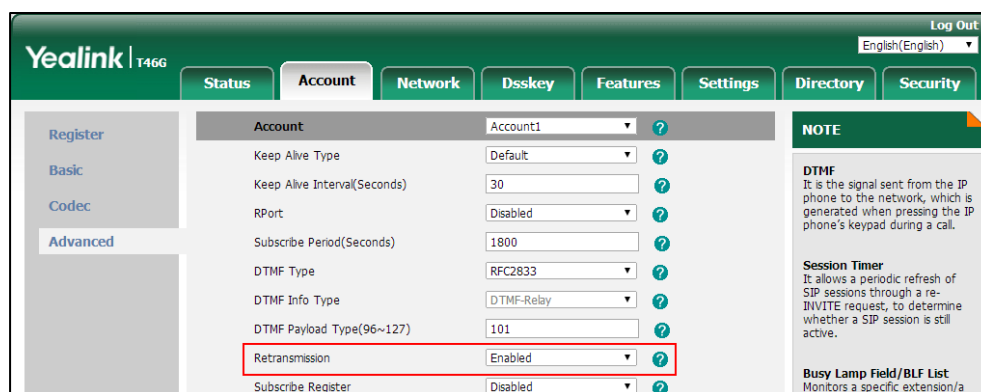
**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>account.X.100rel_enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the 100 reliable retransmission feature for a specific account.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;Retransmission</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure 100 reliable retransmission via web user interface:**

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Retransmission**.



4. Click **Confirm** to accept the change.

## Reboot in Talking

Reboot in talking feature allows IP phones to reboot during an active call when it receives a reboot request by action URI. For more information on action URI, refer to [Action URI](#) on page 641.

IP phones do not receive and handle HTTP/HTTPS GET requests by default. To use this feature,



you need to specify the trusted IP address(es) for action URI in advance. For more information, refer to [Configuring Trusted IP Address for Action URI](#) on page 645.

## Procedure

Reboot in talking can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure reboot in talking. <b>Parameter:</b> features.reboot_in_talk_enable
<b>Web User Interface</b>		Configure reboot in talking. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-general&q=load

## Details of Configuration Parameter:

Parameter	Permitted Values	Default
features.reboot_in_talk_enable	0 or 1	0
<b>Description:</b> Enables or disables the phone to reboot during a call when it receives a reboot request by action URI. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "features.action_uri_limit_ip" is set to "any" or trusted IP address(es) and it is not the first time for the IP phone to receive HTTP/HTTPS GET request from the trusted IP address(es). <b>Web User Interface:</b> Features->General Information->Reboot in Talking <b>Phone User Interface:</b> None		

## To configure reboot in talking via web user interface:

1. Click on **Features->General Information**.

2. Select the desired value from the pull-down list of **Reboot in Talking** field.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected. In the 'General Information' section, the 'Reboot in Talking' field is highlighted with a red box and set to 'Enabled'. Other fields include Call Waiting (Enabled), Call Waiting On Code, Call Waiting Off Code, Auto Redial (Disabled), Auto Redial Interval (1~300s) (10), Auto Redial Times (1~300) (10), Voice Mail Tone (Enabled), DHCP Hostname (SIP-T46G), Hide Feature Access Codes (Disabled), Display Method on Dialing (User Name), and Auto Linekeys (Disabled). A 'NOTE' section on the right provides details about various features: Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

3. Click **Confirm** to accept the change.

A dialog box pops up to prompt that the settings will take effect after a reboot.

4. Click **OK** to reboot the phone.

## Answer By Hand

Answer by hand feature allows you to answer an incoming call by picking up the handset, pressing the Speakerphone key or pressing the HEADSET key directly.

If you disable answer by hand feature, you need to press the corresponding line key or the **Answer** soft key to answer an incoming call after picking up the handset, pressing the Speakerphone key or pressing the HEADSET key.

### Procedure

Answer by hand can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure answer by hand. <b>Parameter:</b> features.off_hook_answer.enable
------------------------------------------------------	---------------------	-----------------------------------------------------------------------------------

### Details of Configuration Parameter:

Parameter	Permitted Values	Default
features.off_hook_answer.enable	0 or 1	1
<b>Description:</b> Enables or disables the IP phone to answer an incoming call by picking up the handset,		

Parameter	Permitted Values	Default
<p>pressing the Speakerphone key or pressing the HEADSET key directly.</p> <p><b>0</b>-Disabled, you need to press the corresponding line key, the <b>Answer</b> soft key or the OK key to answer an incoming call after picking up the handset, pressing the Speakerphone key or pressing the HEADSET key.</p> <p><b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## USB Recording

Yealink IP phones support recording during a call. Before recording, ensure that the USB flash drive has been inserted into the USB port of the phone. Call recording is only applicable to SIP-T54S, SIP-T52S, SIP-T48G/S, SIP-T46G/S, SIP-T42S, SIP-T41S, SIP-T29G and SIP-T27G IP phones.

You need to press the **Start REC** soft key during a call to record the audio call or conference. Yealink IP phones also support recording calls by pressing record/URL record DSS key. For more information, refer to [Dsskey Recording \(Record and URL Record\)](#) on page 607.

### Note

Before recording any call, especially those involving PSTN, it is necessary to know about the rules and restrictions of any governing call-recording in the place where you are. It is also very important to have the consent of the person you are calling before recording the conversation.

The recorded calls are saved in \*.wav format and include a date/time stamp, the other party's number/IP address/name (or the first person's number/IP address/name you called), duration of the call and the recording file size. For example, 20160422-1515-Bob was created on Apr. 22, 2016, at 15:15 and you have a call with Bob. Recorded calls can be played on either the phone itself or on a computer using an application capable of playing \*.wav files.

For more information, refer to [Yealink phone-specific user guide](#).

### Procedure

USB Recording feature can be only configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx> .cfg	<p>Configure the call recording (using a USB flash drive) feature on a phone basis.</p> <p><b>Parameter:</b></p> <p>features.usb_call_recording.enable</p>
------------------------------------------------------	-------------------------	------------------------------------------------------------------------------------------------------------------------------------------------------------

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.usb_call_recording.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the call recording (using a USB flash drive) feature for the IP phone.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, you can record the active audio call for the phone by pressing the <b>Start REC</b> soft key, and the recorded calls will be saved to the USB flash drive.</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## CSTA Control

User Agent Computer Supported Telecommunications Applications (uaCSTA) is explained in detail in [Using CSTA for SIP Phone User Agents \(uaCSTA\)](#) and [Services for Computer Supported Telecommunications Applications Phase III](#).

The uaCSTA feature on the phone may be used for remote control of the phone from computer applications such as PC softphone. You can use the application to control the phone to perform basic call operations. For example, place a call, answer a call, end a call and transfer a call to another party.

It is not applicable to SIP-T19(P) E2 IP phones.

**Procedure**

The uaCSTA feature can be configured using the following methods.

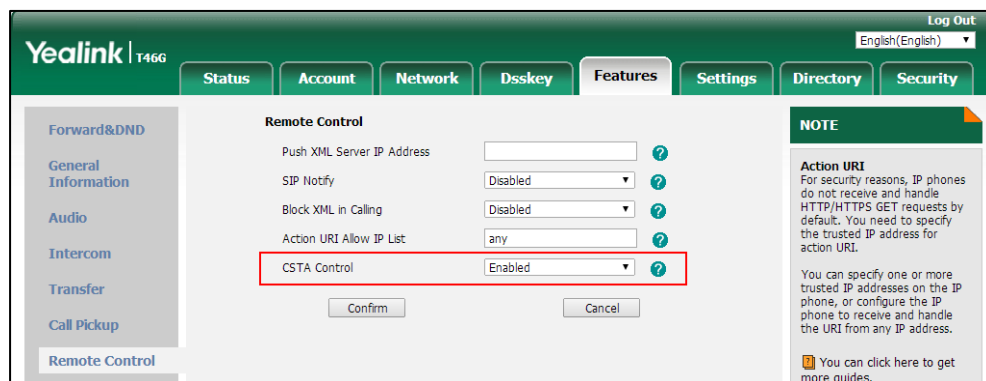
<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure uaCSTA feature. <b>Parameter:</b> features.csta_control.enable
<b>Web User Interface</b>		Configure uaCSTA feature. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-remotecontrol&q=load

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.csta_control.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the uaCSTA (User Agent Computer Supported Telecommunications Applications) feature on the IP phone.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Remote Control-&gt;CSTA Control</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure uaCSTA feature via web user interface:**

1. Click on **Features->Remote Control**.
2. Select the desired value from the pull-down list of the **CSTA Control**.

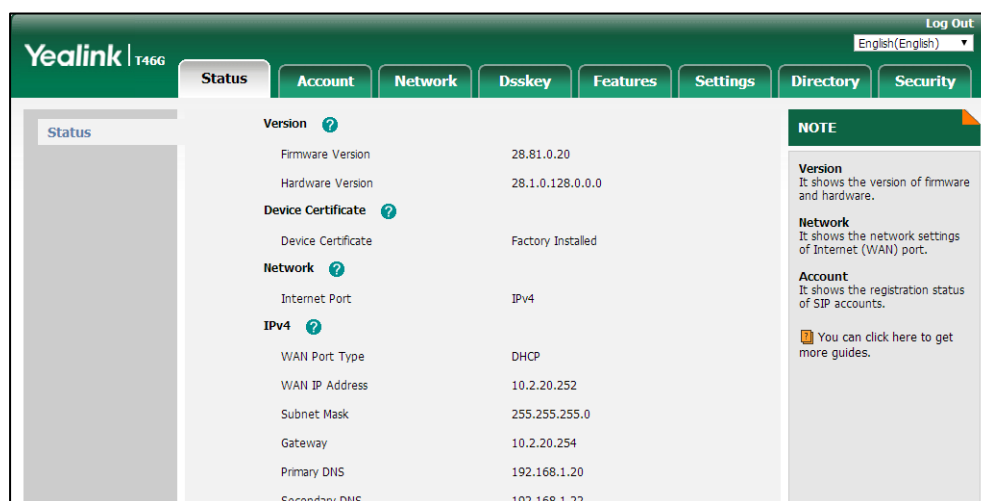


3. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
4. Click **OK** to reboot the phone.

## Quick Login

Quick login feature allows users to fast access to web user interface using the request URI "https://username:password@phoneIPAddress" (for example, https://admin:admin@192.168.0.10). You will navigate to the **Status** web page after accessing

the web user interface. It is helpful for users to quickly log into the web user interface without entering the username and password in the login page.



**Note** The use of the quick login feature may be restricted by the web explorer (e.g., Internet Explorer). For security purposes, we recommend you to use this feature in a secure network environment.

## Procedure

Quick login can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure quick login. <b>Parameter:</b> wui.quick_login
------------------------------------------------------	---------------------	----------------------------------------------------------------

## Details of the Configuration Parameter:

Parameter	Permitted Values	Default
<b>wui.quick_login</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the quick login feature. <b>0</b> -Disabled <b>1</b> -Enabled, you can quickly log into the web user interface using a request URI (for example, https://admin:admin@192.168.0.10). <b>Note:</b> It works only if the value of the parameter "static.wui.https_enable" is set to 1 (Enabled). <b>Web User Interface:</b> None		

Parameter	Permitted Values	Default
<b>Phone User Interface:</b> None		

## CFG File Version Information Customization

You can customize the version information for the CFG configuration file and then check the version information via phone/web user interface. With this feature, you can easily know which version of the CFG configuration file the IP phone is configured.

### Procedure

CFG file version information can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the CFG configuration file version information.  <b>Parameter:</b> features.custom_version_info
------------------------------------------------------	---------------------	--------------------------------------------------------------------------------------------------------------------

### Details of Configuration Parameter:

Parameter	Permitted Values	Default
features.custom_version_info	String	Blank
<b>Description:</b> Configures the version information of the CFG configuration file. After configuration, you can check the configuration file version information at the path: <b>Menu-&gt;Status-&gt;CFG Version</b> (phone user interface) or <b>Status-&gt;Status-&gt;Version-&gt;Configuration Version</b> (web user interface). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		





## Configuring Advanced Features

---

This chapter provides information for making configuration changes for the following advanced features:

- [Remote Phone Book](#)
- [Lightweight Directory Access Protocol \(LDAP\)](#)
- [Busy Lamp Field \(BLF\)](#)
- [Busy Lamp Field \(BLF\) List](#)
- [Hide Feature Access Codes](#)
- [Automatic Call Distribution \(ACD\)](#)
- [Shared Call Appearance \(SCA\)](#)
- [Bridge Lines Appearance \(BLA\)](#)
- [Message Waiting Indicator \(MWI\)](#)
- [Short Message Service \(SMS\)](#)
- [Multicast Paging](#)
- [Dsskey Recording \(Record and URL Record\)](#)
- [Hot Desking](#)
- [Logon Wizard](#)
- [Action URL](#)
- [Action URI](#)
- [Server Redundancy](#)
- [Static DNS Cache](#)
- [Real-Time Transport Protocol \(RTP\) Ports](#)
- [TR-069 Device Management](#)
- [XML Browser](#)
- [Enhanced DSS Keys](#)

## Remote Phone Book

Remote phone book is a centrally maintained phone book, stored on the remote server. Users only need the access URL of the remote phone book. The IP phone can establish a connection with the remote server and download the phone book, and then display the remote phone book entries on the phone user interface. IP phones support up to 5 remote phone books. Remote phone book is customizable.

**Note** For SIP-T19(P) E2/T21(P) E2/T23P/T23G/T40P/T40G IP phones, we recommend you to download less than 3000 remote contacts from the remote server. For other IP phones, we recommend you to download less than 5000 remote contacts from the remote server.

## Customizing Remote Phone Book Template File

You can customize the remote phone book for IP phones as required. You can also add multiple remote contacts at a time and/or share remote contacts between IP phones using the supplied template files (Menu.xml and Department.xml). The Menu.xml file defines departments of a remote phone book. The Department.xml file defines contact lists for a department, which is nested in Menu.xml file. After setup, place the files (Menu.xml and Department.xml) to the provisioning server, and specify the access URL of the file (Menu.xml) in the configuration file.

You can ask the distributor or Yealink FAE for remote XML phone book template. You can also obtain the remote XML phone book template online:

<http://support.yealink.com/documentFront/forwardToDocumentFrontDisplayPage>. For more information on obtaining the remote phone book template, refer to [Obtaining Boot Files/Configuration Files/Resource Files](#) on page 105.

When creating a Department.xml file, learn the following:

- `<YealinkIPPhoneDirectory>` indicates the start of a department file and `</YealinkIPPhoneDirectory>` indicates the end of a department file.
- Create contact lists for a department between `<DirectoryEntry>` and `</DirectoryEntry>`.

### To customize a Department.xml file:

1. Open the template file using an ASCII editor.
2. For each contact that you want to add, add the following strings to the file. Each starts on a separate line:

```
<Name> Test1</Name>
<Telephone> 23000</Telephone>
```

Where:

Specify the contact name between <Name> and </Name>.

Specify the contact number between <Telephone> and </Telephone>.

```

1 <YealinkIPPhoneDirectory>
2
3
4 <DirectoryEntry>
5   <Name>Test1</Name>
6   <Telephone>23000</Telephone>
7 </DirectoryEntry>
8
9
10 <DirectoryEntry>
11   <Name>Test2</Name>
12   <Telephone>303</Telephone>
13   <Telephone>915980830849</Telephone>
14 </DirectoryEntry>
15
16
17 <DirectoryEntry>
18   <Name>Test3</Name>
19   <Telephone>6650</Telephone>
20   <Telephone>915980830849</Telephone>
21 </DirectoryEntry>
22
23
24 </YealinkIPPhoneDirectory>

```

3. Save the file and place this file to the provisioning server.

When creating a Menu.xml file, learn the following:

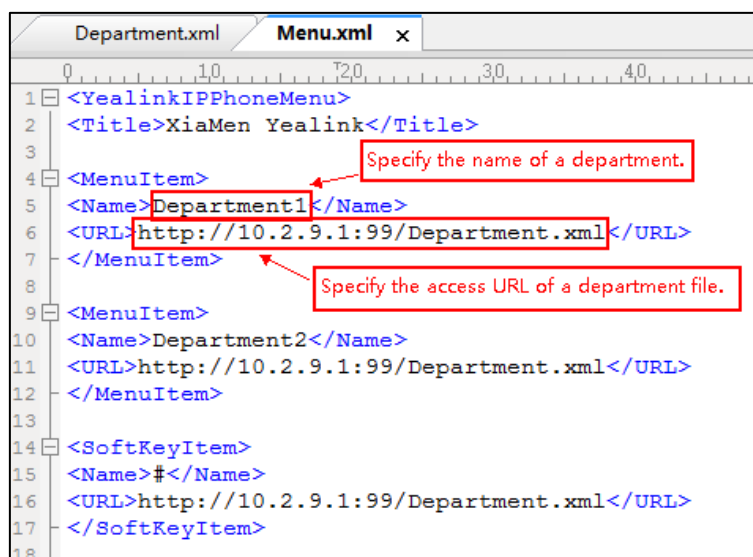
- <YealinkIPPhoneMenu> indicates the start of a remote phone book file and </YealinkIPPhoneMenu> indicates the end of a remote phone book file.
- Create the title of a remote phone book between <Title> and </Title>.
- <MenuItem> indicates the start of specifying a department file and </MenuItem> indicates the end of specifying a department file.
- <SoftKeyItem> indicates the start of specifying an XML file and </SoftKeyItem> indicates the end of specifying an XML file for the digit keys, # key or \* key. In the remote phone book contacts screen, pressing the configured digit keys/# key/\* key can access the subdirectory. If not configured, the LCD screen displays "URL is empty" when pressing the desired digit keys, # key or \* key.

#### To customize a Menu.xml file:

1. Open the template file using an ASCII editor.

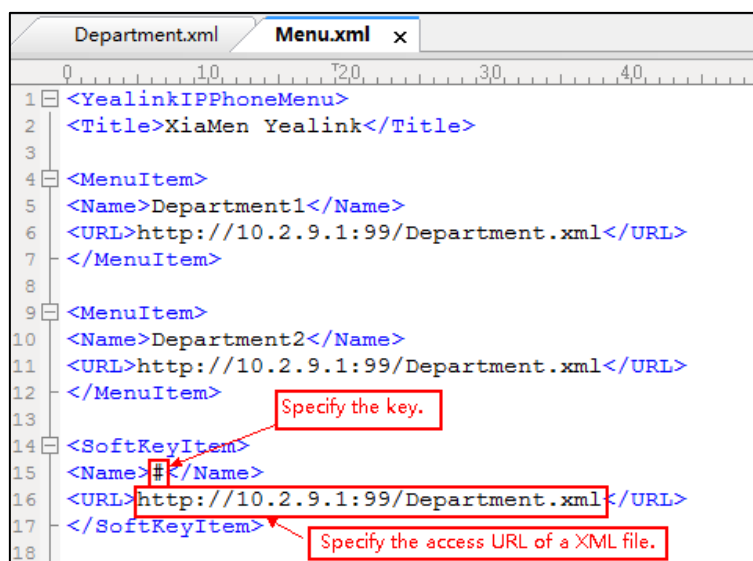
- For each department that you want to add, add the following strings to the file. Each starts on a separate line:

```
<MenuItem>
<Name> Department1</Name>
<URL> http://10.2.9.1:99/Department.xml</URL>
</MenuItem>
```



- For each XML file that you want to add, add the following strings to the file. Each starts on a separate line:

```
<SoftKeyItem>
<Name> #</Name>
<URL> http://10.2.9.1:99/Department.xml</URL>
</SoftKeyItem>
```



4. Save the file and place this file to the provisioning server.
5. Specify the access URL of the remote phone book (remote\_phonebook.data.1.url = http://192.168.1.20/Menu.xml).

During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.20", and downloads the remote phone book file "Menu.xml".

#### Note

Yealink supplies a phonebook generation tool to generate a remote XML phone book. For more information, refer to [Yealink Phonebook Generation Tool User Guide](#).

Incoming/Outgoing Call Lookup allows IP phones to search the entry names from the remote phone book for incoming/outgoing calls. Update Time Interval specifies how often IP phones refresh the local cache of the remote phone book.

### Procedure

Remote phone book can be configured using the following methods.

Central Provisioning (Configuration File)	<y0000000000xx>.cfg	<p>Specify the access URL and the display name of the remote phone book.</p> <p><b>Parameters:</b></p> <p>remote_phonebook.data.X.url</p> <p>remote_phonebook.data.X.name</p> <p>remote_phonebook.display_name</p>
		<p>Specify whether to query the entry name from the remote phone book for outgoing/incoming calls.</p> <p><b>Parameter:</b></p> <p>features.remote_phonebook.enable</p>
		<p>Specify how often the IP phone refreshes the local cache of the remote phone book.</p> <p><b>Parameter:</b></p> <p>features.remote_phonebook.flash_time</p>
		<p>Specify whether to refresh the local cache of the remote phone book at a time when accessing the remote phone book.</p> <p><b>Parameter:</b></p> <p>features.remote_phonebook.enter_update_enable</p>
Web User Interface		<p>Specify the access URL and the display name of the remote phone book.</p> <p>Specify whether to query the entry name from</p>

	<p>the remote phone book for outgoing/incoming calls.</p> <p>Specify how often the IP phone refreshes the local cache of the remote phone book.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=contacts-remote&amp;q=load</p>
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### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>remote_phonebook.data.X.url</b> (X ranges from 1 to 5)	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the access URL of the remote phone book.</p> <p><b>Example:</b></p> <p>remote_phonebook.data.1.url = http://192.168.1.20/phonebook.xml</p> <p><b>Note:</b> For T19(P) E2/T21(P) E2/T23P/T23G/T40P/T40G IP phones, the size of a remote phone book file should be less than 750K. For T27P/T27G/T41P/T41S/T42G/T42S/T46G/T46S/T48G/T48S IP phones, the size of a remote phone book file should be less than 1.5M.</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;Remote Phone Book-&gt;Remote URL</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>remote_phonebook.data.X.name</b> (X ranges from 1 to 5)	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the display name of the remote phone book item.</p> <p><b>Example:</b></p> <p>remote_phonebook.data.1.name = Xmyl</p> <p>"Xmyl" will be displayed on the LCD screen at the phone path <b>Menu-&gt;Directory-&gt;Remote Phone Book</b>. The name of <b>Remote Phone Book</b> can be configured by the parameter "remote_phonebook.display_name".</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;Remote Phone Book-&gt;Display Name</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

Parameters	Permitted Values	Default
<b>remote_phonebook.display_name</b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the display name of the remote phone book.</p> <p><b>Example:</b> remote_phonebook.display_name = Friends "Friends" will be displayed on the LCD screen at the phone path <b>Menu-&gt;Directory</b>. If it is left blank, "Remote Phone Book" will be displayed on the LCD screen at the phone path: <b>Menu-&gt;Directory</b>.</p> <p><b>Note:</b> It is not applicable to SIP-T42G/T42S/T41P/T41S IP phones.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>features.remote_phonebook.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the IP phone to perform a remote phone book search for an incoming or outgoing call and display the matched results on the LCD screen.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Web User Interface:</b> Directory-&gt;Remote Phone Book-&gt;Incoming/Outgoing Call Lookup</p> <p><b>Phone User Interface:</b> None</p>		
<b>features.remote_phonebook.flash_time</b>	<b>0, Integer from 3600 to 1296000</b>	<b>21600</b>
<p><b>Description:</b> It configures how often to refresh the local cache of the remote phone book. If it is set to 3600, the IP phone will refresh the local cache of the remote phone book every 3600 seconds (1 minute). If it is set to 0, the IP phone will not refresh the local cache of the remote phone book.</p> <p><b>Web User Interface:</b> Directory-&gt;Remote Phone Book-&gt;Update Time Interval(Seconds)</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>features.remote_phonebook.enter_update_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to refresh the local cache of the remote phone book at a time when accessing the remote phone book. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

To specify access URL of the remote phone book via web user interface:

1. Click on **Directory->Remote Phone Book**.
2. Enter the access URL in the **Remote URL** field.
3. Enter the name in the **Display Name** field.

The screenshot shows the Yealink T46G web interface. The 'Directory' tab is selected, and the 'Remote Phone Book' sub-tab is active. A table lists five entries for remote phone books. The first entry is highlighted with a red box, showing 'Index' 1, 'Remote URL' 'http://192.168.1.20/phonebook.xml', and 'Display Name' 'Xmyl'. Below the table, there are two settings: 'Incoming/Outgoing Call Lookup' set to 'Enabled' and 'Update Time Interval(Seconds)' set to '21600'. A 'NOTE' box on the right states: 'Remote Phone Book It is a centrally maintained phone book, stored on the remote server. Users only need the access URL of the remote phone book. The IP phone can establish a connection with the remote server and download the phone book, and then display the remote phone book entries on the phone user interface.' There are 'Confirm' and 'Cancel' buttons at the bottom.

4. Click **Confirm** to accept the change.

To configure incoming/outgoing call lookup and update time interval via web user interface:

1. Click on **Directory->Remote Phone Book**.
2. Select the desired value from the pull-down list of **Incoming/Outgoing Call Lookup**.



3. Enter the desired time in the **Update Time Interval(Seconds)** field.

The screenshot shows the Yealink 1466 web interface. The 'Directory' tab is selected. A table lists remote phone books with columns for Index, Remote URL, and Display Name. Below the table, the 'Incoming/Outgoing Call Lookup' is set to 'Enabled' and the 'Update Time Interval(Seconds)' is set to '21600'. A red box highlights these two settings. To the right, a 'NOTE' section explains the Remote Phone Book feature.

Index	Remote URL	Display Name
1	http://192.168.1.20/phonebook.xml	Xmyl
2		
3		
4		
5		

NOTE

**Remote Phone Book**  
It is a centrally maintained phone book, stored on the remote server.  
Users only need the access URL of the remote phone book. The IP phone can establish a connection with the remote server and download the phone book, and then display the remote phone book entries on the phone user interface.

You can click here to get more guides.

4. Click **Confirm** to accept the change.

## Lightweight Directory Access Protocol (LDAP)

LDAP is an application protocol for accessing and maintaining information services for the distributed directory over an IP network. IP phones can be configured to interface with a corporate directory server that supports LDAP version 2 or 3. The following LDAP servers are supported:

- Microsoft Active Directory
- Sun ONE Directory Server
- Open LDAP Directory Server
- Microsoft Active Directory Application Mode (ADAM)

The biggest plus for LDAP is that users can access the central LDAP directory of the corporation using IP phones. Therefore they do not have to maintain the directory locally. Users can search and dial out from the LDAP directory, and save LDAP entries to the local directory. LDAP entries displayed on the IP phone are read only, which cannot be added, edited or deleted by users. When an LDAP server is properly configured, the IP phone can look up entries from the LDAP server in a wide variety of ways. The LDAP server indexes all the data in its entries, and "filters" can be used to select the desired entry or group, and return the desired information.

Configurations on the IP phone limit the amount of the displayed entries when querying from the LDAP server, and decide how attributes are displayed and sorted.

You can set a DSS key to be an LDAP DSS key, and then press the LDAP DSS key to enter the LDAP search screen when the IP phone is idle.

## LDAP Attributes

The following table lists the most common attributes used to configure the LDAP lookup on IP phones.

Abbreviation	Name	Description
gn	givenName	First name
cn	commonName	LDAP attribute is made up from given name joined to surname.
sn	surname	Last name or family name
dn	distinguishedName	Unique identifier for each entry
dc	dc	Domain component
-	company	Company or organization name
-	telephoneNumber	Office phone number
mobile	mobilephoneNumber	Mobile or cellular phone number
ipPhone	IPphoneNumber	Home phone number

For more information on LDAP, refer to [LDAP Directory on Yealink IP Phones](#).

## Procedure

LDAP can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure LDAP. <b>Parameters:</b> ldap.enable ldap.search_type ldap.name_filter ldap.number_filter ldap.tls_mode ldap.host ldap.port ldap.base ldap.user ldap.password ldap.max_hits ldap.name_attr ldap.numb_attr ldap.display_name ldap.version
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		ldap.call_in_lookup ldap.call_out_lookup ldap.ldap_sort ldap.incoming_call_special_search.enable ldap.numb_display_mode ldap.customize_label
		Assign an LDAP DSS key. <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Configure LDAP. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?pname=contacts-LDAP&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?pname=contacts-LDAP&amp;q=load</a>
		Assign an LDAP DSS key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?pname=dsskey&amp;q=load&amp;model=0">http://&lt;phoneIPAddress&gt;/servlet?pname=dsskey&amp;q=load&amp;model=0</a>
<b>Phone User Interface</b>		Assign an LDAP DSS key.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>ldap.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables LDAP feature on the IP phone. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Directory->LDAP->Enable LDAP		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>ldap.search_type</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Configures the search type for LDAP contact look up. <b>0</b> -Prefix matching, the IP phone will search the LDAP contact numbers or names start with the entered character(s). <b>1</b> -Approximate string matching, the IP phone will search the LDAP contact numbers or names contain the entered character(s). <b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>ldap.name_filter</b>	<b>String within 99 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the search criteria for looking up LDAP contact names. The "*" symbol in the filter stands for any character. The "%" symbol in the filter stands for the name entered by the user. <b>Example:</b> ldap.name_filter = ( (cn=*)(sn=*)) When the cn or sn of the LDAP contact matches the entered name, the record will be displayed on the LCD screen. ldap.name_filter = (&(cn=*)(sn=*)) When the cn of the LDAP contact is set and the sn of the LDAP contact matches the entered name, the records will be displayed on the phone LCD screen. ldap.name_filter = (!(cn=*)) When the cn of the LDAP contact does not match the entered name, the records will be displayed on the phone LCD screen. <b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Directory->LDAP->LDAP Name Filter <b>Phone User Interface:</b>		

Parameters	Permitted Values	Default
None		
<b>ldap.number_filter</b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the search criteria for looking up LDAP contact numbers.</p> <p>The "*" symbol in the filter stands for any number. The "%" symbol in the filter stands for the number entered by the user.</p> <p><b>Example:</b></p> <p>ldap.number_filter = (&amp;(telephoneNumber=%)(mobile=%)(ipPhone=%))</p> <p>When the number of the telephoneNumber, mobile or ipPhone of the contact record matches the search criteria, the record will be displayed on the LCD screen.</p> <p>ldap.number_filter = (&amp;(telephoneNumber=*)(mobile=%))</p> <p>When the telephoneNumber of the LDAP contact is set and the mobile of the LDAP contact matches the entered number, the record will be displayed on the phone LCD screen.</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;LDAP-&gt;LDAP Number Filter</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>ldap.tls_mode</b>	<b>0, 1 or 2</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the connection mode between the LDAP server and the IP phone.</p> <p><b>0</b>-LDAP-Unencrypted connection between LDAP server and the IP phone (port 389 is used by default).</p> <p><b>1</b>-LDAP TLS Start-TLS/SSL connection between LDAP server and the IP phone (port 389 is used by default).</p> <p><b>2</b>-LDAPS-TLS/SSL connection between LDAP server and the IP phone (port 636 is used by default).</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;LDAP-&gt;LDAP TLS Mode</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

Parameters	Permitted Values	Default
<b>ldap.host</b>	<b>IP address or domain name</b>	<b>Blank</b>
<p><b>Description:</b> Configures the IP address or domain name of the LDAP server.</p> <p><b>Example:</b> ldap.host = 10.2.1.55</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Directory-&gt;LDAP-&gt;Server Address</p> <p><b>Phone User Interface:</b> None</p>		
<b>ldap.port</b>	<b>Integer from 1 to 65535</b>	<b>389</b>
<p><b>Description:</b> Configures the port of the LDAP server.</p> <p><b>Example:</b> ldap.port = 389</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Directory-&gt;LDAP-&gt;Port</p> <p><b>Phone User Interface:</b> None</p>		
<b>ldap.base</b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the LDAP search base which corresponds to the location of the LDAP phone book from which the LDAP search request begins. The search base narrows the search scope and decreases directory search time.</p> <p><b>Example:</b> ldap.base = dc=yealink,dc=cn</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Directory-&gt;LDAP-&gt;Base</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
None		
<b>ldap.user</b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the user name used to login the LDAP server.</p> <p>This parameter can be left blank in case the server allows anonymous to login. Otherwise you will need to provide the user name to login the LDAP server.</p> <p><b>Example:</b></p> <p>ldap.user = cn=manager,dc=yealink,dc=cn</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;LDAP-&gt;Username</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>ldap.password</b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the password used to login the LDAP server.</p> <p>This parameter can be left blank in case the server allows anonymous to login. Otherwise you will need to provide the password to login the LDAP server.</p> <p><b>Example:</b></p> <p>ldap.password = secret</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;LDAP-&gt;Password</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>ldap.max_hits</b>	<b>Integer from 1 to 32000</b>	<b>50</b>
<p><b>Description:</b></p> <p>Configures the maximum number of search results to be returned by the LDAP server.</p> <p>If it is set to blank, the LDAP server will return all searched results.</p> <p><b>Example:</b></p> <p>ldap.max_hits = 50</p>		

Parameters	Permitted Values	Default
<p><b>Note:</b> A very large value of this parameter will slow down the LDAP search speed, therefore it should be configured according to the available bandwidth. It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Directory-&gt;LDAP-&gt;Max Hits (1~32000)</p> <p><b>Phone User Interface:</b> None</p>		
<b>ldap.name_attr</b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the name attributes of each record to be returned by the LDAP server. It compresses the search results. You can configure multiple name attributes separated by spaces.</p> <p><b>Example:</b> ldap.name_attr = cn sn This requires the "cn" and "sn" attributes set for each contact record on the LDAP server.</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Directory-&gt;LDAP-&gt;LDAP Name Attributes</p> <p><b>Phone User Interface:</b> None</p>		
<b>ldap.numb_attr</b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the number attributes of each record to be returned by the LDAP server. It compresses the search results. You can configure multiple number attributes separated by spaces.</p> <p><b>Example:</b> ldap.numb_attr = mobile ipPhone This requires the "mobile" and "ipPhone" attributes set for each contact record on the LDAP server.</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Directory-&gt;LDAP-&gt;LDAP Number Attributes</p> <p><b>Phone User Interface:</b></p>		



Parameters	Permitted Values	Default
None		
<b>ldap.display_name</b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the display name of the contact record displayed on the LCD screen. The value must start with “%” symbol.</p> <p><b>Example:</b></p> <p>ldap.display_name = %cn</p> <p>The cn of the contact record is displayed on the LCD screen.</p> <p><b>Note:</b> It works only if the value of the parameter “ldap.enable” is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;LDAP-&gt;LDAP Display Name</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>ldap.version</b>	<b>2 or 3</b>	<b>3</b>
<p><b>Description:</b></p> <p>Configures the LDAP protocol version supported by the IP phone. The version must be the same as the version assigned on the LDAP server.</p> <p><b>Note:</b> It works only if the value of the parameter “ldap.enable” is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;LDAP-&gt;Protocol</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>ldap.call_in_lookup</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to perform an LDAP search when receiving an incoming call.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter “ldap.enable” is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;LDAP-&gt;LDAP Lookup For Incoming Call</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>ldap.call_out_lookup</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to perform an LDAP search when placing a call. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Directory->LDAP->LDAP Lookup For Callout <b>Phone User Interface:</b> None		
<b>ldap.ldap_sort</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to sort the search results in alphabetical order or numerical order. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Directory->LDAP->LDAP Sorting Results <b>Phone User Interface:</b> None		
<b>ldap.incoming_call_special_search.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to search the telephone numbers starting with "+" symbol and "00" from the LDAP server if the incoming phone number starts with "+" or "00". When completing the LDAP search, all the search results will be displayed on the LCD screen. <b>0</b> -Disabled <b>1</b> -Enabled For example, If the phone receives an incoming call from the phone number 0044123456789, it will search		

Parameters	Permitted Values	Default
<p>0044123456789 from the LDAP sever first, if no result found, it will search +44123456789 from the server again. The phone will display all the search results.</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.call_in_lookup" is set to 1 (Enabled). You may need to set the value of the parameter "ldap.name_filter" to be <code>((cn=*)(sn=*)(telephoneNumber=*)(mobile=*))</code> for searching the telephone numbers starting with "+" symbol. It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>ldap.numb_display_mode</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Configures the display mode of the attribute name for the LDAP contact number.</p> <p><b>0</b>-NumberN (N is an increasing number), for example: Number1, Number2, Number3...</p> <p><b>1</b>-Attribute name pushed by server</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>ldap.customize_label</b>	<b>String within 99 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the display name of the LDAP phone book.</p> <p><b>Example:</b> ldap.customize_label = Friends "Friends" will be displayed on the LCD screen at the path <b>Menu-&gt;Directory</b>. If it is left blank, "LDAP" will be the displayed on the LCD screen at the path <b>Menu-&gt;Directory</b>.</p> <p><b>Note:</b> It works only if the value of the parameter "ldap.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
None		

## LDAP DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

LDAP DSS key can be configured only if the LDAP feature is enabled.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>programablekey.X.type<sup>[2]</sup></b> <b>expansion_module.X.key.Y.type<sup>[3]</sup></b>	38
<b>Description:</b> Configures a DSS key as an LDAP DSS key on the IP phone. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/ Programable Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.label<sup>[1]</sup></b> <b>programablekey.X.label<sup>[2]</sup></b> <b>expansion_module.X.key.Y.label<sup>[3]</sup></b>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b>	String within 99 characters

Parameters	Permitted Values
(X ranges from 1 to 21)	
<p><b>Description:</b></p> <p>(Optional.) Configures the short label displayed on the LCD screen for line key.</p> <p><b>Note:</b> It is only applicable to SIP-T52S IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key-&gt;Short</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Short Label</p>	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programmable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure LDAP via web user interface:

1. Click on **Directory->LDAP**.
2. Enter the values in the corresponding fields.
3. Select the desired values from the corresponding pull-down lists.

The screenshot shows the Yealink T46G web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'Dsskey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Directory' tab is selected, and the 'LDAP' sub-tab is active. The left sidebar lists 'Local Directory', 'Remote Phone Book', 'Phone Call Info', 'LDAP', 'Multicast IP', and 'Setting'. The main content area displays the LDAP configuration form with the following fields:

- Enable LDAP: Enabled
- LDAP Name Filter: ((cn=%)(sn=%))
- LDAP Number Filter: ((telephoneNumber=%)(n
- LDAP TLS Mode: LDAP
- Server Address: 10.2.1.55
- Port: 389
- Base: dc=yealink,dc=cn
- User name: cn=manager,dc=yealink,dc=
- Password: \*\*\*\*\*
- Max Hits (1~32000): 50
- LDAP Name Attributes: cn sn
- LDAP Number Attributes: mobile ipPhone
- LDAP Display Name: %cn
- Protocol: Version 3
- LDAP Lookup For Incoming Call: Enabled
- LDAP Lookup For Callout: Enabled
- LDAP Sorting Results: Enabled

At the bottom of the form are 'Confirm' and 'Cancel' buttons. On the right side, there is a 'NOTE' section with the following text:

**NOTE**  
LDAP (Lightweight Directory Access Protocol) is an application protocol for accessing and maintaining information services for the distributed directory over an IP network.  
Yealink IP phone can interface with a corporate directory server that supports LDAP version 2 or 3, such as OpenLDAP, Microsoft Active Directory, Microsoft Active Directory Application Mode (ADAM) or Sun One Directory Server.  
You can click here to get more guides.

4. Click **Confirm** to accept the change.

**To configure an LDAP DSS key via web user interface:**

1. Click on **Dsskey**->**Line Key** (or **Programable Key**).
2. In the desired DSS key field, select **LDAP** from the pull-down list of **Type**.
3. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

Key	Type	Value	Label	Line	Extension
Line Key1	Line	Default	101	Line 1	
Line Key2	LDAP		14/A		
Line Key3	Line	Default		Line 3	
Line Key4	Line	Default		Line 4	

4. Click **Confirm** to accept the change.

**To configure an LDAP DSS key via phone user interface:**

1. Press **Menu**->**Features**->**DSS Keys**.
2. Select the desired DSS key.
3. Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select **LDAP** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Press the **Save** soft key to accept the change.

## Busy Lamp Field (BLF)

BLF is used to monitor a specific user for status changes on IP phones. For example, you can configure a BLF DSS key on a supervisor's phone to monitor the IP phone user status (busy or idle). When the monitored user places a call, a busy indicator on the supervisor's phone indicates that the user's phone is in use. You can customize BLF LED status when the supervisor's phone and the monitored phone are in different statuses.

When the monitored user is idle, the supervisor can press the BLF DSS key to dial out the phone number. When the monitored user receives an incoming call, the supervisor can press the BLF DSS key to pick up the call directly. When the monitored user is in a call, the supervisor can press the BLF DSS key to interrupt and set up a conference call. You can customize the key behavior when pressing the BLF DSS key.

You can also configure the IP phone to automatically initiate an outgoing intercom call to a monitored extension when pressing the BLF DSS key.

**Note**

BLF is not applicable to SIP-T19(P) E2 IP phones.

## BLF Subscription

IP phones support BLF using a SUBSCRIBE/NOTIFY mechanism as specified in [RFC 3265](#). This feature depends on support from a SIP server.

When the IP phone is configured to monitor a specific user, it sends a SUBSCRIBE message to the server. A NOTIFY message which includes XML in the message body is sent to the IP phone to inform the current state of monitored user. Once status of the monitored user is changed from idle to busy or vice versa, the IP phone is notified from the server with a NOTIFY message. You can manually configure the period of the BLF subscription.

Example of a SUBSCRIBE message:

```
SUBSCRIBE sip:1011@10.3.20.2:5060 SIP/2.0
Via: SIP/2.0/UDP 10.3.20.1:5060;branch=z9hG4bK2940676338
From: "1010" <sip:1010@10.2.1.48:5060>;tag=2493044525
To: <sip:1011@10.2.1.48:5060>;tag=2527548726
Call-ID: 0_3538292381@10.3.20.1
CSeq: 2 SUBSCRIBE
Contact: <sip:1010@10.3.20.1:5060>
Accept: application/dialog-info+xml
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Expires: 30
Event: dialog
Content-Length: 0
```

Example of a NOTIFY message (<state>confirmed</state> shows the call has been established):

```
NOTIFY sip:1010@10.3.20.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.3.20.2:5060;branch=z9hG4bK276311022
From: <sip:1011@10.2.1.48:5060>;tag=3436332841
To: "1010" <sip:1010@10.2.1.48:5060>;tag=3098567568
Call-ID: 0_4117916748@10.3.20.1
CSeq: 4 NOTIFY
Contact: <sip:1011@10.3.20.2:5060>
Content-Type: application/dialog-info+xml
Max-Forwards: 70
User-Agent: Yealink SIP-T27P 45.80.0.20
Subscription-State: active;expires=17
Event: dialog
Content-Length: 534

<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="3" state="partial"
entity="sip:1011@10.2.1.48:5060">
```

```

<dialog id="74" call-id="0_2561109579@10.3.20.1" local-tag="2778958897" remote-tag="1132018898"
direction="recipient">
<state>confirmed</state>
<local>
<identity>sip:1011@10.2.1.48:5060</identity>
<target uri="sip:1011@10.2.1.48:5060"/>
</local>
<remote>
<identity>sip:1010@10.2.1.48:5060</identity>
<target uri="sip:1010@10.2.1.48:5060"/>
</remote>
</dialog>
</dialog-info>

```

## Procedure

BLF can be configured using the following methods.

Central Provisioning (Configuration File)	<MAC>.cfg	Configure the period of the BLF subscription. <b>Parameter:</b> account.X.blf.subscribe_period
		Configure the event of the BLF subscription. <b>Parameter:</b> account.X.blf.subscribe_event
		Configure whether to handle NOTIFY messages out of the BLF dialog. <b>Parameter:</b> account.X.out_dialog_blf_enable
Web User Interface		Configure the period of the BLF subscription.  Configure whether to handle NOTIFY messages out of the BLF dialog. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>



**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>account.X.blf.subscribe_period<sup>[1]</sup></b>	<b>Integer from 30 to 2147483647</b>	<b>1800</b>
<p><b>Description:</b></p> <p>Configures the period (in seconds) of the BLF subscription for a specific account.</p> <p>The IP phone is able to successfully refresh the SUBSCRIBE before expiration of the SUBSCRIBE dialog.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;Subscribe Period(Seconds)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.blf.subscribe_event<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the event of the BLF subscription for a specific account.</p> <p><b>0</b>-dialog</p> <p><b>1</b>-presence</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.out_dialog_blf_enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to handle NOTIFY messages out of the BLF dialog for a specific account.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;Out Dialog BLF</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2.

#### To configure BLF subscription via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Enter the desired period of BLF subscription in the **Subscribe Period(Seconds)** field.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected, and the 'Advanced' sub-tab is active. The 'Account' dropdown is set to 'Account1'. The 'Subscribe Period(Seconds)' field is highlighted with a red box and contains the value '1800'. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), and 'DTMF Type' (RFC2833). A 'NOTE' box on the right explains the DTMF signal and the Session Timer.

4. Click **Confirm** to accept the change.

#### To configure out dialog BLF via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Out Dialog BLF**.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected, and the 'Advanced' sub-tab is active. The 'Account' dropdown is set to 'Account 1'. The 'Out Dialog BLF' field is highlighted with a red box and is set to 'Enabled'. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'DTMF Type' (RFC2833), 'DTMF Info Type' (DTMF-Relay), and 'DTMF Payload Type(96~127)' (101). A 'NOTE' box on the right explains the DTMF signal, the Session Timer, the Busy Lamp Field/BLF List, and the Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA).

4. Click **Confirm** to accept the change.

## Visual Alert and Audio Alert for BLF Pickup

Visual and audio alert for BLF pickup allow the supervisor's phone to play a specific ring tone and display a visual prompt (for example, "6001<-6002", 6001 is the monitored extension which receives an incoming call from 6002) when the monitored user receives an incoming call. You can configure the display method of the call party information.

In addition to the BLF DSS key, visual alert for BLF pickup feature enables the supervisor to pick up the monitored user's incoming call by pressing the **Pick Up** soft key. You may need to configure the directed call pickup code in advance. For more information on how to configure the directed call pickup code for the **DPickup** soft key, refer to [Directed Call Pickup](#) on page 434.

### Ring Type for BLF Pickup

Ring type for BLF pickup allows you to configure a different ring tone to distinguish the incoming call on your phone from that on the monitored phone. It helps clearly notify you of the ringing state of a monitored phone. To use this feature, you need to enable audio alert for BLF pickup feature in advance.

### Procedure

Configuration changes can be performed using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify whether to use visual alert and audio alert for BLF pickup.  <b>Parameters:</b> features.pickup.blf_visual_enable features.pickup.blf_audio_enable
		Configure ring type for BLF pickup.  <b>Parameter:</b> features.blf.ring_type
		Configure the display method of the call party information.  <b>Parameter:</b> features.pickup_display.method
<b>Web User Interface</b>		Specify whether to use visual alert and audio alert for BLF pickup.  Configure ring type for BLF pickup.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-callpickup&q=load

<b>Phone User Interface</b>	Configure ring type for BLF pickup.
-----------------------------	-------------------------------------

**Details of Configuration Parameters:**

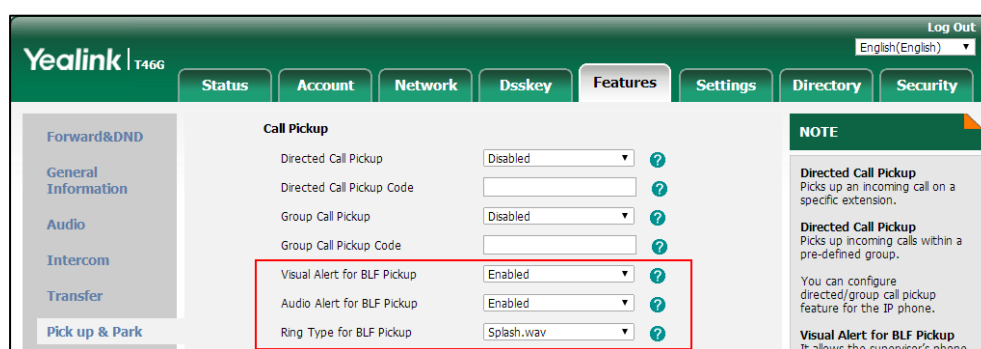
Parameters	Permitted Values	Default
<b>features.pickup.blf_visual_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to display a visual alert when the monitored user receives an incoming call.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Pick up &amp; Park-&gt;Visual Alert for BLF Pickup</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.pickup.blf_audio_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to play an audio alert when the monitored user receives an incoming call.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Pick up &amp; Park-&gt;Audio Alert for BLF Pickup</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.blf.ring_type</b>	<b>Refer to the following content</b>	<b>Splash.wav</b>
<p><b>Description:</b></p> <p>Configures a ring tone for BLF pickup.</p> <p><b>Permitted Values:</b></p> <p>Ring1.wav, Ring2.wav, Ring3.wav, Ring4.wav, Ring5.wav, Ring6.wav, Ring7.wav, Ring8.wav, Silent.wav, Splash.wav or custom ring tone name (for example, Customring.wav).</p>		

Parameters	Permitted Values	Default
<p><b>Example:</b></p> <pre>features.blf.ring_type = Ring5.wav</pre> <p><b>Note:</b> It works only if the value of the parameter "features.pickup.blf_audio_enable" is set to 1 (Enabled). It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Pick up &amp; Park-&gt;Ring Type for BLF Pickup</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Sound-&gt;BLF Ring Type</p>		
<b>features.pickup_display.method</b>	<b>0, 1, 2 or 3</b>	<b>2</b>
<p><b>Description:</b></p> <p>Configures the display method of the call party information on the visual prompt.</p> <p><b>0</b>-Name  <b>1</b>-Number  <b>2</b>-Name Number  <b>3</b>-Number Name</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "features.pickup.blf_visual_enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure visual alert and audio alert for BLF pickup via web user interface:**



1. Click on **Features->Pick up & Park**.
2. Select the desired value from the pull-down list of **Visual Alert for BLF Pickup**.
3. Select the desired value from the pull-down list of **Audio Alert for BLF Pickup**.

- Select the desired ring tone from the pull-down list of **Ring Type for BLF Pickup**.



- Click **Confirm** to accept the change.

**To select a ring tone for BLF pickup via web user interface:**

- Press **Menu->Basic->Sound->BLF Ring Type**.
- Press  or  to select the desired ring tone.
- Press the **Save** soft key to accept the change.

## BLF LED Mode

BLF LED Mode provides five kinds of definition for the BLF/BLF List DSS key LED status. As there is no hard line key on SIP-T48G/S IP phones, BLF LED mode configuration is only applicable to SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones. BLF LED mode is also applicable to the expansion module EXP50 (connected to SIP-T54S/T52S IP phones), EXP40 (connected to SIP-T48G/T48S/T46G/T46S IP phones), EXP20 (connected to SIP-T29G/T27P/T27G IP phones). The following table lists the LED statuses of the BLF DSS key when BLF LED Mode is set to 0, 1, 2, 3 or 4 respectively. The default value of BLF LED mode is 0.

BLF LED mode feature is also applicable to BLF list DSS key. For more information on BLF List DSS key, refer to [Busy Lamp Field \(BLF\)](#) on page 532.

**Line key/Expansion Module Key LED** (configured as a BLF DSS key or a BLF List DSS key and BLF LED Mode is set to 0)

LED Status	Description
Solid green	The monitored user is idle.
Fast-flashing red (200ms)	The monitored user receives an incoming call.
Solid red	The monitored user is dialing. The monitored user is talking. The monitored user's conversation is placed on hold (This LED status requires server support).
Slow-flashing red (1s)	The call is parked against the monitored user's phone number.

LED Status	Description
Off	The monitored user does not exist.

**Line Key/Expansion Module Key LED** (configured as a BLF DSS key or a BLF List DSS key and BLF LED Mode is set to 1)

LED Status	Description
Fast-flashing red (200ms)	The monitored user receives an incoming call.
Solid red	The monitored user is dialing. The monitored user is talking. The monitored user's conversation is placed on hold (This LED status requires server support).
Slow-flashing red (1s)	The call is parked against the monitored user's phone number.
Off	The monitored user is idle. The monitored user does not exist.

**Line Key/Expansion Module Key LED** (configured as a BLF DSS key or a BLF List DSS key and BLF LED Mode is set to 2)

LED Status	Description
Fast-flashing red (200ms)	The monitored user receives an incoming call.
Solid red	The monitored user is dialing. The monitored user is talking. The monitored user's conversation is placed on hold (This LED status requires server support).
Slow-flashing red (1s)	The call is parked against the monitored user's phone number.
Off	The monitored user is idle. The monitored user does not exist.

**Line Key/Expansion Module Key LED** (configured as a BLF DSS key or a BLF List DSS key and BLF LED Mode is set to 3)

LED Status	Description
Fast-flashing green (200ms)	The monitored user receives an incoming call.
Solid red	The monitored user is dialing. The monitored user is talking. The monitored user's conversation is placed on hold (This LED status requires server support).
Slow-flashing red (1s)	The call is parked against the monitored user's phone number.
Off	The monitored user is idle.

LED Status	Description
	The monitored user does not exist.

**Line Key/Expansion Module Key LED** (configured as a BLF DSS key or a BLF List DSS key and BLF LED Mode is set to 4. This mode is specifically designed for the Genband server.)

LED Status	Description
Solid green	The monitored user is talking.
Slow-flashing green (1s)	The monitored user does not exist.
Off	The monitored user is idle.

## Procedure

BLF LED mode can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure BLF LED mode. <b>Parameter:</b> features.blf_led_mode
<b>Web User Interface</b>		Configure BLF LED mode. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load</a>

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>features.blf_led_mode</b>	<b>0, 1, 2, 3 or 4</b>	<b>0</b>
<p><b>Description:</b> Configures BLF LED mode and provides five kinds of definition for the BLF/BLF List DSS key LED status.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. For the Genband server, you can set the value of this parameter to 4.</p> <p><b>Web User Interface:</b> Features-&gt;General Information-&gt;BLF LED Mode</p> <p><b>Phone User Interface:</b> None</p>		

### To configure BLF LED mode via web user interface:

1. Click on **Features->General Information**.



2. Select the desired value from the pull-down list of **BLF LED Mode**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected. Under 'General Information', the 'BLF LED Mode' dropdown is highlighted with a red box and shows the value '0'. Other settings include 'Call Waiting' (Enabled), 'Auto Redial' (Disabled), 'Diversion/History-Info' (Enabled), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right provides details about various features like Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

3. Click **Confirm** to accept the change.

## Configuring BLF LED Status and BLF DSS Key Behavior

You can customize the BLF LED status (or BLF icon colors for SIP-T48G/T48S IP phones) and BLF/BLF List DSS key behavior using the EDK macros if required. For more information on EDK macros, refer to [Understanding Macro Action Strings](#) on page 688.

The following behaviors are supported when pressing the BLF/BLF List DSS keys:

- **newcall** - dial out the phone number
- **btransfer** - transfer the active call to the monitored user without consulting
- **ctransfer** - transfer the active call to the monitored user with prior consulting
- **pickup** - pick up the call directly
- **bargein** - barge in and set up a conference call
- **custom EDK macros** (for example, 1234\$Tinvite\$) - execute a series of macro action strings

### Procedure

BLF LED status and BLF DSS key behavior can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<code>&lt;y0000000000xx&gt;.cfg</code>	Configure BLF LED status/icon colors and BLF/BLF List DSS key behavior when the monitored user is idle. <b>Parameters:</b>
--------------------------------------------------	----------------------------------------	-------------------------------------------------------------------------------------------------------------------------------

		blf.enhanced.idle.enable blf.enhanced.idle.led blf.enhanced.idle.callin.action blf.enhanced.idle.talking.action blf.enhanced.idle.idle.action
		Configure BLF LED status/icon colors and BLF/BLF List DSS key behavior when the monitored user is ringing. <b>Parameters:</b> blf.enhanced.callin.enable blf.enhanced.callin.led blf.enhanced.callin.callin.action blf.enhanced.callin.talking.action blf.enhanced.callin.idle.action
		Configure BLF LED status/icon colors and BLF/BLF List DSS key behavior when the monitored user is calling out. <b>Parameters:</b> blf.enhanced.callout.enable blf.enhanced.callout.led blf.enhanced.callout.callin.action blf.enhanced.callout.talking.action blf.enhanced.callout.idle.action
		Configure BLF LED status/icon colors and BLF/BLF List DSS key behavior when the monitored user is talking. <b>Parameters:</b> blf.enhanced.talking.enable blf.enhanced.talking.led blf.enhanced.talking.callin.action blf.enhanced.talking.talking.action blf.enhanced.talking.idle.action
		Configure BLF LED status/icon colors and BLF/BLF List DSS key behavior when a call is being parked against the monitored phone. <b>Parameters:</b> blf.enhanced.parked.enable

		blf.enhanced.parked.led blf.enhanced.parked.callin.action blf.enhanced.parked.talking.action blf.enhanced.parked.idle.action
--	--	---------------------------------------------------------------------------------------------------------------------------------------

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>blf.enhanced.idle.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the custom configuration when the monitored user is idle.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, the IP phone will display the custom BLF/BLF List DSS key LED status/icon colors and perform the custom behavior when pressing the BLF/BLF List DSS key.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.idle.led</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the custom BLF/BLF List DSS key LED status/icon colors when the monitored user is idle.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <p>blf.enhanced.idle.led = \$LEDg1000o\$</p> <p>For SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones, the BLF/BLF list DSS key LED illuminates solid green for 1000ms and then goes out.</p> <p>For SIP-T48G/T48S IP phones, the BLF/BLF List DSS key field displays a green icon for 1000ms and then the icon color turns to white.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.idle.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
None <b>Phone User Interface:</b> None		
<b>blf.enhanced.idle.idle.action</b>	<b>String</b>	<b>Blank</b>
<b>Description:</b> Configures the behavior when pressing the BLF/BLF list DSS key if the monitor and monitored user are idle. This value uses the same macro action string syntax as an Enhanced DSS key. <b>Example:</b> blf.enhanced.idle.idle.action = 1234\$Tinvite\$ <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.idle.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>blf.enhanced.idle.callin.action</b>	<b>String</b>	<b>Blank</b>
<b>Description:</b> Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is ringing and monitored user is idle. This value uses the same macro action string syntax as an Enhanced DSS key. <b>Example:</b> blf.enhanced.idle.callin.action = newcall <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.idle.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>blf.enhanced.idle.talking.action</b>	<b>String</b>	<b>Blank</b>
<b>Description:</b>		

Parameters	Permitted Values	Default
<p>Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is talking and monitored user is idle.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <pre>blf.enhanced.idle.talking.action = newcall</pre> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.idle.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.callin.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the custom configuration when the monitored user is ringing.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, the IP phone will display the custom BLF/BLF List DSS key LED status/icon colors and perform the custom behavior when pressing the BLF/BLF List DSS key.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.callin.led</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the custom BLF/BLF List DSS key LED status/icon colors when the monitored user is ringing.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <pre>blf.enhanced.callin.led = \$LEDg1000o\$</pre> <p>For</p> <p>SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones, the BLF/BLF List DSS key LED illuminates solid green for 1000ms and then goes out.</p>		

Parameters	Permitted Values	Default
<p>For SIP-T48G/T48S IP phones, the BLF/BLF List DSS key field displays a green icon for 1000ms and then the icon color turns to white.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.callin.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.callin.idle.action</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is idle and monitored user is ringing.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <p>blf.enhanced.callin.idle.action = newcall</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.callin.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.callin.callin.action</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the behavior when pressing the BLF/BLF list DSS key if the monitor and monitored user are ringing.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <p>blf.enhanced.callin.callin.action = newcall</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.callin.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
None		
<b>blf.enhanced.callin.talking.action</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is talking and monitored user is ringing.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <p>blf.enhanced.callin.talking.action = newcall</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.callin.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.callout.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the custom configuration when the monitored user is calling out.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, the IP phone will display the custom BLF/BLF List DSS key LED status/icon colors and perform the custom behavior when pressing the BLF/BLF List DSS key.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.callout.led</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the custom BLF/BLF List DSS key LED status/icon colors when the monitored user is calling out.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p>		

Parameters	Permitted Values	Default
<p>blf.enhanced.callout.led = \$LEDg1000o\$</p> <p>For SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones, the BLF/BLF list DSS key LED illuminates solid green for 1000ms and then goes out.</p> <p>For SIP-T48G/T48S IP phones, the BLF/BLF List DSS key field displays a green icon for 1000ms and then the icon color turns to white.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.callout.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.callout.idle.action</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is idle and monitored user is calling out.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <p>blf.enhanced.callout.idle.action = newcall</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.callout.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.callout.callin.action</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is ringing and monitored user is calling out.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <p>blf.enhanced.callout.callin.action = newcall</p>		



Parameters	Permitted Values	Default
<p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.callout.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>blf.enhanced.callout.talking.action</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b> Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is talking and monitored user is calling out. This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b> blf.enhanced.callout.talking.action = newcall</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.callout.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>blf.enhanced.talking.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the custom configuration when the monitored user is talking. <b>0</b>-Disabled <b>1</b>-Enabled, the IP phone will display the custom BLF/BLF List DSS key LED status/icon colors and perform the custom behavior when pressing the BLF/BLF List DSS key.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>blf.enhanced.talking.led</b>	<b>String</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the custom BLF/BLF List DSS key LED status/icon colors when the monitored user is talking.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <p>blf.enhanced.talking.led = \$LEDg1000o\$</p> <p>For SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones, the BLF/BLF list DSS key LED illuminates solid green for 1000ms and then goes out.</p> <p>For SIP-T48G/T48S IP phones, the BLF/BLF List DSS key field displays a green icon for 1000ms and then the icon color turns to white.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.talking.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.talking.idle.action</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is idle and monitored user is talking.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <p>blf.enhanced.talking.idle.action = newcall</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.talking.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.talking.callin.action</b>	<b>String</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is ringing and monitored user is talking.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <p>blf.enhanced.talking.callin.action = newcall</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.talking.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.talking.talking.action</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the behavior when pressing the BLF/BLF list DSS key if the monitor and monitored user are talking.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b></p> <p>blf.enhanced.talking.talking.action = newcall</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.talking.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>blf.enhanced.parked.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the custom configuration when a call is being parked against the monitored phone.</p> <p><b>0-Disabled</b></p> <p><b>1-Enabled</b>, the IP phone will display the custom BLF/BLF List DSS key LED status/icon colors and perform the custom behavior when pressing the BLF/BLF List DSS key.</p>		

Parameters	Permitted Values	Default
<p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>blf.enhanced.parked.led</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b> Configures the custom BLF/BLF List DSS key LED status/icon colors when a call is being parked against the monitored phone. This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b> blf.enhanced.parked.led = \$LEDg1000o\$ For SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones, the BLF/BLF list DSS key LED illuminates solid green for 1000ms and then goes out. For SIP-T48G/T48S IP phones, the BLF/BLF List DSS key field displays a green icon for 1000ms and then the icon color turns to white.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.parked.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>blf.enhanced.parked.idle.action</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b> Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is idle and a call is being parked against the monitored phone. This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p><b>Example:</b> blf.enhanced.parked.idle.action = newcall</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.parked.enable" is set to 1 (Enabled).</p>		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>blf.enhanced.parked.callin.action</b>	<b>String</b>	<b>Blank</b>
<b>Description:</b> Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is ringing and a call is being parked against the monitored phone. This value uses the same macro action string syntax as an Enhanced DSS key. <b>Example:</b> blf.enhanced.parked.callin.action = newcall <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.parked.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>blf.enhanced.parked.talking.action</b>	<b>String</b>	<b>Blank</b>
<b>Description:</b> Configures the behavior when pressing the BLF/BLF list DSS key if the monitor is talking and a call is being parked against the monitored phone. This value uses the same macro action string syntax as an Enhanced DSS key. <b>Example:</b> blf.enhanced.parked.talking.action = newcall <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. It works only if the value of the parameter "blf.enhanced.parked.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Configuring a BLF DSS Key

You can configure a BLF DSS key on a supervisor's phone to monitor the IP phone user status (busy or idle). For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Procedure

BLF DSS key can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Assign a BLF DSS key. <b>Parameters:</b> linekey.X.type/ expansion_module.X.key.Y.type linekey.X.line/ expansion_module.X.key.Y.line linekey.X.value/ expansion_module.X.key.Y.value linekey.X.pickup_value/ expansion_module.X.key.Y.pickup_value linekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Assign a BLF DSS key. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=dsskey &q=load&model=0
<b>Phone User Interface</b>		Assign a BLF DSS key.

### Details of Configuration Parameters:

Parameters	Permitted Values
linekey.X.type <sup>[1]</sup> expansion_module.X.key.Y.type <sup>[2]</sup>	16
<b>Description:</b> Configures a DSS key as a BLF DSS key on the IP phone.  <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.  <b>Web User Interface:</b>	

Parameters	Permitted Values
Dsskey->Line Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.line<sup>[1]</sup></b> <b>expansion_module.X.key.Y.line<sup>[2]</sup></b>	Refer to the following content
<b>Description:</b> Configures the desired line to apply the BLF DSS key. <b>Permitted Values:</b> 1 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G) 1 to 12 (for SIP-T52S/T42G/T42S) 1 to 6 (for SIP-T41P/T41S/T27P/T27G) 1 to 3 (for SIP-T40P/T40G/T23P/T23G) 1 to 2 (for SIP-T21(P) E2) 1-Line 1 2-Line 2 ... 16-Line 16 <b>Example:</b> linekey.2.line = 1 <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Line <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Account ID	
<b>linekey.X.value<sup>[1]</sup></b> <b>expansion_module.X.key.Y.value<sup>[2]</sup></b>	String within 99 characters
<b>Description:</b> Configures the phone number or extension of the monitored user. <b>Example:</b> linekey.2.value = 1008 <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Value	

Parameters	Permitted Values
<b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Value	
<b>linekey.X.pickup_value</b> <sup>[1]</sup> <b>expansion_module.X.key.Y.pickup_value</b> <sup>[2]</sup>	String within 256 characters
<b>Description:</b> Configures the pickup code for BLF feature. <b>Example:</b> line.2.pickup_value = *88 <b>Note:</b> This parameter only applies to intercom and BLF feature. It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Extension <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Extension	
<b>linekey.X.label</b> <sup>[1]</sup> <b>expansion_module.X.key.Y.label</b> <sup>[2]</sup>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Short <b>Phone User Interface:</b>	



Parameters	Permitted Values
Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure a BLF DSS key via web user interface:

1. Click on **Dsskey->Line Key**.
2. In the desired DSS key field, select **BLF** from the pull-down list of **Type**.
3. Enter the phone number or extension you want to monitor in the **Value** field.
4. Select the desired line from the pull-down list of **Line**.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. (Optional.) Enter the directed call pickup code in the **Extension** field.

7. Click **Confirm** to accept the change.

#### To configure a BLF DSS key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press **◀** or **▶**, or the **Switch** soft key to select **BLF** from the **Type** field.
4. Press **◀** or **▶**, or the **Switch** soft key to select the desired line from the **Account ID** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the phone number or extension you want to monitor in the **Value** field.
7. (Optional.) Enter the directed call pickup code in the **Extension** field.
8. Press the **Save** soft key to accept the change.

## Using BLF DSS Key to Initiate an Intercom Call

The BLF DSS key can also be used to initiate an intercom call.

## Procedure

Initiate an intercom call using BLF DSS key can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure the IP phone to initiate an intercom call using BLF DSS key. <b>Parameter:</b> features.blf.intercom_mode.enable
		Configure the intercom mode. <b>Parameters:</b> features.intercom.mode features.intercom.feature_access_code

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>features.blf.intercom_mode.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to initiate an outgoing intercom call with a monitored user when pressing the BLF DSS key. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> To use this feature, you also need to configure the intercom mode (configured by the parameter "features.intercom.mode"). It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.intercom.mode</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Configures the intercom mode. <b>0</b> -SIP <b>1</b> -FAC, the feature access code is configured by the parameter "features.intercom.feature_access_code". <b>Web User Interface:</b> None		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>features.intercom.feature_access_code</b>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the intercom feature access code. <b>Note:</b> It works only if the value of parameter "features.intercom.mode" is set to 1 (FAC). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Busy Lamp Field (BLF) List

BLF List allows a list of specific extensions to be monitored for status changes. It enables the monitoring phone to subscribe to a list of users, and receive status notifications of the monitored users. Different indicators on the monitoring phone show the status of monitored users. The monitoring user can also be notified about calls being parked/no longer parked against any monitored user. IP phones support BLF list using a SUBSCRIBE/NOTIFY mechanism as specified in [RFC 3265](#). This feature depends on support from a SIP server. You can customize BLF LED status when the supervisor's phone and the monitored phone are in different statuses. For more information, refer to [Configuring BLF LED Status and BLF DSS Key Behavior](#) on page 543.

**Note** BLF list is not applicable to SIP-T19(P) E2 IP phones.

### Procedure

BLF List can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<MAC>.cfg	Configure BLF List. <b>Parameters:</b> account.X.blf.blf_list_uri account.X.blf_list_code account.X.blf_list_barge_in_code account.X.blf_list_retrieve_call_parked_code
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	<y0000000000xx>.cfg	Specify whether to automatically configure the BLF list DSS keys. <b>Parameter:</b> phone_setting.auto_blf_list_enable
		Configure the order of BLF list DSS keys assigned automatically. <b>Parameter:</b> phone_setting.blf_list_sequence_type
		Assign a BLF List DSS key. <b>Parameters:</b> linekey.X.type/ expansion_module.X.key.Y.type linekey.X.line/ expansion_module.X.key.Y.line
<b>Web User Interface</b>	Configure BLF List. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>	
	Assign a BLF List DSS key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0">http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0</a>	
<b>Phone User Interface</b>	Assign a BLF List DSS key.	

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
phone_setting.auto_blf_list_enable	0 or 1	1
<b>Description:</b> Enables or disables the IP phone to automatically configure the BLF list DSS keys. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> None <b>Phone User Interface:</b>		

Parameters	Permitted Values	Default
None		
<b>phone_setting.blf_list_sequence_type</b>	<b>0, 1, 2 or 3</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the order of BLF list DSS keys assigned automatically.</p> <p><b>0</b>-Line Key-&gt;Ext Key (Expansion1-&gt;Expansion2...)</p> <p><b>1</b>-Ext Key (Expansion1-&gt;Expansion2...)-&gt;Line Key</p> <p><b>2</b>-Line Key (first page)-&gt;Ext Key (first page on Expansion1-&gt;first page on Expansion2...)-&gt;Line Key (second page)-&gt;Ext Key (second page on Expansion1-&gt;second page on Expansion2...)</p> <p><b>3</b>-Ext Key (first page on Expansion1-&gt;first page on Expansion2...)-&gt;Line Key (first page)-&gt;Ext Key (second page on Expansion1-&gt;second page on Expansion2...)-&gt;Line Key (second page)</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.auto_blf_list_enable" is set to 1 (Enabled). To assign Ext Key, make sure the expansion module has been connected to the phone in advance. It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.blf.blf_list_uri<sup>[1]</sup></b>	<b>String within 256 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the BLF List URI to monitor a list of users for a specific account.</p> <p><b>Example:</b></p> <p>account.1.blf.blf_list_uri = 4609@pbx.test.com</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;BLF List URI</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.blf_list_code<sup>[1]</sup></b>	<b>String within 32 characters</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b> Configures the feature access code for directed call pickup for a specific account.</p> <p><b>Example:</b> account.1.blf_list_code = *97</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b> Account-&gt;Advanced-&gt;BLF List Pickup Code</p> <p><b>Phone User Interface:</b> None</p>		
account.X.blf_list_barge_in_code <sup>[1]</sup>	String within 32 characters	Blank
<p><b>Description:</b> Configures the feature access code for directed call pickup with barge-in for a specific account.</p> <p><b>Example:</b> account.1.blf_list_barge_in_code = *33</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b> Account-&gt;Advanced-&gt;BLF List Barge In Code</p> <p><b>Phone User Interface:</b> None</p>		
account.X.blf_list_retrieve_call_parked_code <sup>[1]</sup>	String within 32 characters	Blank
<p><b>Description:</b> Configures the feature access code for the call park retrieve for a specific account.</p> <p><b>Example:</b> account.1.blf_list_retrieve_call_parked_code = *88</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b> Account-&gt;Advanced-&gt;BLF List Retrieve Call Parked Code</p> <p><b>Phone User Interface:</b> None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12;

for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2.

## BLF List DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877. To manually configure BLF list DSS key, make sure the value of the parameter "phone\_setting.auto\_blf\_list\_enable" is set to 0 (Disabled).

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>expansion_module.X.key.Y.type<sup>[2]</sup></b>	39
<b>Description:</b> Configures a DSS key as a BLF List DSS key on the IP phone. <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.line<sup>[1]</sup></b> <b>expansion_module.X.key.Y.line<sup>[2]</sup></b>	Refer to the following content
<b>Description:</b> Configures the desired line to apply the BLF List DSS key. <b>Permitted Values:</b> 1 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G) 1 to 12 (for SIP-T52S/T42G/T42S) 1 to 6 (for SIP-T41P/T41S/T27P/T27G) 1 to 3 (for SIP-T40P/T40G/T23P/T23G) 1 to 2 (for SIP-T21(P) E2) 1-Line 1 2-Line 2 ... 16-Line 16 <b>Example:</b> linekey.2.line = 1 <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to	

Parameters	Permitted Values
SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.	
<b>Web User Interface:</b>	
Dsskey->Line Key->Line	
<b>Phone User Interface:</b>	
Menu->Features->DSS Keys->Line Key X->Account ID	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure the BLF List settings via web user interface:

1. Click on **Account->Advanced**.
2. Select the account (for example, account 1) from the pull-down list of **Account**.
3. Enter the BLF List URI in the **BLF List URI** field.
4. (Optional.) Enter the directed pickup code in the **BLF List Pickup Code** field.
5. (Optional.) Enter the barge-in code in the **BLF List Barge In Code** field.
6. (Optional.) Enter the retrieve call parked code in the **BLF List Retrieve Call Parked Code** field.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected, and the 'Advanced' sub-tab is active. The 'Account' dropdown is set to 'Account 1'. The 'BLF List' section is highlighted with a red box, showing the following fields:

- BLF List URI: 4609@pbx.yealink.com
- BLF List Pickup Code: \*97
- BLF List Barge In Code: \*33
- BLF List Retrieve Call Parked Code: \*88

Other visible fields include:

- Keep Alive Type: Default
- Keep Alive Interval(Seconds): 30
- RPort: Disabled
- Subscribe Period(Seconds): 1800
- RTP Encryption(SRTP): Optional
- PTime(ms): 20
- Shared Line: Shared Call Appearance
- Call Pull Feature Access Code: \*11
- Dialog Info Call Pickup: Enabled
- VQ RTPC-XR Collector Name: collector
- VQ RTPC-XR Collector Address: 10.2.1.98
- VQ RTPC-XR Collector Port: 5060

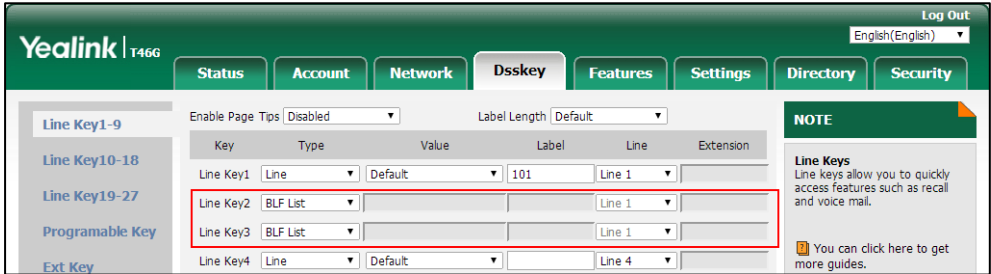
On the right side, there is a 'NOTE' section with information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

7. Click **Confirm** to accept the change.



To configure BLF List DSS keys manually via web user interface:

1. Click on **Dsskey**->**Line Key**.
2. In the desired DSS key field, select **BLF List** from the pull-down list of **Type**.
3. Repeat the step 2, configure more BLF list DSS keys.



4. Click **Confirm** to accept the change.

## Hide Feature Access Codes

Hide Feature Access Codes feature enables the IP phone to display the feature name instead of the dialed feature access code automatically. For example, the dialed call park code will be replaced by the identifier “Call Park” when you park an active call.

The hide feature access codes feature is applicable to the following features:

- Voice Mail
- Pick up
- Group Pick up
- Barge In (not applicable to SIP-T19(P) E2 IP phones)
- Retrieve
- Call Park
- Call Pull (not applicable to SIP-T19(P) E2 IP phones)

### Procedure

The hide feature access codes feature can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the hide feature access codes feature. <b>Parameter:</b> features.hide_feature_access_codes.enable
<b>Web User Interface</b>		Configure the hide feature access codes feature. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p

=features-general&amp;q=load

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.hide_feature_access_codes.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to display feature name instead of the feature access code when dialing or in talk.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Hide Feature Access Codes</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To enable hide feature access codes feature via web user interface:**

1. Click on **Features->General Information**.
2. Select **Enabled** from the pull-down list of **Hide Feature Access Codes**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' section is active. The 'Hide Feature Access Codes' option is set to 'Enabled' and is highlighted with a red box. Other settings include Call Waiting (Enabled), Call Waiting On Code, Call Waiting Off Code, Auto Redial (Disabled), Auto Redial Interval (1~300s) set to 10, Auto Redial Times (1~300) set to 10, Voice Mail Tone (Enabled), DHCP Hostname (SIP-T46G), Reboot in Talking (Disabled), Display Method on Dialing (User Name), and Auto Linekeys (Disabled). A 'NOTE' section on the right provides details for Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

3. Click **Confirm** to accept the change.

## Automatic Call Distribution (ACD)

ACD enables organizations to manage a large number of phone calls on an individual basis.

ACD enables the use of IP phones in a call-center role by automatically distributing incoming calls to available users, or agents. ACD depends on support from a SIP server. ACD is disabled on the IP phone by default. You need to enable it on a per-line basis before logging into the ACD system.

After the IP phone user logs into the ACD system, the server monitors the IP phone status and then decides whether to assign an incoming call to the user's IP phone. When the IP phone status is changed to unavailable, the server stops distributing calls to the IP phone. The IP phone will remain in the unavailable status until the user manually changes the IP phone status or the ACD auto available timer (if configured) expires. How long the IP phone remains unavailable is configurable by the auto available timer. When the timer expires, the IP phone status is automatically changed to available. ACD auto available timer feature depends on support from a SIP server.

You can configure an ACD DSS key for the user to log into the ACD system. The ACD DSS key on the IP phone indicates the ACD status. ACD DSS key is not applicable to SIP-T19(P) E2 IP phones.

## Procedure

ACD can be configured using the following methods.

Central Provisioning (Configuration Files)	<MAC>.cfg	Configure ACD feature on a per-line basis.  <b>Parameters:</b> account.X.acd.enable account.X.acd.available account.X.subscribe_acd_expires
	<y0000000000xx>.cfg	Configure ACD auto available.  <b>Parameters:</b> acd.enable acd.auto_available_timer
		Assign an ACD DSS key.  <b>Parameters:</b> linekey.X.type/ expansion_module.X.key.Y.type linekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
Web User Interface		Configure ACD auto available.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-acd&q=load

	Assign an ACD DSS key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?parameter=dsskey&amp;q=load&amp;model=0">http://&lt;phoneIPAddress&gt;/servlet?parameter=dsskey&amp;q=load&amp;model=0</a>
<b>Phone User Interface</b>	Assign an ACD DSS key.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>account.X.acd.enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables ACD feature for a specific account. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>account.X.acd.available<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to display the <b>Available/Avail</b> or <b>Unavailable/Unavail</b> soft key for a specific account after the IP phone logs into the ACD system. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "account.X.acd.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>account.X.subscribe_acd_expires<sup>[1]</sup></b>	<b>Integer from 120 to 3600</b>	<b>3600</b>
<b>Description:</b> Configures the period (in seconds) of ACD subscription for a specific account.		

Parameters	Permitted Values	Default
<p><b>Note:</b> It works only if the value of the parameter "account.X.acd.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Account-&gt;Advanced-&gt;ACD Subscribe Period(120~3600s)</p> <p><b>Phone User Interface:</b> None</p>		
<b>acd.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the IP phone to automatically change the status of the ACD agent to available after the designated time.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.acd.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Features-&gt;ACD-&gt;ACD Auto Available</p> <p><b>Phone User Interface:</b> None</p>		
<b>acd.auto_available_timer</b>	<b>Integer from 0 to 120</b>	<b>60</b>
<p><b>Description:</b> Configures the interval (in seconds) for the status of the ACD agent to be automatically changed to available.</p> <p><b>Note:</b> It works only if the values of parameters "account.X.acd.enable" and "acd.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Features-&gt;ACD-&gt;ACD Auto Available Timer (0~120s)</p> <p><b>Phone User Interface:</b> None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## ACD DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>expansion_module.X.key.Y.type<sup>[2]</sup></b>	42
<b>Description:</b> Configures a DSS key to be an ACD DSS key on the IP phone. <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.label<sup>[1]</sup></b> <b>expansion_module.X.key.Y.label<sup>[2]</sup></b>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Short <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure an ACD DSS key via web user interface:

1. Click on **Dsskey**->**Line Key**.
2. In the desired DSS key field, select **ACD** from the pull-down list of **Type**.
3. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

The screenshot shows the Yealink T46G web interface with the 'Dsskey' tab selected. On the left, a sidebar lists 'Line Key1-9', 'Line Key10-18', 'Line Key19-27', 'Programable Key', and 'Ext Key'. The main area displays a table for configuring line keys. The table has columns: Key, Type, Value, Label, Line, and Extension. The 'Line Key2' row is highlighted with a red box, showing 'Type' as 'ACD' and 'Label' as 'N/A'. Other rows show 'Line' as the type and 'Default' as the value. A 'NOTE' box on the right explains that line keys allow quick access to features like recall and voice mail.

4. Click **Confirm** to accept the change.

#### To configure the ACD auto available timer feature via web user interface:

1. Click on **Features**->**ACD**.
2. Select the desired value from the pull-down list of **ACD Auto Available**.
3. Enter the desired timer in the **ACD Auto Available Timer(0~120s)** field.

The screenshot shows the Yealink T46G web interface with the 'Features' tab selected and the 'ACD' sub-tab active. The left sidebar lists various features like 'Forward&DND', 'General Information', 'Audio', 'Intercom', 'Transfer', 'Call Pickup', 'Remote Control', 'Phone Lock', and 'ACD'. The main area shows the 'ACD' configuration. A red box highlights the 'ACD Auto Available' dropdown set to 'Enabled' and the 'ACD Auto Available Timer(0~120s)' field set to '60'. Below these fields are 'Confirm' and 'Cancel' buttons. A 'NOTE' box on the right explains that ACD (Automatic Call Distribution) enables organizations to manage calls on an individual basis by distributing incoming calls to available users or agents.

4. Click **Confirm** to accept the change.

#### To configure the ACD subscribe period via web user interface:

1. Click on **Account**->**Advanced**.
2. Select the desired account from the pull-down list of **Account**.

- Enter the desired timer in the **ACD Subscribe Period(120~3600s)** field.

The screenshot shows the Yealink T466 web interface. The 'Account' tab is selected, and the 'Account 1' configuration page is displayed. The 'ACD Subscribe Period(120~3600s)' field is highlighted with a red box and contains the value 3600. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'DTMF Type' (RFC2833), 'Conference URI', 'Early Media' (Disabled), 'Out Dialog BLF' (Disabled), 'VQ RTPC-XR Collector Name', 'VQ RTPC-XR Collector Address', and 'VQ RTPC-XR Collector Port' (5060). A 'NOTE' section on the right provides additional information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

- Click **Confirm** to accept the change.

#### To configure an ACD DSS key via phone user interface:

- Press **Menu->Features->DSS Keys**.
- Select the desired DSS key.
- Press **◀** or **▶**, or the **Switch** soft key to select **ACD** from the **Type** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- Press the **Save** soft key to accept the change.

## Shared Call Appearance (SCA)

SCA allows users to share an extension which can be registered on two or more IP phones at the same time. For more information on how to register accounts, refer to [Account Registration](#) on page 192. If you want to customize multiple DSS keys to associate with an account, refer to [Multiple Line Keys per Account](#) on page 199. If you want to configure multiple concurrent calls for each registered line, refer to [Multiple Call Appearances](#) on page 206.

Any IP phone can be used to originate or receive calls on the shared line. An incoming call can be presented to multiple phones simultaneously. The incoming call can be answered on any IP phone but not all. A call that is active on one IP phone will be presented visually to other IP phones that share the call appearance.

IP phones support SCA using a SUBSCRIBE/NOTIFY mechanism as specified in [RFC 3265](#). The events used are:

- "call-info" for call appearance state notification
- "line-seize" for the IP phone to ask to seize the line



SCA supports the IP phones barging in an active call. In addition, SCA has the call pull capability. Call pull feature allows users to retrieve an existing call from another shared phone that is in active or public hold status.

If the call is placed on public hold, the held call is available for any shared party to retrieve. If the call is placed on private hold, the held call is only available for the hold party to retrieve. You need to configure either the private hold soft key or a private hold DSS key before you place the call on private hold.

## Procedure

SCA can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<MAC>.cfg	Configure the registration line type. <b>Parameter:</b> account.X.shared_line
		Configure the call pull feature access code. <b>Parameter:</b> account.X.shared_line_callpull_code
	<y0000000000xx>.cfg	Configure the private hold soft key. <b>Parameters:</b> phone_setting.custom_softkey_enable custom_softkey_talking.url
		Assign a private hold DSS key. <b>Parameters:</b> linekey.X.type/ expansion_module.X.key.Y.type linekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>	Configure the registration line type. Configure the call pull feature access code. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>	
	Configure the private hold soft key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=s">http://&lt;phoneIPAddress&gt;/servlet?p=s</a>	

	ettings-softkey&q=load
	Assign a private hold DSS key. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0
<b>Phone User Interface</b>	Assign a private hold DSS key.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>account.X.shared_line</b> <sup>[1]</sup>	<b>0, 1 or 3</b>	<b>0</b>
<b>Description:</b> Configures the registration line type for a specific account. <b>0</b> -Disabled, the shared line feature is disabled. <b>1</b> -Shared Call Appearance <b>3</b> -Draft BLA <b>Web User Interface:</b> Account->Advanced->Shared Line <b>Phone User Interface:</b> None		
<b>account.X.shared_line_callpull_code</b> <sup>[1]</sup>	<b>String within 32 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the call pull feature access code to retrieve an existing call from another shared phone that is in active or public hold status for a specific account. <b>Note:</b> It works only if the value of the parameter "account.X.shared_line" is set to 1 (Share Call Appearance). It is not applicable to SIP-T19(P) E2 IP phones. <b>Web User Interface:</b> Account->Advanced->Call Pull Feature Access Code <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## Private Hold Soft Key

Configuring the private hold soft key may affect the softkey layout in the Talking state. For more information, refer to [Softkey Layout](#) on page 252.

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>phone_setting.custom_softkey_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables custom soft keys layout feature. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Settings->Softkey Layout->Custom Softkey <b>Phone User Interface:</b> None		
<b>custom_softkey_talking.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Talking state. <b>Example:</b> custom_softkey_talking.url = http://192.168.1.20/XMLfiles/Talking.xml During the auto provisioning process, the IP phone connects to the provisioning server "192.168.1.20", and downloads the Talking state file from the "XMLfiles" directory. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Private Hold DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

**Details of Configuration Parameters:**

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>expansion_module.X.key.Y.type<sup>[2]</sup></b>	20
<b>Description:</b> Configures a DSS key to be a private hold DSS key on the IP phone. <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.label<sup>[1]</sup></b> <b>expansion_module.X.key.Y.label<sup>[2]</sup></b>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Short <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6,

Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

**To configure the shared line settings on the primary phone via web user interface:**

1. Register the primary account (for example, 4609).

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected. The 'SIP Server 1' section is highlighted with a red box, showing fields for Server Host (pbx.yealink.com), Port (5060), Transport (UDP), Server Expires (3600), and Server Retry Counts (3). The 'Enable Outbound Proxy Server' section is also highlighted with a red box, showing fields for Outbound Proxy Server 1 (10.1.8.11), Port (5060), and Outbound Proxy Server 2 (Port 5060). The 'NAT' dropdown is set to 'Disabled'. The 'Register' status is 'Registered'.

2. Click on **Advanced**, select **Shared Call Appearance** from the pull-down list of **Shared Line**.

The screenshot shows the Yealink T46G web interface. The 'Advanced' tab is selected. The 'Shared Line' dropdown is highlighted with a red box, showing the selected option 'Shared Call Appearance'. The 'BLF List Retrieve Call Parked Code' field is empty. The 'Call Pull Feature Access Code' field is empty. The 'Dialog Info Call Pickup' dropdown is set to 'Disabled'. The 'BLA Number' field is empty. The 'Out Dialog BLF' dropdown is set to 'Disabled'. The 'VQ RTPC-XR Collector Name' field is empty. The 'VQ RTPC-XR Collector Address' field is empty. The 'VQ RTPC-XR Collector Port' field is set to 5060.

3. Click **Confirm** to accept the change.

**To configure the shared line settings on alternate phone via web user interface:**

1. Register the alternate account (for example, 4609\_1).  
(Enter the primary account 4609 in the **Register Name** field.)

The screenshot shows the Yealink T46G web interface with the 'Account' tab selected. The 'Account' dropdown is set to 'Account1'. The 'Register Status' is 'Registered'. The 'SIP Server 1' section is highlighted with a red box, showing fields for 'Line Active' (Enabled), 'Label' (4609\_1), 'Display Name' (4609\_1), 'Register Name' (4609), 'User Name' (4609\_1), 'Password' (\*\*\*\*\*), 'Server Host' (pbx.yealink.com), 'Port' (5060), 'Transport' (UDP), 'Server Expires' (3600), and 'Server Retry Counts' (3). The 'Outbound Proxy Server' section is also highlighted with a red box, showing fields for 'Enable Outbound Proxy Server' (Enabled), 'Outbound Proxy Server 1' (10.1.8.11), 'Port' (5060), 'Outbound Proxy Server 2', 'Proxy Fallback Interval' (3600), and 'NAT' (Disabled). The 'SIP Server 2' section is also visible but not highlighted. The 'NOTE' section on the right contains information about 'Account Registration', 'Server Redundancy', and 'NAT Traversal'.

2. Click on **Advanced**, select **Shared Call Appearance** from the pull-down list of **Shared Line**.

The screenshot shows the Yealink T46G web interface with the 'Advanced' tab selected. The 'Account' dropdown is set to 'Account 1'. The 'Shared Line' dropdown is highlighted with a red box and set to 'Shared Call Appearance'. The 'NOTE' section on the right contains information about 'DTMF', 'Session Timer', 'Busy Lamp Field/BLF List', and 'Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA)'. The 'Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA)' section explains that it allows users to share a SIP line on several IP phones and that any IP phone can be used to originate or receive calls on the shared line.

3. Click **Confirm** to accept the change.


**To configure the call pull feature access code via web user interface:**

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Enter the call pull feature access code (for example, \*11) in the **Call Pull Feature Access Code** field.

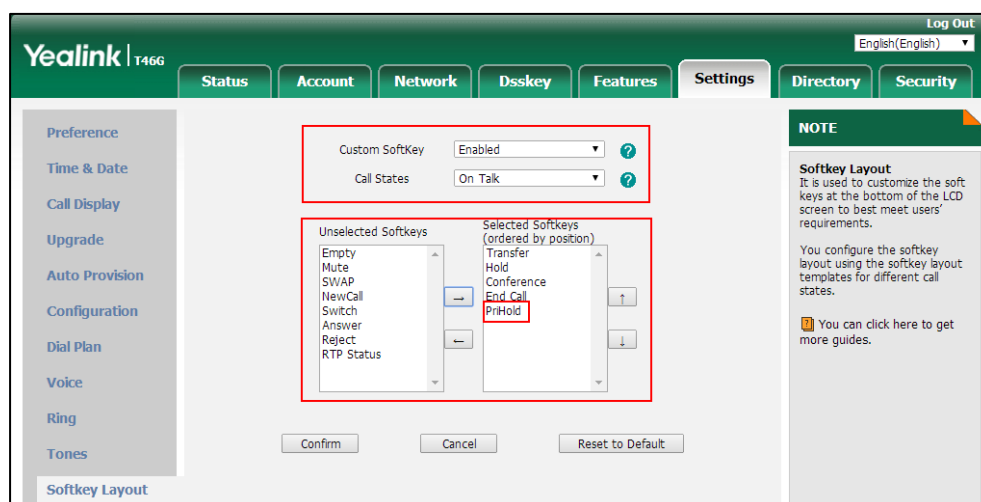
The screenshot shows the Yealink T46G web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'Dsskey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Account' tab is selected, and the 'Advanced' sub-tab is active. The 'Account' dropdown menu shows 'Account 1'. The 'Call Pull Feature Access Code' field is highlighted with a red box and contains the value '\*11'. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'BLF List Barge In Code', 'BLF List Retrieve Call Parked Code', 'Shared Line' (Shared Call Appearance), 'Dialog Info Call Pickup' (Disabled), 'Unregister When Reboot' (Disabled), 'Out Dialog BLF' (Disabled), 'VQ RTPC-XR Collector Name', 'VQ RTPC-XR Collector Address', and 'VQ RTPC-XR Collector Port' (5060). A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

4. Click **Confirm** to accept the change.

**To configure the private hold soft key via web user interface:**

1. Click on **Settings->Softkey Layout**.
2. Select **Enabled** from the pull-down list of **Custom Softkey**.
3. Select **On Talk** from the pull-down list of **Call States**.
4. Select **PriHold** from the **Unselected Softkeys** column and then click .

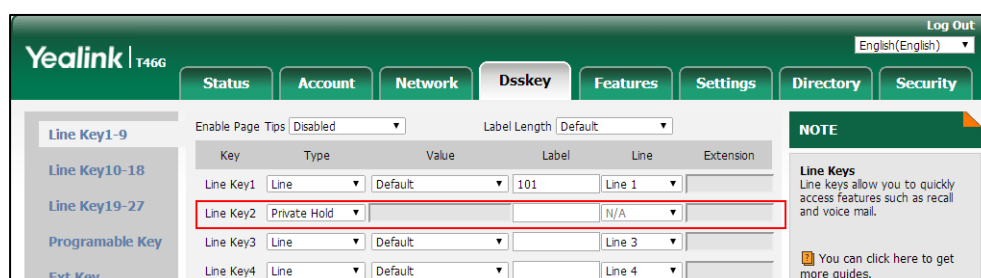
The **PriHold** appears in the **Selected Softkeys** column.



- Click **Confirm** to accept the change.

**To configure a private hold DSS key via web user interface:**

- Click on **Dsskey**->**Line Key**.
- In the desired DSS key field, select **Private Hold** from the pull-down list of **Type**.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.



- Click **Confirm** to accept the change.

**To configure a private hold DSS key via phone user interface:**

- Press **Menu**->**Features**->**DSS Keys**.
- Press **Left Arrow** or **Right Arrow**, or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press **Left Arrow** or **Right Arrow**, or the **Switch** soft key to select **Private Hold** from the **Key Type** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- Press the **Save** soft key to accept the change.

## Bridge Lines Appearance (BLA)

BLA allows users to share a SIP line on two or more IP phones. Users can monitor the specific extension (BLA number) for status changes on each IP phone. To use this feature, a BLA group should be pre-configured on the server and one of them is specified as a BLA number. BLA



depends on support from a SIP server.

Any IP phone can be used to originate or receive calls on the bridge line. An incoming call to the BLA number can be presented to multiple phones in the group simultaneously. The incoming call can be answered on any IP phone of the group but not all.

IP phones support BLA using a SUBSCRIBE/NOTIFY mechanism as specified in [RFC 3265](#). The event used is:

- "dialog" for bridged line appearance subscribe and notify

If the call is placed on public hold, the held call is available for all phones in the group to retrieve.

## Procedure

BLA can be configured using the following methods.

Central Provisioning (Configuration File)	<MAC>.cfg	Configure the registration line type. <b>Parameter:</b> account.X.shared_line
		Configure the BLA number. <b>Parameter:</b> account.X.bla_number
		Configure the period of BLA subscription. <b>Parameter:</b> account.X.bla_subscribe_period
Web User Interface		Configure the registration line type. Configure the BLA number. Configure the period of BLA subscription. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>account.X.shared_line</b> <sup>[1]</sup>	<b>0, 1 or 3</b>	<b>0</b>
<b>Description:</b> Configures the registration line type for a specific account.		

Parameters	Permitted Values	Default
<b>0</b> -Disabled, the shared line feature is disabled. <b>1</b> -Shared Call Appearance <b>3</b> -Draft BLA <b>Web User Interface:</b> Account->Advanced->Shared Line <b>Phone User Interface:</b> None		
<b>account.X.bla_number<sup>[1]</sup></b>	<b>String within 99 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the BLA number for a specific account. <b>Example:</b> account.1.bla_number = 14084588327 <b>Note:</b> It works only if the value of the parameter "account.X.shared_line" is set to 3 (Draft BLA). <b>Web User Interface:</b> Account->Advanced->BLA Number <b>Phone User Interface:</b> None		
<b>account.X.bla_subscribe_period<sup>[1]</sup></b>	<b>Integer from 60 to 7200</b>	<b>300</b>
<b>Description:</b> Configures the period (in seconds) of the BLA subscription for a specific account. <b>Note:</b> It works only if the value of the parameter "account.X.shared_line" is set to 3 (Draft BLA). <b>Web User Interface:</b> Account->Advanced->BLA Subscription Period <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure the BLA feature via web user interface:**

1. Click on **Account->Advanced**.

2. Select the desired account from the pull-down list of **Account**.
3. Select **Draft BLA** from the pull-down list of **Shared Line**.
4. Enter the desired value in the **BLA Number** field.
5. Enter the desired value in the **BLA Subscription Period** field.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is active. The 'Account' dropdown is set to 'Account 1'. The 'Keep Alive Type' is 'Default', 'Keep Alive Interval(Seconds)' is '30', 'RPort' is 'Disabled', and 'Subscribe Period(Seconds)' is '1800'. The 'Shared Line' dropdown is set to 'Draft BLA'. The 'Dialog Info Call Pickup' is 'Enabled'. The 'BLA Number' field contains '14084588327' and the 'BLA Subscription Period' field contains '300'. The 'SIP Send MAC' is 'Disabled', 'BLF List Retrieve Call Parked Code' is '\*88', 'Shared Line' is 'Shared Call Appearance', 'Call Pull Feature Access Code' is '\*11', 'Dialog Info Call Pickup' is 'Enabled', 'VQ RTPC-XR Collector Name' is 'collector', 'VQ RTPC-XR Collector Address' is '10.2.1.98', and 'VQ RTPC-XR Collector Port' is '5060'. The 'Confirm' button is at the bottom.

6. Click **Confirm** to accept the change.

## Message Waiting Indicator (MWI)

Message Waiting Indicator (MWI) informs users that how many messages are waiting in their mailbox without calling the mailbox. IP phones support both audio and visual MWI when receiving new voice messages. MWI will be indicated in four ways: a warning tone, an indicator message (including a voice mail icon) on the LCD screen, the power indicator LED slowly flashes red and the MESSAGE key LED lights up (MESSAGE key LED is only applicable to SIP-T29G/T27P/T27G/T23P/T23G/T21(P) E2 IP phones). For more information on power indicator LED, refer to [Power Indicator LED](#) on page 133.

IP phones support both solicited and unsolicited MWI.

### Unsolicited MWI

Unsolicited MWI is a server related feature. The IP phone sends a SUBSCRIBE message to the server for message-summary updates. The server sends a message-summary NOTIFY within the subscription dialog each time the MWI status changes.

### Solicited MWI

For solicited MWI, you must enable MWI subscription feature on IP phones. IP phones support subscribing the MWI messages to the account or the voice mail number.

## Procedure

Configuration changes can be performed using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure subscribe for MWI. <b>Parameters:</b> account.X.subscribe_mwi account.X.subscribe_mwi_expires
		Configure subscribe MWI to voice mail. <b>Parameter:</b> account.X.subscribe_mwi_to_vm
		Configure the voice mail number on a per-line basis. <b>Parameter:</b> voice_mail.number.X
		Configure the presentation of audio and visual MWI. <b>Parameter:</b> account.X.display_mwi.enable
<b>Web User Interface</b>		Configure subscribe for MWI. Configure subscribe MWI to voice mail. Configure the voice mail number on a per-line basis. Configure the presentation of audio and visual MWI. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>
<b>Phone User Interface</b>		Configure the voice mail number on a per-line basis.

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.subscribe_mwi <sup>[1]</sup>	0 or 1	0
<b>Description:</b> Enables or disables the IP phone to subscribe the message waiting indicator for a specific		

Parameters	Permitted Values	Default
<p>account.</p> <p><b>0</b>-Disabled, the server automatically sends a message-summary NOTIFY in a new dialog each time the MWI status changes. (This requires server support).</p> <p><b>1</b>-Enabled, the IP phone will send a SUBSCRIBE message to the server for message-summary updates.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;Subscribe for MWI</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.subscribe_mwi_expires<sup>[1]</sup></b>	<b>Integer from 0 to 84600</b>	<b>3600</b>
<p><b>Description:</b></p> <p>Configures MWI subscribe expiry time (in seconds) for a specific account.</p> <p>The IP phone is able to successfully refresh the SUBSCRIBE for message-summary events before expiration of the subscription dialog.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.subscribe_mwi" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;MWI Subscription Period (Seconds)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.subscribe_mwi_to_vm<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to subscribe the message waiting indicator to the voice mail number for a specific account.</p> <p><b>0</b>-Disabled, the IP phone will subscribe the message waiting indicator to the account X.</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.subscribe_mwi" is set to 1 (Enabled) and "voice_mail.number.X" is configured.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;Subscribe MWI To Voice Mail</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>voice_mail.number.X<sup>[1]</sup></b>	<b>String within 99</b>	<b>Blank</b>

Parameters	Permitted Values	Default
	<b>characters</b>	
<p><b>Description:</b> Configures the voice mail number for a specific account.</p> <p><b>Example:</b> voice_mail.number.1 = 1234</p> <p><b>Web User Interface:</b> Account-&gt;Advanced-&gt;Voice Mail</p> <p><b>Phone User Interface:</b> Menu-&gt;Message-&gt;Voice Mail-&gt;Set Voice Mail-&gt;AccountX Code</p>		
<b>account.X.display_mwi.enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the IP phone to present audio and visual MWI when receiving new voice messages.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It always works at the time of Unsolicited MWI; at the time of solicited MWI, MWI subscription feature should be configured in advance. It works only if the values of parameters "features.voice_mail_tone_enable", "features.voice_mail_popup.enable" and "phone_setting.mail_power_led_flash_enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Account-&gt;Advanced-&gt;Voice Mail Display</p> <p><b>Phone User Interface:</b> None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure subscribe for MWI via web user interface:**

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Subscribe for MWI**.

- Enter the period time in the **MWI Subscription Period(Seconds)** field.

The screenshot shows the Yealink T466 web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Account' dropdown is set to 'Account1'. The 'Subscribe for MWI' field is set to 'Enabled' and the 'MWI Subscription Period(Seconds)' field is set to '3600'. A red box highlights these two fields. The 'Subscribe MWI To Voice Mail' field is set to 'Disabled'.

- Click **Confirm** to accept the change.

**To configure subscribe MWI to voice mail via web user interface:**

- Click on **Account->Advanced**.
- Select the desired account from the pull-down list of **Account**.
- Select **Enabled** from the pull-down list of **Subscribe for MWI**.
- Select the desired value from the pull-down list of **Subscribe MWI To Voice Mail**.
- Enter the desired voice number in the **Voice Mail** field.

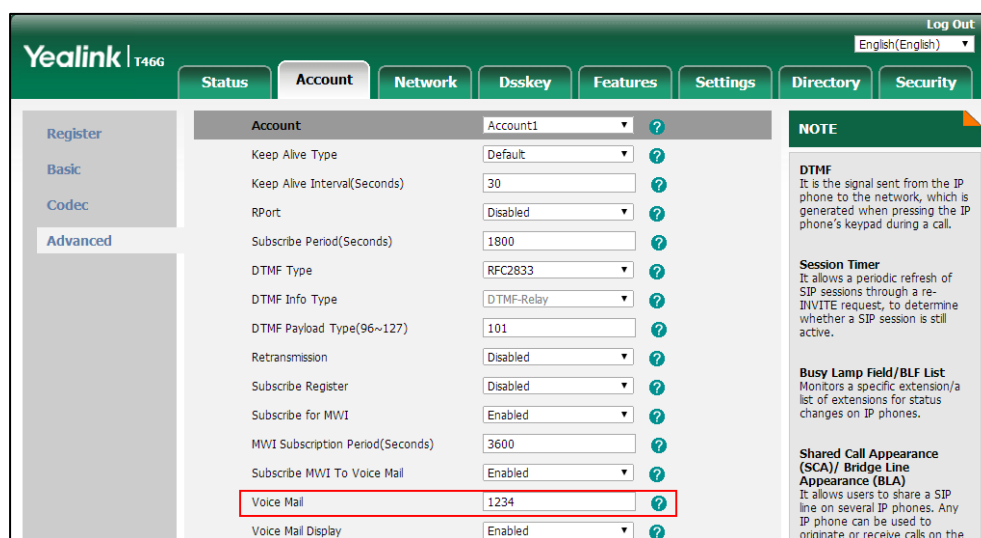
The screenshot shows the Yealink T466 web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Account' dropdown is set to 'Account1'. The 'Subscribe for MWI' field is set to 'Enabled' and the 'MWI Subscription Period(Seconds)' field is set to '3600'. The 'Subscribe MWI To Voice Mail' field is set to 'Enabled' and the 'Voice Mail' field is set to '1234'. A red box highlights these four fields. The 'Voice Mail Display' field is set to 'Enabled'.

- Click **Confirm** to accept the change.

**To configure the presentation of audio and visual MWI via web user interface:**

- Click on **Account->Advanced**.
- Select the desired account from the pull-down list of **Account**.

3. Select the desired value from the pull-down list of **Voice Mail Display**.



The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected. The 'Voice Mail' field is highlighted with a red box, and the 'Voice Mail Display' field is also highlighted with a red box. The 'Voice Mail Display' field is currently set to 'Enabled'.

Field	Value
Account	Account1
Keep Alive Type	Default
Keep Alive Interval(Seconds)	30
RPort	Disabled
Subscribe Period(Seconds)	1800
DTMF Type	RFC2833
DTMF Info Type	DTMF-Relay
DTMF Payload Type(96~127)	101
Retransmission	Disabled
Subscribe Register	Disabled
Subscribe for MWI	Enabled
MWI Subscription Period(Seconds)	3600
Subscribe MWI To Voice Mail	Enabled
Voice Mail	1234
Voice Mail Display	Enabled

**NOTE**

**DTMF**  
It is the signal sent from the IP phone to the network, which is generated when pressing the IP phone's keypad during a call.

**Session Timer**  
It allows a periodic refresh of SIP sessions through a re-INVITE request, to determine whether a SIP session is still active.

**Busy Lamp Field/BLF List**  
Monitors a specific extension/a list of extensions for status changes on IP phones.

**Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA)**  
It allows users to share a SIP line on several IP phones. Any IP phone can be used to originate or receive calls on the

4. Click **Confirm** to accept the change.

## Short Message Service (SMS)

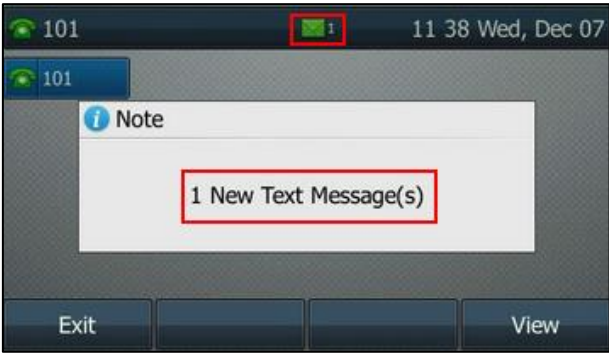
SMS feature allows users to send and receive text messages using Yealink IP phones. It depends on support from a SIP server.

You can send text messages by initiating a new dialogue or by replying to a received message. By default, this feature is enabled. You will find the **Text Message** option on the IP phone user interface: **Menu->Message**, where you can send text messages. If this feature is disabled, the **Text Message** option will disappear.





When receiving a new text message, the phone will play a warning tone. The power indicator LED will slowly flash red, and the LCD screen will prompt receiving new text messages with the number of waiting messages or a flashing icon. You can customize the warning tone or select specialized tone sets (vary from country to country) for your IP phone. For more information, refer to [Tones](#) on page 724.



**Procedure**

Configuration changes can be performed using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure SMS for the IP phone. <b>Parameter:</b> features.text_message.enable
--------------------------------------------------	---------------------	--------------------------------------------------------------------------------------

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
features.text_message.enable	0 or 1	1
<b>Description:</b> Enables or disables the IP phone to send or receive text message(s). <b>0</b> -Disabled, the IP phone can neither send nor receive text message(s). <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

**Multicast Paging**

Multicast paging allows IP phones to send/receive Real-time Transport Protocol (RTP) streams to/from the pre-configured multicast address(es) on the desired channel without involving SIP

signaling. Up to 31 listening multicast addresses can be specified on the IP phone.

Yealink IP phones support the following 31 channels:

- **0:** Broadcasts are sent to channel 0. Note that the Yealink IP phones running firmware version 80 or earlier can be regarded as listening to channel 0. It is the default channel.
- **1 to 25:** Broadcasts are sent to channel 1 to 25. We recommend that you specify these channels when broadcasting with Polycom IP phones which have 25 channels you can listen to.
- **26 to 30:** Broadcasts are sent to channel 26 to 30. We recommend that you specify these channels when broadcasting with Yealink IP phones running firmware version 81 or later.

The IP phones can only send and receives broadcasts to/from the listened channels. Other channels' broadcasts will be ignored automatically by the IP phone.

## Sending RTP Stream

Users can send an RTP stream without involving SIP signaling by pressing a configured multicast paging DSS key or a paging list DSS key. A multicast address (IP: Port) and a channel (0 to 30) should be assigned to the multicast paging DSS key, which is defined to transmit RTP stream to a group of designated IP phones on the desired channel.

When the IP phone sends the RTP stream to a pre-configured multicast address belonging to a desired channel, each IP phone preconfigured to listen to the multicast address on the same channel can receive the RTP stream. When the originator stops sending the RTP stream, the subscribers stop receiving it.

### Procedure

Configuration changes can be performed using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify a multicast codec for the IP phone to send the RTP stream. <b>Parameter:</b> multicast.codec
		Configure the multicast IP address and port number for a paging list DSS key. <b>Parameter:</b> multicast.paging_address.X.ip_address
		Configure the multicast paging group name for a paging list DSS key. <b>Parameter:</b> multicast.paging_address.X.label
		Configure the channel of the multicast paging group for a paging list DSS key.

		<b>Parameter:</b> multicast.paging_address.X.channel
		Assign a multicast paging DSS key. <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type linekey.X.value/ programablekey.X.value/ expansion_module.X.key.Y.value linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel linekey.X.extension/ programablekey.X.extension/ expansion_module.X.key.Y.extension
		Assign a paging list DSS key. <b>Parameter:</b> linekey.X.type/ programmable.X.type/ expansion_module.X.key.Y.type linekey.X.label/ programmable.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Specify a multicast codec for the IP phone to send the RTP stream. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load</a>
		Configure the multicast IP address and port number for a paging list DSS key. Configure the multicast paging group name for a paging list DSS key. Configure the channel of the multicast paging group for a paging list DSS key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=contacts-multicastIP&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=contacts-multicastIP&amp;q=load</a>
		Assign a multicast paging DSS key or a paging list DSS key. <b>Navigate to:</b>

	http://<phoneIPAddress>/servlet?p=dsskey &q=load&model=0
<b>Phone User Interface</b>	<p>Configure the multicast IP address and port number for a paging list DSS key.</p> <p>Configure the multicast paging group name for a paging list DSS key.</p> <p>Configure the channel of the multicast paging group for a paging list DSS key.</p> <p>Assign a multicast paging DSS key or a paging list DSS key.</p>

**Details of the Configuration Parameters:**

Parameters	Permitted Values	Default
<b>multicast.codec</b>	<b>PCMU, PCMA, G729, G722</b>	<b>G722</b>
<p><b>Description:</b> Configures the codec of multicast paging.</p> <p><b>Example:</b> multicast.codec = G722</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones.</p> <p><b>Web User Interface:</b> Features-&gt;General Information-&gt;Multicast Codec</p> <p><b>Phone User Interface:</b> None</p>		
<b>multicast.paging_address.X.ip_address</b> (X ranges from 1 to 31)	<b>String</b>	<b>Blank</b>
<p><b>Description:</b> Configures the IP address and port number of the multicast paging group in the paging list. It will be displayed on the LCD screen when placing the multicast paging call.</p> <p><b>Example:</b> multicast.paging_address.1.ip_address = 224.5.6.20:10008 multicast.paging_address.2.ip_address = 224.1.6.25:1001</p> <p><b>Note:</b> The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.</p> <p><b>Web User Interface:</b> Directory-&gt;Multicast IP-&gt;Paging List-&gt;Paging Address</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> Menu->Features->Paging List->Option->Edit->Address		
<b>multicast.paging_address.X.label</b> (X ranges from 1 to 31)	<b>String</b>	<b>Blank</b>
<b>Description:</b> Configures the name of the multicast paging group to be displayed in the paging list. It will be displayed on the LCD screen when placing the multicast paging calls. <b>Example:</b> multicast.paging_address.1.label = Product multicast.paging_address.2.label = Sales <b>Web User Interface:</b> Directory->Multicast IP->Paging List->Label <b>Phone User Interface:</b> Menu->Features->Paging List->Option->Edit->Label		
<b>multicast.paging_address.X.channel</b> (X ranges from 1 to 31)	<b>Integer from 0 to 30</b>	<b>0</b>
<b>Description:</b> Configures the channel of the multicast paging group in the paging list. If it is set to 0, all the Yealink IP phones running firmware version 80 or prior or Yealink IP phones listens to channel 0 or third-party available devices (for example, Cisco IP phones) in the paging group can receive the RTP stream. If it is set to 1 to 25, the Polycom or Yealink IP phones preconfigured to listen to the channel can receive the RTP stream. If it is set to 26 to 30, the Yealink IP phones preconfigured to listen to the channel can receive the RTP stream. <b>Example:</b> multicast.paging_address.1.channel = 3 multicast.paging_address.2.channel = 5 <b>Web User Interface:</b> Directory->Multicast IP->Paging List->Channel <b>Phone User Interface:</b> Menu->Features->Paging List->Option->Edit->Channel		

## Multicast Paging DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type</b> <sup>[1]</sup> <b>programmablekey.X.type</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.type</b> <sup>[3]</sup>	24
<b>Description:</b> Configures a DSS key as a multicast paging DSS key on the IP phone. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.value</b> <sup>[1]</sup> <b>programmablekey.X.value</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.value</b> <sup>[3]</sup>	String within 99 characters
<b>Description:</b> Configures the multicast IP address and port number. <b>Note:</b> The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Value <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Value	
<b>linekey.X.label</b> <sup>[1]</sup> <b>programmablekey.X.label</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.label</b> <sup>[3]</sup>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> EXT key is only applicable to	

<p>SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key/Programable Key-&gt;Label</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Label</p>	
<p><b>linekey.X.shortlabel</b></p> <p>(X ranges from 1 to 21)</p>	<p><b>String within 99 characters</b></p>
<p><b>Description:</b></p> <p>(Optional.) Configures the short label displayed on the LCD screen for line key.</p> <p><b>Note:</b> It is only applicable to SIP-T52S IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key-&gt;Short</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Short Label</p>	
<p><b>linekey.X.extension<sup>[1]</sup></b></p> <p><b>programmablekey.X.extension<sup>[2]</sup></b></p> <p><b>expansion_module.X.key.Y.extension<sup>[3]</sup></b></p>	<p><b>String within 256 characters</b></p>
<p><b>Description:</b></p> <p>Configures the channel of multicast paging group.</p> <p><b>Example:</b></p> <p>linekey.2.extension = 2</p> <p><b>Note:</b> This parameter only applies to multicast paging feature. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key/Programable Key-&gt;Extension</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Channel</p>	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programmable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

## Paging List DSS key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type</b> <sup>[1]</sup> <b>programmablekey.X.type</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.type</b> <sup>[3]</sup>	66
<b>Description:</b> Configures a DSS key as a paging list DSS key on the IP phone. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.label</b> <sup>[1]</sup> <b>programmablekey.X.label</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.label</b> <sup>[3]</sup>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones. <b>Web User Interface:</b>	



Parameters	Permitted Values
Dsskey->Line Key->Short	
<b>Phone User Interface:</b>	
Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programmable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure a codec for multicast paging via web user interface:

1. Click on **Features->General Information**.
2. Select the desired codec from the pull-down list of **Multicast Codec**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'Multicast Codec' is set to 'G722'. Other settings include 'Call Waiting' (Enabled), 'Auto Redial' (Disabled), 'DTMF Repetition' (3), and 'Play Hold Tone' (Enabled). A 'NOTE' section on the right provides additional information about various features like Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

3. Click **Confirm** to accept the change.

#### To configure two sending multicast addresses via web user interface:

1. Click on **Directory->Multicast IP**.
2. Enter the sending multicast address and port number in the **Paging Address** field.
3. Enter the label in the **Label** field.

The label will appear on the LCD screen when sending the RTP multicast.

- Select the desired channel from the pull-down list **Channel**.

- Click **Confirm** to accept the change.

**To configure a multicast paging DSS key via web user interface:**

- Click on **Dsskey**->**Line Key** (or **Programmable Key**).
- In the desired DSS key field, select **Paging** from the pull-down list of **Type**.
- Enter the multicast IP address and port number in the **Value** field.  
The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- Enter the desired channel in the **Extension** field.  
The valid channel ranges from 0 to 30.

- Click **Confirm** to accept the change.

**To configure a paging list DSS key via web user interface:**

- Click on **Dsskey**->**Line Key** (or **Programmable Key**).
- In the desired DSS key field, select **Paging List** from the pull-down list of **Type**.

- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

- Click **Confirm** to accept the change.

#### To configure a multicast paging DSS key via phone user interface:

- Press **Menu->Features->DSS Keys**.
- Select the desired DSS key.
- Press or , or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press or , or the **Switch** soft key to select **Multicast Paging** from the **Key Type** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- Enter the multicast IP address and port number in the **Value** field.
- Enter the desired channel in the **Channel** field.
- Press the **Save** soft key to accept the change.

#### To configure a paging list DSS key via phone user interface:

- Press **Menu->Features->DSS Keys**.
- Select the desired DSS key.
- Press or , or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press or , or the **Switch** soft key to select **Paging List** from the **Key Type** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- Press the **Save** soft key to accept the change.

## Receiving RTP Stream

IP phones can receive an RTP stream from the pre-configured multicast address(es) on the desired channel without involving SIP signaling, and can handle the incoming multicast paging calls differently depending on the configurations of Paging Barge and Paging Priority Active.

### Paging Barge

Paging Barge feature defines the lowest priority of the multicast paging call that can be received when there is a voice call (a normal phone call rather than a multicast paging call) in progress. If it is disabled, all incoming multicast paging calls will be automatically ignored. If it is set to a

specify priority value, the incoming multicast paging calls with higher or equal priority are automatically answered and the ones with lower priority are ignored.

### Ignore DND

Ignore DND feature defines the lowest priority of the multicast paging call that can be received when DND is activated in phone mode. If it is disabled, all incoming multicast paging calls will be automatically ignored when DND is activated in phone mode. If it is set to a specify priority value, the incoming multicast paging calls with higher or equal priority are automatically answered and the ones with lower priority are ignored. The phone will automatically answer all incoming multicast paging calls when DND is activated in custom mode.

### Paging Priority Active

Paging Priority Active feature decides how the IP phone handles the incoming multicast paging calls when there is already a multicast paging call in progress. If it is disabled, the IP phone will automatically ignore all incoming multicast paging calls. If it is enabled, an incoming multicast paging call with higher priority or equal is automatically answered, and the one with lower priority is ignored.

### Procedure

Configuration changes can be performed using the following methods.

Central Provisioning (Configuration File)	<y0000000000xx>.cfg	Configure the listening multicast address. <b>Parameters:</b> multicast.listen_address.X.ip_address multicast.listen_address.X.label multicast.listen_address.X.channel multicast.listen_address.X.volume multicast.receive.use_speaker
		Configure Paging Priority Active, Paging Barge and Ignore DND features. <b>Parameters:</b> multicast.receive_priority.enable multicast.receive_priority.priority multicast.receive.ignore_dnd.priority
Web User Interface		Configure the listening multicast address. Configure Paging Priority Active, Paging Barge and Ignore DND features. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=contacts-multicastIP&q=load

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>multicast.listen_address.X.ip_address</b> (X ranges from 1 to 31)	<b>IP address: port</b>	<b>Blank</b>
<b>Description:</b> Configures the multicast address and port number that the IP phone listens to. <b>Example:</b> multicast.listen_address.1.ip_address = 224.5.6.20:10008 <b>Note:</b> The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255. <b>Web User Interface:</b> Directory->Multicast IP->Multicast Listening->Listening Address <b>Phone User Interface:</b> None		
<b>multicast.listen_address.X.label</b> (X ranges from 1 to 31)	<b>String within 99 characters</b>	<b>Blank</b>
<b>Description:</b> (Optional.) Configures the label to be displayed on the LCD screen when receiving the multicast paging calls. <b>Example:</b> multicast.listen_address.1.label = Paging1 <b>Web User Interface:</b> Directory->Multicast IP->Multicast Listening->Label <b>Phone User Interface:</b> None		
<b>multicast.listen_address.X.channel</b> (X ranges from 1 to 31)	<b>Integer from 0 to 30</b>	<b>0</b>
<b>Description:</b> Configures the channel that the IP phone listens to. If it is set to 0, the IP phone can receive an RTP stream of the pre-configured multicast address from the IP phones running old firmware version (old provisioning mechanism), from the IP phones listen to the channel 0, or from the available third-party devices (for example, Cisco IP phones). If it is set to 1 to 25, the IP phone can receive an RTP stream of the pre-configured multicast address on the channel 1 to 25 respectively from Yealink or Polycom IP phones. If it is set to 26 to 30, the IP phone can receive the RTP stream of the pre-configured		

Parameters	Permitted Values	Default
<p>multicast address on the channel 26 to 30 respectively from Yealink IP phones.</p> <p><b>Example:</b></p> <p>multicast.listen_address.1.channel = 2</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;Multicast IP-&gt;Multicast Listening-&gt;Channel</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>multicast.listen_address.X.volume</b></p> <p>(X ranges from 1 to 31)</p>	<p><b>Integer from 0 to 15</b></p>	<p><b>0</b></p>
<p><b>Description:</b></p> <p>Configures the volume of the speaker when receiving the multicast paging calls.</p> <p>If it is set to 0, the current volume of the speaker takes effect. The volume of the speaker can be adjusted by pressing the Volume key in advance when the phone is during a call. You can also adjust the volume of the speaker during the paging call.</p> <p>If it is set to 1 to 15, the configured volume takes effect and the current volume of the speaker will be ignored. You are not allowed to adjust the volume of the speaker during the paging call.</p> <p><b>Example:</b></p> <p>multicast.listen_address.1.volume = 1</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>multicast.receive.use_speaker</b></p>	<p><b>0 or 1</b></p>	<p><b>0</b></p>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to always use the speaker as the audio device when receiving the multicast paging calls.</p> <p><b>0</b>-Disabled, the engaged audio device will be used when receiving the multicast paging calls.</p> <p><b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

Parameters	Permitted Values	Default
<b>multicast.receive_priority.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to handle the incoming multicast paging calls when there is an active multicast paging call on the IP phone.</p> <p><b>0</b>-Disabled, the IP phone will ignore the incoming multicast paging calls when there is an active multicast paging call on the IP phone.</p> <p><b>1</b>-Enabled, the IP phone will receive the incoming multicast paging call with a higher or equal priority and ignore that with a lower priority.</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;Multicast IP-&gt;Paging Priority Active</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>multicast.receive_priority.priority</b>	<b>Integer from 0 to 31</b>	<b>31</b>
<p><b>Description:</b></p> <p>Configures the lowest priority of the multicast paging call that can be received when there is a voice call (a normal phone call rather than a multicast paging call) in progress.</p> <p>1 is the highest priority, 31 is the lowest priority.</p> <p><b>0</b>-Disabled, all incoming multicast paging calls will be automatically ignored when a voice call is in progress.</p> <p><b>1-1</b></p> <p><b>2-2</b></p> <p><b>3-3</b></p> <p>...</p> <p><b>31-31</b></p> <p>If it is set to other values, the IP phone will receive the incoming multicast paging call with a higher or equal priority and ignore that with a lower priority when a voice call is in progress.</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;Multicast IP-&gt;Paging Barge</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>multicast.receive.ignore_dnd.priority</b>	<b>Integer from 0 to 31</b>	<b>0</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures the lowest priority of the multicast paging call that can be received when DND is activated in phone mode.</p> <p>1 is the highest priority, 31 is the lowest priority.</p> <p><b>0</b>-Disabled, all incoming multicast paging calls will be automatically ignored when DND is activated in phone mode.</p> <p><b>1-1</b></p> <p><b>2-2</b></p> <p><b>3-3</b></p> <p>...</p> <p><b>31-31</b></p> <p>If it is not set to 0 (Disabled), the IP phone will receive the incoming multicast paging call with a higher or same priority than this value and ignore that with a lower priority than this value when DND is activated in phone mode.</p> <p><b>Web User Interface:</b></p> <p>Directory-&gt;Multicast IP-&gt;Ignore DND</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure multicast listening addresses via web user interface:**

1. Click on **Directory->Multicast IP**.
2. Select the desired value from the pull-down list of **Paging Barge**.
3. Select the desired value from the pull-down list of **Ignore DND**.
4. Select the desired value from the pull-down list of **Paging Priority Active**.
5. Enter the multicast IP address(es) and port number (for example, 224.5.6.20:10008) which the phone listens to for incoming RTP multicast in the **Listening Address** field.  
1 is the highest priority and 31 is the lowest priority.
6. Enter the label in the **Label** field.  
Label will appear on the LCD screen when receiving the multicast RTP stream.



7. Select the desired channel from the pull-down list of **Channel**.

**Multicast Listening**

Paging Barge: 31 ?

Ignore DND: 6 ?

Paging Priority Active: Enabled ?

IP Address	Listening Address	Label	Channel	Priority
1 IP Address	224.5.6.20:10008	Product	2	1
2 IP Address			0	2
3 IP Address			0	3
4 IP Address			0	4

**NOTE**

**Multicast Paging**  
Multicast paging allows IP phones to send/receive Real-time Transport Protocol (RTP) streams to/from the pre-configured multicast address(es) without involving SIP signaling. Up to 10 listening multicast addresses can be specified on the IP phone.

You can click here to get more guides.

8. Click **Confirm** to accept the change.

## Dsskey Recording (Record and URL Record)

Yealink IP phones support record calls by pressing the call record DSS key. It depends on support from a SIP server. When the user presses the call record DSS key, the IP phone sends a record request to the server. IP phones themselves do not have memory to store the recording, what they can do is to trigger the recording and indicate the recording status.

Normally, there are 2 main methods to trigger a recording on a certain server. We call them record and URL record. Record is for the IP phone to send the server a SIP INFO message containing a specific header. URL record is for the IP phone to send the server an HTTP GET message containing a specific URL. The server processes these messages and decides to start or stop a recording.

**Note** It is not applicable to SIP-T19(P) E2 IP phones.

### Record

When a user presses a record DSS key during a call, the IP phone sends a SIP INFO message to the server with the specific header "Record: on", and then the recording starts.

Example of a SIP INFO message:

```
Via: SIP/2.0/UDP 10.3.20.14:5060;branch=z9hG4bK1870385345
From: "1009" <sip:1009@10.2.1.48:5060>;tag=1385842459
To: <sip:1006@10.2.1.48:5060>;tag=2383911905
Call-ID: 0_1289812066@10.3.20.14
CSeq: 2 INFO
Contact: <sip:1009@10.3.20.14:5060>
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Record: on
```

```
Content-Length: 0
```

When the user presses the record DSS key again, the IP phone sends a SIP INFO message to the server with the specific header "Record: off", and then the recording stops.

Example of a SIP INFO message:

```
Via: SIP/2.0/UDP 10.3.20.14:5060;branch=z9hG4bK175716007
From: "1009" <sip:1009@10.2.1.48:5060>;tag=1385842459
To: <sip:1006@10.2.1.48:5060>;tag=2383911905
Call-ID: 0_1289812066@10.3.20.14
CSeq: 3 INFO
Contact: <sip:1009@10.3.20.14:5060>
Max-Forwards: 70
User-Agent: Yealink SIP-T46G 28.82.0.20
Record: off
Content-Length: 0
```

## URL Record

When a user presses a URL record DSS key during a call, the IP phone sends an HTTP GET message to the server.

Example of an HTTP GET message:

```
GET /URLRecord/record.xml HTTP/1.1\r\n
Request Method: GET
Request URI: /URLRecord/record.xml
Request version: HTTP/1.1
Host: 10.3.5.97:8080\r\n
User-Agent: Yealink SIP-T46G 28.82.0.20 00:15:65:74:B1:50\r\n
```

If the recording is successfully started, the server will respond with a 200 OK message.

Example of a 200 OK message:

```
<YealinkIPPhoneText>
<Title>
  </Title>
<Text>
  The recording session is successfully started.
</Text>
</YealinkIPPhoneText>
```

If the recording fails for some reasons, for example, the recording box is full, the server will respond with a 200 OK message.

Example of a 200 OK message:

```
<YealinkIPPhoneText>
```

```
<Title>
  </Title>
<Text>
  Probably the recording box is full.
  </Text>
<YealinkIPPhoneText>
```

When the user presses the URL record DSS key again, the IP phone sends an HTTP GET message to the server, and then the server will respond with a 200 OK message.

Example of a 200 OK message:

```
<YealinkIPPhoneText>
<Title>
  </Title>
<Text>
  The recording session is successfully stopped.
  </Text>
<YealinkIPPhoneText>
```

### Procedure

Record or URL record DSS key can be configured using the following methods.

Central Provisioning (Configuration File)	<y0000000000xx>.cfg	Assign a record DSS key. <b>Parameters:</b> linekey.X.type/ expansion_module.X.key.Y.type linekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
		Assign a URL record DSS key. <b>Parameters:</b> linekey.X.type/ expansion_module.X.key.Y.type linekey.X.value/ expansion_module.X.key.Y.value linekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
Web User Interface		Assign a record DSS key and URL record DSS key. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p

	=dsskey&q=load&model=0
<b>Phone User Interface</b>	Assign a record DSS key and URL record DSS key.

## Record DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>expansion_module.X.key.Y.type<sup>[2]</sup></b>	25
<b>Description:</b> Configures a DSS key as a record DSS key on the IP phone.  <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.  <b>Web User Interface:</b> Dsskey->Line Key->Type  <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.label<sup>[1]</sup></b> <b>expansion_module.X.key.Y.label<sup>[2]</sup></b>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key.  <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.  <b>Web User Interface:</b> Dsskey->Line Key->Label  <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key.	

Parameters	Permitted Values
<p><b>Note:</b> It is only applicable to SIP-T52S IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key-&gt;Short</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Short Label</p>	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

## URL Record DSS Key

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>expansion_module.X.key.Y.type<sup>[2]</sup></b>	35
<p><b>Description:</b></p> <p>Configures a DSS key as a URL record DSS key on the IP phone.</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key-&gt;Type</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Type</p>	
<b>linekey.X.value<sup>[1]</sup></b> <b>expansion_module.X.key.Y.value<sup>[2]</sup></b>	String within 99 characters
<p><b>Description:</b></p> <p>Configures the URL to record a call.</p> <p><b>Example:</b></p> <p>linekey.2.value = http://10.3.5.97:8080/URLRecord/record.xml</p> <p><b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b></p>	

Parameters	Permitted Values
Dsskey->Line Key->Value <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Value	
<b>linekey.X.label<sup>[1]</sup></b> <b>expansion_module.X.key.Y.label<sup>[2]</sup></b>	<b>String within 99 characters</b>
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> It is not applicable to SIP-T19(P) E2 IP phones. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	<b>String within 99 characters</b>
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Short <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

**To configure a record DSS key via web user interface:**

1. Click on **Dsskey->Line Key**.
2. In the desired DSS key field, select **Record** from the pull-down list of **Type**.

- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

The screenshot shows the Yealink T46G web interface. The 'DSSkey' tab is selected. On the left, a sidebar lists 'Line Key1-9', 'Line Key10-18', 'Line Key19-27', 'Programable Key', and 'Ext Key'. The main area has a table with columns: Key, Type, Value, Label, Line, and Extension. The 'Line Key2' row is highlighted with a red box. In this row, 'Record' is selected for Type, and 'N/A' is selected for Line. The 'Label' field is empty. Above the table, there are dropdowns for 'Enable Page Tips' (set to Disabled) and 'Label Length' (set to Default). On the right, there is a 'NOTE' section titled 'Line Keys' explaining their function and a link to guides.

- Click **Confirm** to accept the change.

#### To configure a URL record DSS key via web user interface:

- Click on **DSSkey->Line Key**.
- In the desired DSS key field, select **URL Record** from the pull-down list of **Type**.
- Enter the URL in the **Value** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

This screenshot is similar to the previous one, but the 'Line Key2' row is now configured with 'URL Record' as the Type and 'http://10.3.5.97:8080/UR' as the Value. The 'Label' field remains empty. The rest of the interface, including the sidebar and the 'NOTE' section, is identical to the previous screenshot.

- Click **Confirm** to accept the change.

#### To configure a record DSS key via phone user interface:

- Press **Menu->Features->DSS Keys**.
- Select the desired DSS key.
- Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select **Record** from the **Key Type** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- Press the **Save** soft key to accept the change.

#### To configure a URL record DSS key via phone user interface:

- Press **Menu->Features->DSS Keys**.
- Select the desired DSS key.
- Press  $\leftarrow$  or  $\rightarrow$ , or the **Switch** soft key to select **URL Record** from the **Type** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- Enter the URL in the **Value** field.
- Press the **Save** soft key to accept the change.

## Hot Desking

Hot desking originates from the definition of being the temporary physical occupant of a work station or surface by a particular employee. A primary motivation for hot desking is cost reduction. Hot desking is regularly used in places where not all employees are in the office at the same time, or not in the office for a long time, which means actual personal offices would often be vacant, consuming valuable space and resources.

Hot desking allows a user to clear registration configurations of all accounts on the IP phone, and then register his account on line 1. To use this feature, you need to assign a hot desking DSS key.

### Procedure

Hot Desking feature can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the hot desking logon wizard. <b>Parameters:</b> hotdesking.dsskey_register_name_enable hotdesking.dsskey_username_enable hotdesking.dsskey_password_enable hotdesking.dsskey_sip_server_enable hotdesking.dsskey_outbound_enable
		Assign a hot desking DSS key. <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Assign a hot desking DSS key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0">http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0</a>
<b>Phone User Interface</b>		Assign a hot desking DSS key.



**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>hotdesking.dsskey_register_name_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to provide input field of register name on the hot desking logon wizard when pressing the Hot Desking DSS key. <b>0</b> -Disabled <b>1</b> -Enabled, the original register name information will be cleared. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>hotdesking.dsskey_username_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to provide input field of user name on the hot desking logon wizard when pressing the Hot Desking DSS key. <b>0</b> -Disabled <b>1</b> -Enabled, the original user name information will be cleared. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>hotdesking.dsskey_password_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to provide input field of password on the hot desking logon wizard when pressing the Hot Desking DSS key. <b>0</b> -Disabled <b>1</b> -Enabled, the original password information will be cleared. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>hotdesking.dsskey_sip_server_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to provide input field of SIP server on the hot desking logon wizard when pressing the Hot Desking DSS key. <b>0</b> -Disabled <b>1</b> -Enabled, the original SIP server information will be cleared. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>hotdesking.dsskey_outbound_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to provide input field of outbound server on the hot desking logon wizard when pressing the Hot Desking DSS key. <b>0</b> -Disabled <b>1</b> -Enabled, the original outbound server information will be cleared. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Hot Desking DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>programablekey.X.type<sup>[2]</sup></b> <b>expansion_module.X.key.Y.type<sup>[3]</sup></b>	<b>34</b>
<b>Description:</b>	

Parameters	Permitted Values
<p>Configures a DSS key as a hot desking DSS key on the IP phone.</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key-&gt;Type</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Type</p>	
<b>linekey.X.label<sup>[1]</sup></b> <b>programablekey.X.label<sup>[2]</sup></b> <b>expansion_module.X.key.Y.label<sup>[3]</sup></b>	String within 99 characters
<p><b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key.</p> <p><b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key-&gt;Label</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Label</p>	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<p><b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key.</p> <p><b>Note:</b> It is only applicable to SIP-T52S IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key-&gt;Short</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Short Label</p>	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

**To configure a hot desking DSS key via web user interface:**

1. Click on **Dsskey**->**Line Key**.
2. In the desired DSS key field, select **Hot Desking** from the pull-down list of **Type**.
3. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

The screenshot shows the Yealink T466 web interface. The 'Dsskey' tab is selected. On the left, a sidebar lists 'Line Key1-9', 'Line Key10-18', 'Line Key19-27', 'Programable Key', and 'Ext Key'. The main area shows a table for configuring DSS keys. The table has columns: Key, Type, Value, Label, Line, and Extension. The 'Line Key2' row is highlighted with a red box. In this row, 'Type' is set to 'Hot Desking', 'Value' is empty, 'Label' is empty, and 'Line' is set to 'N/A'. Other rows show 'Line' as the type with 'Default' values. On the right, there is a 'NOTE' section about Line Keys and a link to get more guides.

4. Click **Confirm** to accept the change.

**To configure a hot desking DSS key via phone user interface:**

1. Press **Menu**->**Features**->**DSS Keys**.
2. Select the desired DSS key.
3. Press or , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press or , or the **Switch** soft key to select **Hot Desking** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Press the **Save** soft key to accept the change.

## Logon Wizard

Logon wizard allows IP phones to provide the logon wizard during the first startup.

**Note** Logon wizard feature works only if there is no registered account on the IP phone.

### Procedure

Logon wizard can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the logon wizard. <b>Parameters:</b> phone_setting.logon_wizard hotdesking.startup_register_name_enable hotdesking.startup_username_enable hotdesking.startup_password_enable hotdesking.startup_sip_server_enable hotdesking.startup_outbound_enable
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<b>Web User Interface</b>	Configure the logon wizard. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-general&amp;q=load</a>
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**Details of the Configuration Parameters:**

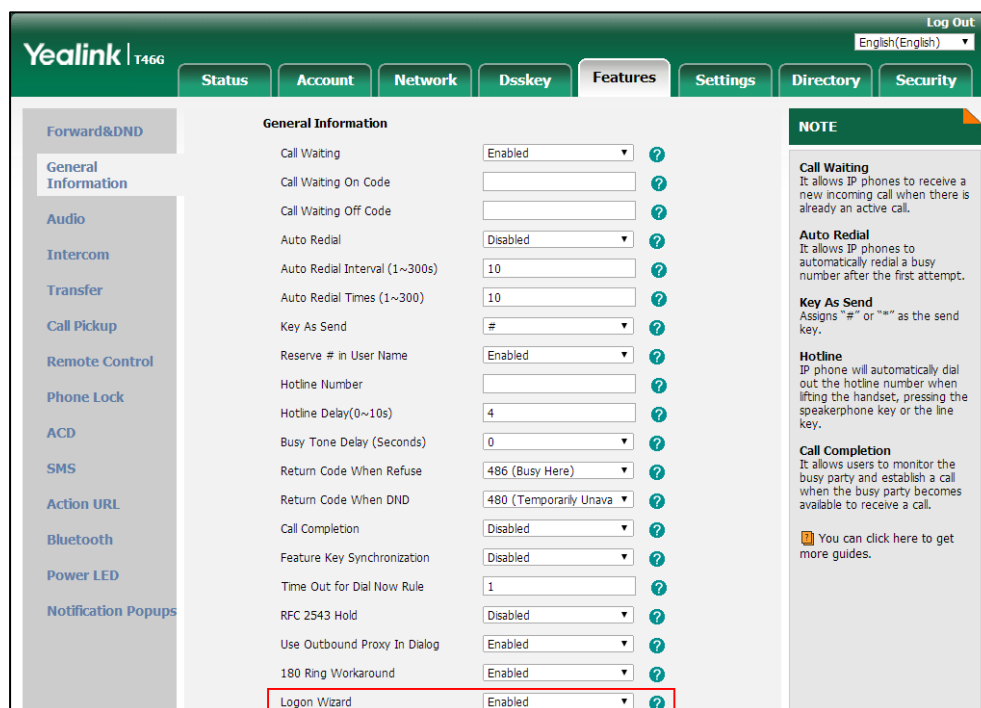
Parameters	Permitted Values	Default
<b>phone_setting.logon_wizard</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to provide the logon wizard after startup when there is no registered account. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Features->General Information->Logon Wizard <b>Phone User Interface:</b> None		
<b>hotdesking.startup_register_name_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to provide input field of register name on the logon wizard after startup when there is no registered account. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.logon_wizard" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>hotdesking.startup_username_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to provide input field of user name on the logon wizard after startup when there is no registered account.		

Parameters	Permitted Values	Default
<b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.logon_wizard" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>hotdesking.startup_password_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to provide input field of password on the logon wizard after startup when there is no registered account. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.logon_wizard" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>hotdesking.startup_sip_server_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to provide input field of SIP server on the logon wizard after startup when there is no registered account. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.logon_wizard" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>hotdesking.startup_outbound_enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to provide input field of outbound server on the logon wizard after startup when there is no registered account.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.logon_wizard" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure logon wizard feature via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Logon Wizard**.



3. Click **Confirm** to accept the change.

## Action URL

Action URL allows IP phones to interact with web server applications by sending an HTTP or HTTPS GET request. You can specify a URL that triggers a GET request when a specified event occurs. Action URL can only be triggered by the pre-defined events (for example, Open DND). The valid URL format is: *http(s)://IP address of the server/help.xml?*

The following table lists the pre-defined events for action URL.

Event	Description
Setup Completed	When the IP phone completes startup.
Registered	When the IP phone successfully registers an account.
Unregistered	When the IP phone logs off the registered account.
Register Failed	When the IP phone fails to register an account.
Off Hook	When the IP phone is off hook.
On Hook	When the IP phone is on hook.
Incoming Call	When the IP phone receives an incoming call.
Outgoing Call	When the IP phone places a call.
Established	When the IP phone establishes a call.
Terminated	When the IP phone terminates a call.
Open DND	When the IP phone enables the DND mode.
Close DND	When the IP phone disables the DND mode.
Open Always Forward	When the IP phone enables the always forward.
Close Always Forward	When the IP phone disables the always forward.
Open Busy Forward	When the IP phone enables the busy forward.
Close Busy Forward	When the IP phone disables the busy forward.
Open NoAnswer Forward	When the IP phone enables the no answer forward.
Close NoAnswer Forward	When the IP phone disables the no answer forward.
Transfer Call	When the IP phone transfers a call.
Blind Transfer	When the IP phone blind transfers a call.
Attended Transfer	When the IP phone performs the semi-attended/attended transfer.
Hold	When the IP phone places a call on hold.
UnHold	When the IP phone resumes a hold call.
Held	When a call of the IP phone is held.



Event	Description
UnHeld	When a held call is resumed.
Mute	When the IP phone mutes a call.
UnMute	When the IP phone un-mutes a call.
Missed Call	When the IP phone misses a call.
IP Changed	When the IP address of the IP phone changes.
Idle To Busy	When the state of the IP phone changes from idle to busy.
Busy To Idle	When the state of phone changes from busy to idle.
Reject Incoming Call	When the IP phone rejects an incoming call.
Answer New-In Call	When the IP phone answers a new call.
Transfer Failed	When the IP phone fails to transfer a call.
Transfer Finished	When the IP phone completes to transfer a call.
Forward Incoming Call	When the IP phone forwards an incoming call.
Autop Finish	When the IP phone completes auto provisioning via power on.
Open Call Waiting	When the IP phone enables the call waiting.
Close Call Waiting	When the IP phone disables the call waiting.
Headset	When the IP phone presses the HEADSET key.
Handfree	When the IP phone presses the Speakerphone key.
Cancel Call Out	When the IP phone cancels an outgoing call in the ring-back state.
Remote Busy	When an outgoing call is rejected.
Call Remote Canceled	When the remote party cancels the outgoing call in the ringing state.

An HTTP or HTTPS GET request may contain variable name and variable value, separated by “=”. Each variable value starts with \$ in the query part of the URL. The valid URL format is: `http(s)://IP address of server/help.xml?variable name=$variable value`. Variable name can be customized by users, while the variable value is pre-defined. For example, a URL `"http://192.168.1.10/help.xml?mac=$mac"` is specified for the event Mute, \$mac will be dynamically replaced with the MAC address of the IP phone when the IP phone mutes a call.

The following table lists pre-defined variable values.

Variable Value	Description
\$mac	The MAC address of the IP phone.
\$ip	The IP address of the IP phone.
\$model	The IP phone model.
\$firmware	The firmware version of the IP phone.
\$active_url	The SIP URI of the current account when the IP phone places a call, receives an incoming call or establishes a call.
\$active_user	The user part of the SIP URI for the current account when the IP phone places a call, receives an incoming call or establishes a call.
\$active_host	The host part of the SIP URI for the current account when the IP phone places a call, receives an incoming call or establishes a call.
\$local	The SIP URI of the caller when the IP phone places a call. The SIP URI of the callee when the IP phone receives an incoming call.
\$remote	The SIP URI of the callee when the IP phone places a call. The SIP URI of the caller when the IP phone receives an incoming call.
\$display_local	The display name of the caller when the IP phone places a call. The display name of the callee when the IP phone receives an incoming call.
\$display_remote	The display name of the callee when the IP phone places a call. The display name of the caller when the IP phone receives an incoming call.
\$call_id	The call-id of the active call.
\$callerID	The display name of the caller when the IP phone receives an incoming call.
\$calledNumber	The phone number of the callee when the IP phone places a call.

## Procedure

Action URL can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure action URL. <b>Parameters:</b> action_url.setup_completed action_url.registered action_url.unregistered action_url.register_failed action_url.off_hook action_url.on_hook action_url.incoming_call action_url.outgoing_call action_url.call_established action_url.dnd_on action_url.dnd_off action_url.always_fwd_on action_url.always_fwd_off action_url.busy_fwd_on action_url.busy_fwd_off action_url.no_answer_fwd_on action_url.no_answer_fwd_off action_url.transfer_call action_url.blind_transfer_call action_url.attended_transfer_call action_url.hold action_url.unhold action_url.held action_url.unheld action_url.mute action_url.unmute action_url.missed_call action_url.call_terminated action_url.busy_to_idle action_url.idle_to_busy action_url.ip_change action_url.forward_incoming_call
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		action_url.reject_incoming_call action_url.answer_new_incoming_call action_url.transfer_finished action_url.transfer_failed action_url.setup_autop_finish action_url.call_waiting_on action_url.call_waiting_off action_url.headset action_url.handfree action_url.cancel_callout action_url.remote_busy action_url.call_remote_canceled
<b>Web User Interface</b>		Configure action URL. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-actionurl&q=load

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>action_url.setup_completed<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends after startup. <b>Example:</b> action_url.setup_completed = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Setup Completed <b>Phone User Interface:</b> None		
<b>action_url.registered<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends after an account is registered. <b>Example:</b> action_url.registered = http://192.168.0.20/help.xml?IP=\$ip		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Features->Action URL->Registered <b>Phone User Interface:</b> None		
<b>action_url.unregistered</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends after an account is unregistered. <b>Example:</b> action_url.unregistered = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Unregistered <b>Phone User Interface:</b> None		
<b>action_url.register_failed</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends after a register failed. <b>Example:</b> action_url.register_failed = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Register Failed <b>Phone User Interface:</b> None		
<b>action_url.off_hook</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when off hook. <b>Example:</b> action_url.off_hook = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Off Hook		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b>  None		
<b>action_url.on_hook<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when on hook.  <b>Example:</b> action_url.on_hook = http://192.168.0.20/help.xml?IP=\$ip  <b>Web User Interface:</b> Features->Action URL->On Hook  <b>Phone User Interface:</b> None		
<b>action_url.incoming_call<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when receiving an incoming call.  <b>Example:</b> action_url.incoming_call = http://192.168.0.20/help.xml?IP=\$ip  <b>Web User Interface:</b> Features->Action URL->Incoming Call  <b>Phone User Interface:</b> None		
<b>action_url.outgoing_call<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when placing a call.  <b>Example:</b> action_url.outgoing_call = http://192.168.0.20/help.xml?IP=\$ip  <b>Web User Interface:</b> Features->Action URL->Outgoing Call  <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>action_url.call_established<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the action URL the IP phone sends when establishing a call.</p> <p><b>Example:</b> action_url.call_established = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b> Features-&gt;Action URL-&gt;Established</p> <p><b>Phone User Interface:</b> None</p>		
<b>action_url.dnd_on<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the action URL the IP phone sends when DND feature is activated.</p> <p><b>Example:</b> action_url.dnd_on = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b> Features-&gt;Action URL-&gt;Open DND</p> <p><b>Phone User Interface:</b> None</p>		
<b>action_url.dnd_off<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the action URL the IP phone sends when DND feature is deactivated.</p> <p><b>Example:</b> action_url.dnd_off = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b> Features-&gt;Action URL-&gt;Close DND</p> <p><b>Phone User Interface:</b> None</p>		
<b>action_url.always_fwd_on<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<b>Description:</b> Configures the action URL the IP phone sends when always forward feature is activated. <b>Example:</b> action_url.always_fwd_on = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Open Always Forward <b>Phone User Interface:</b> None		
<b>action_url.always_fwd_off<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when always forward feature is deactivated. <b>Example:</b> action_url.always_fwd_off = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Close Always Forward <b>Phone User Interface:</b> None		
<b>action_url.busy_fwd_on<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when busy forward feature is activated. <b>Example:</b> action_url.busy_fwd_on = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Open Busy Forward <b>Phone User Interface:</b> None		
<b>action_url.busy_fwd_off<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when busy forward feature is deactivated. <b>Example:</b>		



Parameters	Permitted Values	Default
<p>action_url.busy_fwd_off = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Action URL-&gt;Close Busy Forward</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>action_url.no_answer_fwd_on<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the action URL the IP phone sends when no answer forward feature is activated.</p> <p><b>Example:</b></p> <p>action_url.no_answer_fwd_on = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Action URL-&gt;Open NoAnswer Forward</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>action_url.no_answer_fwd_off<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the action URL the IP phone sends when no answer forward feature is deactivated.</p> <p><b>Example:</b></p> <p>action_url.no_answer_fwd_off = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Action URL-&gt;Close NoAnswer Forward</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>action_url.transfer_call<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the action URL the IP phone sends when performing a transfer.</p> <p><b>Example:</b></p> <p>action_url.transfer_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
Features->Action URL->Transfer Call <b>Phone User Interface:</b> None		
<b>action_url.blind_transfer_call</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when performing a blind transfer. <b>Example:</b> action_url.blind_transfer_call = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Blind Transfer <b>Phone User Interface:</b> None		
<b>action_url.attended_transfer_call</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when performing an attended/semi-attended transfer. <b>Example:</b> action_url.attended_transfer_call = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Attended Transfer <b>Phone User Interface:</b> None		
<b>action_url.hold</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when placing a call on hold. <b>Example:</b> action_url.hold = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Hold <b>Phone User Interface:</b>		

Parameters	Permitted Values	Default
None		
<b>action_url.unhold<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the action URL the IP phone sends when resuming a hold call.</p> <p><b>Example:</b> action_url.unhold = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b> Features-&gt;Action URL-&gt;UnHold</p> <p><b>Phone User Interface:</b> None</p>		
<b>action_url.held<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the action URL the IP phone sends when a call is held.</p> <p><b>Example:</b> action_url.held = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>action_url.unheld<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the action URL the IP phone sends when a held call is resumed.</p> <p><b>Example:</b> action_url.unheld = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>action_url.mute<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<b>Description:</b> Configures the action URL the IP phone sends when muting a call. <b>Example:</b> action_url.mute = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Mute <b>Phone User Interface:</b> None		
<b>action_url.unmute<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when un-muting a call. <b>Example:</b> action_url.unmute = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->UnMute <b>Phone User Interface:</b> None		
<b>action_url.missed_call<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when missing a call. <b>Example:</b> action_url.missed_call = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Missed Call <b>Phone User Interface:</b> None		
<b>action_url.call_terminated<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when terminating a call. <b>Example:</b>		

Parameters	Permitted Values	Default
<p>action_url.call_terminated = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b> Features-&gt;Action URL-&gt;Terminated</p> <p><b>Phone User Interface:</b> None</p>		
<b>action_url.busy_to_idle</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the action URL the IP phone sends when changing the state of the IP phone from busy to idle.</p> <p><b>Example:</b> action_url.busy_to_idle = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b> Features-&gt;Action URL-&gt;Busy To Idle</p> <p><b>Phone User Interface:</b> None</p>		
<b>action_url.idle_to_busy</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the action URL the IP phone sends when changing the state of the IP phone from idle to busy.</p> <p><b>Example:</b> action_url.idle_to_busy = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b> Features-&gt;Action URL-&gt;Idle To Busy</p> <p><b>Phone User Interface:</b> None</p>		
<b>action_url.ip_change</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the action URL the IP phone sends when changing the IP address of the IP phone.</p> <p><b>Example:</b></p>		

Parameters	Permitted Values	Default
<p>action_url.ip_change = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Action URL-&gt;IP Changed</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>action_url.forward_incoming_call<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the action URL the IP phone sends when forwarding an incoming call.</p> <p><b>Example:</b></p> <p>action_url.forward_incoming_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Action URL-&gt;Forward Incoming Call</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>action_url.reject_incoming_call<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the action URL the IP phone sends when rejecting an incoming call.</p> <p><b>Example:</b></p> <p>action_url.reject_incoming_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Action URL-&gt;Reject Incoming Call</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>action_url.answer_new_incoming_call<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the action URL the IP phone sends when answering a new incoming call.</p> <p><b>Example:</b></p> <p>action_url.answer_new_incoming_call = http://192.168.0.20/help.xml?IP=\$ip</p>		

Parameters	Permitted Values	Default
<b>Web User Interface:</b>  Features->Action URL->Answer New Incoming Call  <b>Phone User Interface:</b>  None		
<b>action_url.transfer_finished<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when completing a call transfer.  <b>Example:</b> action_url.transfer_finished = http://192.168.0.20/help.xml?IP=\$ip  <b>Web User Interface:</b>  Features->Action URL->Transfer Finished  <b>Phone User Interface:</b>  None		
<b>action_url.transfer_failed<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when failing to transfer a call.  <b>Example:</b> action_url.transfer_failed = http://192.168.0.20/help.xml?IP=\$ip  <b>Web User Interface:</b>  Features->Action URL->Transfer Failed  <b>Phone User Interface:</b>  None		
<b>action_url.setup_autop_finish<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when completing auto provisioning via power on.  <b>Example:</b> action_url.setup_autop_finish = http://192.168.0.20/help.xml?IP=\$ip		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Features->Action URL->Autop Finish <b>Phone User Interface:</b> None		
<b>action_url.call_waiting_on<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when call waiting feature is enabled. <b>Example:</b> action_url.call_waiting_on = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Open Call Waiting <b>Phone User Interface:</b> None		
<b>action_url.call_waiting_off<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when call waiting feature is disabled. <b>Example:</b> action_url.call_waiting_off = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Close Call Waiting <b>Phone User Interface:</b> None		
<b>action_url.headset<sup>[1]</sup></b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when pressing the HEADSET key. <b>Example:</b> action_url.headset = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Headset		



Parameters	Permitted Values	Default
<b>Phone User Interface:</b>  None		
<b>action_url.handfree</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when pressing the Speakerphone key. <b>Example:</b> action_url.handfree = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Handfree <b>Phone User Interface:</b> None		
<b>action_url.cancel_callout</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when cancels the outgoing call in the ring-back state. <b>Example:</b> action_url.cancel_callout = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Cancel Call Out <b>Phone User Interface:</b> None		
<b>action_url.remote_busy</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the action URL the IP phone sends when the outgoing call is rejected. <b>Example:</b> action_url.remote_busy = http://192.168.0.20/help.xml?IP=\$ip <b>Web User Interface:</b> Features->Action URL->Remote Busy <b>Phone User Interface:</b>		

Parameters	Permitted Values	Default
None		
<b>action_url.call_remote_canceled</b> <sup>[1]</sup>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the action URL the IP phone sends when the remote party cancels the outgoing call in the ringing state.</p> <p><b>Example:</b></p> <p>action_url.call_remote_canceled = http://192.168.0.20/help.xml?IP=\$ip</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Action URL-&gt;Call Remote Canceled</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>The value format is: http(s)://IP address of server/help.xml?variable name=variable value. And the valid variable values are: \$mac, \$ip, \$model, \$firmware, \$active\_url, \$active\_user, \$active\_host, \$local, \$remote, \$display\_local, \$display\_remote, \$call\_id, \$callerID and \$calledNumber.

#### To configure action URL via web user interface:

1. Click on **Features->Action URL**.
2. Enter the action URLs in the corresponding fields.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'Action URL' sub-tab is active. The 'Setup Completed' field is highlighted with a red box and contains the URL 'http://192.168.0.20/help.xml?IP=\$ip'. Other fields like 'Registered', 'Unregistered', 'Register Failed', 'Off Hook', 'On Hook', 'Incoming Call', 'Outgoing Call', 'Established', 'Terminated', 'Open DND', 'Close DND', 'Open Always Forward', and 'Close Always Forward' are also visible. A 'NOTE' box on the right explains the Action URL format and provides a link to more guides.

3. Click **Confirm** to accept the change.

## Action URI

### HTTP/HTTPS GET Request

Opposite to action URL, action URI allows IP phones to interact with web server application by receiving and handling an HTTP or HTTPS GET request. When receiving a GET request, the IP phone will perform the specified action and respond with a 200 OK message. A GET request may contain variable named as "key" and variable value, which are separated by "=". The valid URI format is: *http(s)://<phoneIPAddress>/servlet?key=variable value*. For example:  
http://10.3.20.10/servlet?key=OK.

#### Note

Yealink IP phones are compatible with other two old valid URI formats:  
*http(s)://<phoneIPAddress>/cgi-bin/ConfigManApp.com?key=variable value* and  
*http(s)://<phoneIPAddress>/cgi-bin/cgiServer.exx?key=variable value*.

### SIP Notify Message

In addition, Yealink IP phones support performing the specified action immediately by accepting a SIP NOTIFY message with the "Event: ACTION-URI" header from a SIP proxy server. The message body of the SIP NOTIFY message may contain variable named as "key" and variable value, which are separated by "=".

This method is especially useful for users who always working in the small office/home office where a secure firewall may prevent the HTTP or HTTPS GET request from the external network.

#### Note

If you want to only accept the SIP NOTIFY message from your SIP server and outbound proxy server, you have to enable the Accept SIP Trust Server Only feature. For more information, refer to [Accept SIP Trust Server Only](#) on page 352.

Example of a SIP Notify with the variable value (OK):

#### Message Header

```
NOTIFY sip:3583@10.2.40.10:5062 SIP/2.0
Via: SIP/2.0/UDP 10.2.40.27:5063;branch=z9hG4bK4163876675
From: <sip:3586@10.2.1.48>;tag=2900480538
To: "3583" <sip:3583@10.2.1.48>;tag=490600926
Call-ID: 2923387519@10.2.40.10
CSeq: 4 NOTIFY
Contact: <sip:3586@10.2.40.27:5063>
Max-Forwards: 70
User-Agent: Yealink SIP-T46G
Event: ACTION-URI
Content-Type: message/sipfrag
Content-Length: 6
```

**Message Body**

key=OK

The following table lists pre-defined variable values:

Variable Value	Phone Action
OK	Press the OK/ ✓ key.
ENTER	Press the Enter soft key. (not applicable to SIP-T48S/T48G IP phones)
SPEAKER	Press the Speakerphone key.
F_TRANSFER	Transfers a call to another party.
VOLUME_UP	Increase the volume.
VOLUME_DOWN	Decrease the volume.
MUTE	Mute a call.
F_HOLD/HOLD	Place an active call on hold.
F_CONFERENCE	Press the Conf/Conference soft key.
Cancel/CANCEL	Cancel actions or reject incoming calls or end a call.
X	Cancel actions or reject incoming calls or mute or un-mute calls.
0-9/*/POUND	Press the keypad (0-9, * or #).
L1-LX	Press the line keys (for SIP-T48G/S, X=29; for SIP-T54S/T46G/T46S/T29G, X=27; for SIP-T42G/T42S/T41P/T41S, X=15; for SIP-T52S/T27P/T27G, X=21; for SIP-T40P/T40G/T23P/T23G, X=3; for SIP-T21(P) E2, X=2).
F1-F4	Press the soft keys.
MSG	Press the MESSAGE key.
HEADSET	Press the HEADSET key.
RD	Press the RD/Redial key.
UP/DOWN/LEFT/RIGHT	Press the navigation keys.
Reboot	Reboot the phone.

Variable Value	Phone Action
AutoP	Perform auto provisioning.
DNDOn	Activate the DND feature. <b>Note:</b> For Yealink IP phones, it works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled).
DNDOff	Deactivate the DND feature. <b>Note:</b> For Yealink IP phones, it works only if the value of the parameter "features.dnd.allow" is set to 1 (Enabled).
number=xxx&outgoing_uri=y	Place a call to xxx from SIP URI y. <b>Example:</b> http://10.3.20.10/servlet?key=number=1234&outgoing_uri=1006@10.2.1.48 (1234 means the number you dial out; 1006@10.2.1.48 means the SIP URL you dial from.)
OFFHOOK	Pick up the handset.
ONHOOK	Hang up the handset.
ANSWER/ASW/Asw	Answer a call.
Reset	Reset a phone.
ATrans=xxx	Perform a semi-attended/attended transfer to xxx.
BTrans=xxx	Perform a blind transfer to xxx.
phonecfg=get[&accounts=x][&dnd=x][&fw=x]	Get firmware version, registration, DND or forward configuration information. The valid value of "x" is 0 or 1, 0 means you do not need to get configuration information. 1 means you want to get configuration information. <b>Note:</b> The valid URI is: http(s)://<phoneIPAddress>/servlet?phonecfg=get[&accounts=x][&dnd=x][&fw=x] <b>Example:</b> http://10.3.20.10/servlet?phonecfg=get[&accounts=1][&dnd=0][&fw=1]
CallWaitingOn	Activate the call waiting feature.
CallWaitingOff	Deactivate the call waiting feature.

Variable Value	Phone Action
AlwaysFwdOn/BusyFwdOn/NoAnswFwdOn=xxx=n	<p>Activate an always/busy/no answer forward feature to xxx for the IP phone ("xxx" means the destination number)</p> <p>The valid value of "n" means the duration time (seconds) before forwarding incoming calls (n is the times of 6, for example, 24). It is only applicable to no answer forward feature.</p> <p><b>Note:</b> For Yealink IP phones, it works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and call forward mode is Phone, the always/busy/no answer forward feature will apply to all the accounts on the phone.</p> <p><b>Example:</b></p> <p>http://10.10.20.10/servlet?key=NoAnswFwdOn=1001=24</p>
AlwaysFwdOff/BusyFwdOff/NoAnswFwdOff	<p>Deactivate the always/busy/no answer forward feature for the IP phone.</p> <p><b>Note:</b> For Yealink IP phones, it works only if the value of the parameter "features.fwd.allow" is set to 1 (Enabled) and call forward mode is Phone, the always/busy/no answer forward feature will apply to all the accounts on the phone.</p> <p><b>Example:</b></p> <p>http://10.10.20.10/servlet?key=NoAnswFwdOff</p>
CALLEND/CallEnd	End a call.
ASW/CANCEL/HOLD/UNHOLD:xxx	<p>Answer/end/hold/unhold a call (xxx refers to the call-id of the active call).</p> <p><b>Example:</b></p> <p>http://10.10.20.10/servlet?key=ASW:33093</p> <p><b>Note:</b> To get the call-id of the active call, configure the action URL:</p> <p><i>http(s)://&lt;phoneIPAddress&gt;/help.xml?CallId=\$call_id</i>. For more information, refer to <a href="#">Action URL</a> on page 622.</p>

**Note**

The variable value is not applicable to all events. For example, the variable value "MUTE" is only applicable when the IP phone is during a call.

When authentication is required, you can use the following URI format:  
*http(s)://username:password@<phoneIPAddress>/servlet?key=variable value* or  
*http(s)://<phoneIPAddress>/servlet?key=variable value&@username:password*.

Yealink IP phones also support a combination of the variable values in the URI, but the order of the variable value is determined by the operation of the phone. The valid URI format is:  
*http(s)://<phoneIPAddress>/servlet?key=variable value[:variable value].* Variable values are separated by a semicolon from each other. This method is not applicable to SIP-T48G/S IP phones.

The following shows an example for deleting all entries from the call history list when the phone is idle:

`http://10.3.20.10/servlet?key=F1;F3;DOWN;DOWN;DOWN;OK;OK.`

### Configuring Trusted IP Address for Action URI

For security reasons, IP phones do not handle HTTP/HTTPS GET requests by default. You need to specify the trusted IP address for action URI. When the IP phone receives a GET request from the trusted IP address for the first time, the LCD screen prompts the message "Allow remote control?". Press the **OK** soft key on the phone to allow remote control. You can specify one or more trusted IP addresses on the IP phone, or configure the IP phone to receive and handle the URI from any IP address.

If you are using SIP NOTIFY message method, you do not need to specify the trusted IP address for action URI. But you should enable the IP phone to receive the action URI requests. When the IP phone receives a SIP NOTIFY message with the "Event: ACTION-URI" header from a SIP proxy server for the first time, the LCD screen also prompts the message "Allow remote control?".

You can use action URI feature to capture the phone's current screen. For more information, refer to [Scenario A - Capturing the Current Screen of the Phone](#) on page 648.

#### Procedure

Specify the trusted IP address for action URI using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the IP phone to receive the action URI requests. <b>Parameter:</b> features.action_uri.enable
		Configure the IP phone to pop up the Allow Remote Control prompt. <b>Parameter:</b> features.show_action_uri_option
		Specify the trusted IP address(es) for sending the action URI to the IP phone. <b>Parameter:</b> features.action_uri_limit_ip

<b>Web User Interface</b>	<p>Specify the trusted IP address(es) for sending the action URI to the IP phone.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=features-remotecontrl&amp;q=load</p>
---------------------------	--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

**Details of the Configuration Parameters:**

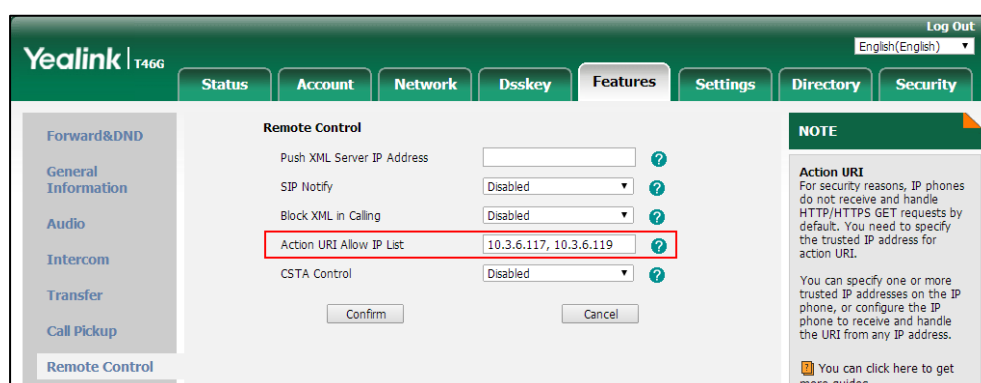
Parameters	Permitted Values	Default
<b>features.action_uri.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to receive the action URI requests.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.show_action_uri_option</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the phone to pop up the Allow Remote Control prompt.</p> <p><b>0</b>-Disabled, the phone will not pop up the Allow Remote Control prompt when receiving an HTTP or HTTPS GET request, or receiving a SIP NOTIFY message with the "Event: ACTION-URI" header. The phone will directly perform the specified action.</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "features.action_uri.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.action_uri_limit_ip</b>	<b>IP address or any</b>	<b>Blank</b>
<p><b>Description:</b></p>		



Parameters	Permitted Values	Default
<p>Configures the IP address of the server from which the IP phone receives the action URI requests.</p> <p>For discontinuous IP addresses, multiple IP addresses are separated by commas.</p> <p>For continuous IP addresses, the format likes *.*.* and the "*" stands for the values 0~255.</p> <p>For example: 10.10.*.* stands for the IP addresses that range from 10.10.0.0 to 10.10.255.255.</p> <p>If left blank, the IP phone will reject any HTTP GET request.</p> <p>If it is set to "any", the IP phone will accept and handle HTTP GET requests from any IP address.</p> <p><b>Example:</b></p> <p>features.action_uri_limit_ip = 10.3.6.117, 10.3.6.119</p> <p><b>Note:</b> It works only if the value of the parameter "features.action_uri.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Remote Control-&gt;Action URI Allow IP List</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure the trusted IP address(es) for action URI via web user interface:**

1. Click on **Features->Remote Control**.
2. Enter the IP address or any in the **Action URI Allow IP List** field.  
Multiple IP addresses are separated by commas. If you enter "any" in this field, the IP phone can receive and handle GET requests from any IP address. If you leave the field blank, the IP phone cannot receive or handle any HTTP GET request.



3. Click **Confirm** to accept the change.

## Scenario A - Capturing the Current Screen of the Phone

You can capture the screen display of the IP phone using the action URI. IP phones support handling an HTTP or HTTPS GET request. The URI format is `http(s)://<phoneIPAddress>/screencapture`. The captured picture can be saved as a BMP or JPEG file.

You can also use the URI `"http(s)://<phoneIPAddress>/screencapture/download"` to capture the screen display first, and then download the image (which is saved as a JPG file and named with the phone model and the capture time) to the local system. Before capturing the phone's current screen, ensure that the IP address of the computer is included in the trusted IP address for Action URI on the phone. For more information on the trusted IP address, refer to [Configuring Trusted IP Address for Action URI](#) on page 645.

When you capture the screen display, the IP phone may prompt you to enter the user name and password of the administrator if web browser does not remember the user name and password for web user interface login.

### Note

IP phones also support capturing the screen display using the old URI `"http://<phoneIPAddress>/servlet?command=screenshot"`.

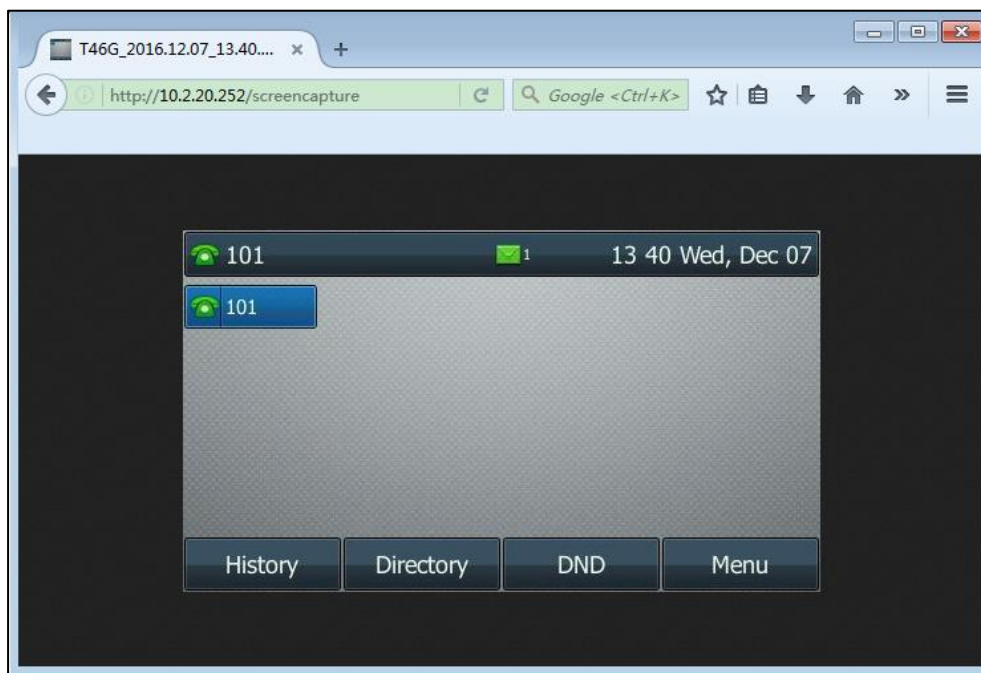
### To capture the current screen of the phone:

1. Enter request URI (for example, `http://10.2.20.252/screencapture`) in the browser's address bar and press the Enter key on the keyboard.
2. Do one of the following:
  - If it is the first time you capture the phone's current screen using the computer, the browser will display "Remote control forbidden", and the LCD screen will prompt the message "Allow remote control?".

Press the **OK** soft key on the phone to allow remote control. The phone will return to the previous screen.

Refresh the web page.

The browser will display an image of the phone's current screen. You can save the image to your local system.



- Else, the browser will display an image of the phone's current screen directly. You can save the image to your local system.

#### Note

Frequent capture may affect the phone performance. Yealink recommend you to capture the phone screen display within a minimum interval of 4 seconds.

## Scenario B - Placing a Call via Web User Interface

You can place a call via web user interface. Before doing it, ensure that the IP address of your computer is included in the trusted IP address for Action URI on the phone. For more information on the trusted IP address, refer to [Configuring Trusted IP Address for Action URI](#) on page 645.

If you place a call via web user interface but the trusted IP address has not been configured, the web user interface prompts "Call fail".

#### To place a call via web user interface:

1. Click on **Directory**->**Phone Call Info**.
2. Select the desired account from the pull-down list of **Outgoing Identity**.

3. Enter the callee's number in the **Dial Number** field.

The screenshot shows the Yealink web user interface. At the top, there are tabs for Status, Account, Network, Dsskey, Features, Settings, Directory, and Security. On the left, there is a sidebar with links for Local Directory, Remote Phone Book, Phone Call Info, LDAP, Multicast IP, and Setting. The main area is divided into two sections: Call Panel and Call List. The Call Panel has a 'Dial Number' field with '102' entered and a 'Dial' button. Below it is an 'Outgoing Identity' dropdown menu showing '101@10.2.1.48'. The Call List section has a 'Dial List' table with columns for Index, Date, Time, Local Identity, Name, and Number. The table contains six rows of call data. On the right, there is a 'NOTE' section with a 'Call Log' description and a link to get more guides.

Index	Date	Time	Local Identity	Name	Number
1	Wed Dec 07	10:37	101@10.2.1.48	102	102@10.2.1.48
2	Wed Dec 07	10:37	101@10.2.1.48	102	102@10.2.1.48
3	Wed Dec 07	10:36	101@10.2.1.48	102	102@10.2.1.48
4	Wed Dec 07	10:35	101@10.2.1.48	11	11@10.2.1.48
5	Wed Dec 07	10:35	101@10.2.1.48	112	112@10.2.1.48
6	Wed Dec 07	10:35	101@10.2.1.48	11	11@10.2.1.48

4. Click **Dial** to dial out the number.

The web user interface prompts "Call Success" and the phone will automatically dial out the number. You can click **Hang Up** to end the call.

If it is the first time you place a call via web user interface, the LCD screen will prompt the message "Allow remote control?". Press **OK** on the phone to allow remote control and then the phone will automatically dial out the number.

**Note** You can also place an IP direct call via web user interface. The IP phone supports either IPv4 or IPv6 address.

## Server Redundancy

Server redundancy is often required in VoIP deployments to ensure continuity of phone service, for events where the server needs to be taken offline for maintenance, the server fails, or the connection between the IP phone and the server fails.

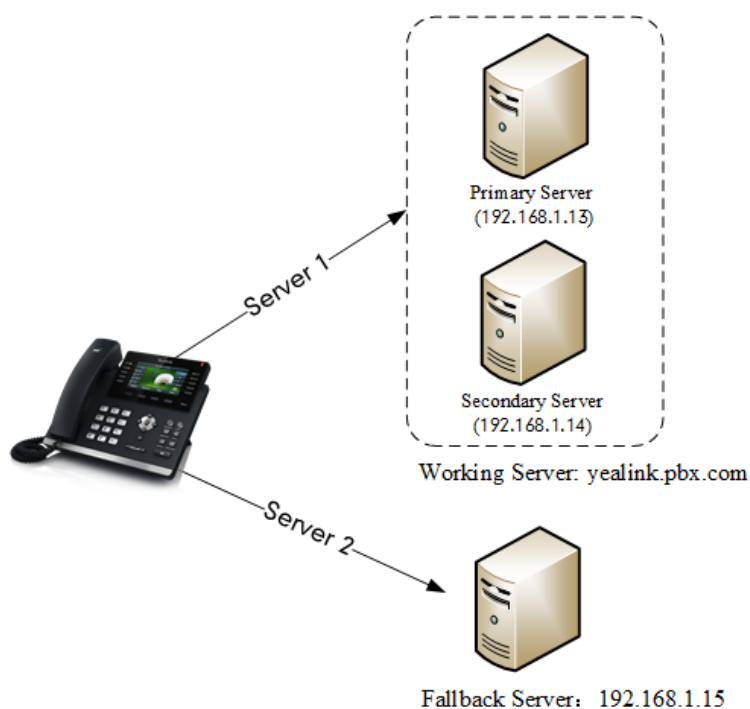
Two types of redundancy are possible. In some cases, a combination of the two may be deployed:

- **Failover:** In this mode, the full phone system functionality is preserved by having a second equivalent capability call server take over from the one that has gone down/off-line. This mode of operation should be done using the DNS mechanism from the primary to the secondary server. Therefore, if you want to use this mode, the server must be configured with a domain name.
- **Fallback:** In this mode, a second less featured call server with SIP capability takes over call control to provide basic calling capability, but without some advanced features (for example, shared line and MWI) offered by the working server. IP phones support configuration of two servers per SIP registration for fallback purpose.

**Note** For concurrent registration mode, it has certain limitation when using some advanced features, and for successive registration mode, the phone service may have a brief interrupt while the server fails. So we recommend you to use the failover mode for server redundancy because this mode can ensure the continuity of the phone service and you can use all the call features while the server fails.

## Phone Configuration for Redundancy Implementation

To assist in explaining the redundancy behavior, an illustrative example of how an IP phone may be configured is shown as below. In the example, server redundancy for fallback and failover purposes is deployed. Two separate servers (a working server and a fallback server) are configured for per line registration.



**Working Server:** Server 1 is configured with the domain name of the working server. For example: yealink.pbx.com. DNS mechanism is used such that the working server is resolved to multiple servers with different IP addresses for failover purpose. The working server is deployed in redundant pairs, designated as primary and secondary servers. The primary server (for example, 192.168.1.13) has the highest priority server in a cluster of servers resolved by the DNS server. The secondary server (for example, 192.168.1.14) backs up a primary server when the primary server fails and offers the same functionality as the primary server.

**Fallback Server:** Server 2 is configured with the IP address of the fallback server. For example, 192.168.1.15. A fallback server offers less functionality than the working server.

## Outgoing Call When the Working Server Connection Fails

When a user initiates a call, the IP phone will go through the following steps to connect the call:

1. Sends the INVITE request to the primary server.
2. If the primary server does not respond correctly to the INVITE (that is, the primary server responds to the INVITE with 503 message or the request for responding with 100 Trying message times out (64\*T1 seconds, defined in [RFC 3261](#))), then tries to make the call using the secondary server.

3. If the secondary server is also unavailable, the IP phone will try the fallback server until it either succeeds in making a call or exhausts all servers at which point the call will fail.

At the start of a call, server availability is determined by SIP signaling failure. SIP signaling failure depends on the SIP protocol being used as described below:

- If TCP is used, then the signaling fails if the connection or the send fails.
- If UDP is used, then the signaling fails if ICMP is detected or if the signal times out. If the signaling has been attempted through all servers in the list (this list contains all the server addresses resolved by the DNS server) and this is the last server, then the signaling fails after the complete UDP timeout defined in [RFC 3261](#). If it is not the last server in the list, the maximum number of retries depends on the configured retry counts (configured by the parameter "account.X.sip\_server.Y.retry\_counts").

## Phone Registration

Registration method of the failover mode:

The IP phone must always register to the primary server first except in failover conditions. If this is unsuccessful, the phone will re-register as many times as configured until the registration is successful. When the primary server registration is unavailable, the secondary server will serve as the working server. As soon as the primary server registration succeeds, it returns to being the working server.

Registration methods of the fallback mode include (not applicable to outbound proxy servers):

- **Concurrent registration (default):** The IP phone registers to SIP server 1 and SIP server 2 (working server and fallback server) at the same time. Note that although the IP phone registers to two SIP servers, only one server works at the same time. In a failure situation, a fallback server can take over the basic calling capability, but without some advanced features (for example, shared lines and MWI) offered by the working server.
- **Successive registration:** The IP phone only registers to one server at a time. The IP phone first registers to the working server. In a failure situation, the IP phone registers to the fallback server, and the fallback server can take over all calling capabilities.

For more information on server redundancy, refer to [Server Redundancy on Yealink IP Phones](#).

## Procedure

Server redundancy can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure the SIP server redundancy. <b>Parameters:</b> account.X.sip_server.Y.address account.X.sip_server.Y.port account.X.sip_server.Y.expires account.X.sip_server.Y.retry_counts
--------------------------------------------------	-----------	------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

		<p>Configure the outbound proxy server redundancy.</p> <p><b>Parameters:</b></p> <p>account.X.outbound_proxy.enable</p> <p>account.X.outbound_proxy.Y.address</p> <p>account.X.outbound_proxy.Y.port</p> <p>account.X.outbound_proxy.Y.retry_counts</p>
		<p><b>Fallback Mode</b></p> <p><b>Parameters:</b></p> <p>account.X.fallback.redundancy_type</p> <p>account.X.fallback.timeout</p> <p>account.X.outbound_proxy_fallback_interval</p>
		<p><b>Failover Mode for SIP server</b></p> <p><b>Parameters:</b></p> <p>account.X.sip_server.Y.register_on_enable</p> <p>account.X.sip_server.Y.only_signal_with_registered</p> <p>account.X.sip_server.Y.invite_retry_counts</p> <p>account.X.sip_server.Y.failback_mode</p> <p>account.X.sip_server.Y.failback_timeout</p> <p>account.X.sip_server.Y.failback_subscribe.enable</p>
		<p><b>Failover Mode for outbound proxy server</b></p> <p><b>Parameters:</b></p> <p>account.X.outbound_proxy.Y.register_on_enable</p> <p>account.X.outbound_proxy.Y.only_signal_with_registered</p> <p>account.X.outbound_proxy.Y.invite_retry_counts</p> <p>account.X.outbound_proxy.Y.failback_mode</p> <p>account.X.outbound_proxy.Y.failback_timeout</p> <p>account.X.outbound_proxy.Y.failback_subscribe.enable</p>
<b>Web User Interface</b>		<p>Configure the server redundancy on the IP phone.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=account-register&amp;q=load&amp;acc=0</p>

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>account.X.sip_server.Y.address<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>String within 256 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the IP address or domain name of the SIP server Y that accepts registrations for a specific account.  <b>Example:</b> account.1.sip_server.1.address = yealink.pbx.com  <b>Web User Interface:</b> Account->Register->SIP Server Y->Server Host  <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Accounts->SIP ServerY		
<b>account.X.sip_server.Y.port<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>Integer from 0 to 65535</b>	<b>5060</b>
<b>Description:</b> Configures the port of the SIP server Y that specifies registrations for a specific account.  <b>Example:</b> account.1.sip_server.1.port = 5060  <b>Note:</b> If the value of this parameter is set to 0, the port used depends on the value specified by the parameter "account.X.sip_server.Y.transport_type".  <b>Web User Interface:</b> Account->Register->SIP Server Y->Port  <b>Phone User Interface:</b> None		
<b>account.X.sip_server.Y.expires<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>Integer from 30 to 2147483647</b>	<b>3600</b>
<b>Description:</b> Configures the registration expiration time (in seconds) of the SIP server Y for a specific account.  <b>Example:</b> account.1.sip_server.1.expires = 3600  <b>Web User Interface:</b> Account->Register->SIP Server Y->Server Expires		



Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>account.X.sip_server.Y.retry_counts<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>Integer from 0 to 20</b>	<b>3</b>
<b>Description:</b> Configures the retry times for the IP phone to resend requests when the SIP server Y is unavailable or there is no response from the SIP server Y for a specific account.  <b>Example:</b> account.1.sip_server.1.retry_counts= 3  The IP phone moves to the next available server after three failed attempts.  <b>Web User Interface:</b> Account->Register->SIP Server Y->Server Retry Counts  <b>Phone User Interface:</b> None		
<b>account.X.sip_server.Y.register_on_enable<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to register to the secondary server before sending requests to it for a specific account when encountering a failover.  <b>0</b> -Disabled, the IP phone won't attempt to register to the secondary server, since the phone assumes that the primary and secondary servers share registration information. So the IP phone will directly send the requests to the secondary server.  <b>1</b> -Enabled, the IP phone will register to the secondary server first, and then send the requests to it.  <b>Web User Interface:</b> None  <b>Phone User Interface:</b> None		
<b>account.X.sip_server.Y.only_signal_with_registered<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to only send requests to the registered server for a specific account when encountering a failover.  <b>0</b> -Disabled		

Parameters	Permitted Values	Default
<b>1-Enabled</b>  <b>Note:</b> It works only if the value of the parameter "account.X.sip_server.Y.register_on_enable" is set to 1 (Enabled) and the value of the parameter "account.X.sip_server.Y.failback_mode" is set to 1, 2 or 3.  <b>Web User Interface:</b> None  <b>Phone User Interface:</b> None		
<b>account.X.sip_server.Y.invite_retry_counts<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>Integer from 1 to 10</b>	<b>3</b>
<b>Description:</b> Configures the number of retries attempted before sending requests to the next available server for a specific account when encountering a failover.  <b>Web User Interface:</b> None  <b>Phone User Interface:</b> None		
<b>account.X.outbound_proxy_enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to send requests to the outbound proxy server for a specific account.  <b>0-Disabled</b> <b>1-Enabled</b>  <b>Web User Interface:</b> Account->Register->Enable Outbound Proxy Server  <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Accounts->Outbound Status		
<b>account.X.outbound_proxy.Y.address<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>IP address or domain name</b>	<b>Blank</b>
<b>Description:</b> Configures the IP address or domain name of the outbound proxy server Y for a specific account.  <b>Note:</b> It works only if the value of the parameter "account.X.outbound_proxy_enable" is set		

Parameters	Permitted Values	Default
<p>to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Register-&gt;Outbound Proxy Server Y</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Advanced (default password: admin) -&gt;Accounts-&gt;Outbound ProxyY</p>		
<p><b>account.X.outbound_proxy.Y.port<sup>[1]</sup></b></p> <p>(Y ranges from 1 to 2)</p>	<p><b>Integer from 0 to 65535</b></p>	<p><b>5060</b></p>
<p><b>Description:</b></p> <p>Configures the port of the outbound proxy server Y for a specific account.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.outbound_proxy_enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Register-&gt;Outbound Proxy Server Y-&gt;Port</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>account.X.outbound_proxy.Y.retry_counts<sup>[1]</sup></b></p> <p>(Y ranges from 1 to 2)</p>	<p><b>Integer from 0 to 20</b></p>	<p><b>-1</b></p>
<p><b>Description:</b></p> <p>Configures the retry times for the IP phone to resend requests when the outbound proxy server Y is unavailable or there is no response from the outbound proxy server Y for a specific account.</p> <p>If it is set to <b>-1</b>, the IP phone will invoke the value of the parameter "account.X.sip_server.Y.retry_counts" to take effect.</p> <p><b>Example:</b></p> <p>account.1.outbound_proxy.1.retry_counts = 3</p> <p>The IP phone moves to the next available outbound proxy server after three failed attempts.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>account.X.outbound_proxy.Y.register_on_enable<sup>[1]</sup></b></p> <p>(Y ranges from 1 to 2)</p>	<p><b>0 or 1</b></p>	<p><b>-1</b></p>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to register to the secondary outbound proxy server before</p>		

Parameters	Permitted Values	Default
<p>sending requests to it for a specific account when encountering a failover.</p> <p><b>0</b>-Disabled, the IP phone won't attempt to register to the secondary outbound proxy server, since the phone assumes that the primary and secondary outbound proxy servers share registration information. So the IP phone will directly send the requests to the secondary outbound proxy server.</p> <p><b>1</b>-Enabled, the IP phone will register to the secondary outbound proxy server first, and then send the requests to it.</p> <p>If it is set to <b>-1</b>, the IP phone will invoke the value of the parameter "account.X.sip_server.Y.register_on_enable" to take effect.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.outbound_proxy.Y.only_signal_with_registered<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0 or 1</b>	<b>-1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to only send requests to the registered outbound proxy server for a specific account when encountering a failover.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p>If it is set to <b>-1</b>, the IP phone will invoke the value of the parameter "account.X.sip_server.Y.only_signal_with_registered" to take effect.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.outbound_proxy.Y.register_on_enable" is set to 1 (Enabled) and the value of the parameter "account.X.outbound_proxy.Y.failback_mode" is set to 1, 2 or 3.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.outbound_proxy.Y.invite_retry_counts<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>Integer from 1 to 10</b>	<b>-1</b>
<p><b>Description:</b></p> <p>Configures the number of retries attempted before sending requests to the next available outbound proxy server for a specific account when encountering a failover.</p> <p>If it is set to <b>-1</b>, the IP phone will invoke the value of the parameter</p>		

Parameters	Permitted Values	Default
<p>"account.X.sip_server.Y.invite_retry_counts" to take effect.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.fallback.redundancy_type<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the registration mode for a specific account in fallback mode.</p> <p><b>0</b>-Concurrent Registration</p> <p><b>1</b>-Successive Registration</p> <p><b>Note:</b> It is not applicable to outbound proxy servers.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.fallback.timeout<sup>[1]</sup></b>	<b>Integer from 10 to 2147483647</b>	<b>120</b>
<p><b>Description:</b></p> <p>Configures the time interval (in seconds) for the IP phone to detect whether the working server is available by sending the registration request for a specific account after the fallback server takes over call control.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.fallback.redundancy_type" is set to 1 (Successive Registration). It is not applicable to outbound proxy servers.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.outbound_proxy_fallback_interval<sup>[1]</sup></b>	<b>Integer from 0 to 65535</b>	<b>3600</b>
<p><b>Description:</b></p> <p>Configures the time interval (in seconds) for the IP phone to detect whether the working outbound proxy server is available by sending the registration request after the fallback server takes over call control.</p>		

Parameters	Permitted Values	Default
<b>Example:</b> account.1.outbound_proxy_fallback_interval = 3600 <b>Note:</b> It is only applicable to outbound proxy servers. <b>Web User Interface:</b> Account->Register->Proxy Fallback Interval <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Accounts->Proxy Fallback Interval		
<b>account.X.sip_server.Y.fallback_mode<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0, 1, 2 or 3</b>	<b>0</b>
<b>Description:</b> Configures the fallback mode for the IP phone to retry the primary server in failover for a specific account. <b>0</b> -newRequests: all requests are sent to the primary server first, regardless of the last server that was used. If the primary server does not respond correctly, the IP phone will try to send requests to the secondary server. <b>1</b> -DNSTTL: the IP phone will send requests to the last registered server first. If the TTL for the DNS A records on the registered server expires, the phone will retry to send requests to the primary server. <b>2</b> -Registration: the IP phone will send requests to the last registered server first. If the registration expires, the phone will retry to send requests to the primary server. <b>3</b> -duration: the IP phone will send requests to the last registered server first. If the time defined by the parameter "account.X.sip_server.Y.fallback_timeout" expires, the phone will retry to send requests to the primary server. <b>Note:</b> DNSTTL, Registration and duration mode can only be processed when the IP phone is idle (that is, no incoming/outbound calls, no active calls or meetings, and so on.). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>account.X.sip_server.Y.fallback_timeout<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0, Integer from 60 to 65535</b>	<b>3600</b>
<b>Description:</b> Configures the timeout (in seconds) for the phone to retry to send requests to the primary server after failing over to the current working server for a specific account. If you set the parameter to 0, the IP phone will not send requests to the primary server until		

Parameters	Permitted Values	Default
<p>a failover event occurs with the current working server.</p> <p>If you set the parameter from 1 to 59, the timeout will be 60 seconds.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.sip_server.Y.failback_mode" is set to 3 (duration).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.sip_server.Y.failback_subscribe.enable<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to retry to re-subscribe after registering to the secondary server with different IP address for a specific account when encountering a failover.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, the IP phone will immediately re-subscribe to the secondary server, for ensuring the normal use of the features associated with subscription (for example, BLF, SCA).</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.sip_server.Y.failback_mode" is set to 1, 2 or 3.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.outbound_proxy.Y.failback_mode<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0, 1, 2 or 3</b>	<b>-1</b>
<p><b>Description:</b></p> <p>Configures the failback mode for the IP phone to retry the primary outbound proxy server in failover for a specific account.</p> <p>If it is set to <b>-1</b>, the IP phone will invoke the value of the parameter "account.X.sip_server.Y.failback_mode" to take effect.</p> <p><b>0</b>-newRequests: all requests are sent to the primary outbound proxy server first, regardless of the last server that was used. If the primary outbound proxy server does not respond correctly, the IP phone will try to send requests to the secondary outbound proxy server.</p> <p><b>1</b>-DNSTTL: the IP phone will send requests to the last registered outbound proxy server first. If the TTL for the DNS A records on the registered outbound proxy server expires, the phone will retry to send requests to the primary outbound proxy server.</p>		

Parameters	Permitted Values	Default
<p><b>2-Registration:</b> the IP phone will send requests to the last registered outbound proxy server first. If the registration expires, the phone will retry to send requests to the primary outbound proxy server.</p> <p><b>3-duration:</b> the IP phone will send requests to the last registered outbound proxy server first. If the time defined by the parameter "account.X.outbound_proxy.Y.failback_timeout" expires, the phone will retry to send requests to the primary outbound proxy server.</p> <p><b>Note:</b> DNSTTL, Registration and duration mode can only be processed when the IP phone is idle (that is, no incoming/outbound calls, no active calls or meetings, and so on.).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.outbound_proxy.Y.failback_timeout<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0, Integer from 60 to 65535</b>	<b>-1</b>
<p><b>Description:</b></p> <p>Configures the timeout (in seconds) for the phone to retry to send requests to the primary outbound proxy server after failing over to the current working server for a specific account.</p> <p>If it is set to <b>-1</b>, the IP phone will invoke the value of the parameter "account.X.sip_server.Y.failback_timeout" to take effect.</p> <p>If you set the parameter to 0, the IP phone will not send requests to the primary outbound proxy server until a failover event occurs with the current working server.</p> <p>If you set the parameter from 1 to 59, the timeout will be 60 seconds.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.outbound_proxy.Y.failback_mode" is set to 3 (duration).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.outbound_proxy.Y.failback_subscribe.enable<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0 or 1</b>	<b>-1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to retry to re-subscribe after registering to the secondary outbound proxy server with different IP address for a specific account when encountering a failover.</p> <p><b>0-Disabled</b></p> <p><b>1-Enabled,</b> the IP phone will immediately re-subscribe to the secondary outbound proxy</p>		



Parameters	Permitted Values	Default
<p>server, for ensuring the normal use of the features associated with subscription (for example, BLF, SCA).</p> <p>If it is set to <b>-1</b>, the IP phone will invoke the value of the parameter "account.X.sip_server.Y.failback_subscribe.enable" to take effect.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.outbound_proxy.Y.failback_mode" is set to 1, 2 or 3.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

#### To configure server redundancy for fallback purpose via web user interface:

1. Click on **Account->Register**.
2. Select the desired account from the pull-down list of **Account**.
3. Configure registration parameters of the selected account in the corresponding fields.
4. Configure parameters of SIP server 1 and SIP server 2 in the corresponding fields.

**Yealink | T46G** Log Out English(English)

**Status** **Account** **Network** **Dsskey** **Features** **Settings** **Directory** **Security**

**Register**

**Basic**  
**Codec**  
**Advanced**

**Account** Account1 ?

Register Status: Registered

Line Active: Enabled ?

Label: 4605 ?

Display Name: 4605 ?

Register Name: 4605 ?

User Name: 4605 ?

Password: \*\*\*\*\* ?

**SIP Server 1** ?

Server Host: 192.168.1.14 Port: 5060 ?

Transport: UDP ?

Server Expires: 3600 ?

Server Retry Counts: 3 ?

**SIP Server 2** ?

Server Host: 192.168.1.15 Port: 5060 ?

Transport: UDP ?

Server Expires: 3600 ?

Server Retry Counts: 3 ?

**NOTE**

**Account Registration**  
Registers account(s) for the IP phone.

**Server Redundancy**  
It is often required in VoIP deployments to ensure continuity of phone service, for events where the server needs to be taken offline for maintenance, the server fails, or the connection between the IP phone and the server fails.

**NAT Traversal**  
A general term for techniques that establish and maintain IP connections traversing NAT gateways. STUN is one of the NAT traversal techniques.

You can configure NAT traversal for this account.

You can click here to get more guides.

5. If you use outbound proxy servers, do the following:
  - 1) Select **Enabled** from the pull-down list of **Enable Outbound Proxy Server**.
  - 2) Configure parameters of outbound proxy server 1 and outbound proxy server 2 in the corresponding fields.

The screenshot shows the Yealink T46G web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'Dsskey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Account' tab is selected, showing the configuration for 'Account1'. The left sidebar has 'Register', 'Basic', 'Codec', and 'Advanced' options. The main content area is divided into two columns. The left column contains registration parameters: Register Status (Registered), Line Active (Enabled), Label (4605), Display Name (4605), Register Name (4605), User Name (4605), and Password (\*\*\*\*\*). The right column contains SIP server and proxy settings. A red box highlights the 'SIP Server 1' and 'SIP Server 2' sections, which include fields for Server Host, Transport (UDP), Server Expires (3600), and Server Retry Counts (3). Below these are the 'Outbound Proxy Servers' settings, where 'Enable Outbound Proxy Server' is set to 'Enabled', and both 'Outbound Proxy Server 1' (10.1.8.11) and 'Outbound Proxy Server 2' (10.1.8.12) are set to Port 5060. On the far right, a 'NOTE' section provides additional information about account registration, server redundancy, and NAT traversal.

6. Click **Confirm** to accept the change.

**To configure server redundancy for failover purpose via web user interface:**

1. Click on **Account**->**Register**.
2. Select the desired account from the pull-down list of **Account**.
3. Configure registration parameters of the selected account in the corresponding fields.
4. Configure parameters of the SIP server 1 or SIP server 2 in the corresponding fields.

You must set the port of SIP server to 0 for NAPTR, SRV and A queries.

5. Select **DNS-NAPTR** from the pull-down list of **Transport**.

The screenshot shows the Yealink T466 web interface. The 'Account' tab is selected. Under 'SIP Server 1', the 'Transport' dropdown is set to 'DNS NAPTR'. The 'Server Host' is 'yealink.pbx.com' and 'Port' is '0'. 'Server Expires' is '3600' and 'Server Retry Counts' is '3'. A red box highlights the 'SIP Server 1' section.

6. If you use outbound proxy servers, do the following:
- 1) Select **Enabled** from the pull-down list of **Enable Outbound Proxy Server**.
  - 2) Configure parameters of outbound proxy server 1/2 in the corresponding fields.
- You must set the port of outbound proxy server to 0 for NAPTR, SRV and A queries.

The screenshot shows the Yealink T466 web interface. The 'Account' tab is selected. Under 'SIP Server 1', the 'Transport' dropdown is set to 'DNS NAPTR'. The 'Server Host' is 'yealink.pbx.com' and 'Port' is '0'. 'Server Expires' is '3600' and 'Server Retry Counts' is '3'. Under 'SIP Server 2', the 'Server Host' is empty and 'Port' is '5060'. 'Transport' is 'UDP', 'Server Expires' is '3600', and 'Server Retry Counts' is '3'. At the bottom, the 'Enable Outbound Proxy Server' dropdown is set to 'Enabled'. The 'Outbound Proxy Server 1' section is highlighted with a red box, showing 'Outbound Proxy Server 1' as 'yealink.sbc.com' and 'Port' as '0'.

7. Click **Confirm** to accept the change.

## Server Domain Name Resolution

If a domain name is configured for a server, the IP address(es) associated with that domain name will be resolved through DNS as specified by [RFC 3263](#). The DNS query involves NAPTR, SRV and A queries, which allows the IP phone to adapt to various deployment environments.

The IP phone performs NAPTR query for the NAPTR pointer and transport protocol (UDP, TCP and TLS), the SRV query on the record returned from the NAPTR for the target domain name and the port number, and the A query for the IP addresses.

If an explicit port (except 0) is specified, A query will be performed only. If a server port is set to 0 and the transport type is set to DNS-NAPTR, NAPTR and SRV queries will be tried before falling to A query. If no port is found through the DNS query, 5060 will be used.

The following details the procedures of DNS query for the IP phone to resolve the domain name (for example, yealink.pbx.com) of working server into the IP address, port and transport protocol.

### NAPTR (Naming Authority Pointer)

First, the IP phone sends NAPTR query to get the NAPTR pointer and transport protocol.

Example of NAPTR records:

	order	pref	flags	service	regexp	replacement
IN NAPTR	90	50	"s"	"SIP+D2T"	""	_sip._tcp.yealink.pbx.com
IN NAPTR	100	50	"s"	"SIP+D2U"	""	_sip._udp.yealink.pbx.com

Parameters are explained in the following table:

Parameter	Description
order	Specify preferential treatment for the specific record. The order is from lowest to highest, lower order is more preferred.
pref	Specify the preference for processing multiple NAPTR records with the same order value. Lower value is more preferred.
Flags	The flag "s" means to perform an SRV lookup.
service	Specify the transport protocols: SIP+D2U: SIP over UDP SIP+D2T: SIP over TCP SIP+D2S: SIP over SCTP SIPS+D2T: SIPS over TCP
regexp	Always empty for SIP services.
replacement	Specify a domain name for the next query.

The IP phone picks the first record because its order of 90 is lower than 100. The pref parameter is unimportant as there is no other record with order 90. The flag "s" indicates performing the SRV query next. TCP will be used, targeted to a host determined by an SRV query of "\_sip.\_tcp.yealink.pbx.com". If the flag of the NAPTR record returned is empty, the IP phone will perform NAPTR query again according to the previous NAPTR query result.

### SRV (Service Location Record)

The IP phone performs an SRV query on the record returned from the NAPTR for the host name and the port number. Example of SRV records:

	Priority	Weight	Port	Target
IN SRV	0	1	5060	server1.yealink.pbx.com
IN SRV	0	2	5060	server2.yealink.pbx.com

Parameters are explained in the following table:

Parameter	Description
Priority	Specify preferential treatment for the specific host entry. Lower priority is more preferred.
Weight	When priorities are equal, weight is used to differentiate the preference. The preference is from highest to lowest. Keep the same to load balance.
Port	Identify the port number to be used.
Target	Identify the actual host for an A query.

SRV query returns two records. The two SRV records point to different hosts and have the same priority 0. The weight of the second record is higher than the first one, so the second record will be picked first. The two records also contain a port "5060", the IP phone uses this port. If the Target is not a numeric IP address, the IP phone performs an A query. So in this case, the IP phone uses "server1.yealink.pbx.com" and "server2.yealink.pbx.com" for the A query.

### A (Host IP Address)

The IP phone performs an A query for the IP address of each target host name. Example of A records:

Server1.yealink.pbx.com IN A 192.168.1.13

Server2.yealink.pbx.com IN A 192.168.1.14

The IP phone picks the IP address "192.168.1.14" first.

### Procedure

SIP Server Domain Name Resolution can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure the transport method on the IP phone.  <b>Parameters:</b> account.X.sip_server.Y.transport_type account.X.naptr_build
<b>Web User Interface</b>		Configure the transport type on the IP phone.

	<b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-register&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-register&amp;q=load&amp;acc=0</a>
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**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>account.X.sip_server.Y.transport_type<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0, 1, 2 or 3</b>	<b>0</b>
<b>Description:</b> Configures the transport method the IP phone uses to communicate with the SIP server for a specific account. <b>0</b> -UDP <b>1</b> -TCP <b>2</b> -TLS <b>3</b> -DNS-NAPTR If the value of this parameter is set to 3 (DNS-NAPTR), the value of the parameter "account.X.sip_server.Y.address" is set to a host name and the value of the parameter "account.X.sip_server.Y.port" is set to 0, the IP phone will perform the DNS NAPTR and SRV queries for the transport protocol, ports and servers. If the value of this parameter is set to 3 (DNS-NAPTR), the value of the parameter "account.X.sip_server.Y.address" is set to an IP address and the value of the parameter "account.X.sip_server.Y.port" is set to an explicit port (except 0), then UDP is used. <b>Web User Interface:</b> Account->Register->SIP Server Y->Transport <b>Phone User Interface:</b> None		
<b>account.X.naptr_build<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Configures the way of SRV query for the IP phone to be performed when no result is returned from NAPTR query for a specific account. <b>0</b> -SRV query using UDP only <b>1</b> -SRV query using UDP, TCP and TLS <b>Web User Interface:</b> None <b>Phone User Interface:</b>		

Parameters	Permitted Values	Default
None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## Static DNS Cache

Failover redundancy can only be utilized when the configured domain name of the server is resolved to multiple IP addresses. If the IP phone is not configured with a DNS server, or the DNS query returns no result from a DNS server, you can statically configure a set of DNS NAPTR/SRV/A records into the IP phone. The IP phone will attempt to resolve the domain name of the SIP server with static DNS cache.

When the IP phone is configured with a DNS server, it will behave as follows to resolve domain name of the server:

- The IP phone performs a DNS query to resolve the domain name from the DNS server.
- If the DNS query returns no results for the domain name, or the returned record cannot be contacted, the values in the static DNS cache (if configured) are used when their configured time intervals are not elapsed.
- If the configured time interval is elapsed, the IP phone will attempt to perform a DNS query again.
- If the DNS query returns a result, the IP phone will use the returned record from the DNS server and ignore the statically configured cache values.

When the IP phone is not configured with a DNS server, it will behave as follows:

- The IP phone attempts to resolve the domain name within the static DNS cache.
- The IP phone will always use the results returned from the static DNS cache.

Support for negative caching of DNS queries as described in [RFC 2308](#) is also provided to allow faster failover when prior DNS queries have returned no results from the DNS server.

IP phones can be configured to use static DNS cache preferentially. Static DNS cache is configurable on a per-line basis.

## Procedure

Static DNS cache can be configured only using the configuration file.

Central Provisioning (Configuration Files)	<y0000000000xx>.cfg	<p>Configure NAPTR/SRV/A records.</p> <p><b>Parameters:</b></p> <p>dns_cache_naptr.X.name</p> <p>dns_cache_naptr.X.order</p> <p>dns_cache_naptr.X.preference</p> <p>dns_cache_naptr.X.replace</p> <p>dns_cache_naptr.X.service</p> <p>dns_cache_naptr.X.ttl</p> <p>dns_cache_srv.X.name</p> <p>dns_cache_srv.X.port</p> <p>dns_cache_srv.X.priority</p> <p>dns_cache_srv.X.target</p> <p>dns_cache_srv.X.weight</p> <p>dns_cache_srv.X.ttl</p> <p>dns_cache_a.X.name</p> <p>dns_cache_a.X.ip</p> <p>dns_cache_a.X.ttl</p>
	<MAC>.cfg	<p>Configure the IP phone whether to cache the additional DNS records.</p> <p><b>Parameter:</b></p> <p>account.X.dns_cache_type</p>
		<p>Configure the IP phone whether to use static DNS cache preferentially.</p> <p><b>Parameter:</b></p> <p>account.X.static_cache_pri</p>

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>dns_cache_naptr.X.name</b> (X ranges from 1 to 12)	<b>Domain name</b>	<b>Blank</b>
<b>Description:</b> Configures the domain name to which NAPTR record X refers.		



Parameters	Permitted Values	Default
<b>Example:</b> dns_cache_naptr.1.name = yealink.pbx.com  <b>Web User Interface:</b> None  <b>Phone User Interface:</b> None		
<b>dns_cache_naptr.X.order</b> (X ranges from 1 to 12)	<b>Integer from 0 to 65535</b>	<b>0</b>
<b>Description:</b> Configures the order of NAPTR record X.  NAPTR record with lower order is more preferred. For example, NAPTR record with the order 90 has the higher priority than that with the order 100 because 90 is lower than 100.  <b>Example:</b> dns_cache_naptr.1.order = 90  <b>Web User Interface:</b> None  <b>Phone User Interface:</b> None		
<b>dns_cache_naptr.X.preference</b> (X ranges from 1 to 12)	<b>Integer from 0 to 65535</b>	<b>0</b>
<b>Description:</b> Configures the preference of NAPTR record X.  NAPTR record with lower value is more preferred when the multiple NAPTR records have the same order value.  <b>Example:</b> dns_cache_naptr.1.preference = 50  <b>Web User Interface:</b> None  <b>Phone User Interface:</b> None		
<b>dns_cache_naptr.X.replace</b> (X ranges from 1 to 12)	<b>Domain name with SRV prefix</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Configures a domain name to be used for the next SRV query in NAPTR record X.</p> <p><b>Example:</b></p> <p>dns_cache_naptr.1.replace = _sip_tcp.yealink.pbx.com</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>dns_cache_naptr.X.service</b></p> <p>(X ranges from 1 to 12)</p>	<p><b>String within 32 characters</b></p>	<p><b>Blank</b></p>
<p><b>Description:</b></p> <p>Configures the transport protocol available for the SIP server in NAPTR record X.</p> <p><b>SIP+D2U:</b> SIP over UDP</p> <p><b>SIP+D2T:</b> SIP over TCP</p> <p><b>SIP+D2S:</b> SIP over SCTP</p> <p><b>SIPS+D2T:</b> SIPS over TCP</p> <p><b>Example:</b></p> <p>dns_cache_naptr.1.service = SIP+D2T</p> <p><b>Note:</b> For more information, refer to <a href="#">RFC 2915</a>.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>dns_cache_naptr.X.ttl</b></p> <p>(X ranges from 1 to 12)</p>	<p><b>Integer from 30 to 2147483647</b></p>	<p><b>300</b></p>
<p><b>Description:</b></p> <p>Configures the time interval (in seconds) that NAPTR record X may be cached before the record should be consulted again.</p> <p><b>Example:</b></p> <p>dns_cache_naptr.1.ttl = 3600</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
None		
<b>dns_cache_srv.X.name</b> (X ranges from 1 to 12)	<b>Domain name with SRV prefix</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the domain name in SRV record X.</p> <p><b>Example:</b></p> <p>dns_cache_srv.1.name = _sip._tcp.yealink.pbx.com</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>dns_cache_srv.X.port</b> (X ranges from 1 to 12)	<b>Integer from 0 to 65535</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the port to be used in SRV record X.</p> <p><b>Example:</b></p> <p>dns_cache_srv.1.port = 5060</p> <p><b>Note:</b> For more information, refer to <a href="#">RFC 2782</a>.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>dns_cache_srv.X.priority</b> (X ranges from 1 to 12)	<b>Integer from 0 to 65535</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the priority for the target host in SRV record X.</p> <p>Lower priority is more preferred. For example, SRV record with the priority value 0 is more preferred than that with the priority value 1 because 0 is lower than 1.</p> <p><b>Note:</b> For more information, refer to <a href="#">RFC 2782</a>.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
None		
<b>dns_cache_srv.X.target</b> (X ranges from 1 to 12)	Domain name	Blank
<b>Description:</b> Configures the domain name of the target host for an A query in SRV record X. <b>Example:</b> dns_cache_srv.1.target = server1.yealink.pbx.com <b>Note:</b> For more information, refer to <a href="#">RFC 2782</a> . <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>dns_cache_srv.X.weight</b> (X ranges from 1 to 12)	Integer from 0 to 65535	0
<b>Description:</b> Configures the weight of the target host in SRV record X. When priorities are equal, weight is used to differentiate the preference. Higher weight value is more preferred. <b>Example:</b> dns_cache_srv.1.weight = 1 <b>Note:</b> For more information, refer to <a href="#">RFC 2782</a> . <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>dns_cache_srv.X.ttl</b> (X ranges from 1 to 12)	Integer from 30 to 2147483647	300
<b>Description:</b> Configures the time interval (in seconds) that SRV record X may be cached before the record should be consulted again. <b>Example:</b> dns_cache_srv.1.ttl = 3600 <b>Web User Interface:</b>		

Parameters	Permitted Values	Default
None <b>Phone User Interface:</b> None		
<b>dns_cache_a.X.name</b> (X ranges from 1 to 12)	<b>Domain name</b>	<b>Blank</b>
<b>Description:</b> Configures the domain name in A record X. <b>Example:</b> dns_cache_a.1.name = yealink.pbx.com <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>dns_cache_a.X.ip</b> (X ranges from 1 to 12)	<b>IP address</b>	<b>Blank</b>
<b>Description:</b> Configures the IP address that the domain name in A record X maps to. <b>Example:</b> dns_cache_a.1.ip = 192.168.1.13 <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>dns_cache_a.X.ttl</b> (X ranges from 1 to 12)	<b>Integer from 30 to 2147483647</b>	<b>300</b>
<b>Description:</b> Configures the time interval (in seconds) that A record X may be cached before the record should be consulted again. <b>Example:</b> dns_cache_a.1.ttl = 3600 <b>Web User Interface:</b> None		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>account.X.dns_cache_type<sup>[1]</sup></b>	<b>0, 1 or 2</b>	<b>1</b>
<b>Description:</b> Configures whether the IP phone uses the DNS cache for domain name resolution of the server and caches the additional DNS records for a specific account. <b>0</b> -Perform real-time DNS query rather than using DNS cache. <b>1</b> -Use DNS cache, but do not cache the additional DNS records. <b>2</b> -Use DNS cache and cache the additional DNS records. <b>Example:</b> account.1.dns_cache_type = 1 <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>account.X.static_cache_pri<sup>[1]</sup></b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Configures whether preferentially to use the static DNS cache for domain name resolution of the SIP server for a specific account. <b>0</b> -Use domain name resolution from the DNS server preferentially <b>1</b> -Use static DNS cache preferentially <b>Example:</b> account.1.static_cache_pri = 1 <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

## Real-Time Transport Protocol (RTP) Ports

The Real-time Transport Protocol (RTP) is a network protocol for delivering audio over IP networks. The phone is compatible with [RFC 1889 - RTP: A Transport Protocol for Real-Time Applications](#) - and the updated [RFC 3550](#). It treats all RTP streams as bi-directional from a control perspective and expects that both RTP end points will negotiate the respective destination IP addresses and ports.

You can specify the IP phone's RTP port range. Since the IP phone supports conferencing and multiple RTP streams, it can use several ports concurrently. The UDP port used for RTP streams is traditionally an even-numbered port. For example, the default RTP min port on the IP phones is 11780. The first voice session sends RTP on port 11780. Additional calls would then use ports 11782, 11784, 11786, and so on. up to the max port.

### Procedure

RTP ports can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure RTP ports. <b>Parameters:</b> static.network.port.max_rtpport static.network.port.min_rtpport
<b>Web User Interface</b>		Configure RTP ports. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?parameter=network-adv&q=load

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.network.port.min_rtpport</b>	<b>Integer from 1024 to 65535</b>	<b>11780</b>
<b>Description:</b> Configures the minimum local RTP port.  <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.  <b>Web User Interface:</b> Network->Advanced->Local RTP Port->Min RTP Port (1024~65535)  <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>static.network.port.max_rtpport</b>	<b>Integer from 1024 to 65535</b>	<b>12780</b>
<p><b>Description:</b> Configures the maximum local RTP port.</p> <p><b>Note:</b> The value of the maximum local RTP port cannot be less than that of the minimum local RTP port (configured by the parameter "static.network.port.min_rtpport"). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b> Network-&gt;Advanced-&gt;Local RTP Port-&gt;Max RTP Port (1024~65535)</p> <p><b>Phone User Interface:</b> None</p>		

**To configure the minimum and maximum RTP port via web user interface:**

1. Click on **Network->Advanced**.
2. In the **Local RTP Port** block, enter the max and min RTP port in the **Max RTP Port (1024~65535)** and **Min RTP Port (1024~65535)** field respectively.

The screenshot displays the Yealink T46G web management interface. The 'Network' tab is selected, and the 'Advanced' sub-tab is active. On the left sidebar, 'Advanced' is highlighted. The main content area shows various network settings. The 'Local RTP Port' section is enclosed in a red rectangular box. Within this section, the 'Max RTP Port (1024~65535)' is configured to 12780 and the 'Min RTP Port (1024~65535)' is configured to 11780. Other settings visible include LLDP (Active, Enabled), CDP (Active, Disabled), Voice QoS (Voice QoS 0~63 set to 46, SIP QoS 0~63 set to 26), and VPN (Active, Disabled). A 'NOTE' panel on the right provides additional information about VLAN, NAT Traversal, Quality of Service (QoS), Web Server Type, and 802.1X Authentication.

3. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
4. Click **OK** to reboot the phone.



## TR-069 Device Management

TR-069 is a technical specification defined by the Broadband Forum, which defines a mechanism that encompasses secure auto-configuration of a CPE (Customer-Premises Equipment), and incorporates other CPE management functions into a common framework. TR-069 uses common transport mechanisms (HTTP and HTTPS) for communication between CPE and ACS (Auto Configuration Servers). The HTTP(S) messages contain XML-RPC methods defined in the standard for configuration and management of the CPE.

TR-069 is intended to support a variety of functionalities to manage a collection of CPEs, including the following primary capabilities:

- Auto-configuration and dynamic service provisioning
- Software or firmware image management
- Status and performance monitoring
- Diagnostics

The following table provides a description of RPC methods supported by IP phones.

RPC Method	Description
GetRPCMethods	This method is used to discover the set of methods supported by the CPE.
SetParameterValues	This method is used to modify the value of one or more CPE parameters.
GetParameterValues	This method is used to obtain the value of one or more CPE parameters.
GetParameterNames	This method is used to discover the parameters accessible on a particular CPE.
GetParameterAttributes	This method is used to read the attributes associated with one or more CPE parameters.
SetParameterAttributes	This method is used to modify attributes associated with one or more CPE parameters.
Reboot	This method causes the CPE to reboot.
Download	This method is used to cause the CPE to download a specified file from the designated location. File types supported by IP phones are: <ul style="list-style-type: none"> <li>• Firmware Image</li> <li>• Configuration File</li> </ul>
Upload	This method is used to cause the CPE to upload a specified file to the designated location.

RPC Method	Description
	File types supported by IP phones are: <ul style="list-style-type: none"> <li>• Configuration File</li> <li>• Log File</li> </ul>
ScheduleInform	This method is used to request the CPE to schedule a one-time Inform method call (separate from its periodic Inform method calls) sometime in the future.
FactoryReset	This method resets the CPE to its factory default state.
TransferComplete	This method informs the ACS of the completion (either successful or unsuccessful) of a file transfer initiated by an earlier Download or Upload method call.
AddObject	This method is used to add a new instance of an object defined on the CPE.
DeleteObject	This method is used to remove a particular instance of an object.

For more information on TR-069, refer to [Yealink TR-069 Technote](#).

## Procedure

TR-069 can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	<p>Configure TR-069 feature.</p> <p><b>Parameters:</b></p> <p>static.managementserver.enable</p> <p>static.managementserver.username</p> <p>static.managementserver.password</p> <p>static.managementserver.url</p> <p>static.managementserver.connection_request_username</p> <p>static.managementserver.connection_request_password</p> <p>static.managementserver.periodic_inform_enable</p> <p>static.managementserver.periodic_inform_interval</p>
<b>Web User Interface</b>		<p>Configure TR-069 feature.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=settings-tr069&amp;q=load</p>

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>static.managementserver.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the TR-069 feature. <b>0</b> -Disabled <b>1</b> -Enabled  <b>Web User Interface:</b> Settings->TR069->Enable TR069  <b>Phone User Interface:</b> None		
<b>static.managementserver.username</b>	<b>String within 128 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the user name for the IP phone to authenticate with the ACS (Auto Configuration Servers). Leave it blank if no authentication is required. <b>Example:</b> static.managementserver.username = tr69  <b>Web User Interface:</b> Settings->TR069->ACS Username  <b>Phone User Interface:</b> None		
<b>static.managementserver.password</b>	<b>String within 64 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the password for the IP phone to authenticate with the ACS (Auto Configuration Servers). Leave it blank if no authentication is required. <b>Example:</b> static.managementserver.password = tr69		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Settings->TR069->ACS Password <b>Phone User Interface:</b> None		
<b>static.managementserver.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the access URL of the ACS (Auto Configuration Servers). <b>Example:</b> static.managementserver.url = http://officetelprov.orangero.net:8080/ftacs-digest/ACS <b>Web User Interface:</b> Settings->TR069->ACS URL <b>Phone User Interface:</b> None		
<b>static.managementserver.connection_request_username</b>	<b>String within 128 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the user name for the IP phone to authenticate the incoming connection requests of the ACS (Auto Configuration Servers). <b>Example:</b> static.managementserver.connection_request_username = accuser <b>Web User Interface:</b> Settings->TR069->Connection Request Username <b>Phone User Interface:</b> None		
<b>static.managementserver.connection_request_password</b>	<b>String within 64 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the password for the IP phone to authenticate the incoming connection requests of the ACS (Auto Configuration Servers).		

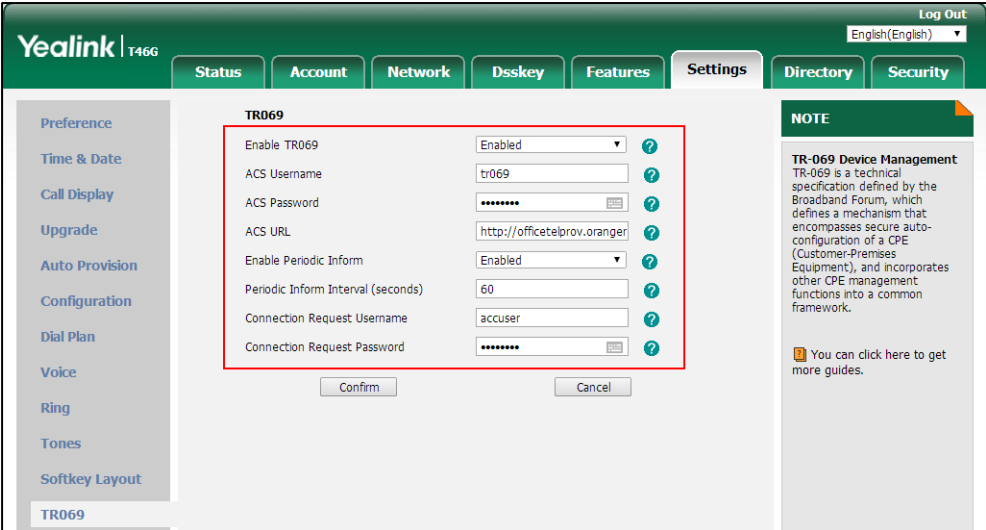
Parameters	Permitted Values	Default
<b>Example:</b> static.managementserver.connection_request_password = acspwd  <b>Web User Interface:</b> Settings->TR069->Connection Request Password  <b>Phone User Interface:</b> None		
<b>static.managementserver.periodic_inform_enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to periodically report its configuration information to the ACS (Auto Configuration Servers).  <b>0</b> -Disabled <b>1</b> -Enabled  <b>Web User Interface:</b> Settings->TR069->Enable Periodic Inform  <b>Phone User Interface:</b> None		
<b>static.managementserver.periodic_inform_interval</b>	<b>Integer from 5 to 4294967295</b>	<b>60</b>
<b>Description:</b> Configures the interval (in seconds) for the IP phone to report its configuration to the ACS (Auto Configuration Servers).  <b>Note:</b> It works only if the value of the parameter "static.managementserver.periodic_inform_enable" is set to 1 (Enabled).  <b>Web User Interface:</b> Settings->TR069->Periodic Inform Interval (seconds)  <b>Phone User Interface:</b> None		

**To configure TR-069 via web user interface:**

1. Click on **Settings->TR069**.
2. Select **Enabled** from the pull-down list of **Enable TR069**.
3. Enter the user name and password authenticated by the ACS in the **ACS Username** and

**ACS Password** fields.

- 4. Enter the URL of the ACS in the **ACS URL** field.
- 5. Select the desired value from the pull-down list of **Enable Periodic Inform**.
- 6. Enter the desired time in the **Periodic Inform Interval (seconds)** field.
- 7. Enter the user name and password authenticated by the IP phone in the **Connection Request Username** and **Connection Request Password** fields.



- 8. Click **Confirm** to accept the change.

## XML Browser

XML browser simply means that the SIP phones' LCD screen display can be managed by external applications. The XML browser feature allows users to develop and deploy custom services which meet user functional requirements on the server. Users can customize practical applications, such as weather report, stock information, Google search, news service, and so on.

To use the XML browser feature, you must configure an XML key in advance.

For more information on XML browser, refer to [Yealink IP Phones XML Browser Developer's Guide](#).

### Procedure

XML browser DSS key can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Assign an XML browser DSS key. <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type linekey.X.value/ programablekey.X.value/ expansion_module.X.key.Y.value
--------------------------------------------------	---------------------	------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

		linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Assign an XML browser DSS key.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=dss key&q=load&model=0
<b>Phone User Interface</b>		Assign an XML browser DSS key.

## XML Browser DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type<sup>[1]</sup></b> <b>programablekey.X.type<sup>[2]</sup></b> <b>expansion_module.X.key.Y.type<sup>[3]</sup></b>	27
<b>Description:</b> Configures a DSS key as an XML Browser DSS key on the IP phone. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Type	
<b>linekey.X.value<sup>[1]</sup></b> <b>programablekey.X.value<sup>[2]</sup></b> <b>expansion_module.X.key.Y.value<sup>[3]</sup></b>	String within 99 characters
<b>Description:</b> Configures the available access URL to browser the XML object. <b>Example:</b> linekey.2.value = http://10.2.1.158/TextMenu.xml <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b>	

Parameters	Permitted Values
Dsskey->Line Key/Programable Key->Value <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Value	
<b>linekey.X.label<sup>[1]</sup></b> <b>programmablekey.X.label<sup>[2]</sup></b> <b>expansion_module.X.key.Y.label<sup>[3]</sup></b>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Short <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programmable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

**To configure an XML Browser DSS key via web user interface:**

1. Click on **Dsskey->Line Key** (or **Programmable Key**).
2. In the desired DSS key field, select **XML Browser** from the pull-down list of **Type**.



3. Enter the available access URL in the **Value** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

5. Click **Confirm** to accept the change.

#### To configure an XML Browser DSS key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS Key.
3. Press or , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press or , or the **Switch** soft key to select **XML Browser** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the available access URL in the **Value** field.
7. Press the **Save** soft key to accept the change.

## Enhanced DSS Keys

Enhanced DSS Keys (EDK) enables users to customize the functions of a phone's DSS keys and soft keys. You can use EDK to assign frequently used function to DSS keys and soft keys or to create menu shortcuts to frequently used phone settings to suit your needs.

Application scenarios involve the following:

- Adding new DSS keys or soft keys to simplify the operation of common telephony tasks that may need more than one key press with the default configuration.
- Removing certain default DSS keys or soft keys for functions that may be redundant or never used.

The rules for configuring EDK for DSS keys and soft keys are different. Before using EDK, you are advised to become familiar with the macro language. For more information, refer to [Understanding Macro Action Strings](#) on page 688.

For more information on Enhanced DSS Keys, refer to [Using Enhanced DSS Keys on Yealink IP Phones](#).

## Some Guidelines for Configuring Enhanced DSS Keys

The following guidelines will help you to configure EDK efficiently:

- Activation of EDK functions requires valid macro construction.

- "\$" character delimits the parts of the macro string (except the digits only) and it must exist in pairs. For more information, refer to [Understanding Macro Action Strings](#) on page 688.
- Macros can be invoked in the format "!<macro name>", where <macro name> is defined by the parameter "edk.edklist.X.mname".
- If there are two or more same macros, the soft key or DSS key will invoke the macro with a smallest value of X defined by the parameter "edk.edklist.X.mname".

## Understanding Macro Action Strings

Before using EDK, you must be familiar with macro language shown in this section.

The following table shows the basic macro action string syntax when creating EDK macros:

Macro Action	Description
<b>Digits</b>	<p>The digits to be sent. You can use only *, #, +, 0-9. The appearance of this parameter depends on the action string.</p> <p>Example: <b>*981135</b>.</p>
<b>\$C&lt;command&gt;\$</b>	<p>This is the command. It can appear anywhere in the action string. Supported commands (or shortcuts) include:</p> <ul style="list-style-type: none"> <li>• hang up (hu)</li> <li>• hold (h)</li> <li>• waitconnect (wc)</li> <li>• pause &lt;number of seconds&gt; (p &lt;num sec&gt;) where the maximum value is 10</li> </ul> <p>Example: <b>4411\$Cwc\$\$Cp10\$</b> defines dialing 4411&gt;&gt;waitconnect&gt;&gt;connected and lasts 10 seconds.</p>
<b>\$T&lt;type&gt;\$</b>	<p>The embedded action type. Multiple actions can be defined. Supported action types include:</p> <ul style="list-style-type: none"> <li>• invite</li> <li>• dtmf</li> <li>• refer</li> </ul> <p>Example: <b>*338\$Tdtmf\$</b> defines sending the *338 by the type of dtmf.</p> <p><b>Note:</b> We recommend that you always define this field. If it is not defined, the supplied digits are dialed using INVITE.</p>
<b>\$M&lt;macro&gt;\$</b>	<p>The embedded macro. The &lt;macro&gt; string must begin with a letter. If the macro name is not defined, the execution of the action string will be ignored.</p>

Macro Action	Description
	Example: <b>\$MAA\$</b> means invoking the EDK macro AA.
<b>\$S&lt;softkey ID&gt;\$</b>	<p>The functionality of performing this action is the same as that of pressing the desired soft key.</p> <p>Each soft key has a unique identifier on the IP phone, you can configure this parameter according to the system-defined softkey ID. If the softkey ID is not defined on the phone or there is no matched soft key on the current screen, the execution of the action string will be ignored. The softkey ID is case-insensitive.</p> <p>Example: <b>\$Sanswer\$</b> means pressing the <b>Answer</b> soft key.</p> <p>Yealink IP phones support customizing soft keys. When invoking a custom soft key, the prefix “#” must be added.</p> <p>Example: If the custom softkey label is IVR1, the custom softkey id is custom_macro, then</p> <p><b>\$S#custom_macro\$</b> means pressing the <b>IVR1</b> soft key.</p> <p><b>Note:</b> To view the softkey ID, you can configure the values of parameters “features.enhanced_dss_keys.enable” and “edk.id_mode.enable” to 1 (Enabled) and then long press the Volume Up key when the phone is idle.</p>
<b>\$K&lt;key name&gt;\$</b>	<p>The functionality of performing this action is the same as that of pressing the desired hard key.</p> <p>Supported key names include:</p> <ul style="list-style-type: none"> <li>LineKeyX (for SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2).</li> <li>SoftKeyX (X ranges from 1 to 4)</li> <li>ArrowUp</li> <li>ArrowDown</li> <li>ArrowLeft</li> <li>ArrowRight</li> <li>VolDown</li> <li>VolUp</li> <li>Cancel</li> <li>OK</li> <li>DialPadX (X ranges from 0 to 9)</li> <li>DialPadPound</li> </ul>

Macro Action	Description
	<ul style="list-style-type: none"> <li>• DialPadStar</li> <li>• Headset</li> <li>• Mute</li> <li>• Message</li> <li>• Hold</li> <li>• Redial</li> <li>• Transfer</li> <li>• Speaker</li> <li>• Conference</li> <li>• ExtX@Y (X stands for the serial number of expansion module key, Y stands for the serial number of expansion module; For SIP-T48S/T48G/T46S/T46G/T29G: X ranges from 1 to 40, Y ranges from 1 to 6; For SIP-T54S/T52S: X ranges from 1 to 60, Y ranges from 1 to 3; @Y can be omitted if there is only one expansion module connected to the phone.)</li> <li>• Menu (You can enter menu by executing this command at any interface except the non-executable situations. For example, entering menu is blocked during an active call.)</li> <li>• Home (You can return back to idle screen by executing this command at dialing screen for SIP-T48G/S IP phones)</li> </ul> <p>Example: <b>\$KDialPadPound\$</b> means pressing the pound key.</p> <p><b>Note:</b> If a key (for example, Redial key) is not found but the function is available, perform the corresponding action. If a key is not found and the function is unavailable, the execution of the action string will be ignored.</p>
<b>\$I&lt;menu item ID&gt;\$</b>	<p>The action to be performed for positioning and entering the desired menu item.</p> <p>Each menu item has a unique identifier on the IP phone, you can configure this parameter according to the system-defined menu item ID. If the menu item ID is not defined on the phone or there is no matched menu item on the current screen, the execution of the action string will be ignored. The menu item ID is case-insensitive.</p>

Macro Action	Description
	<p>Example: <b>\$Istatus_list&amp;</b> means entering the <b>Status</b> menu.</p> <p><b>Note:</b> To view the menu item ID, you can configure the value of the parameter "edk.id_mode.enable" to 1 (Enabled) and then long press the Volume Up key when the phone is idle.</p>
<b>\$P&lt;label&gt;&amp;C&lt;characters number allowed&gt;&amp;N&amp;M\$</b>	<p>The user input prompt string. Enable to prompt the pop-up box, specify the label for the input box, specify the maximum input characters, specify the character type for the input box, and specify whether to mask the input by the "*".</p> <p>"label" means the specified label for pop-up box.</p> <p>"characters number allowed" defines the maximum number of input characters.</p> <p>If &amp;N is included, the character type is Number (default input method: 123). If &amp;N is not included, the character type is Text (default input method: abc), you can manually change input method.</p> <p>If &amp;M is included, the input are masked by the "*".</p> <p>Example: <b>\$PEnter name&amp;C3&amp;N&amp;M</b> means prompting an Enter name pop-up box, the maximum number of input characters is 3, the input type is Number, and the input is masked by the "*".</p>
<b>\$P&lt;EDK prompt X&gt;N&lt;characters number allowed&gt;\$</b>	<p>This is a macro substitution string.</p> <p>"EDK prompt X" means which EDK prompt is invoked. It indicates the X defined by "edk.edkprompt.X.enable".</p> <p>"characters number allowed" defines the maximum number of input characters. The user needs to press the <b>Enter</b> soft key to complete data entry.</p> <p>Example: <b>\$P2N5\$</b> means invoking the EDK prompt 2 and the maximum number of input characters is 5.</p> <p><b>Note:</b> For more information on EDK prompt, refer to <a href="#">Configuring EDK Prompt</a> on page 696.</p>
<b>\$L&lt;label&gt;\$</b>	<p>This is the label for the entire operation. The value can be any string including the null string (in this case, no label displays).</p> <p>This label is used if no label is configured for a custom key DSS key or soft key, otherwise this one is ignored. Make this the first entry in the action string.</p> <p>Example: <b>\$LEDK2\$1234\$Tinvite\$</b> defines calling out the number 1234 and using the label "EDK2" for a Custom Key</p>

Macro Action	Description
	DSS key or soft key.
<b>\$LED&lt;color and time&gt;&amp;L&lt;label&gt;\$</b> (not applicable to SIP-T19(P) E2 IP phones)	<p>The status of the BLF/BLF list DSS key LED. Enable to specify the LED color for the BLF/BLF list DSS key, and specify the duration time (in milliseconds) for the corresponding status of the BLF/BLF list DSS key. The valid value of the duration time ranges from 100 to 60000ms. Can be composed of multiple combination of "color" and "time". The status will be stuck in an infinite loop until triggered by other macros.</p> <p>Supported colors include (must be lowercase):</p> <ul style="list-style-type: none"> <li>• r (red)</li> <li>• ri (red; red with incoming mark for SIP-T48G/S)</li> <li>• ro (red; red with outgoing mark for SIP-T48G/S)</li> <li>• g (green)</li> <li>• gi (green; green with incoming mark for SIP-T48G/S)</li> <li>• go (green; green with outgoing mark for SIP-T48G/S)</li> <li>• o (off)</li> </ul> <p>&amp;L&lt;label&gt; (Optional.): You have to put the "&amp;L&lt;label&gt;" last in the command flow.</p> <p>Example: <b>\$LEDg1000o100r300&amp;Lidle\$</b> means an infinite loop for BLF/BLF list DSS key LED status: illuminates solid green for 1000ms, goes out for 100ms and then illuminates solid red for 300ms. At the same time, the label of the BLF/BLF list DSS key is changed to "idle".</p> <p><b>Note:</b> The last "color" can be configured without "time", and it means permanently displaying the last color until triggered by other macros. This macro can be only used for BLF/BLF list feature. For more information, refer to <a href="#">Configuring BLF LED Status and BLF DSS Key Behavior</a> on page 543.</p>

## Configuring EDK Feature

To configure EDK list, EDK prompt, EDK soft keys and custom DSS keys, you have to enable EDK feature. By default, the EDK feature is disabled.

Yealink IP phones support displaying the softkey id or menu item id by long pressing the Volume Up key. It is especially useful for those users who need to view the softkey id or menu item id when configuring EDK macros.

The following graphic shows an example for displaying the softkey id and menu item id after

accessing the menu of the phone:



Procedure

EDK feature can be configured using the configuration file.

Central Provisioning (Configuration File)	<y0000000000xx>.cfg	Configure enhanced DSS keys. <b>Parameter:</b> features.enhanced_dss_keys.enable
		Configure whether to display the softkey id or menu item id by long pressing the Volume Up key. <b>Parameter:</b> edk.id_mode.enable

Details of Configuration Parameters:

Parameters	Permitted Values	Default
features.enhanced_dss_keys.enable	0 or 1	0
<b>Description:</b> Enables or disables the Enhanced DSS Keys (EDK) feature. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
edk.id_mode.enable	0 or 1	0
<b>Description:</b>		

Parameters	Permitted Values	Default
<p>Enables or disables to view the softkey id or menu item id by long pressing the Volume Up key for three seconds at any interface.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> The menu item id displays on the status bar of the phone. Long pressing the Volume Up key again for three seconds to exit. It works only if the value of the parameter "features.enhanced_dss_keys.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## Configuring EDK List

Using the Enhanced DSS Keys (EDK) List parameters to define a macro is useful when defining more than one soft key or DSS key.

### Procedure

EDK list can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure EDK list. <b>Parameters:</b> edk.edklist.X.enable edk.edklist.X.mname edk.edklist.X.action
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### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>edk.edklist.X.enable</b> (X ranges from 1 to 255)	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables Enhanced DSS Keys (EDK) macro X.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "features.enhanced_dss_keys.enable" is set to 1 (Enabled).</p>		



Parameters	Permitted Values	Default
<b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>edk.edklist.X.mname</b> (X ranges from 1 to 255)	String	Blank
<b>Description:</b> Configures the unique identifier used by the soft key or DSS key configuration to reference the enhanced DSS keys entry for macro X. It cannot start with a digit. This parameter must have a value, it cannot be left blank. <b>Example:</b> edk.edklist.2.mname = macro2 <b>Note:</b> If there are two or more same macros, the soft key or DSS key will invoke the macro with a smallest value of X. It works only if the value of the parameter "features.enhanced_dss_keys.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>edk.edklist.X.action</b> (X ranges from 1 to 255)	String	Blank
<b>Description:</b> Configures the action string that contains a macro definition of the action that the softkey or DSS key performs. If EDK is enabled, this parameter must have a value. <b>Example:</b> edk.edklist.2.action = 1013\$Tinvite\$ <b>Note:</b> It works only if the value of the parameter "features.enhanced_dss_keys.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Configuring EDK Prompt

The Enhanced DSS Keys (EDK) Prompt parameters must be used if interactivity with user is implemented as part of any macro.

### Procedure

EDK prompt can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure EDK prompt. <b>Parameters:</b> edk.edkprompt.X.enable edk.edkprompt.X.label edk.edkprompt.X.type edk.edkprompt.X.userfeedback
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### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>edk.edkprompt.X.enable</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to display the Enhanced DSS Keys (EDK) prompt X for data input. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> If a macro attempts to use an EDK prompt that is disabled, the macro execution will be ignored. It works only if the value of the parameter "features.enhanced_dss_keys.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>edk.edkprompt.X.label</b> (X ranges from 1 to 10)	<b>String</b>	<b>Blank</b>
<b>Description:</b> Configures the title of the input box which is displayed on the Enhanced DSS Keys (EDK) prompt X. If it is left blank, no prompt displays.		

Parameters	Permitted Values	Default
<p><b>Example:</b></p> <p>edk.edkprompt.1.label = Enter Password</p> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "edk.edkprompt.X.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>edk.edkprompt.X.type</b></p> <p>(X ranges from 1 to 10)</p>	text or numeric	text
<p><b>Description:</b></p> <p>Configures the type of characters entered by the user for Enhanced DSS Keys (EDK) prompt X.</p> <p>If it is set to numeric, the default input method is 123, and you can switch to abc/ABC/2aB input method.</p> <p>If it is set to text, the default input method is abc, you can switch to ABC/2aB/123 input method.</p> <p><b>Example:</b></p> <p>edk.edkprompt.1.type = numeric</p> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "edk.edkprompt.X.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>edk.edkprompt.X.userfeedback</b></p> <p>(X ranges from 1 to 10)</p>	visible or masked	visible
<p><b>Description:</b></p> <p>Configures whether to hide the entered text for Enhanced DSS Keys (EDK) prompt X.</p> <p>If it is set to visible, the entered text is visible.</p> <p>If it is set to masked, the entered text displays as asterisk characters (*). It can be used to mask password fields.</p> <p><b>Example:</b></p> <p>edk.edkprompt.1.userfeedback = masked</p> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and</p>		

Parameters	Permitted Values	Default
"edk.edkprompt.X.enable" are set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Configuring EDK Soft Keys

System administrator can customize the soft keys to suit user's need. You can assign functions to soft keys using macros. This feature is typically used to access frequently used functions or to create menu shortcuts to frequently used phone settings.

You can configure the soft keys to display functions depending on the phone's menu level or call state. For example, you can make a **DPickup** soft key available when the phone is in an active call state.

Custom soft keys can be added in the following call states:

- **Idle** - There are no active calls on the phone.
- **Alerting (or ringing)** - There is an incoming call on the phone.
- **Connecting** - There is an outgoing call on the phone. And the call is connecting.
- **Transfer connecting** - There is a call being transferred to another phone. And the call is connecting.
- **Talk** - There is an active call on the phone.
- **Call failed** - The outgoing call encounters a failure.
- **Ring back** - There is an outgoing call on the phone. And the phone is in the ringback state.
- **Transfer ring back** - There is a call being transferred to another phone. And the phone is in the ringback state.
- **Hold** - The call is placed on hold on the phone.
- **Held** - The call is held.
- **Conference** - The phone sets up a conference call.
- **Dial tone** - There is a dial tone you can hear on the phone.
- **Dialing** - The phone is on the dialing screen.

Configuring the custom soft keys may affect the softkey layout in different states. For more information, refer to [Softkey Layout](#) on page 252.

## Procedure

Soft keys can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure a custom soft key. <b>Parameters:</b> softkey.X.enable softkey.X.label softkey.X.position softkey.X.action softkey.X.softkey_id softkey.X.use.idle softkey.X.use.incoming_call softkey.X.use.connecting softkey.X.use.transfer_connecting softkey.X.use.on_talk softkey.X.use.call_failed softkey.X.use.ring_back softkey.X.use.transfer_ring_back softkey.X.use.hold softkey.X.use.held softkey.X.use.conferenced softkey.X.use.dialtone softkey.X.use.dialing
------------------------------------------------------	---------------------	------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

## Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>softkey.X.enable</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the custom soft key X. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "features.enhanced_dss_keys.enable" is set to 1 (Enabled). <b>Web User Interface:</b> None		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>softkey.X.label</b> (X ranges from 1 to 10)	<b>String</b>	<b>Blank</b>
<b>Description:</b> Configures the text displayed on the soft key label.  <b>Example:</b> softkey.1.label = IVR1  <b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled).  <b>Web User Interface:</b> None  <b>Phone User Interface:</b> None		
<b>softkey.X.position</b> (X ranges from 1 to 10)	<b>Integer from 0 to 10</b>	<b>0</b>
<b>Description:</b> Configures the position on the LCD screen for soft key X.  If it is set to 0, the soft key X is positioned in the first empty position from the left.  If it is set to a value that is greater than the number of the soft keys, a <b>More</b> soft key appears and the soft key X is positioned in the desired position from the left. The original soft key moves to the next space, and so forth. Press the <b>More</b> soft key to view the remaining soft keys.  <b>Example:</b> softkey.1.position = 3  <b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled).  <b>Web User Interface:</b> None  <b>Phone User Interface:</b> None		
<b>softkey.X.action</b> (X ranges from 1 to 10)	<b>String</b>	<b>Blank</b>
<b>Description:</b>		

Parameters	Permitted Values	Default
<p>Configures the action or function for custom soft key X.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS key.</p> <p>You can also invoke the EDK macro that was already defined. The macro name follows the character "!".</p> <p><b>Example:</b></p> <pre>softkey.1.action = !macro1</pre> <p>In this example, macro1 stands for the macro name configured by the parameter "edk.edklist.X.mname".</p> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>softkey.X.softkey_id</b> (X ranges from 1 to 10)	String	Blank
<p><b>Description:</b></p> <p>Configures the softkey id for custom softkey X.</p> <p><b>Example:</b></p> <pre>softkey.1.softkey_id = custom_macro1</pre> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled). To distinguish the custom softkey id and system-defined softkey id, the custom softkey id displayed on the LCD screen will add a "#" prefix (for example, #custom_macro1).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>softkey.X.use.idle</b> (X ranges from 1 to 10)	0 or 1	0
<p><b>Description:</b></p> <p>Enables or disables the custom soft key X to be displayed in the idle state.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable"</p>		

Parameters	Permitted Values	Default
<p>and "softkey.X.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>softkey.X.use.incoming_call</b></p> <p>(X ranges from 1 to 10)</p>	<p><b>0 or 1</b></p>	<p><b>0</b></p>
<p><b>Description:</b></p> <p>Enables or disables the custom soft key X to be displayed in the alerting (ringing) state.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>softkey.X.use.connecting</b></p> <p>(X ranges from 1 to 10)</p>	<p><b>0 or 1</b></p>	<p><b>0</b></p>
<p><b>Description:</b></p> <p>Enables or disables the custom soft key X to be displayed in the connecting state.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<p><b>softkey.X.use.transfer_connecting</b></p> <p>(X ranges from 1 to 10)</p>	<p><b>0 or 1</b></p>	<p><b>0</b></p>
<p><b>Description:</b></p> <p>Enables or disables the custom soft key X to be displayed in the transfer connecting state.</p>		



Parameters	Permitted Values	Default
<b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>softkey.X.use.on_talk</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the custom soft key X to be displayed in the talk state. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>softkey.X.use.call_failed</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the custom soft key X to be displayed in the call failed state. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>softkey.X.use.ring_back</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>

Parameters	Permitted Values	Default
<b>Description:</b> Enables or disables the custom soft key X to be displayed in the ring back state. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>softkey.X.use.transfer_ring_back</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the custom soft key X to be displayed in the transfer ring back state. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>softkey.X.use.hold</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the custom soft key X to be displayed in the hold state. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>softkey.X.use.held</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the custom soft key X to be displayed in the held state.</p> <p><b>0</b>-Disabled  <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>softkey.X.use.conferenced</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the custom soft key X to be displayed in the conference state.</p> <p><b>0</b>-Disabled  <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>softkey.X.use.dialtone</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the custom soft key X to be displayed in the dial tone (no numbers entered) state.</p> <p><b>0</b>-Disabled  <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
None <b>Phone User Interface:</b> None		
<b>softkey.X.use.dialing</b> (X ranges from 1 to 10)	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the custom soft key X to be displayed in the dialing state. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the values of the parameters "features.enhanced_dss_keys.enable" and "softkey.X.enable" are set to 1 (Enabled). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Configuring a Custom Key

As with EDK soft keys, you can assign functions to DSS keys using macros. Before assigning functions, you have to configure a custom key DSS key in advance.

### Procedure

Custom key DSS key can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Assign a custom key DSS key. <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type linekey.X.value/ programablekey.X.value/ expansion_module.X.key.Y.value linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Assign a custom key DSS key. <b>Navigate to:</b>

	http://<phoneIPAddress>/servlet?p=dss key&q=load&model=0
<b>Phone User Interface</b>	Assign a custom key DSS key.

## Custom Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type</b> <sup>[1]</sup> <b>programablekey.X.type</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.type</b> <sup>[3]</sup>	<b>73</b>
<p><b>Description:</b> Configures a DSS key as a Custom Key DSS key on the IP phone.</p> <p><b>Note:</b> To configure a Custom Key DSS key, make sure the value of the parameter "features.enhanced_dss_keys.enable" is set to 1 (Enabled) in advance. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p> <p><b>Web User Interface:</b> Dsskey-&gt;Line Key/Programable Key-&gt;Type</p> <p><b>Phone User Interface:</b> Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Type</p>	
<b>linekey.X.value</b> <sup>[1]</sup> <b>programablekey.X.value</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.value</b> <sup>[3]</sup>	<b>String within 99 characters</b>
<p><b>Description:</b> Configures the action or function for the DSS key.</p> <p>This value uses the same macro action string syntax as an Enhanced DSS Key. You can also invoke the EDK macro. The macro name follows the character "!".</p> <p><b>Example:</b> linekey.2.value = !macro2</p> <p>In this example, macro1 stands for the macro name configured by the parameter "edk.edklist.X.mname".</p> <p><b>Note:</b> To configure a Custom Key DSS key, make sure the value of the parameter "features.enhanced_dss_keys.enable" is set to 1 (Enabled) in advance. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones.</p>	

Parameters	Permitted Values
<b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Value <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Value	
<b>linekey.X.label</b> <sup>[1]</sup> <b>programablekey.X.label</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.label</b> <sup>[3]</sup>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> To configure a Custom Key DSS key, make sure the value of the parameter "features.enhanced_dss_keys.enable" is set to 1 (Enabled) in advance. EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programable Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters
<b>Description:</b> (Optional.) Configures the short label displayed on the LCD screen for line key. <b>Note:</b> It is only applicable to SIP-T52S IP phones. <b>Web User Interface:</b> Dsskey->Line Key->Short <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Short Label	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

**To configure a Custom Key DSS key via web user interface:**

1. Click on **Dsskey**->**Line Key** (or **Programable Key**).
2. In the desired DSS key field, select **Custom Key** from the pull-down list of **Type**.
3. Enter the desired macro in the **Value** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

Key	Type	Value	Label	Line	Extension
Line Key1	Line	Default	101	Line 1	
Line Key2	Custom Key	Imacro2	N/A		
Line Key3	Line	Default		Line 3	
Line Key4	Line	Default		Line 4	

5. Click **Confirm** to accept the change.

**To configure a Custom Key DSS key via phone user interface:**

1. Press **Menu**->**Features**->**DSS Keys**.
2. Select the desired DSS Key.
3. Press **Left Arrow** or **Right Arrow** , or the **Switch** soft key to select **Custom Key** from the **Type** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
5. Enter the desired macro in the **Value** field.
6. Press the **Save** soft key to accept the change.

## Example Macros

Macros can be applied to various scenarios. The following takes an example to help you understand how to execute the macro action string.

**Scenario Operations:**

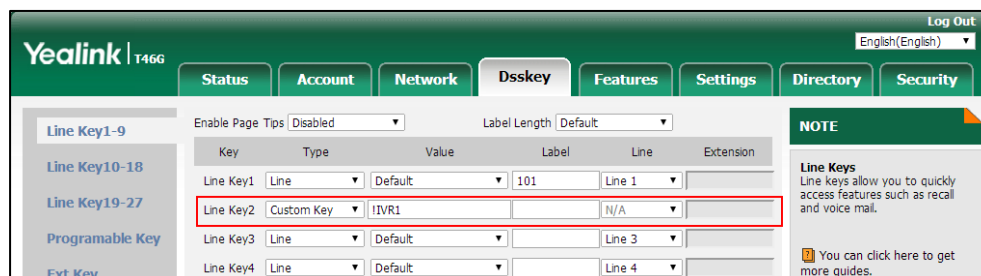
1. Add/Edit the following parameters in the y000000000028.cfg file or <MAC>.cfg (for example, 001565770984.cfg) file you want the IP phone to download:

```
features.enhanced_dss_keys.enable = 1
edk.edklist.1.enable = 1
edk.edklist.1.mname = IVR1
edk.edklist.1.action = 1013$Tinvite$$Cwc$$Cp10$$Penter
number&C4&N$$Tdtmf$$Cp3$$Chu$
```

The macro "IVR1" can be invoked by DSS keys or soft keys. Once triggered, the action string "1013\$Tinvite\$\$Cwc\$\$Cp10\$\$Penter number&C4&N\$\$Tdtmf\$\$Cp3\$\$Chu\$" will be executed.

2. Configure a Custom Key via web user interface.

If you want to invoke a macro action string using a Custom Key, a macro name you enter in the **Value** field must match the name you configure in the configuration file. And the macro name you enter in **Value** field must begin with the "!" prefix. "IVR1" represents invoking the macro "IVR1".



Of course, you can also enter the macro action string "1013\$Tinvite\$\$Cwc\$\$Cp10\$\$Penter number&C4&N\$\$Tdtmf\$\$Cp3\$\$Chu\$" in the **Value** field instead of invoking the macro "IVR1". The execution effect is the same.

The Custom Key displays on the phone:



3. Press the Custom Key when the phone is idle.

The action string 1013\$Tinvite\$\$Cwc\$\$Cp10\$\$Penter number&C4&N\$\$Tdtmf\$\$Cp3\$\$Chu\$ is executed in order as follows:

- 1) A prompt appears on the LCD screen.
- 2) Enter a number for up to 4 digits. For example, 1234. Press the **OK** soft key to continue.
- 3) The IP phone places a call to 1013 using the default account, and then waits for connection.
- 4) After connection, there is a 10 second pause, and then the entered number 1234 is using DTMF dialing on the active call.
- 5) After sending DTMF digits, there is a 3 second pause, and then the active call is automatically disconnected.



## Configuring Audio Features

This chapter provides information for making configuration changes for the following audio features:

- [Redial Tone](#)
- [Ring Tones](#)
- [Distinctive Ring Tones](#)
- [Tones](#)
- [Voice Mail Tone](#)
- [Ringer Device for Headset](#)
- [Handset/Headset/Speakerphone Mode](#)
- [Headset Prior](#)
- [Dual Headset](#)
- [Sending Volume](#)
- [Ringer Volume](#)
- [Audio Codecs](#)
- [Acoustic Clarity Technology](#)
- [DTMF](#)
- [Voice Quality Monitoring \(VQM\)](#)

### Redial Tone

Redial tone allows IP phones to continue to play the dial tone after inputting the preset numbers on the dialing screen.

#### Procedure

Redial tone can be configured using the following methods.

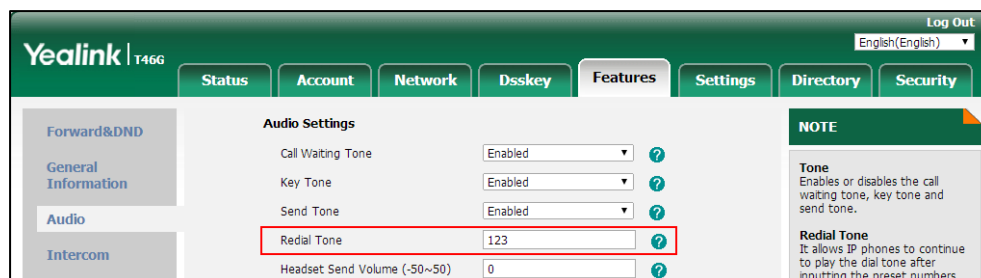
<b>Central Provisioning (Configuration File)</b>	<code>&lt;y0000000000xx&gt;.cfg</code>	Configure redial tone feature. <b>Parameter:</b> features.redial_tone
<b>Web User Interface</b>		Configure redial tone feature. <b>Navigate to:</b> <code>http://&lt;phoneIPAddress&gt;/servlet?p=features-audio&amp;q=load</code>

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.redial_tone</b>	<b>Integer within 6 digits</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configure a number which you enter on the dialing screen, the IP phone will replay the dial tone.</p> <p><b>Example:</b></p> <p>features.redial_tone = 123</p> <p>The IP phone will continue to play the dial tone after inputting "123" on the dialing screen.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Audio-&gt;Redial Tone</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure redial tone via web user interface:**

1. Click on **Features->Audio**.
2. Enter the desired value in the **Redial Tone** field.



3. Click **Confirm** to accept the change.

## Ring Tones

Ring tones are used to indicate incoming calls acoustically. Users can select a built-in system ring tone or a custom ring tone for the phone system or specify line registration. To set the custom ring tones, you need to upload the custom ring tones to the IP phone in advance.

The ring tone format must meet the following:

Phone Model	Format	Single File Size	Note
SIP-T54S/T52S/T48S/T46S/T42S/T41S/T27G	.wav	<=8MB	2MB of space should be reserved for the phone
SIP-T48G/T46G/T29G	.wav	<=8MB	2.4MB of space should be reserved for the
SIP-T42G/T41P/T40P/T40G/T27P/T23P/T23G/T21(P) E2/T19(P) E2	.wav	<=100KB	5KB of space should be reserved for the phone

#### Note

The ring tone file must be PCMU audio format, mono channel, 8K sample rate and 16 bit resolution.

## Procedure

Ring tones can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<y0000000000xx>.cfg	Configure a ring tone for the IP phone. <b>Parameter:</b> phone_setting.ring_type
		Specify the access URL of the custom ring tone. <b>Parameter:</b> ringtone.url
		Delete all custom ring tone files. <b>Parameter:</b> ringtone.delete
	<MAC>.cfg	Configure a ring tone for the SIP account. <b>Parameter:</b> account.X.ringtone.ring_type
<b>Web User Interface</b>		Upload the custom ring tones. Configure a ring tone for the IP phone. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=settings-preference&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=settings-preference&amp;q=load</a>

	Configure a ring tone on a per-line basis. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-basic&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-basic&amp;q=load&amp;acc=0</a>
<b>Phone User Interface</b>	Configure a ring tone for the IP phone. Configure a ring tone for the account.

### Details of the Configuration Parameters:

Parameters	Permitted Values	Default
<b>phone_setting.ring_type</b>	<b>Refer to the following content</b>	<b>Ring1.wav</b>
<p><b>Description:</b> Configures a ring tone for the IP phone.</p> <p><b>Permitted Values:</b> Ring1.wav, Ring2.wav, Ring3.wav, Ring4.wav, Ring5.wav, Ring6.wav, Ring7.wav, Ring8.wav, Silent.wav, Splash.wav or custom ring tone name (for example, Customring.wav).</p> <p><b>Example:</b> phone_setting.ring_type = Ring1.wav</p> <p><b>Web User Interface:</b> Settings-&gt;Preference-&gt;Ring Type</p> <p><b>Phone User Interface:</b> Menu-&gt;Basic-&gt;Sound-&gt;Ring Tones-&gt;Common</p>		
<b>account.X.ringtone.ring_type<sup>[1]</sup></b>	<b>Refer to the following content</b>	<b>Common</b>
<p><b>Description:</b> Configures a ring tone for a specific account.</p> <p><b>Permitted Values:</b> Common, Ring1.wav, Ring2.wav, Ring3.wav, Ring4.wav, Ring5.wav, Ring6.wav, Ring7.wav, Ring8.wav, Silent.wav, Splash.wav or custom ring tone name (for example, Customring.wav).</p> <p><b>Example:</b> account.1.ringtone.ring_type = Ring3.wav It means account1 will use the Ring3.wav as the ring tone. account.1.ringtone.ring_type = Common It means account1 will use the ring tone selected for the IP phone configured by the parameter "phone_setting.ring_type".</p> <p><b>Web User Interface:</b></p>		

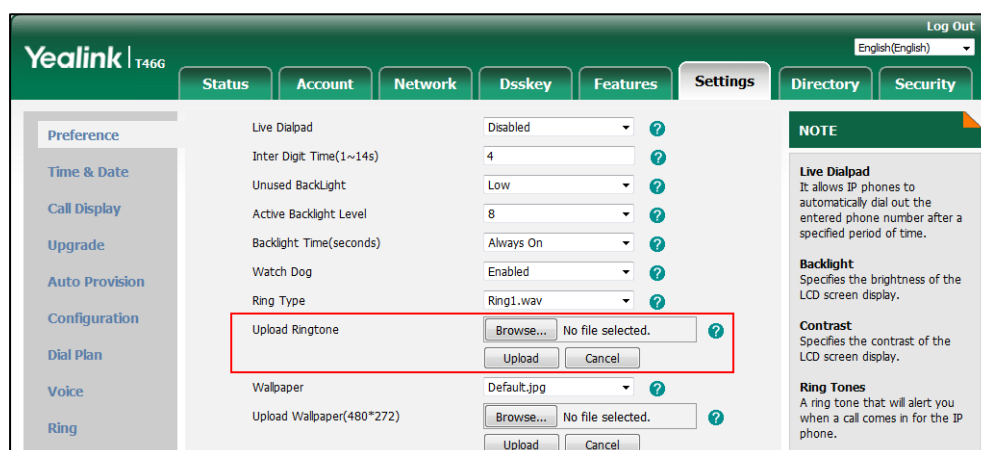
Parameters	Permitted Values	Default
Account->Basic->Ring Type <b>Phone User Interface:</b> Menu->Basic->Sound->Ring Tones->Account X		
<b>ringtone.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the access URL of the custom ring tone file. <b>Example:</b> ringtone.url = tftp://192.168.1.100/Customring.wav <b>Web User Interface:</b> Settings->Preference->Upload Ringtone <b>Phone User Interface:</b> None		
<b>ringtone.delete</b>	<b>http://localhost/all</b>	<b>Blank</b>
<b>Description:</b> Delete all custom ring tone files. <b>Example:</b> ringtone.delete = http://localhost/all <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To upload a custom ring tone via web user interface:**

1. Click on **Settings->Preference**.
2. In the **Upload Ringtone** field, click **Browse** to locate a ring tone file (the file format must be \*.wav) from your local system.

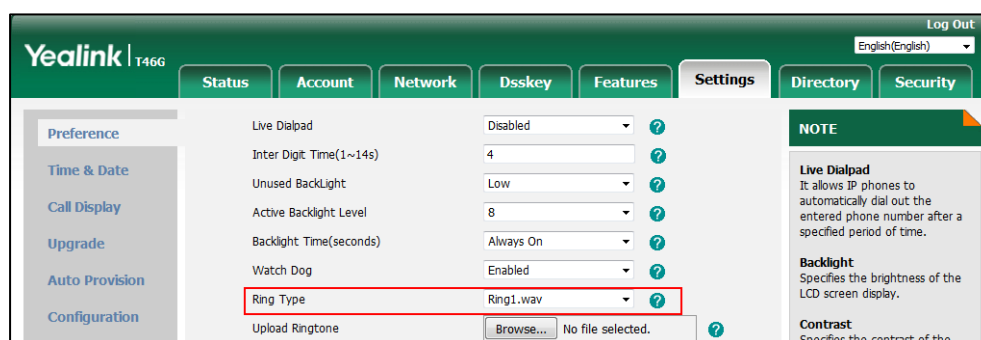
- Click **Upload** to upload the file.



The custom ring tone appears in the pull-down list of **Ring Type**.

#### To change the ring tone for the phone via web user interface:

- Click on **Settings->Preference**.
- Select the desired ring tone from the pull-down list of **Ring Type**.



- Click **Confirm** to accept the change.

#### To change the ring tone for the account via web user interface:

- Click on **Account->Basic**.
- Select the desire account from the pull-down list of **Account**.
- Select the desired ring tone from the pull-down list of **Ring Type**.

If **Common** is selected, this account will use the ring tone selected for the phone.

The screenshot shows the Yealink T466 web interface. The 'Account' tab is selected. The 'Ring Type' dropdown menu is highlighted with a red box and is set to 'Common'. Other settings visible include 'Proxy Require', 'Local Anonymous', 'Local Anonymous Rejection', 'Send Anonymous Code', 'On Code', 'Off Code', 'Send Anonymous Rejection Code', 'Missed Call Log', and 'Auto Answer'.

4. Click **Confirm** to accept the change.

**To select a ring tone for the phone via phone user interface:**

1. Press **Menu->Basic->Sound->Ring Tones->Common**.
2. Press  $\uparrow$  or  $\downarrow$  to select the desired ring tone.
3. Press the **Save** soft key to accept the change.

**To select a ring tone for the account via phone user interface:**

1. Press **Menu->Basic->Sound->Ring Tones**.
2. Press  $\uparrow$  or  $\downarrow$  to select the desired account and then press the **Enter** soft key.
3. Press  $\uparrow$  or  $\downarrow$  to select the desired ring tone.

If **Common** is selected, this account will use the ring tone selected for the phone.

4. Press the **Save** soft key to accept the change.

## Distinctive Ring Tones

Distinctive ring tones allows certain incoming calls to trigger IP phones to play distinctive ring tones. The IP phone inspects the INVITE request for an "Alert-Info" header when receiving an incoming call. If the INVITE request contains an "Alert-Info" header, the IP phone strips out the URL or keyword parameter and maps it to the appropriate ring tone.

### Note

If the caller already exists in the local directory, the ring tone assigned to the caller should be preferentially played.

Alert-Info headers in the following four formats:

- 1) Alert-Info: Bellcore-drN
- 2) Alert-Info: ringtone-N (or Alert-Info: MyMelodyN)

- 3) Alert-Info: <URL>
- 4) Alert-Info: info=info text;x-line-id=0

### 1) Alert-Info: Bellcore-drN

When the Alert-Info header contains the keyword "Bellcore-drN", the IP phone will play the desired ring tone.

The following table identifies the corresponding ring tone:

Value of N	Ring Tone (features.alert_info_tone = 1)	Ring Tone (features.alert_info_tone = 0)
1	Bellcore-dr1	Ring1.wav
2	Bellcore-dr2	Ring2.wav
3	Bellcore-dr3	Ring3.wav
4	Bellcore-dr4	Ring4.wav
5	Bellcore-dr5	Ring5.wav
6	Ring6.wav	
7	Ring7.wav	
8	Ring8.wav	
9	Silent.wav	
10	Splash.wav	
N<1 or N>10	Ring1.wav	

#### Examples:

```
Alert-Info: http://127.0.0.1/Bellcore-dr1
Alert-Info: test/Bellcore-dr1
Alert-Info: Bellcore-dr1
Alert-Info: Bellcore-dr1;x-line-id=1
Alert-Info: <http://10.1.0.31>;info=Bellcore-dr1
```



The following table identifies the different Bellcore ring tone patterns and cadences (These ring tones are designed for the BroadWorks server).

Bellcore Tone	Pattern ID	Pattern	Cadence	Minimum Duration (ms)	Nominal Duration (ms)	Maximum Duration (ms)
Bellcore-dr1(standard)	1	Ring		1800	2000	2200
		Silent		3600	4000	4400
Bellcore-dr2	2	Ring	Long	630	800	1025
		Silent		315	400	525
		Ring	Long	630	800	1025
		Silent		3475	4000	4400
Bellcore-dr3	3	Ring	Short	315	400	525
		Silent		145	200	525
		Ring	Short	315	400	525
		Silent		145	200	525
		Ring	Long	630	800	1025
		Silent		2975	4000	4400
Bellcore-dr4	4	Ring	Short	200	300	525
		Silent		145	200	525
		Ring	Long	800	1000	1100
		Silent		145	200	525
		Ring	Short	200	300	525
		Silent		2975	4000	4400
Bellcore-dr5	5	Ring		450	500	550

#### Note

If the user is waiting for a call, "Bellcore-dr5" is a ring splash tone that reminds the user that the DND or Always Call Forward feature is enabled on the server side.

## 2) Alert-Info: ringtone-N (or Alert-Info: MyMelodyN)

When the Alert-Info header contains the keyword "ringtone-N" or "MyMelodyN", the IP phone will play the corresponding local ring tone (RingN.wav), or play the first local ring tone (Ring1.wav) in about 10 seconds if "N" is greater than 10 or less than 1.

#### Examples:

Alert-Info: ringtone-2

Alert-Info: ringtone-2;x-line-id=1

```
Alert-Info: <http://10.1.0.31>;info=ringtone-2
Alert-Info: <http://127.0.0.1/ringtone-2>
Alert-Info: MyMelody2
Alert-Info: MyMelody2;x-line-id=1
Alert-Info: <http://10.1.0.31>;x-line-id=0;info=MyMelody2
```

The following table identifies the corresponding local ring tone:

Value of N	Ring Tone
1	Ring1.wav
2	Ring2.wav
3	Ring3.wav
4	Ring4.wav
5	Ring5.wav
6	Ring6.wav
7	Ring7.wav
8	Ring8.wav
9	Silent.wav
10	Splash.wav
N<1 or N>10	Ring1.wav

### 3) Alert-Info: <URL>

When the Alert-Info header contains a remote URL, the IP phone will try to download the WAV ring tone file from the URL and then play the remote ring tone if the value of the parameter "account.X.alert\_info\_url\_enable" is set to 1 (or the item called "Distinctive Ring Tones" on the web user interface is Enabled), or play the preconfigured local ring tone in about 10 seconds if the value of the parameter "account.X.alert\_info\_url\_enable" is set to 0 or if the IP phone fails to download the remote ring tone.

#### Example:

```
Alert-Info: http://192.168.0.12:8080/Custom.wav
```

### 4) Alert-Info: info=info text;x-line-id=0

When the Alert-Info header contains an info text, the IP phone will map the text with the Internal Ringer Text preconfigured (or the value of the parameter "distinctive\_ring\_tones.alert\_info.X.text" is configured) on the IP phone, and then play the ring tone associated with the Internal Ringer

Text (the ring tone can be configured by the parameter “distinctive\_ring\_tones.alert\_info.X.ringer”). If no internal ringer text maps, the IP phone will play the preconfigured local ring tone in about 10 seconds.

**Example:**

```
Alert-Info: info=family;x-line-id=0
Alert-Info: <http://10.1.0.31>;info=family
Alert-Info: <http://10.1.0.31>;info=family;x-line-id=0
```

## Auto Answer

If the INVITE request contains the following type of strings, the IP phone will answer incoming calls automatically without playing the ring tone:

- Alert-Info: Auto Answer
- Alert-Info: info = alert-autoanswer
- Alert-Info: answer-after = 0 (or Alert-Info: Answer-After = 0)

If enable auto answer tone feature is enabled, the phone plays a warning tone to alert the user before answering the incoming call. For more information on enable auto answer tone, refer to [Auto Answer](#) on page 345.

**Note**

If the Alert-Info header contains multiple types of keywords, the IP phone will process the keywords in the following order: AutoAnswer>URL>info text/Bellcore-drN/ringtone-N>MyMelodyN.

## Procedure

Distinctive ring tones can be configured using the following methods.

<b>Central Provisioning (Configuration Files)</b>	<MAC>.cfg	Configure distinctive ring tones. <b>Parameter:</b> account.X.alert_info_url_enable
	<y0000000000x x>.cfg	Configure the internal ringer text and internal ringer file. <b>Parameters:</b> features.alert_info_tone distinctive_ring_tones.alert_info.X.text distinctive_ring_tones.alert_info.X.ringer
<b>Web User Interface</b>		Configure distinctive ring tones. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

	<p>Configure the internal ringer text and internal ringer file.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?m=mod_data&amp;p=settings-ring&amp;q=load</p>
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### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>account.X.alert_info_url_enable<sup>[1]</sup></b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to download the ring tone from the URL contained in the Alert-Info header for a specific account.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;Distinctive Ring Tones</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.alert_info_tone</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to map the keywords in the Alert-info header to the specified Bellcore ring tones.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>distinctive_ring_tones.alert_info.X.text</b> (X ranges from 1 to 10)	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the internal ringer text to map the keywords contained in the Alert-Info header.</p> <p><b>Example:</b></p> <p>distinctive_ring_tones.alert_info.1.text = Family</p>		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Settings->Ring->Internal Ringer Text  <b>Phone User Interface:</b> None		
<b>distinctive_ring_tones.alert_info.X.ringer</b> (X ranges from 1 to 10)	<b>Integer from 1 to 10</b>	<b>1</b>
<b>Description:</b> Configures the desired ring tone for each internal ringer text. The value ranges from 1 to 10, the digit stands for the appropriate ring tone. <b>1</b> -Ring1.wav <b>2</b> -Ring2.wav <b>3</b> -Ring3.wav <b>4</b> -Ring4.wav <b>5</b> -Ring5.wav <b>6</b> -Ring6.wav <b>7</b> -Ring7.wav <b>8</b> -Ring8.wav <b>9</b> -Silent.wav <b>10</b> -Splash.wav  <b>Web User Interface:</b> Settings->Ring->Internal Ringer File  <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure distinctive ring tones via web user interface:**

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.

3. Select the desired value from the pull-down list of **Distinctive Ring Tones**.

The screenshot shows the 'Account' configuration page for a Yealink T46G IP phone. The 'Distinctive Ring Tones' field is highlighted with a red box and set to 'Enabled'. The page includes tabs for Status, Account, Network, Dsskey, Features, Settings, Directory, and Security. A sidebar on the left lists options like Register, Basic, Codec, and Advanced. A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

4. Click **Confirm** to accept the change.

**To configure the internal ringer text and internal ringer file via web user interface:**

1. Click on **Settings->Ring**.
2. Enter the keywords in the **Internal Ringer Text** fields.
3. Select the desired ring tones for each text from the pull-down lists of **Internal Ringer File**.

The screenshot shows the 'Settings > Ring' configuration page for a Yealink T46G IP phone. The first row of the 'Internal Ringer Text' and 'Internal Ringer File' fields is highlighted with a red box. The page includes tabs for Status, Account, Network, Dsskey, Features, Settings, Directory, and Security. A sidebar on the left lists options like Preference, Time & Date, Call Display, Upgrade, Auto Provision, Configuration, Dial Plan, Voice, Ring, and Tones. A 'NOTE' section on the right provides information about Distinctive Ring Tones.

4. Click **Confirm** to accept the change.

## Tones

When receiving a message, the IP phone will play a warning tone. You can customize tones or select specialized tone sets (vary from country to country) to indicate different conditions of the

IP phone. The default tones used on IP phones are the US tone sets. Available tone sets for IP phones:

- Australia
- Austria
- Brazil
- Belgium
- China
- Czech
- Denmark
- Finland
- France
- Germany
- Great Britain
- Greece
- Hungary
- Lithuania
- India
- Italy
- Japan
- Mexico
- New Zealand
- Netherlands
- Norway
- Portugal
- Spain
- Switzerland
- Sweden
- Russia
- United States
- Chile
- Czech ETSI

Configured tones can be heard on IP phones for the following conditions.

Condition	Description
Dial	When in the dialing interface
Secondary Dial	When adding a comma “,” to the digit map (For more information on digit map, refer to <a href="#">Dial Plan using Digit</a> )

Condition	Description
	<a href="#">Map String Rules</a> )
Ring Back	Ring-back tone
Busy	When the callee is busy
Congestion	When the network is congested
Call Waiting	Call waiting tone (For more information on call waiting, refer to <a href="#">Call Waiting</a> )
Dial Recall	When receiving a call back
Info	When receiving a special message
Stutter	When receiving a voice mail (For more information on voice mail tone, refer to <a href="#">Voice Mail Tone</a> )
Message	When receiving a text message (For more information on text message, refer to <a href="#">Short Message Service (SMS)</a> )
Auto Answer	When automatically answering a call (For more information on auto answer, refer to <a href="#">Auto Answer</a> )

## Procedure

Tones can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	<p>Configure the tones for the IP phone.</p> <p><b>Parameters:</b></p> <p>voice.tone.country</p> <p>voice.tone.dial</p> <p>voice.tone.secondary_dial</p> <p>voice.tone.ring</p> <p>voice.tone.busy</p> <p>voice.tone.congestion</p> <p>voice.tone.callwaiting</p> <p>voice.tone.dialrecall</p> <p>voice.tone.info</p> <p>voice.tone.stutter</p> <p>voice.tone.message</p> <p>voice.tone.autoanswer</p>
<b>Web User Interface</b>		Configure the tones for the IP phone.



	<b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=settings-tones&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=settings-tones&amp;q=load</a>
--	--------------------------------------------------------------------------------------------------------------------------------------------------------------------------

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>voice.tone.country</b>	<b>Refer to the following content</b>	<b>Custom</b>
<p><b>Description:</b> Configures the country tone for the IP phone.</p> <p><b>Permitted Values:</b> Custom, Australia, Austria, Brazil, Belgium, Chile, China, Czech, Czech ETSI, Denmark, Finland, France, Germany, Great Britain, Greece, Hungary, Lithuania, India, Italy, Japan, Mexico, New Zealand, Netherlands, Norway, Portugal, Spain, Switzerland, Sweden, Russia, United States.</p> <p><b>Example:</b> voice.tone.country = Custom</p> <p><b>Web User Interface:</b> Settings-&gt;Tones-&gt;Select Country</p> <p><b>Phone User Interface:</b> None</p>		
<b>voice.tone.dial</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b> Customizes the dial tone.</p> <p>tonelist = element[,element] [,element]...</p> <p>Where</p> <p><b>element</b> = [!]Freq1[+Freq2][+Freq3][+Freq4] /Duration</p> <p><b>Freq</b>: the frequency of the tone (ranges from 200 to 4000Hz). If it is set to 0Hz, it means the tone is not played.</p> <p><b>For SIP-T40P/T40G/T27G/T23P/T23G/T21(P) E2/T19(P) E2:</b> A tone is comprised of at most two different frequencies.</p> <p><b>For SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T29G/T27P/T27G:</b> A tone is comprised of at most four different frequencies.</p> <p><b>Duration</b>: the duration (in milliseconds) of the dial tone, ranges from 0 to 30000ms.</p> <p>You can configure at most eight different tones for one condition, and separate them by commas. (for example, 250/200,0/1000,200+300/500,200+500+800+1500/1000).</p> <p>If you want the IP phone to play tones once, add an exclamation mark "!" before tones (for</p>		

Parameters	Permitted Values	Default
<p>example, !250/200,0/1000,200+300/500,200+500+800+1500/1000).</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b> Settings-&gt;Tones-&gt;Dial</p> <p><b>Phone User Interface:</b> None</p>		
<b>voice.tone.secondary_dial</b>	<b>String</b>	<b>350+440/300 0</b>
<p><b>Description:</b> Customizes the secondary dial tone.</p> <p>The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b> Settings-&gt;Tones-&gt;Secondary Dial</p> <p><b>Phone User Interface:</b> None</p>		
<b>voice.tone.ring</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b> Customizes the ringback tone.</p> <p>The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b> Settings-&gt;Tones-&gt;Ring Back</p> <p><b>Phone User Interface:</b> None</p>		
<b>voice.tone.busy</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b> Customizes the tone when the callee is busy.</p>		

Parameters	Permitted Values	Default
<p>The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b> Settings-&gt;Tones-&gt;Busy</p> <p><b>Phone User Interface:</b> None</p>		
<b>voice.tone.congestion</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b> Customizes the tone when the network is congested.</p> <p>The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b> Settings-&gt;Tones-&gt;Congestion</p> <p><b>Phone User Interface:</b> None</p>		
<b>voice.tone.callwaiting</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b> Customizes the call waiting tone.</p> <p>The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b> Settings-&gt;Tones-&gt;Call Waiting</p> <p><b>Phone User Interface:</b> None</p>		
<b>voice.tone.dialrecall</b>	<b>String</b>	<b>Blank</b>

Parameters	Permitted Values	Default
<p><b>Description:</b></p> <p>Customizes the call back tone.</p> <p>The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Tones-&gt;Dial Recall</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>voice.tone.info</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Customizes the info tone. The phone will play the info tone with the special information, for example, the number you are calling is not in service.</p> <p>The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Tones-&gt;Info</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>voice.tone.stutter</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Customizes the tone when the IP phone receives a voice mail.</p> <p>The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Tones-&gt;Stutter</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

Parameters	Permitted Values	Default
<b>voice.tone.message</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Customizes the tone when the IP phone receives a text message.</p> <p>The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Tones-&gt;Message</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>voice.tone.autoanswer</b>	<b>String</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Customizes the warning tone for auto answer.</p> <p>The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p><b>Note:</b> It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Tones-&gt;Auto Answer</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure tones via web user interface:**

1. Click on **Settings->Tones**.
2. Select the desired value from the pull-down list of **Select Country**.

If you select **Custom**, you can customize a tone for each condition of the IP phone.

The screenshot shows the Yealink T46G web interface. The 'Settings' tab is selected. In the 'Tones' section, the 'Select Country' dropdown is set to 'Custom' and is highlighted with a red box. Below this, there are input fields for various tones: Dial, Secondary Dial (350+440/3000), Ring Back, Busy, Congestion, Call Waiting, Dial Recall, Info, Stutter, Message, and Auto Answer. Each field has a question mark icon to its right. On the right side, a 'NOTE' box explains that users can customize tones or select specialized tone sets (vary from country to country) to indicate different conditions of the IP phone. It also includes a link to get more guides.

3. Click **Confirm** to accept the change.

## Voice Mail Tone

Voice mail tone feature allows the IP phone to play a warning tone when receiving a new voice mail. You can customize the warning tone or select specialized tone sets (vary from country to country) for your IP phone. For more information, refer to [Tones](#) on page 724.

### Procedure

Voice mail tone can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure whether to play a warning tone when the IP phone receives a new voice mail. <b>Parameter:</b> features.voice_mail_tone_enable
<b>Web User Interface</b>		Configure whether to play a warning tone when the IP phone receives a new voice mail. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?parameters=features-general&q=load

### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.voice_mail_tone_enable	0 or 1	1

Parameter	Permitted Values	Default
<p><b>Description:</b></p> <p>Enables or disables the IP phone to play a warning tone when it receives a new voice mail.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.display_mwi.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Voice Mail Tone</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure voice mail tone via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Voice Mail Tone**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'Voice Mail Tone' dropdown menu is highlighted with a red box and set to 'Enabled'. Other settings visible include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Call Waiting Off Code' (empty), 'Auto Redial' (Disabled), 'Auto Redial Interval (1~300s)' (10), 'Auto Redial Times (1~300)' (10), 'Accept SIP Trust Server Only' (Enabled), 'Allow IP Call' (Enabled), 'IP Direct Auto Answer' (Enabled), 'Call List Show Number' (Enabled), and 'Auto Linekeys' (Disabled). A 'NOTE' panel on the right provides details for 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion'. At the bottom, there are 'Confirm' and 'Cancel' buttons.

3. Click **Confirm** to accept the change.

## Ringer Device for Headset

The IP phones support either or both speaker and headset ringer devices. Ringer Device for Headset feature allows users to configure which ringer device to be used when receiving an incoming call. For example, if the ringer device is set to Headset, ring tone will be played through your headset.

If the ringer device is set to Headset or Headset&Speaker, the headset (either a wired headset, Bluetooth headset or USB headset) should be connected to the IP phone and the headset mode

also should be activated in advance. The USB headset is only applicable to SIP-T41S/T42S/T46S/T48S/T52S/T54S IP phones.

You can press the HEADSET key to activate the headset mode. For more information, refer to the [Yealink phone-specific user guide](#).

## Procedure

Ringer device for headset can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the ringer device for the IP phone. <b>Parameter:</b> features.ringer_device.is_use_headset
<b>Web User Interface</b>		Configure the ringer device for the IP phone. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-audio&q=load

## Details of Configuration Parameter:

Parameter	Permitted Values	Default
features.ringer_device.is_use_headset	0, 1 or 2	0
<p><b>Description:</b> Configures the ringer device for the IP phone.</p> <p><b>0</b>-Use Speaker <b>1</b>-Use Headset <b>2</b>-Use Headset &amp; Speaker</p> <p>If the ringer device is set to Headset or Headset&amp;Speaker, the headset should be connected to the IP phone and the headset mode also should be activated in advance.</p> <p><b>Note:</b> If you want to user speaker ringer device for the phone, you have to set the value of the parameter "features.speaker_mode.enable" to 1 (Enabled). If you want to user headset ringer device for the phone, you have to set the value of the parameter "features.headset_mode.enable" to 1 (Enabled).</p> <p><b>Web User Interface:</b> Features-&gt;Audio-&gt;Ringer Device for Headset</p> <p><b>Phone User Interface:</b> None</p>		



To configure ringer device for headset via web user interface:

1. Click on **Features->Audio**.
2. Select the desired value from the pull-down list of **Ringer Device for Headset**.

3. Click **Confirm** to accept the change.

## Handset/Headset/Speakerphone Mode

Yealink IP phones support three ways to place/answer a call: using the handset, using the headset or using the speakerphone. You can disable the infrequently used audio device as required.

### Procedure

Handset/headset/speakerphone mode can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the speaker mode. <b>Parameter:</b> features.speaker_mode.enable
		Configure the handset mode. <b>Parameter:</b> features.handset_mode.enable
		Configure the headset mode. <b>Parameter:</b> features.headset_mode.enable

### Details of the Configuration Parameters:

Parameters	Permitted Values	Default
features.speaker_mode.enable	0 or 1	1

Parameters	Permitted Values	Default
<b>Description:</b> Enables or disables the phone's audio speakerphone. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.handset_mode.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the phone's handset. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.headset_mode.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the phone's headset jack. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

## Headset Prior

Headset prior allows users to use headset preferentially if a headset is physically connected to the IP phone. This feature is especially useful for permanent or full-time headset users.

### Note

It is not applicable to the Bluetooth headset and USB headset.

### Procedure

Headset prior can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure headset prior. <b>Parameter:</b> features.headset_prior
<b>Web User Interface</b>		Configure headset prior. <b>Navigate to:</b> http://<phoneIPAddress>/servl et?p=features-general&q=load

### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
<b>features.headset_prior</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables headset prior feature. You need to press the HEADSET key to activate the headset mode in advance.</p> <p><b>0</b>-Disabled, the headset mode can be deactivated by pressing the Speakerphone key or the HEADSET key except off-hook.</p> <p><b>1</b>-Enabled, the headset mode will not be deactivated until the user presses the HEADSET key again.</p> <p><b>Note:</b> It works only if the value of the parameter "features.headset_mode.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Headset Prior</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

### To configure headset prior via web user interface:

1. Click on **Features->General Information**.

2. Select the desired value from the pull-down list of **Headset Prior**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected. In the 'General Information' section, the 'Headset Prior' dropdown menu is highlighted with a red box and is set to 'Enabled'. Other settings like 'Call Waiting', 'Auto Redial', and 'Dual-Headset' are also visible. A 'NOTE' section on the right provides details about 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion'.

3. Click **Confirm** to accept the change.

## Dual Headset

Dual headset allows users to use two headsets on one IP phone. To use this feature, users need to physically connect two headsets to the headset and handset jacks respectively. Once the IP phone connects to a call, the user with the headset connected to the headset jack has full-duplex capabilities, while the user with the headset connected to the handset jack is only able to listen.

**Note** Dual headset is not applicable to the Bluetooth headset and USB headset.

### Procedure

Dual headset can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure dual headset. <b>Parameter:</b> features.headset_training
<b>Web User Interface</b>		Configure dual headset. <b>Navigate to:</b> http://<phoneIPAddress>/servl et?p=features-general&q=load

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.headset_training</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables dual headset feature.</p> <p><b>0-Disabled</b></p> <p><b>1-Enabled</b>, users can use two headsets on one phone. When the IP phone joins in a call, the users with the headset connected to the headset jack have a full-duplex conversation, while the users with the headset connected to the handset jack are only allowed to listen to.</p> <p><b>Note:</b> It works only if the value of the parameter "features.headset_mode.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Features-&gt;General Information-&gt;Dual Headset</p> <p><b>Phone User Interface:</b> None</p>		

**To configure dual headset via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Dual Headset**.

The screenshot displays the Yealink T46G web management interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'Dual Headset' dropdown menu is highlighted with a red rectangle and is currently set to 'Enabled'. Other settings visible include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Allow Mute' (Enabled), 'Auto Answer Delay(1~4s)' (1), 'Enable Auto Answer Tone' (Enabled), 'Headset Prior' (Enabled), 'DTMF Replace Tran' (Disabled), 'Accept SIP Trust Server Only' (Disabled), 'Display Method on Dialing' (User Name), and 'Auto Linekeys' (Disabled). At the bottom of the configuration area are 'Confirm' and 'Cancel' buttons. On the right side, a 'NOTE' section provides additional information about features like Call Waiting, Auto Redial, Key As Send, Hotline, and Call Completion.

3. Click **Confirm** to accept the change.

## Sending Volume

Sending volume allows user to adjust the sending volume of currently engaged audio devices (handset, speakerphone or headset) when the phone is in use.

### Procedure

Sending volume can be configured using the following methods.

Central Provisioning (Configuration File)	<y0000000000xx>.cfg	Configure the sending volume of the speaker. <b>Parameter:</b> voice.handfree_send
		Configure the sending volume of the handset. <b>Parameter:</b> voice.handset_send
		Configure the sending volume of the headset. <b>Parameter:</b> voice.headset_send
Web User Interface		Configure the sending volume of the speaker/handset/headset. <b>Navigate to:</b>  http://<phoneIPAddress>/servlet?p=features-audio&q=load

### Details of the Configuration Parameters:

Parameters	Permitted Values	Default
voice.handfree_send	Integer from -50 to 50	0
<b>Description:</b> Configures the sending volume of the speaker.  <b>Note:</b> We recommend that you modify this parameter cautiously. An unreasonable value may render the voice quality bad. If you change this parameter, the IP phone will reboot to make the change take effect.  <b>Web User Interface:</b> Features->Audio->Handfree Send Volume (-50~50)		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>voice.handset_send</b>	<b>Integer from -50 to 50</b>	<b>0</b>
<b>Description:</b> Configures the sending volume of the handset.  <b>Note:</b> We recommend that you modify this parameter cautiously. An unreasonable value may render the voice quality bad. If you change this parameter, the IP phone will reboot to make the change take effect.  <b>Web User Interface:</b> Features->Audio->Handset Send Volume (-50~50)  <b>Phone User Interface:</b> None		
<b>voice.headset_send</b>	<b>Integer from -50 to 50</b>	<b>0</b>
<b>Description:</b> Configures the sending volume of the headset.  <b>Note:</b> We recommend that you modify this parameter cautiously. An unreasonable value may render the voice quality bad. If you change this parameter, the IP phone will reboot to make the change take effect.  <b>Web User Interface:</b> Features->Audio->Headset Send Volume (-50~50)  <b>Phone User Interface:</b> None		

**To configure sending volume via web user interface:**

1. Click on **Features->Audio**.

- Enter the desired value in the **Headset Send Volume (-50~50)** field.
- Enter the desired value in the **Handset Send Volume (-50~50)** field.
- Enter the desired value in the **Handfree Send Volume (-50~50)** field.

The screenshot shows the Yealink T466 web interface. The 'Audio Settings' section is active. The 'Headset Send Volume (-50~50)', 'Handset Send Volume (-50~50)', and 'Handfree Send Volume (-50~50)' fields are highlighted with a red box. The 'Ringer Device for Headset' is set to 'Use Speaker'. A 'NOTE' section on the right explains the 'Redial Tone' and 'Ringer Device for Headset' settings.

- Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
- Click **OK** to reboot the phone.

## Ringer Volume

You can configure the ringer volume forcedly, so the user cannot adjust the ringer volume on the phone. This feature is used to avoid missing calls when the user turns down the ringer volume.

### Procedure

Ringer volume can be configured using the following method.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the ringer volume forcedly. <b>Parameter:</b> force.voice.ring_vol
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### Details of the Configuration Parameter:

Parameters	Permitted Values	Default
force.voice.ring_vol	Integer from 0 to 15	Blank
<b>Description:</b> Configures the ringer volume forcedly.  If it is set from 0 to 15, the user cannot adjust the ringer volume on the phone. The user can press the <b>Silence</b> soft key to mute the ringer when receiving an incoming call.		



Parameters	Permitted Values	Default
<p>If it is left blank, the user can adjust the ringer volume on the phone.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

## Audio Codecs

CODEC is an abbreviation of COmpress-DEcompress, capable of coding or decoding a digital data stream or signal by implementing an algorithm. The object of the algorithm is to represent the high-fidelity audio signal with minimum number of bits while retaining the quality. This can effectively reduce the frame size and the bandwidth required for audio transmission.

The audio codec that the phone uses to establish a call should be supported by the SIP server. When placing a call, the IP phone will offer the enabled audio codec list to the server and then use the audio codec negotiated with the called party according to the priority.

## Supported Audio Codecs

The following table summarizes the supported audio codecs on IP phones:

Codec	Algorithm	Reference	Bit Rate	Sample Rate	Packetization Time
G722	G.722	RFC 3551	64 Kbps	16 Ksps	20ms
PCMA	G.711	RFC 3551	64 Kbps	8 Ksps	20ms
PCMU	G.711	RFC 3551	64 Kbps	8 Ksps	20ms
G729	G.729	RFC 3551	8 Kbps	8 Ksps	20ms
G726-16	G.726	RFC 3551	16 Kbps	8 Ksps	20ms
G726-24	G.726	RFC 3551	24 Kbps	8 Ksps	20ms
G726-32	G.726	RFC 3551	32 Kbps	8 Ksps	20ms
G726-40	G.726	RFC 3551	40 Kbps	8 Ksps	20ms
G723_53/ G723_63	G.723.1	RFC 3551	5.3 Kbps 6.3 Kbps	8 Ksps	30ms
iLBC_15_2kbps	iLBC	RFC 3952	15.2 Kbps	8 Ksps	20ms
iLBC_13_33kbps	iLBC	RFC 3952	13.33 Kbps	8 Ksps	30ms

Codec	Algorithm	Reference	Bit Rate	Sample Rate	Packetization Time
opus	opus	RFC 6716	16 Kbps 20 Kbps	8 Ksps 16 Ksps	20ms

**Note**

The network bandwidth necessary to send the encoded audio is typically 5~10% higher than the bit rate due to packetization overhead. For example, a two-way G.722 audio call at 64 Kbps consumes about 135 Kbps of network bandwidth.

The Opus codec supports various audio bandwidths, defined as follows:

Abbreviation	Audio Bandwidth	Sample Rate (Effective)
NB (narrowband)	4 kHz	8 kHz
MB (medium-band)	6 kHz	12 kHz
WB (wideband)	8 kHz	16 kHz
SWB (super-wideband)	12 kHz	24 kHz
FB (fullband)	20 kHz	48 kHz

**Note**

The SIP-T40P/T40G/T23P/T23G/T21(P) E2/T19(P) E2 IP phones support using WB for opus encoding; and using NB, MB, WB, SWB and FB for opus decoding.

The SIP-T54S/T52S/T48S/T46S/T42S/T41S/T27G IP phones support using WB (default) or NB for opus encoding; and using NB, MB, WB, SWB and FB for opus decoding.

The following table lists the audio codecs supported by each phone model:

Phone Model	Supported Audio Codecs	Default Audio Codecs
SIP-T48G/T46G/T42G/T41P/T29G	G722, PCMA, PCMU, G729, G726-16, G726-24, G726-32, G726-40, G723_53, G723_63, iLBC_15_2kpbs, iLBC_13_33kpbs	G722, PCMA, PCMU, G729
SIP-T54S/T52S/T48S/T46S/T42S/T41S/T27G	G722, PCMA, PCMU, G729, G726-16, G726-24, G726-32, G726-40, G723_53, G723_63, opus, iLBC_15_2kpbs, iLBC_13_33kpbs	G722, PCMA, PCMU, G729
SIP-T27P	G722, PCMA, PCMU, G729, G726-16, G726-24, G726-32, G726-40, iLBC_15_2kpbs, iLBC_13_33kpbs	G722, PCMA, PCMU, G729
SIP-T40P/T40G/T23P/T23G	G722, PCMA, PCMU, G729,	G722, PCMA, PCMU,

Phone Model	Supported Audio Codecs	Default Audio Codecs
/T21(P) E2/T19(P) E2	G726-16, G726-24, G726-32, G726-40, opus, iLBC_15_2kpbs, iLBC_13_33kpbs	G729

## Audio Codec Configuration

Yealink IP phones running firmware version 81 or later support a new configuration behavior for the audio codecs. It is more efficiently for you to provision a number of different IP phone modules.

The configuration parameters are different for the new configuration behavior and the older one. For more information on old configuration behavior, refer to [Yealink\\_SIP-T2\\_Series\\_T19\(P\)\\_E2\\_T4\\_Series\\_T5\\_Series\\_CP860\\_IP\\_Phones\\_Administrator\\_Guide\\_V81](#).

### Procedure

Configuration changes can be performed using the following methods.

Central Provisioning (Configuration File)	<MAC>.cfg	Configure the codecs to use on a per-line basis.  <b>Parameter:</b> account.X.codec.<payload_type>.enable
		Configure the priority for the enabled codec.  <b>Parameters:</b> account.X.codec.<payload_type>.priority
		Configure opus sample rate.  <b>Parameter:</b> account.X.codec.opus.para
Web User Interface	Configure the codecs to use on a per-line basis.  Configure the priority for the enabled codec.  Configure the opus sample rate.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-codec&q=load&acc=0	

**Details of Configuration Parameters:**





Parameters	Permitted Values	Default																
<b>account.X.codec.&lt;payload_type&gt;.enable<sup>[1]</sup></b> (where <payload_type> should be replaced by the name of audio codec)	<b>0 or 1</b>	<b>Refer to the following content</b>																
<b>Description:</b>  Enables or disables the specified audio codec for a specific account.  <b>0</b> -Disabled <b>1</b> -Enabled  <b>The name of audio codec:</b>  <table><tr><td><b>g722</b>-G722</td><td><b>pcmu</b>-PCMU</td><td><b>pcma</b>-PCMA</td><td><b>g729</b>-G729</td></tr><tr><td><b>g726_16</b>-G726-16</td><td><b>g726_24</b>-G726-24</td><td><b>g726_32</b>-G726-32</td><td><b>g726_40</b>-G726-40,</td></tr><tr><td><b>g723_53</b>-G723_53</td><td><b>g723_63</b>-G723_63</td><td><b>opus</b>-opus</td><td></td></tr><tr><td><b>ilbc_15_2kpbs</b>-iLBC_15_2kpbs</td><td><b>ilbc_13_33kpbs</b>-iLBC_13_33kpbs</td><td></td><td></td></tr></table> <b>Default:</b>  When audio codec is G722, the default value is 1; When audio codec is PCMU, the default value is 1; When audio codec is PCMA, the default value is 1; When audio codec is G729, the default value is 1; When audio codec is G726-16, the default value is 0; When audio codec is G726-24, the default value is 0; When audio codec is G726-32, the default value is 0; When audio codec is G726-40, the default value is 0; When audio codec is G723_53, the default value is 0; When audio codec is G723_63, the default value is 0; When audio codec is opus, the default value is 0; When audio codec is iLBC_15_2kpbs, the default value is 0; When audio codec is iLBC_13_33kpbs, the default value is 0;  <b>Example:</b>  account.1.codec.g722.enable = 1  <b>Note:</b> The name of audio codec in this parameter should be the correct one as listed in the above example, otherwise the corresponding configuration will not take effect.  <b>Web User Interface:</b>  Account->Codec->Audio Codec  <b>Phone User Interface:</b>  None			<b>g722</b> -G722	<b>pcmu</b> -PCMU	<b>pcma</b> -PCMA	<b>g729</b> -G729	<b>g726_16</b> -G726-16	<b>g726_24</b> -G726-24	<b>g726_32</b> -G726-32	<b>g726_40</b> -G726-40,	<b>g723_53</b> -G723_53	<b>g723_63</b> -G723_63	<b>opus</b> -opus		<b>ilbc_15_2kpbs</b> -iLBC_15_2kpbs	<b>ilbc_13_33kpbs</b> -iLBC_13_33kpbs		
<b>g722</b> -G722	<b>pcmu</b> -PCMU	<b>pcma</b> -PCMA	<b>g729</b> -G729															
<b>g726_16</b> -G726-16	<b>g726_24</b> -G726-24	<b>g726_32</b> -G726-32	<b>g726_40</b> -G726-40,															
<b>g723_53</b> -G723_53	<b>g723_63</b> -G723_63	<b>opus</b> -opus																
<b>ilbc_15_2kpbs</b> -iLBC_15_2kpbs	<b>ilbc_13_33kpbs</b> -iLBC_13_33kpbs																	

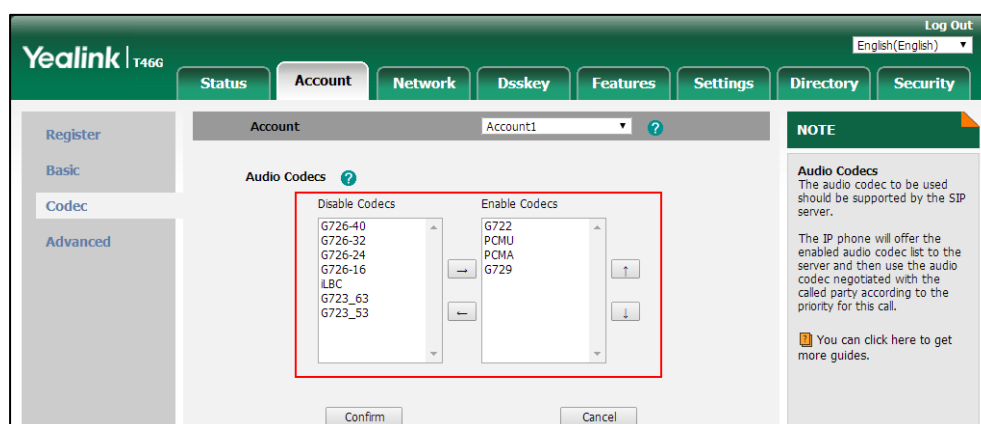
Parameters	Permitted Values	Default																
<b>account.X.codec.&lt;payload_type&gt;.priority<sup>[1]</sup></b> (where <payload_type> should be replaced by the name of audio codec)	<b>Integer from 0 to 13</b>	<b>Refer to the following content</b>																
<b>Description:</b> Configures the priority of the enabled audio codec for a specific account. <b>The name of audio codec:</b> <table><tr><td><b>g722</b>-G722</td><td><b>pcmu</b>-PCMU</td><td><b>pcma</b>-PCMA</td><td><b>g729</b>-G729</td></tr><tr><td><b>g726_16</b>-G726-16</td><td><b>g726_24</b>-G726-24</td><td><b>g726_32</b>-G726-32</td><td><b>g726_40</b>-G726-40</td></tr><tr><td><b>g723_53</b>-G723_53</td><td><b>g723_63</b>-G723_63</td><td><b>opus</b>-opus</td><td></td></tr><tr><td><b>ilbc_15_2kpbs</b>-iLBC_15_2kpbs</td><td><b>ilbc_13_33kpbs</b>-iLBC_13_33kpbs</td><td></td><td></td></tr></table> <b>Default:</b> When audio codec is G722, the default value is 1; When audio codec is PCMU, the default value is 2; When audio codec is PCMA, the default value is 3; When audio codec is G729, the default value is 4; When audio codec is G726_16, the default value is 0; When audio codec is G726_24, the default value is 0; When audio codec is G726_32, the default value is 0; When audio codec is G726_40, the default value is 0; When audio codec is G723_53, the default value is 0; When audio codec is G723_63, the default value is 0; When audio codec is opus, the default value is 0; When audio codec is iLBC_15_2kpbs, the default value is 0; When audio codec is iLBC_13_33kpbs, the default value is 0;  <b>Example:</b> account.1.codec.g722.priority = 1  <b>Note:</b> The priority of codec in disable codec list is not specified, and numerical value 1 is defined as the highest priority in the enable codec list. The name of audio codec in this parameter should be the correct one as listed in the above example, otherwise the corresponding configuration will not take effect.  <b>Web User Interface:</b> Account->Codec->Audio Codec  <b>Phone User Interface:</b> None			<b>g722</b> -G722	<b>pcmu</b> -PCMU	<b>pcma</b> -PCMA	<b>g729</b> -G729	<b>g726_16</b> -G726-16	<b>g726_24</b> -G726-24	<b>g726_32</b> -G726-32	<b>g726_40</b> -G726-40	<b>g723_53</b> -G723_53	<b>g723_63</b> -G723_63	<b>opus</b> -opus		<b>ilbc_15_2kpbs</b> -iLBC_15_2kpbs	<b>ilbc_13_33kpbs</b> -iLBC_13_33kpbs		
<b>g722</b> -G722	<b>pcmu</b> -PCMU	<b>pcma</b> -PCMA	<b>g729</b> -G729															
<b>g726_16</b> -G726-16	<b>g726_24</b> -G726-24	<b>g726_32</b> -G726-32	<b>g726_40</b> -G726-40															
<b>g723_53</b> -G723_53	<b>g723_63</b> -G723_63	<b>opus</b> -opus																
<b>ilbc_15_2kpbs</b> -iLBC_15_2kpbs	<b>ilbc_13_33kpbs</b> -iLBC_13_33kpbs																	

Parameters	Permitted Values	Default
<b>account.X.codec.opus.para</b> <sup>[1]</sup>	<b>opus-wb or opus-nb</b>	<b>opus-wb</b>
<p><b>Description:</b></p> <p>Configures the sample rate of the opus audio codec for a specific account.</p> <p><b>opus-wb</b>-Opus-WB (16KHz)</p> <p><b>opus-nb</b>-Opus-NB (8KHz)</p> <p><b>Note:</b> It is only applicable to SIP-T54S/T52S/T48S/T46S/T42S/T41S/T27G IP phones.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Codec-&gt;Opus Sample Rate</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure the codecs to use and adjust the priority of the enabled codecs via web user interface:**

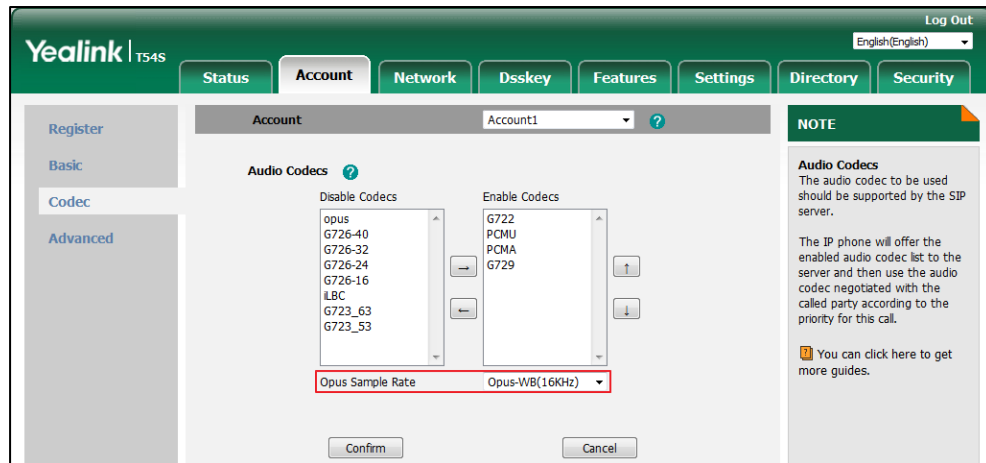
1. Click on **Account->Codec**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired codec from the **Disable Codecs** column and then click  .  
The selected codec appears in the **Enable Codecs** column.
4. Repeat the step 4 to add more codecs to the **Enable Codecs** column.
5. To remove the codec from the **Enable Codecs** column, select the desired codec and then click  .
6. To adjust the priority of codecs, select the desired codec and then click  or  .



7. Click **Confirm** to accept the change.

To configure the opus sample rate via web user interface (take SIP-T54S IP phones for example):

1. Click on **Account->Codec**.
2. Select the desired value from the pull-down list of **Opus Sample Rate**.



3. Click **Confirm** to accept the change.

## Packetization Time (PTime)

Ptime is a measurement of the duration (in milliseconds) of the audio data in each RTP packet sent to the destination, and defines how much network bandwidth is used for the RTP stream transfer. Before establishing a conversation, codec and ptime are negotiated through SIP signaling. The valid values of ptime range from 10 to 60, in increments of 10 milliseconds. The default ptime is 20ms. You can also disable the ptime negotiation.

The following table summarizes the valid values of ptime for each audio codec:

Codec	Packetization Time (Minimun)	Packetization Time (Maximun)
G722	10ms	40ms
PCMA	10ms	40ms
PCMU	10ms	40ms
G729	10ms	80ms
G726-16	10ms	30ms
G726-24	10ms	30ms

Codec	Packetization Time (Minimun)	Packetization Time (Maximun)
G726-32	10ms	30ms (40ms for SIP-T40P/T40G/T23P/T23G/T21(P) E2/T19(P) E2)
G726-40	10ms	30ms
G723_53/ G723_63	30ms	60ms
iLBC	20ms	30ms
opus	10ms	20ms
iLBC_15_2kpbs	20ms, 40ms, 60ms	
iLBC_13_33kpbs	30ms, 60ms	

## Procedure

PTime can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure the ptime. <b>Parameter:</b> account.X.ptime
<b>Web User Interface</b>		Configure the ptime. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0">http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</a>

## Details of Configuration Parameter:

Parameter	Permitted Values	Default
<b>account.X.ptime<sup>[1]</sup></b>	<b>0, 10, 20, 30, 40, 50 or 60</b>	<b>20</b>
<b>Description:</b> Configures the ptime (in milliseconds) for the codec for a specific account. <b>0</b> -Disabled, the ptime negotiation is disabled. <b>10</b> -10 <b>20</b> -20 <b>30</b> -30		



Parameter	Permitted Values	Default
40-40		
50-50		
60-60		
<b>Example:</b>		
account.1.ptime = 20		
<b>Web User Interface:</b>		
Account->Advanced->PTime(ms)		
<b>Phone User Interface:</b>		
None		

To configure the ptime for the account via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **PTime(ms)**.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is selected, and the 'Advanced' sub-tab is active. The 'Account' dropdown menu shows 'Account 1'. The 'PTime(ms)' field is highlighted with a red box, indicating the current value is 20. Other visible fields include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'RTP Encryption(SRTP)' (Optional), 'BLF List URI' (4609@pbx.yealink.com), 'BLF List Pickup Code' (\*97), 'BLF List Barge In Code' (\*33), 'BLF List Retrieve Call Parked Code' (\*88), 'Shared Line' (Shared Call Appearance), 'Call Pull Feature Access Code' (\*11), 'Dialog Info Call Pickup' (Enabled), 'VQ RTPC-XR Collector Name' (collector), 'VQ RTPC-XR Collector Address' (10.2.1.98), and 'VQ RTPC-XR Collector Port' (5060). A 'NOTE' section on the right provides additional information about DTMF, Session Timer, and other features.

4. Click **Confirm** to accept the change.

## Acoustic Clarity Technology

### Acoustic Echo Cancellation (AEC)

Acoustic Echo Cancellation (AEC) is used to reduce acoustic echo from a voice call to provide natural full-duplex communication patterns. It also increases the capacity achieved through silence suppression by preventing echo from traveling across a network. IP phones employ advanced AEC for hands-free operation. AEC is not normally required for calls via the handset. In certain situation, where echo is experienced by the remote party, AEC may be used to reduce/avoid echo when the user uses the handset.

#### Note

Utilizing acoustic echo cancellation will introduce a small delay increase into audio path which might cause a lower voice quality.

#### Procedure

AEC can be configured using the following methods.

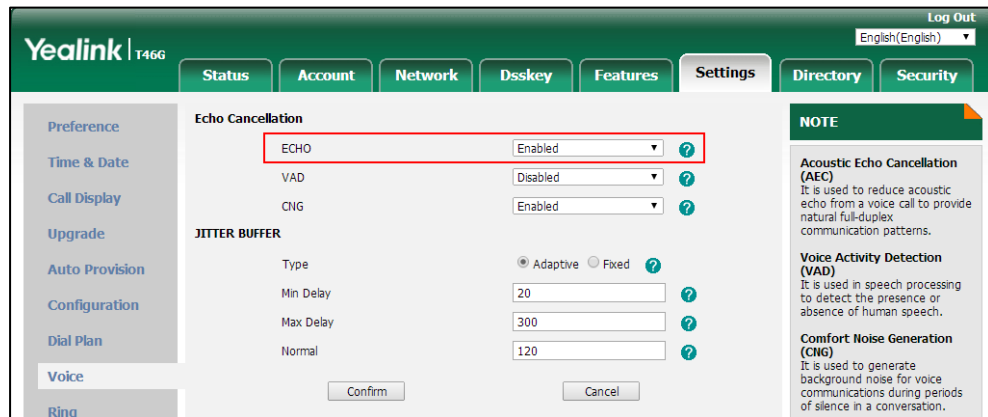
<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure AEC. <b>Parameter:</b> voice.echo_cancellation
<b>Web User Interface</b>		Configure AEC. <b>Navigate to:</b> http://<phoneIPAddress>/serv let?p=settings-voice&q=load

#### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
<b>voice.echo_cancellation</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the AEC (Acoustic Echo Canceller) feature on the IP phone. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Settings->Voice->Echo Cancellation->ECHO <b>Phone User Interface:</b> None		

**To configure AEC via web user interface:**

1. Click on **Settings->Voice**.
2. Select the desired value from the pull-down list of **ECHO**.



3. Click **Confirm** to accept the change.

## Background Noise Suppression (BNS)

Background noise suppression (BNS) is designed primarily for hands-free operation and reduces background noise to enhance communication in noisy environments.

## Automatic Gain Control (AGC)

Automatic Gain Control (AGC) is applicable to hands-free operation and is used to keep audio output at nearly a constant level by adjusting the gain of signals in certain circumstances. This increases the effective user-phone radius and helps with the intelligibility of soft-talkers.

## Voice Activity Detection (VAD)

Voice Activity Detection (VAD) is used in speech processing to detect the presence or absence of human speech. When detecting period of "silence", VAD replaces that silence efficiently with special packets that indicate silence is occurring. It can facilitate speech processing, and deactivate some processes during non-speech section of an audio session. VAD can avoid unnecessary coding or transmission of silence packets in VoIP applications, saving on computation and network bandwidth.

### Procedure

VAD can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure VAD. <b>Parameter:</b>
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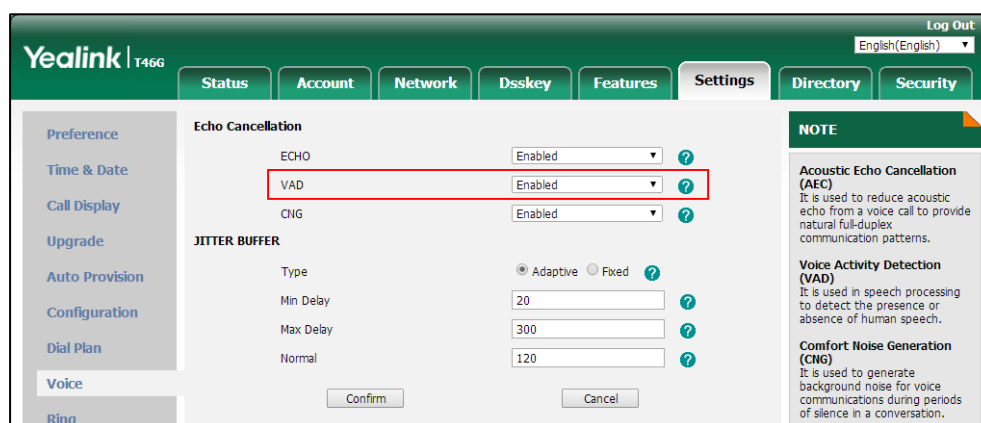
		voice.vad
<b>Web User Interface</b>		Configure VAD. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?parameters=settings-voice&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?parameters=settings-voice&amp;q=load</a>

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>voice.vad</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the VAD (Voice Activity Detection) feature on the IP phone. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Settings->Voice->Echo Cancellation->VAD <b>Phone User Interface:</b> None		

**To configure VAD via web user interface:**

1. Click on **Settings->Voice**.
2. Select the desired value from the pull-down list of **VAD**.



3. Click **Confirm** to accept the change.

## Comfort Noise Generation (CNG)

Comfort Noise Generation (CNG) is used to generate background noise for voice communications during periods of silence in a conversation. It is a part of the silence suppression or VAD handling for VoIP technology. CNG, in conjunction with VAD algorithms, quickly responds when periods of silence occur and inserts artificial noise until voice activity resumes. The insertion of artificial noise gives the illusion of a constant transmission stream, so that background sound is consistent throughout the call and the listener does not think the line has released. The purpose of VAD and CNG is to maintain an acceptable perceived QoS while simultaneously keeping transmission costs and bandwidth usage as low as possible.

### Note

VAD is used to send CN packets when phone detect a "silence" period; CNG is used to generate comfortable noise when phone receives CN packets from the other side.

For example, A is talking with B.

A: VAD=1, CNG=1

B: VAD=0, CNG=1

If A mutes the call, since VAD=1, A will send CN packets to B. When receiving CN packets, B will generate comfortable noise.

If B mutes the call, since VAD=0, B will not send CN packets to A. So even if CNG=1 (B), A will not hear comfortable noise.

### Procedure

CNG can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure CNG. <b>Parameter:</b> voice.cng
<b>Web User Interface</b>		Configure CNG. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=settings-voice&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet? p=settings-voice&amp;q=load</a>

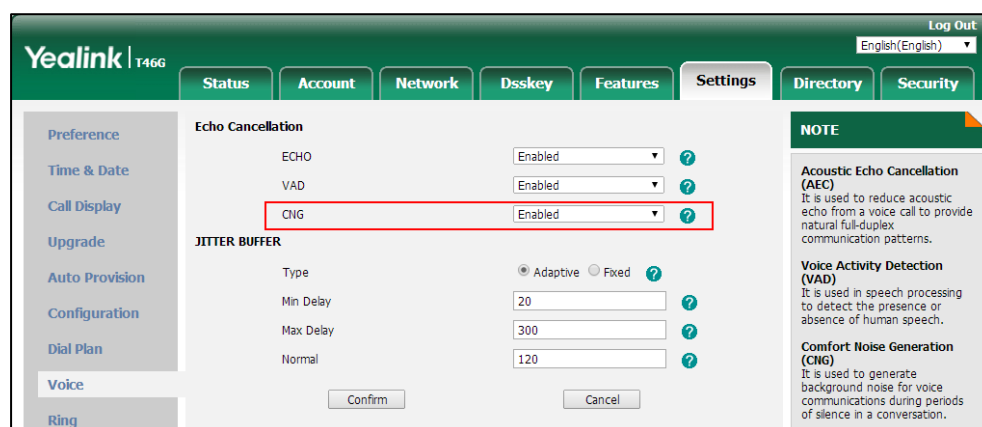
### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
voice.cng	0 or 1	1
<b>Description:</b> Enables or disables the CNG (Comfortable Noise Generation) feature on the IP phone.		

Parameter	Permitted Values	Default
<b>0-Disabled</b> <b>1-Enabled</b> <b>Web User Interface:</b> Settings->Voice->Echo Cancellation->CNG <b>Phone User Interface:</b> None		

To configure CNG via web user interface:

1. Click on **Settings->Voice**.
2. Select the desired value from the pull-down list of **CNG**.



3. Click **Confirm** to accept the change.

## Jitter Buffer

Jitter buffer is a shared data area where voice packets can be collected, stored, and sent to the voice processor in even intervals. Jitter is a term indicating variations in packet arrival time, which can occur because of network congestion, timing drift or route changes. The jitter buffer, located at the receiving end of the voice connection, intentionally delays the arriving packets so that the end user can experience a clear connection with very little sound distortion. IP phones support two types of jitter buffers: fixed and adaptive. A fixed jitter buffer adds the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on IP phones. An adaptive jitter buffer is capable of adapting the changes in the network's delay. The range of the delay time for the dynamic jitter buffer added to packets can be also configured on IP phones.

### Procedure

Jitter buffer can be configured using the following methods.

<b>Central</b>	<y0000000000xx>.cf	Configure the mode of jitter buffer and
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Provisioning  (Configuration File)	g	<p>the delay time for jitter buffer in the wired network.</p> <p><b>Parameters:</b></p> <p>voice.jib.adaptive</p> <p>voice.jib.min</p> <p>voice.jib.max</p> <p>voice.jib.normal</p>
		<p>Configure the mode of jitter buffer and the delay time for jitter buffer in the wireless network.</p> <p><b>Parameters:</b></p> <p>voice.jib.wifi.adaptive</p> <p>voice.iib.wifi.min</p> <p>voice.jib.wifi.max</p> <p>voice.jib.wifi.normal</p>
Web User Interface		<p>Configure the mode of jitter buffer and the delay time for jitter buffer in the wired network.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=settings-voice&amp;q=load</p>

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>voice.jib.adaptive</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Configures the type of jitter buffer in the wired network. <b>0</b> -Fixed <b>1</b> -Adaptive <b>Web User Interface:</b> Settings->Voice->JITTER BUFFER->Type <b>Phone User Interface:</b> None		
<b>voice.jib.min</b>	<b>Integer from 0 to 400</b>	<b>60</b>

Parameters	Permitted Values	Default
<b>Description:</b> Configures the minimum delay time (in milliseconds) of jitter buffer in the wired network. <b>Note:</b> It works only if the value of the parameter "voice.jib.adaptive" is set to 1 (Adaptive). The value of this parameter should be less than the values of the parameters "voice.jib.max" and "voice.jib.normal". <b>Web User Interface:</b> Settings->Voice->JITTER BUFFER->Min Delay <b>Phone User Interface:</b> None		
<b>voice.jib.max</b>	<b>Integer from 0 to 400</b>	<b>240</b>
<b>Description:</b> Configures the maximum delay time (in milliseconds) of jitter buffer in the wired network. <b>Note:</b> It works only if the value of the parameter "voice.jib.adaptive" is set to 1 (Adaptive). The value of this parameter should be greater than the values of the parameters "voice.jib.normal" and "voice.jib.min". <b>Web User Interface:</b> Settings->Voice->JITTER BUFFER->Max Delay <b>Phone User Interface:</b> None		
<b>voice.jib.normal</b>	<b>Integer from 0 to 400</b>	<b>120</b>
<b>Description:</b> Configures the normal delay time (in milliseconds) of jitter buffer in the wired network. <b>Note:</b> It works only if the value of the parameter "voice.jib.adaptive" is set to 0 (Fixed). The value of this parameter should be greater than the value of the parameter "voice.jib.min" and less than the value of the parameter "voice.jib.max". <b>Web User Interface:</b> Settings->Voice->JITTER BUFFER->Normal <b>Phone User Interface:</b> None		
<b>voice.jib.wifi.adaptive</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Configures the type of jitter buffer in the wireless network.		



Parameters	Permitted Values	Default
<b>0</b> -Fixed <b>1</b> -Adaptive <b>Note:</b> It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>voice.jib.wifi.min</b>	<b>Integer from 0 to 500</b>	<b>60</b>
<b>Description:</b> Configures the minimum delay time (in milliseconds) of jitter buffer in the wireless network. <b>Note:</b> It works only if the value of the parameter "voice.jib.wifi.adaptive" is set to 1 (Adaptive). The value of the minimum delay time should be less than or equal to that of the normal delay time (configured by the parameter "voice.jib.wifi.normal"). It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones. The value of this parameter should be less than the values of the parameters "voice.jib.wifi.normal" and "voice.jib.wifi.max". <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>voice.jib.wifi.max</b>	<b>Integer from 0 to 500</b>	<b>500</b>
<b>Description:</b> Configures the maximum delay time (in milliseconds) of jitter buffer in the wireless network. <b>Note:</b> It works only if the value of the parameter "voice.jib.wifi.adaptive" is set to 1 (Adaptive). It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones. The value of this parameter should be greater than the values of the parameters "voice.jib.wifi.normal" and "voice.jib.wifi.min". <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>voice.jib.wifi.normal</b>	<b>Integer from 0 to 500</b>	<b>240</b>
<p><b>Description:</b></p> <p>Configures the normal delay time (in milliseconds) of jitter buffer in the wireless network.</p> <p><b>Note:</b> It works only if the value of the parameter "voice.jib.wifi.adaptive" is set to 0 (Fixed). The value of the normal delay time should be less than or equal to that of the maximum delay time (configured by the parameter "voice.jib.wifi.max"). It is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42S/T41S/T29G/T27G IP phones. The value of this parameter should be greater than the value of the parameter "voice.jib.wifi.min" and less than the value of the parameter "voice.jib.wifi.min".</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure Jitter Buffer in the wired network via web user interface:**

1. Click on **Settings->Voice**.
2. Mark the desired radio box in the **Type** field.
3. Enter the minimum delay time (20-300) for adaptive jitter buffer in the **Min Delay** field.
4. Enter the maximum delay time (20-300) for adaptive jitter buffer in the **Max Delay** field.
5. Enter the fixed delay time (20-300) for fixed jitter buffer in the **Normal** field.

The screenshot shows the Yealink T46G web interface. The 'Settings' tab is selected, and the 'Voice' section is expanded. The 'JITTER BUFFER' section is highlighted with a red box. It contains the following fields:

- Type:** Radio buttons for 'Adaptive' (selected) and 'Fixed'.
- Min Delay:** Input field with value 20.
- Max Delay:** Input field with value 300.
- Normal:** Input field with value 120.

Below the input fields are 'Confirm' and 'Cancel' buttons. To the right, there is a 'NOTE' section with information about Acoustic Echo Cancellation (AEC), Voice Activity Detection (VAD), and Comfort Noise Generation (CNG).

6. Click **Confirm** to accept the change.

## DTMF

DTMF (Dual Tone Multi-frequency), better known as touch-tone, is used for telecommunication signaling over analog telephone lines in the voice-frequency band. DTMF is the signal sent from

the IP phone to the network, which is generated when pressing the IP phone's keypad during a call. Each key pressed on the IP phone generates one sinusoidal tone of two frequencies. One is generated from a high frequency group and the other from a low frequency group.

The DTMF keypad is laid out in a 4×4 matrix, with each row representing a low frequency, and each column representing a high frequency. Pressing a digit key (such as '1') will generate a sinusoidal tone for each of two frequencies (697 and 1209 hertz (Hz)).

#### DTMF Keypad Frequencies:

	1209 Hz	1336 Hz	1477 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

#### Note

The IP phones will not send DTMF sequence when the call is placed on hold or is held,

## Methods of Transmitting DTMF Digit

Three methods of transmitting DTMF digits on SIP calls:

- **RFC 2833** -- DTMF digits are transmitted by RTP Events compliant with RFC 2833.
- **INBAND** -- DTMF digits are transmitted in the voice band.
- **SIP INFO** -- DTMF digits are transmitted by SIP INFO messages.

The method of transmitting DTMF digits is configurable on a per-line basis.

#### RFC 2833

DTMF digits are transmitted using the RTP Event packets that are sent along with the voice path. These packets use RFC 2833 format and must have a payload type that matches what the other end is listening for. The default payload type for RTP Event packets is 101 and the payload type is configurable. The IP phones use the configured value to negotiate with the other end during call establishment.

The RTP Event packet contains 4 bytes. The 4 bytes are distributed over several fields denoted as Event, End bit, R-bit, Volume and Duration. If the End bit is set to 1, the packet contains the end of the DTMF event. You can configure the sending times of the end RTP Event packet.

#### INBAND

DTMF digits are transmitted within the audio of the IP phone conversation. It uses the same codec as your voice and is audible to conversation partners.

## SIP INFO

DTMF digits are transmitted by the SIP INFO messages when the voice stream is established after a successful SIP 200 OK-ACK message sequence. The SIP INFO message is sent along the signaling path of the call. The SIP INFO message can transmit DTMF digits in three ways: DTMF, DTMF-Relay and Telephone-Event.

## Procedure

Configuration changes can be performed using the following methods.

Central Provisioning (Configuration Files)	<MAC>.cfg	Configure the method of transmitting DTMF digit and the payload type. <b>Parameters:</b> account.X.dtmf.type account.X.dtmf.dtmf_payload account.X.dtmf.info_type
	<y0000000000xx>.cfg	Configure the number of times for the IP phone to send the end RTP Event packet. <b>Parameter:</b> features.dtmf.repetition
		Specify how long the phone should play each DTMF tone for. <b>Parameter:</b> features.dtmf.duration
		Configure the frequency level of DTMF digits. <b>Parameter:</b> features.dtmf.volume
		Specify how long the phone should wait between DTMF digits. <b>Parameter:</b> features.dtmf.min_interval
Web User Interface		Configure the method of transmitting DTMF digits and the payload type. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

	<p>Configure the number of times for the IP phone to send the end RTP Event packet.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=feature s-general&amp;q=load</p>
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### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>account.X.dtmf.type<sup>[1]</sup></b>	<b>0, 1, 2 or 3</b>	<b>1</b>
<p><b>Description:</b></p> <p>Configures the DTMF type for a specific account.</p> <p><b>0</b>-INBAND, DTMF digits are transmitted in the voice band.</p> <p><b>1</b>-RFC 2833, DTMF digits are transmitted by RTP Events compliant to RFC 2833.</p> <p><b>2</b>-SIP INFO, DTMF digits are transmitted by the SIP INFO messages.</p> <p><b>3</b>-RFC2833 + SIP INFO, DTMF digits are transmitted by RTP Events compliant to RFC 2833 and the SIP INFO messages.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;DTMF Type</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.dtmf.dtmf_payload<sup>[1]</sup></b>	<b>Integer from 96 to 127</b>	<b>101</b>
<p><b>Description:</b></p> <p>Configures the value of DTMF payload for a specific account.</p> <p><b>Note:</b> It works only if the value of the parameter "account.X.dtmf.type" is set to 1 (RFC2833) or 3 (RFC2833 + SIP INFO).</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;DTMF Payload Type(96~127)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>account.X.dtmf.info_type<sup>[1]</sup></b>	<b>1, 2 or 3</b>	<b>1</b>
<p><b>Description:</b></p> <p>Configures the DTMF info type.</p> <p><b>1</b>-DTMF-Relay</p>		

Parameters	Permitted Values	Default
<b>2-DTMF</b> <b>3-Telephone-Event</b> <b>Note:</b> It works only if the value of the parameter "account.X.dtmf.type" is set to 2 (SIP INFO) or 3 (RFC2833 + SIP INFO). <b>Web User Interface:</b> Account->Advanced->DTMF Info Type <b>Phone User Interface:</b> None		
<b>features.dtmf.repetition</b>	<b>1, 2 or 3</b>	<b>3</b>
<b>Description:</b> Configures the repetition times for the IP phone to send the end RTP Event packet during an active call. <b>Web User Interface:</b> Features->General Information->DTMF Repetition <b>Phone User Interface:</b> None		
<b>features.dtmf.duration</b>	<b>Integer from 0 to 300</b>	<b>100</b>
<b>Description:</b> Configures the duration time (in milliseconds) for each digit when a sequence of DTMF tones is played out automatically. <b>Note:</b> If the time interval between two DTMF digits is less than this value, two or more same DTMF digits could be identified as one DTMF digit. This may cause the loss of one or more DTMF digits. For example, 2662 may be identified as 262. If so, you can modify the value of this parameter to a little lower than the default value. If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.dtmf.volume</b>	<b>Integer from -33 to 0</b>	<b>-10</b>
<b>Description:</b> Configures the volume of the DTMF tone (in db).		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>features.dtmf.min_interval</b>	<b>Integer from 0 to 300</b>	<b>50</b>
<b>Description:</b> Configures the minimum inter-digit time (in milliseconds) between digits when a sequence of DTMF tones is played out automatically. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure the method of transmitting DTMF digits via web user interface:**

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **DTMF Type**.  
If **SIP INFO** or **RFC2833 + SIP INFO** is selected, select the desired value from the pull-down list of **DTMF Info Type**.
4. Enter the desired value in the **DTMF Payload Type(96~127)** field.

The screenshot shows the Yealink T46G web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Account' dropdown is set to 'Account1'. The 'DTMF Type' is set to 'RFC2833', 'DTMF Info Type' is 'DTMF-Relay', and 'DTMF Payload Type(96~127)' is '101'. A red box highlights these three settings. A 'NOTE' section on the right explains DTMF and Session Timer.

5. Click **Confirm** to accept the change.

To configure the number of times to send the end RTP Event packet via web user interface:

1. Click on **Features->General Information**.
2. Select the desired value (1-3) from the pull-down list of **DTMF Repetition**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'DTMF Repetition' dropdown menu is highlighted with a red box and set to '3'. Other settings visible include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Call Waiting Off Code' (empty), 'Auto Redial' (Disabled), 'Auto Redial Interval (1~300s)' (10), 'Auto Redial Times (1~300)' (10), 'Multicast Codec' (G722), 'Play Hold Tone' (Enabled), 'Play Hold Tone Delay' (30), 'Allow Mute' (Enabled), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right provides details about 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion'.

3. Click **Confirm** to accept the change.

## Suppress DTMF Display

Suppress DTMF display allows IP phones to suppress the display of DTMF digits during an active call. DTMF digits are displayed as "\*" on the LCD screen. Suppress DTMF display delay defines whether to display the DTMF digits for a short period of time before displaying as "\*".

### Procedure

Configuration changes can be performed using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure suppress DTMF display and suppress DTMF display delay. <b>Parameters:</b> features.dtmf.hide features.dtmf.hide_delay
<b>Web User Interface</b>		Configure suppress DTMF display and suppress DTMF display delay. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=f eatures-general&q=load



**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>features.dtmf.hide</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the IP phone to suppress the display of DTMF digits during an active call.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled, the DTMF digits are displayed as asterisks.</p> <p><b>Web User Interface:</b> Features-&gt;General Information-&gt;Suppress DTMF Display</p> <p><b>Phone User Interface:</b> None</p>		
<b>features.dtmf.hide_delay</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b> Enables or disables the IP phone to display the DTMF digits for a short period before displaying asterisks during an active call.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "features.dtmf.hide" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Features-&gt;General Information-&gt;Suppress DTMF Display Delay</p> <p><b>Phone User Interface:</b> None</p>		

**To configure suppress DTMF display and suppress DTMF display delay via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Suppress DTMF Display**.

3. Select the desired value from the pull-down list of **Suppress DTMF Display Delay**.

The screenshot shows the Yealink T46G web interface. The 'Features' tab is selected, and the 'General Information' section is expanded. The 'Suppress DTMF Display Delay' dropdown menu is highlighted with a red box, showing 'Enabled' as the selected value. Other settings visible include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Call Waiting Off Code' (empty), 'Auto Redial' (Disabled), 'Play Local DTMF Tone' (Enabled), 'DTMF Repetition' (3), 'Multicast Codec' (G722), 'Play Hold Tone' (Enabled), 'Hide Feature Access Codes' (Enabled), 'Display Method on Dialing' (User Name), and 'Auto Linekeys' (Enabled). A 'NOTE' section on the right provides details for 'Call Waiting', 'Auto Redial', 'Key As Send', 'Hotline', and 'Call Completion'.

4. Click **Confirm** to accept the change.

## Transfer via DTMF

Call transfer is implemented via DTMF on some traditional servers. The IP phone sends specified DTMF digits to the server for transferring calls to third parties.

### Procedure

Configuration changes can be performed using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure transfer via DTMF. <b>Parameters:</b> features.dtmf.replace_tran features.dtmf.transfer
<b>Web User Interface</b>		Configure transfer via DTMF. <b>Navigate to:</b> http://<phoneIPAddress>/servlet? p=features-general&q=load

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>features.dtmf.replace_tran</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to send DTMF sequences for transfer function when pressing the <b>Trans/Transfer</b> soft key or TRAN/TRANSFER key.</p> <p><b>0</b>-Disabled, the IP phone will perform the transfer as normal when pressing the <b>Trans/Transfer</b> soft key or TRAN/TRANSFER key during a call.</p> <p><b>1</b>-Enabled, the IP phone will transmit the designated DTMF digits to the server for performing call transfer when pressing the <b>Trans/Transfer</b> soft key or TRAN/TRANSFER key during a call.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;DTMF Replace Tran</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>features.dtmf.transfer</b>	<b>String within 32 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the DTMF digits to be transmitted to perform call transfer.</p> <p>Valid values are: 0-9, *, #, A-D and E.</p> <p><b>Example:</b></p> <p>features.dtmf.transfer = 123</p> <p><b>Note:</b> It works only if the value of the parameter "features.dtmf.replace_tran" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Tran Send DTMF</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure transfer via DTMF via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **DTMF Replace Tran**.
3. Enter the specified DTMF digits in the **Tran Send DTMF** field.

The screenshot shows the Yealink T466 web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. In the 'General Information' section, the 'DTMF Replace Tran' dropdown menu is set to 'Enabled', and the 'Tran Send DTMF' text field contains the value '123'. These two elements are highlighted with a red rectangular box. Other settings visible include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Call Waiting Off Code' (empty), 'Auto Redial' (Disabled), 'Headset Prior' (Disabled), 'Send Pound Key' (Disabled), 'Fwd International' (Enabled), 'Play Hold Tone' (Enabled), 'Hide Feature Access Codes' (Enabled), 'Display Method on Dialing' (User Name), and 'Auto Linekeys' (Enabled). At the bottom of the form are 'Confirm' and 'Cancel' buttons. On the right side, there is a 'NOTE' section with several call-related features and their descriptions.

4. Click **Confirm** to accept the change.

## Play Local DTMF Tone

Play local DTMF tone allows IP phones to play a local DTMF tone during an active call. If this feature is enabled, you can hear the DTMF tone when pressing the IP phone's keypad during a call.

### Procedure

Configuration changes can be performed using the following methods.

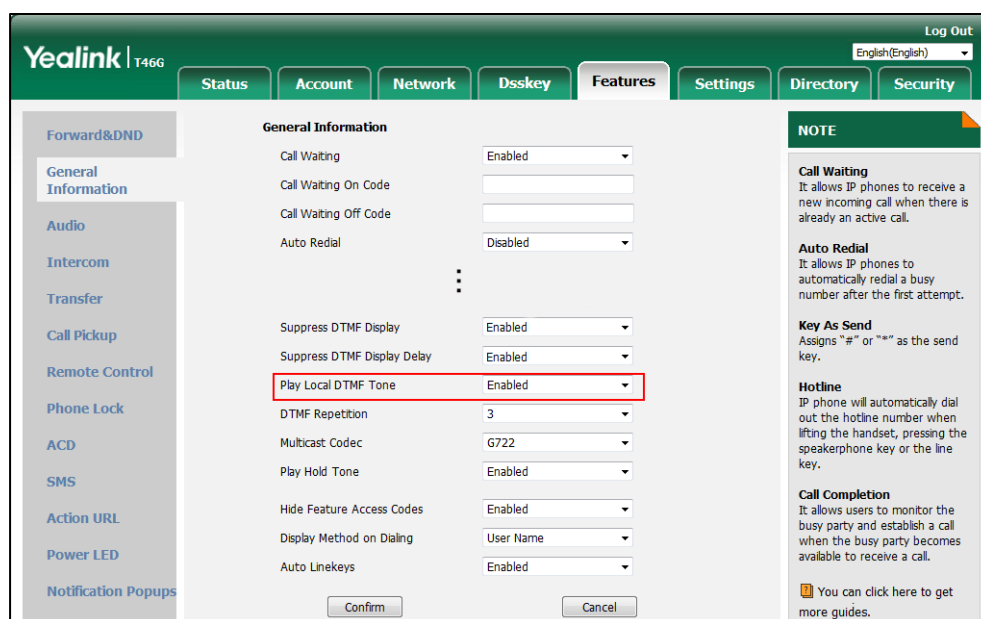
<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure play local DTMF tone. <b>Parameter:</b> features.play_local_dtmf_tone_enable
<b>Web User Interface</b>		Configure play local DTMF tone. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=features-general&q=load

**Details of Configuration Parameter:**

Parameter	Permitted Values	Default
<b>features.play_local_dtmf_tone_enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to play a local DTMF tone.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, you can hear the DTMF tone when pressing the IP phone's keypad during a call.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Play Local DTMF Tone</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure play local DTMF tone via web user interface:**

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Play Local DTMF Tone**.



3. Click **Confirm** to accept the change.

## Voice Quality Monitoring (VQM)

Voice quality monitoring feature allows the IP phones to generate various quality metrics for listening quality and conversational quality. These metrics can be sent between the phones in RTCP-XR packets. These metrics can also be sent in SIP PUBLISH messages to a central voice

quality report collector. Two mechanisms for voice quality monitoring are supported by Yealink IP phones:

- RTCP-XR
- VQ-RTCPXR

## RTCP-XR

The RTCP-XR mechanism, compliant with [RFC 3611-RTP Control Extended Reports \(RTCP XR\)](#), provides the metrics contained in RTCP-XR packets for monitoring the quality of calls. These metrics include network packet loss, delay metrics, analog metrics and voice quality metrics.

### Procedure

RTCP-XR can be configured using the following methods.

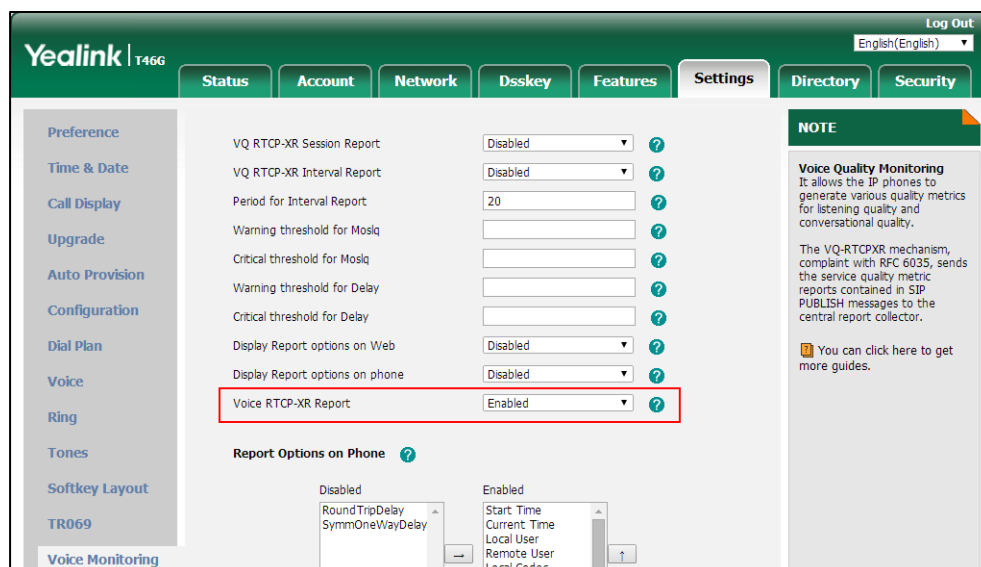
<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure RTCP-XR. <b>Parameter:</b> voice.rtcp_xr.enable
<b>Web User Interface</b>		Configure RTCP-XR. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=settings-voicemonitoring&q=load

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>voice.rtcp_xr.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to send RTCP-XR packets. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect. <b>Web User Interface:</b> Settings->Voice Monitoring->Voice RTCP-XR Report <b>Phone User Interface:</b> None		

To configure RTCP-XR feature via web user interface:

1. Click on **Settings->Voice Monitoring**.
2. Select the desired value from the pull-down list of **Voice RTCP-XR Report**.



3. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
5. Click **OK** to reboot the phone.

## VQ-RTCPXR

The VQ-RTCPXR mechanism, compliant with [RFC 6035](#), sends the service quality metric reports contained in SIP PUBLISH messages to the central report collector. Three types of quality reports can be enabled:

- **Session:** Generated at the end of a call.
- **Interval:** Generated during a call at a configurable period.
- **Alert:** Generated when the call quality degrades below a configurable threshold.

A wide range of performance metrics are generated in the following three ways:

- Based on current values, such as jitter, jitter buffer max and round trip delay.
- Covers the time period from the beginning of the call until the report is sent, such as network packet loss.
- Computed using other metrics as input, such as listening Mean Opinion Score (MOS-LQ) and conversational Mean Opinion Score (MOS-CQ).

To operate with central report collector, IP phones must be configured to forward their voice quality reports to the specified report collector. You can specify the report collector on a per-line basis.

Users can check the voice quality data of the last call via web user interface or phone user interface. Users can also specify the options of the RTP status to be displayed on the phone user interface. Options of the RTP status displayed on the web user interface cannot be specified.

## Procedure

VQ-RTCPXR can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the generation of session packets. <b>Parameter:</b> phone_setting.vq_rtcpxr.session_report.enable
		Configure the generation of interval packets. <b>Parameters:</b> phone_setting.vq_rtcpxr.interval_report.enable phone_setting.vq_rtcpxr_interval_period
		Configure the generation of alert packets. <b>Parameters:</b> phone_setting.vq_rtcpxr_moslq_threshold_warning phone_setting.vq_rtcpxr_moslq_threshold_critical phone_setting.vq_rtcpxr_delay_threshold_warning phone_setting.vq_rtcpxr_delay_threshold_critical
		Configure the phone to display RTP status showing the voice quality report of the last call on the web user interface. <b>Parameter:</b> phone_setting.vq_rtcpxr.states_show_on_web.enable
		Configure the phone to display RTP status showing the voice quality report of the last call or the current call on the phone user interface. <b>Parameter:</b> phone_setting.vq_rtcpxr.states_show_on_gui.enable



		<p>Configure the options of the RTP status displayed on the phone user interface.</p> <p><b>Parameters:</b></p> <p>phone_setting.vq_rtcp_xr_display_start_time.enable</p> <p>phone_setting.vq_rtcp_xr_display_stop_time.enable</p> <p>phone_setting.vq_rtcp_xr_display_local_call_id.enable</p> <p>phone_setting.vq_rtcp_xr_display_remote_call_id.enable</p> <p>phone_setting.vq_rtcp_xr_display_local_codec.enable</p> <p>phone_setting.vq_rtcp_xr_display_remote_codec.enable</p> <p>phone_setting.vq_rtcp_xr_display_jitter.enable</p> <p>phone_setting.vq_rtcp_xr_display_jitter_buffer_max.enable</p> <p>phone_setting.vq_rtcp_xr_display_packets_lost.enable</p> <p>phone_setting.vq_rtcp_xr_display_symm_oneway_delay.enable</p> <p>phone_setting.vq_rtcp_xr_display_round_trip_delay.enable</p> <p>phone_setting.vq_rtcp_xr_display_mos_lq.enable</p> <p>phone_setting.vq_rtcp_xr_display_mos_cq.enable</p>
	<MAC>.cfg	<p>Configure the central report collector.</p> <p><b>Parameters:</b></p> <p>account.X.vq_rtcp_xr_collector_name</p> <p>account.X.vq_rtcp_xr_collector_server_host</p> <p>account.X.vq_rtcp_xr_collector_server_port</p>
<b>Web User Interface</b>		<p>Configure VQ-RTCPXR.</p> <p>Configure the phone to display RTP status showing the voice quality report of the last call on the web user interface.</p> <p>Configure the phone to display RTP status showing the voice quality report of the last call or the current call on the phone user interface.</p> <p>Configure the options of the RTP status displayed on the phone user interface.</p> <p><b>Navigate to:</b></p> <p><a href="http://&lt;phoneIPAddress&gt;/servlet?p=settings-voice">http://&lt;phoneIPAddress&gt;/servlet?p=settings-voice</a></p>

	monitoring&q=load
	<p>Configure the central report collector.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</p>

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>phone_setting.vq_rtcpxr.session_report.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to send a session quality report to the central report collector at the end of each call.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Voice Monitoring-&gt;VQ RTCP-XR Session Report</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>phone_setting.vq_rtcpxr.interval_report.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to send an interval quality report to the central report collector periodically throughout a call.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> To avoid overload, the interval quality reports only generate when the call is abnormal.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Voice Monitoring-&gt;VQ RTCP-XR Interval Report</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>phone_setting.vq_rtcpxr_interval_period</b>	<b>Integer from 5 to 20</b>	<b>20</b>
<p><b>Description:</b></p>		

Parameters	Permitted Values	Default
<p>Configures the interval (in seconds) for the IP phone to send an interval quality report to the central report collector periodically throughout a call.</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_xr.interval_report.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Settings-&gt;Voice Monitoring-&gt;Period for Interval Report</p> <p><b>Phone User Interface:</b> None</p>		
<b>phone_setting.vq_rtcp_xr_moslq_threshold_warning</b>	<b>15 to 40</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the threshold value of listening MOS score (MOS-LQ) multiplied by 10. The threshold value of MOS-LQ causes the phone to send a warning alert quality report to the central report collector.</p> <p>For example, a configured value of 35 corresponds to the MOS score 3.5. When the MOS-LQ value computed by the phone is less than or equal to 3.5, the phone will send a warning alert quality report to the central report collector. When the MOS-LQ value computed by the phone is greater than 3.5, the phone will not send a warning alert quality report to the central report collector.</p> <p>If it is set to blank, warning alerts are not generated due to MOS-LQ.</p> <p><b>Web User Interface:</b> Settings-&gt;Voice Monitoring-&gt;Warning threshold for Moslq</p> <p><b>Phone User Interface:</b> None</p>		
<b>phone_setting.vq_rtcp_xr_moslq_threshold_critical</b>	<b>15 to 40</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the threshold value of listening MOS score (MOS-LQ) multiplied by 10. The threshold value of MOS-LQ causes the phone to send a critical alert quality report to the central report collector.</p> <p>For example, a configured value of 28 corresponds to the MOS score 2.8. When the MOS-LQ value computed by the phone is less than or equal to 2.8, the phone will send a critical alert quality report to the central report collector. When the MOS-LQ value computed by the phone is greater than 2.8, the phone will not send a critical alert quality report to the central report collector.</p> <p>If it is set to blank, critical alerts are not generated due to MOS-LQ.</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
Settings->Voice Monitoring->Critical threshold for Moslq <b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_xr_delay_threshold_warning</b>	<b>10 to 2000</b>	<b>Blank</b>
<b>Description:</b> Configures the threshold value of one way delay (in milliseconds) that causes the phone to send a warning alert quality report to the central report collector. For example, If it is set to 500, when the value of one way delay computed by the phone is greater than or equal to 500, the phone will send a warning alert quality report to the central report collector; when the value of one way delay computed by the phone is less than 500, the phone will not send a warning alert quality report to the central report collector. If it is set to blank, warning alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay. <b>Web User Interface:</b> Settings->Voice Monitoring->Warning threshold for Delay <b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_xr_delay_threshold_critical</b>	<b>10 to 2000</b>	<b>Blank</b>
<b>Description:</b> Configures the threshold value of one way delay (in milliseconds) that causes phone to send a critical alert quality report to the central report collector. For example, If it is set to 500, when the value of one way delay computed by the phone is greater than or equal to 500, the phone will send a critical alert quality report to the central report collector; when the value of one way delay computed by the phone is less than 500, the phone will not send a critical alert quality report to the central report collector. If it is set to blank, critical alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay. <b>Web User Interface:</b> Settings->Voice Monitoring->Critical threshold for Delay <b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_xr.states_show_on_web.enable</b>	<b>0 or 1</b>	<b>0</b>

Parameters	Permitted Values	Default
<b>Description:</b> Enables or disables the voice quality data of the last call to be displayed on web interface at path <b>Status-&gt;RTP Status</b> . <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Settings->Voice Monitoring->Display Report options on Web <b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_xr.states_show_on_gui.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the voice quality data of the last call or current call to be displayed on the LCD screen. You can view the voice quality data of the last call on the phone at the path <b>Menu-&gt;Status-&gt;More-&gt;RTP (RTP Status)</b> . You can view the voice quality data of the current call by pressing <b>RTP/RTP Status</b> soft key during a call. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Settings->Voice Monitoring->Display Report options on phone <b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_xr.display_start_time.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the phone to display Start Time on the LCD screen. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_xr.states_show_on_gui.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Settings->Voice Monitoring->Report options on phone->Start Time <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>phone_setting.vq_rtcp_r_display_stop_time.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the phone to display Current Time or Stop Time on the LCD screen.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_r_states_show_on_gui.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Settings-&gt;Voice Monitoring-&gt;Report options on phone-&gt;Current Time</p> <p><b>Phone User Interface:</b> None</p>		
<b>phone_setting.vq_rtcp_r_display_local_call_id.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the phone to display Local User on the LCD screen.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_r_states_show_on_gui.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Settings-&gt;Voice Monitoring-&gt;Report options on phone-&gt;Local User</p> <p><b>Phone User Interface:</b> None</p>		
<b>phone_setting.vq_rtcp_r_display_remote_call_id.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b> Enables or disables the phone to display Remote User on the LCD screen.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_r_states_show_on_gui.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b> Settings-&gt;Voice Monitoring-&gt;Report options on phone-&gt;Remote User</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
None		
<b>phone_setting.vq_rtcp_xr_display_local_codec.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the phone to display Local Codec on the LCD screen.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Voice Monitoring-&gt;Report options on phone-&gt;Local Codec</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>phone_setting.vq_rtcp_xr_display_remote_codec.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the phone to display Remote Codec on the LCD screen.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Voice Monitoring-&gt;Report options on phone-&gt;Remote Codec</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>phone_setting.vq_rtcp_xr_display_jitter.enable</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the phone to display Jitter on the LCD screen.</p> <p><b>0</b>-Disabled <b>1</b>-Enabled</p> <p><b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Voice Monitoring-&gt;Report options on phone-&gt;Jitter</p>		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_r_display_jitter_buffer_max.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the phone to display Jitter Buffer Max on the LCD screen. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_r_states_show_on_gui.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Settings->Voice Monitoring->Report options on phone->Jitter Buffer Max <b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_r_display_packets_lost.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the phone to display Packets Lost on the LCD screen. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_r_states_show_on_gui.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Settings->Voice Monitoring->Report options on phone->Packets Lost <b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_r_display_symm_oneway_delay.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the phone to display SymmOneWayDelay on the LCD screen. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_r_states_show_on_gui.enable" is set to 1 (Enabled). <b>Web User Interface:</b>		



Parameters	Permitted Values	Default
Settings->Voice Monitoring->Report options on phone->SymmOneWayDelay <b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_xr_display_round_trip_delay.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the phone to display RoundTripDelay on the LCD screen. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Settings->Voice Monitoring->Report options on phone->RoundTripDelay <b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_xr_display_mos_lq.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the phone to display MOS-LQ on the LCD screen. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to 1 (Enabled). <b>Web User Interface:</b> Settings->Voice Monitoring->Report options on phone->MOS-LQ <b>Phone User Interface:</b> None		
<b>phone_setting.vq_rtcp_xr_display_mos_cq.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the phone to display MOS-CQ on the LCD screen. <b>0</b> -Disabled <b>1</b> -Enabled <b>Note:</b> It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to 1 (Enabled).		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> Settings->Voice Monitoring->Report options on phone->MOS-CQ <b>Phone User Interface:</b> None		
<b>account.X.vq_rtcpxr.collector_name</b> <sup>[1]</sup>	String within 32 characters	Blank
<b>Description:</b> Configures the host name of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages for a specific account. <b>Web User Interface:</b> Account->Advanced->VQ RTCP-XR Collector Name <b>Phone User Interface:</b> None		
<b>account.X.vq_rtcpxr.collector_server_host</b> <sup>[1]</sup>	IPv4 Address	Blank
<b>Description:</b> Configures the IP address of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages for a specific account. <b>Web User Interface:</b> Account->Advanced->VQ RTCP-XR Collector Address <b>Phone User Interface:</b> None		
<b>account.X.vq_rtcpxr.collector_server_port</b> <sup>[1]</sup>	Integer from 1 to 65535	5060
<b>Description:</b> Configures the port of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages for a specific account. <b>Web User Interface:</b> Account->Advanced->VQ RTCP-XR Collector Port <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12;

for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

**To configure session report for VQ-RTCPXR via web user interface:**

1. Click on **Settings->Voice Monitoring**.
2. Select the desired value from the pull-down list of **VQ RTCP-XR Session Report**.

The screenshot shows the Yealink T466 web interface. The 'Settings' tab is selected, and the 'Voice Monitoring' sub-tab is active. In the 'VQ RTCP-XR Session Report' dropdown, 'Enabled' is selected and highlighted with a red box. Other settings include 'VQ RTCP-XR Interval Report' (Enabled), 'Period for Interval Report' (20), 'Warning threshold for Mosq' (empty), 'Critical threshold for Mosq' (empty), 'Warning threshold for Delay' (empty), 'Critical threshold for Delay' (empty), 'Display Report options on Web' (Enabled), 'Display Report options on phone' (Enabled), and 'Voice RTCP-XR Report' (Enabled). A 'Report Options on Phone' section is also visible with 'Disabled' and 'Enabled' options.

3. Click **Confirm** to accept the change.

**To configure interval report for VQ-RTCPXR via web user interface:**

1. Click on **Settings->Voice Monitoring**.
2. Select the desired value from the pull-down list of **VQ RTCP-XR Interval Report**.
3. Enter the desired value in the **Period for Interval Report** field.

The screenshot shows the Yealink T466 web interface. The 'Settings' tab is selected, and the 'Voice Monitoring' sub-tab is active. In the 'VQ RTCP-XR Interval Report' dropdown, 'Enabled' is selected and highlighted with a red box. The 'Period for Interval Report' field is set to 20. Other settings are the same as in the previous screenshot.

4. Click **Confirm** to accept the change.

**To configure alert report for VQ-RTCPXR via web user interface:**

1. Click on **Settings->Voice Monitoring**.
2. Enter the desired value in the **Warning threshold for Moslq** field.
3. Enter the desired value in the **Critical threshold for Moslq** field.
4. Enter the desired value in the **Warning threshold for Delay** field.
5. Enter the desired value in the **Critical threshold for Delay** field.

**Yealink | T46G** Log Out English(English)

**Settings** **Directory** **Security**

**Preference** **Time & Date** **Call Display** **Upgrade** **Auto Provision** **Configuration** **Dial Plan** **Voice** **Ring** **Tones** **Softkey Layout** **TR069** **Voice Monitoring**

VQ RTCP-XR Session Report  ?

VQ RTCP-XR Interval Report  ?

Period for Interval Report  ?

Warning threshold for Moslq  ?

Critical threshold for Moslq  ?

Warning threshold for Delay  ?

Critical threshold for Delay  ?

Display Report options on Web  ?

Display Report options on phone  ?

Voice RTCP-XR Report  ?

**Report Options on Phone** ?

Disabled: RoundTripDelay, SymmOneWayDelay

Enabled: Start Time, Current Time, Local User, Remote User, Local Codec

**NOTE**

**Voice Quality Monitoring**  
It allows the IP phones to generate various quality metrics for listening quality and conversational quality.

The VQ-RTCPXR mechanism, compliant with RFC 6035, sends the service quality metric reports contained in SIP PUBLISH messages to the central report collector.

You can click here to get more guides.

6. Click **Confirm** to accept the change.

**To configure RTP status displayed on the web page via web user interface:**

1. Click on **Settings->Voice Monitoring**.
2. Select the desired value from the pull-down list of **Display Report options on Web**.

**Yealink | T46G** Log Out English(English)

**Settings** **Directory** **Security**

**Preference** **Time & Date** **Call Display** **Upgrade** **Auto Provision** **Configuration** **Dial Plan** **Voice** **Ring** **Tones** **Softkey Layout** **TR069** **Voice Monitoring**

VQ RTCP-XR Session Report  ?

VQ RTCP-XR Interval Report  ?

Period for Interval Report  ?

Warning threshold for Moslq  ?

Critical threshold for Moslq  ?

Warning threshold for Delay  ?

Critical threshold for Delay  ?

Display Report options on Web  ?

Display Report options on phone  ?

Voice RTCP-XR Report  ?

**Report Options on Phone** ?

Disabled: RoundTripDelay, SymmOneWayDelay

Enabled: Start Time, Current Time, Local User, Remote User, Local Codec

**NOTE**

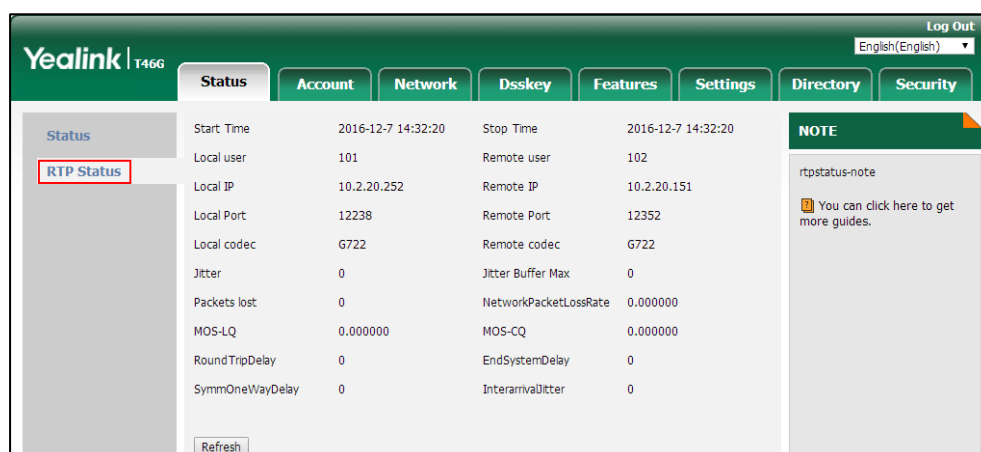
**Voice Quality Monitoring**  
It allows the IP phones to generate various quality metrics for listening quality and conversational quality.

The VQ-RTCPXR mechanism, compliant with RFC 6035, sends the service quality metric reports contained in SIP PUBLISH messages to the central report collector.

You can click here to get more guides.

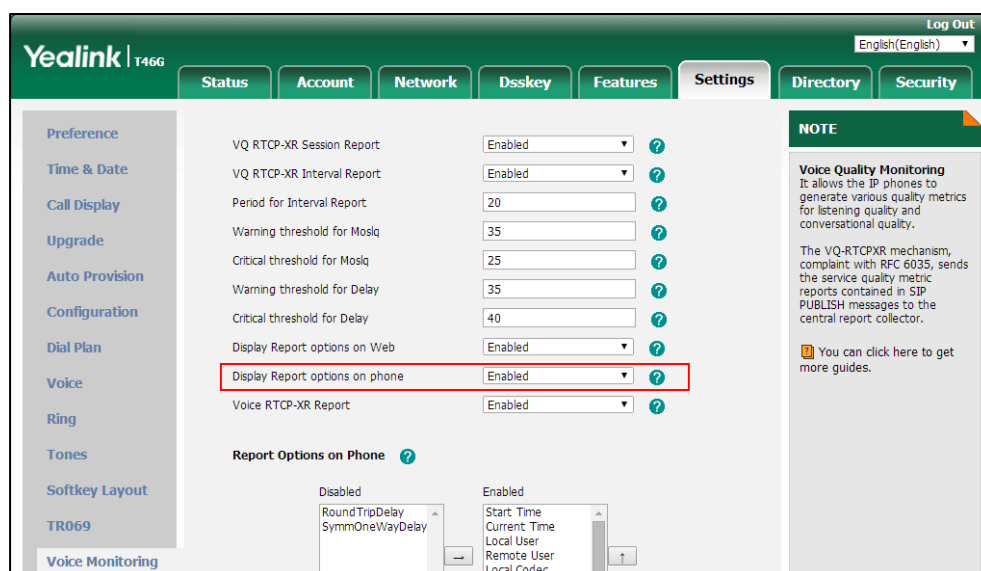
3. Click **Confirm** to accept the change.

The RTP status will appear on the web user interface at the path: **Status->RTP Status**.



To configure RTP status displayed on the LCD screen via web user interface:

1. Click on **Settings->Voice Monitoring**.
2. Select the desired value from the pull-down list of **Display Report options on phone**.




3. Click **Confirm** to accept the change.

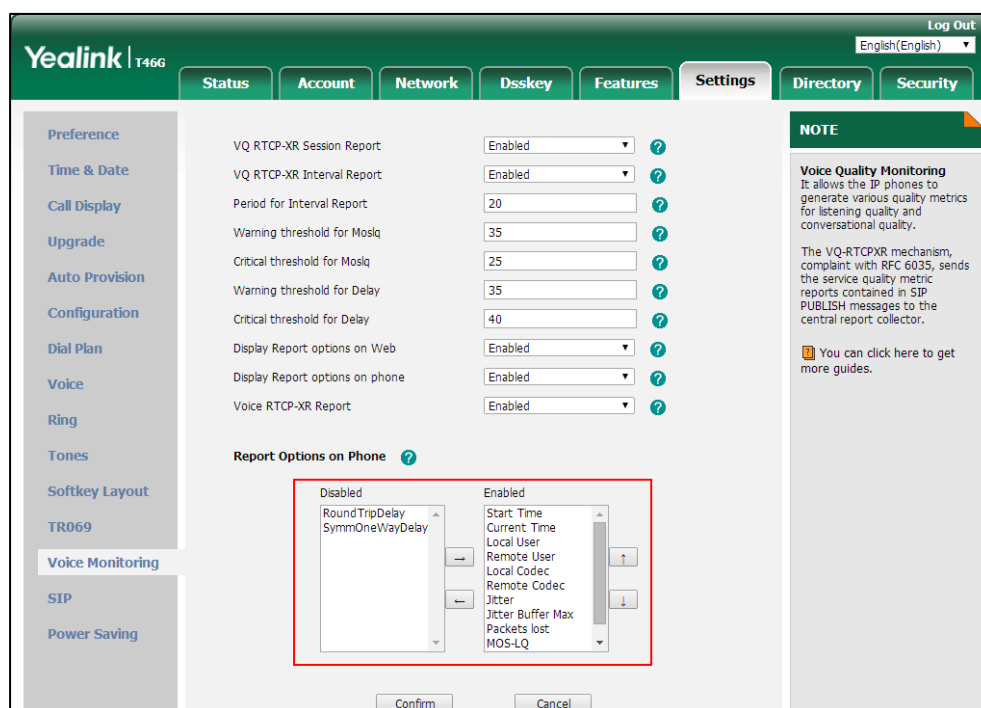
The RTP status will appear on the phone user interface at the path:




**Menu->Status->More->RTP Status.**

To configure the options of the RTP status displayed on the LCD screen via web user interface:

1. Click on **Settings->Voice Monitoring**.
2. In the **Report options on phone** block, select the desired list from the **Disabled** column and then click  .

The selected list appears in the **Enabled** column.



3. Repeat the step 2 to add more items to the **Enabled** column.
4. To remove an item from the **Enabled** column, select the desired item and then click .
5. To adjust the display order of enabled items, select the desired item and then click  or .

The LCD screen will display the item(s) in the adjusted order.

6. Click **Confirm** to accept the change.

#### To configure the central report collector via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Enter the host name of the central report collector in the **VQ RTCP-XR Collector Name** field.
4. Enter the IP address of the central report collector in the **VQ RTCP-XR Collector Address** field.

5. Enter the port of the central report collector in the **VQ RTCP-XR Collector Port** field.

The screenshot shows the Yealink T46G web interface with the 'Account' tab selected. The 'VQ RTCP-XR Collector Port' field is highlighted with a red box. The field contains the value '5060'. Other fields include 'VQ RTCP-XR Collector Name' (collector) and 'VQ RTCP-XR Collector Address' (10.2.1.98). The 'Confirm' button is visible at the bottom.

Field	Value
Keep Alive Type	Default
Keep Alive Interval(Seconds)	30
RPort	Disabled
Subscribe Period(Seconds)	1800
Shared Line	Draft BLA
Dialog Info Call Pickup	Enabled
BLA Number	14084588327
BLA Subscription Period	300
SIP Send MAC	Disabled
BLF List Retrieve Call Parked Code	*88
Shared Line	Shared Call Appearance
Call Pull Feature Access Code	*11
Dialog Info Call Pickup	Enabled
VQ RTCP-XR Collector Name	collector
VQ RTCP-XR Collector Address	10.2.1.98
VQ RTCP-XR Collector Port	5060

6. Click **Confirm** to accept the change.





## Configuring Security Features

This chapter provides information for making configuration changes for the following security-related features:

- [User and Administrator Passwords](#)
- [Auto Logout Time](#)
- [Phone Lock](#)
- [Transport Layer Security \(TLS\)](#)
- [Secure Real-Time Transport Protocol \(SRTP\)](#)
- [Encrypting and Decrypting Files](#)
- [Incoming Signaling Validation](#)

### User and Administrator Passwords

Some menu options are protected by two privilege levels, user and administrator, each with its own password. When logging into the web user interface, you need to enter the username and password to access various menu options. The default user password is "user" and the default administrator password is "admin".

For security reasons, the user or administrator should change the default user or administrator password as soon as possible. A user or an administrator can change the user password. The administrator password can only be changed by an administrator.

Advanced menu options are strictly used by administrators. Users can configure them only if they have administrator privileges.

#### Procedure

User or administrator password can be changed using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Change the user or administrator password of the IP phone. <b>Parameter:</b> static.security.user_password
<b>Web User Interface</b>		Change the user or administrator password of the IP phone. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?password=security&q=load
<b>Phone User Interface</b>		Change the administrator password

	of the IP phone.
--	------------------

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>static.security.user_password</b>	<b>String within 32 characters</b>	<b>user</b>

**Description:**

Configures the password of the user or administrator.

The IP phone uses "user" as the default user password and "admin" as the default administrator password.

The valid value format is <username> : <new password>.

**Example:**

static.security.user\_password = user:123 means setting the password of user to 123.

static.security.user\_password = admin:456 means setting the password of administrator to 456.

**Note:** IP phones support ASCII characters 32-126(0x20-0x7E) in passwords. You can set the password to be empty via web user interface only.

**Web User Interface:**

Security->Password

**Phone User Interface:**

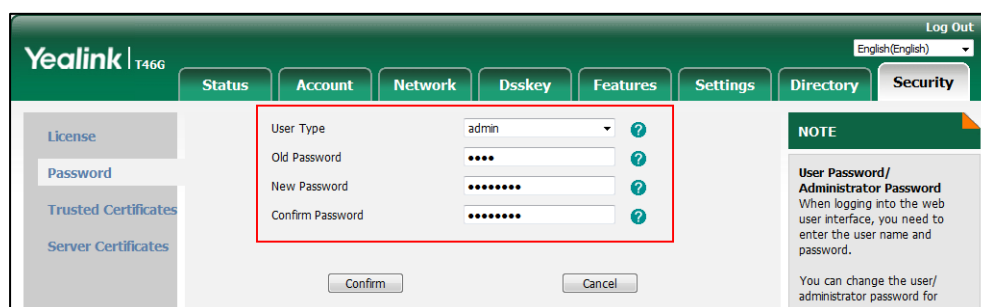
Menu->Advanced (default password: admin) ->Change Password

**Note:** You cannot change the user password via phone user interface.

**To change the user or administrator password via web user interface:**

1. Click on **Security->Password**.
2. Select the desired value (**user** or **admin**) from the pull-down list of **User Type**.
3. Enter new password in the **New Password** and **Confirm Password** fields.

Valid characters are ASCII characters 32-126(0x20-0x7E) except 58(3A).



- Click **Confirm** to accept the change.

**Note**

If logging into the web user interface of the phone with the user credential, you need to enter the old user password in the **Old Password** field.

**To change the administrator password via phone user interface:**

- Press **Menu->Advanced** (default password: admin) ->**Change Password**.
- Enter the current administrator password in the **Old PWD** field.
- Enter new password in the **New PWD** field and **Confirm PWD** field.  
Valid characters are ASCII characters 32-126(0x20-0x7E).
- Press the **Save** soft key to accept the change.

## Auto Logout Time

Auto logout time defines a specific period of time during which the IP phones will automatically log out if you have not performed any actions via web user interface. Once logging out, you must re-enter username and password for web access authentication.

**Procedure**

Auto logout time can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure auto logout time. <b>Parameter:</b> features.relog_offtime
<b>Web User Interface</b>		Configure auto logout time. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p =features-general&q=load

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
features.relog_offtime	Integer from 1 to 1000	5
<b>Description:</b> Configures the timeout interval (in minutes) for web access authentication.  <b>Example:</b> features.relog_offtime = 5  If you log into the web user interface and leave it idle for 5 minutes, it will automatically log		

Parameter	Permitted Values	Default
<p>out.</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;General Information-&gt;Auto Logout Time(1~1000min)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure the auto logout time via web user interface:**

1. Click on **Features->General Information**.
2. Enter the desired auto logout time in **Auto Logout Time(1~1000min)** field.

3. Click **Confirm** to accept the change.

## Phone Lock

Phone lock is used to lock the IP phone to prevent it from unauthorized use. Once the IP phone is locked, a user must enter the password to unlock it. IP phones offer three types of phone lock: Menu Key, Function Keys and All Keys. SIP-T48G/S IP phones only support All Keys type of phone lock. The IP phone will not be locked immediately after the phone lock type is configured. One of the following steps is also needed:

- Long press the pound key when the IP phone is idle.
- Press the phone lock DSS key (if configured) when the IP phone is idle.

In addition to the above steps, you can configure the IP phone to automatically lock the phone after a period of time.

## Phone Lock Type

The following table lists the operation behavior when configuring the type of phone lock:

	All Keys	Function Keys	Menu key
<b>Idle screen</b>	<p><b>Allowable Behavior:</b> You are allowed to press the desired Line Key (key type is Line) or the Speakerphone key to enter the dialing screen.</p> <p><b>Keys not Locked:</b> Line keys (key type is Line), digit keys, HEADSET key, Volume key and Speakerphone key.</p> <p><b>Note:</b> Line key is not applicable to SIP-T19(P) E2.</p>	The same as All Keys.	The Menu key (key type is Menu) is locked.
<b>Incoming call</b>	<p><b>Allowable Behavior:</b> You are allowed to answer or reject incoming calls.</p> <p><b>Keys not Locked:</b> Answer and Reject soft key; OK/ ✓, X, HEADSET key and Speakerphone key.</p> <p><b>Note:</b> Pressing X key to reject the call is not applicable to SIP-T23P/T23G/T21(P) E2/T19(P) E2 IP phones..</p>	The same as All Keys.	The Menu key (key type is Menu) is locked.
<b>Pre-dialing/Dialing screen</b>	<p><b>Allowable Behavior:</b> You are allowed to press the Line Key (key type is Line), input or modify numbers, dial emergency numbers and return to idle screen.</p> <p><b>Keys not Locked:</b> IME, More, Cancel, Send, Delete and Line soft key; line key (key type is Line), X, OK/ ✓, Volume key, Speakerphone key, digit keys, HEADSET key and "*" /"#" (key as send).</p> <p><b>Note:</b> Line key is not applicable to SIP-T19(P) E2.</p>	The same as All Keys, but you can dial any number.	The Menu key (key type is Menu) is locked.
<b>Talking</b>	<p><b>Allowable Behavior:</b> You are allowed to end the call, initiate a new call to the emergency number and resume a call.</p> <p><b>Keys not Locked:</b> EndCall, Cancel, Resume, NewCall soft key; line key (key type is Line), digit keys, X, Volume key, HEADSET key and Speakerphone key.</p> <p><b>Note:</b> Pressing X key to end the call are not applicable to SIP-T23P/T23G/T21(P) E2/T19(P) E2.</p>	The same as All Keys, but you can dial any number.	The Menu key (key type is Menu) is locked.

## Procedure

Phone lock can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the phone lock type. <b>Parameters:</b> phone_setting.phone_lock.enable phone_setting.phone_lock.lock_key_type
		Change the unlock PIN. <b>Parameter:</b> phone_setting.phone_lock.unlock_pin
		Configure the IP phone to automatically lock the phone after a time interval. <b>Parameter:</b> phone_setting.phone_lock.lock_time_out
		Configure emergency numbers. <b>Parameter:</b> phone_setting.emergency.number
		Assign a phone lock DSS key. <b>Parameters:</b> linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type linekey.X.label/ programablekey.X.label/ expansion_module.X.key.Y.label linekey.X.shortlabel
<b>Web User Interface</b>		Configure the phone lock type. Change the unlock PIN. Configure the IP phone to automatically lock the phone after a time interval. Configure emergency numbers. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=features-phonelock&amp;q=load">http://&lt;phoneIPAddress&gt;/servlet?p=features-phonelock&amp;q=load</a>
		Assign a phone lock DSS key. <b>Navigate to:</b> <a href="http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0">http://&lt;phoneIPAddress&gt;/servlet?p=dsskey&amp;q=load&amp;model=0</a>
<b>Phone User Interface</b>		Configure the phone lock type.

	<p>Change the unlock PIN.</p> <p>Configure the IP phone to automatically lock the phone after a time interval.</p> <p>Assign a phone lock DSS key.</p>
--	--------------------------------------------------------------------------------------------------------------------------------------------------------

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>phone_setting.phone_lock.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the phone lock feature.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Phone Lock-&gt;Phone Lock Enable</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Phone Lock-&gt;Lock Enable</p>		
<b>phone_setting.phone_lock.lock_key_type</b>	<b>0, 1 or 2</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the type of phone lock.</p> <p><b>0</b>-All Keys</p> <p><b>1</b>-Function Keys</p> <p><b>2</b>-Menu Keys</p> <p>For more information, refer to <a href="#">Phone Lock Type</a> on page 795.</p> <p><b>Note:</b> It is not applicable to SIP-T48G/S IP phones. It works only if the value of the parameter "phone_setting.phone_lock.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>Features-&gt;Phone Lock-&gt;Phone Lock Type</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Basic-&gt;Phone Lock-&gt;Lock Type</p>		
<b>phone_setting.phone_lock.unlock_pin</b>	<b>Characters within 15 digits</b>	<b>123</b>

Parameters	Permitted Values	Default
<b>Description:</b> Configures the password for unlocking the phone.  <b>Web User Interface:</b> Features->Phone Lock->Phone Unlock PIN(0~15 Digit)  <b>Phone User Interface:</b> Menu->Basic->Change PIN		
<b>phone_setting.phone_lock.lock_time_out</b>	<b>Integer from 0 to 3600</b>	<b>0</b>
<b>Description:</b> Configure the idle time (in seconds) before the IP phone is automatically locked.  The default value is 0 (the phone is locked only by long pressing the pound key or pressing the phone lock DSS key).  <b>Note:</b> It works only if the value of the parameter "phone_setting.phone_lock.enable" is set to 1 (Enabled).  <b>Web User Interface:</b> Features->Phone Lock->Auto Lock(0~3600s)  <b>Phone User Interface:</b> Menu->Basic->Phone Lock->Auto Lock		
<b>phone_setting.emergency.number</b>	<b>String within 99 characters</b>	<b>112,911,110</b>
<b>Description:</b> Configures emergency numbers.  Multiple emergency numbers are separated by commas.  <b>For</b> <b>SIP-T54S/T52S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T23P/T23G/T21(P) E2/T19(P) E2:</b>  If the value of the parameter "phone_setting.phone_lock.enable" is set to 1 (Enabled) and "phone_setting.phone_lock.lock_key_type" is set to 0 (All Keys), you are only allowed to dial emergency numbers configured by "phone_setting.emergency.number".  <b>For SIP-T48G/S:</b>  If the value of the parameter "phone_setting.phone_lock.enable" is set to 1 (Enabled), you are only allowed to dial emergency numbers configured by "phone_setting.emergency.number".  <b>Web User Interface:</b> Features->Phone Lock->Emergency		



Parameters	Permitted Values	Default
<b>Phone User Interface:</b> None		

## Phone Lock DSS Key

For more information on how to configure the DSS Key, refer to [Appendix D: Configuring DSS Key](#) on page 877.

### Details of Configuration Parameters:

Parameters	Permitted Values
<b>linekey.X.type</b> <sup>[1]</sup> <b>programmablekey.X.type</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.type</b> <sup>[3]</sup>	50
<b>Description:</b> Configures a DSS key as a phone lock DSS key on the IP phone. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/ Programmable Key->Type <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Keys X->Type	
<b>linekey.X.label</b> <sup>[1]</sup> <b>programmablekey.X.label</b> <sup>[2]</sup> <b>expansion_module.X.key.Y.label</b> <sup>[3]</sup>	String within 99 characters
<b>Description:</b> (Optional.) Configures the label displayed on the LCD screen for each DSS key. <b>Note:</b> EXT key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones. <b>Web User Interface:</b> Dsskey->Line Key/Programmable Key->Label <b>Phone User Interface:</b> Menu->Features->DSS Keys->Line Key X->Label	
<b>linekey.X.shortlabel</b> (X ranges from 1 to 21)	String within 99 characters

Parameters	Permitted Values
<p><b>Description:</b></p> <p>(Optional.) Configures the short label displayed on the LCD screen for line key.</p> <p><b>Note:</b> It is only applicable to SIP-T52S IP phones.</p> <p><b>Web User Interface:</b></p> <p>Dsskey-&gt;Line Key-&gt;Short</p> <p><b>Phone User Interface:</b></p> <p>Menu-&gt;Features-&gt;DSS Keys-&gt;Line Key X-&gt;Short Label</p>	

<sup>[1]</sup>X is the line key ID. For SIP-T48G/S, X=1-29; for SIP-T54S/T46G/T46S/T29G, X=1-27; for SIP-T42G/T42S/T41P/T41S, X=1-15; for SIP-T52S/T27P/T27G, X=1-21; for SIP-T40P/T40G/T23P/T23G, X=1-3, for SIP-T21(P) E2, X=1-2.

<sup>[2]</sup>X is the programable key ID. For SIP-T54S/T48G/T48S/T46G/T46S, X=1-10, 12-14; for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G, X=1-10, 13; for SIP-T29G/T27P/T27G, X=1-14; for SIP-T23P/T23G/T21(P) E2, X=1-9, 13, 14.

<sup>[3]</sup>X is the Ext key ID. For SIP-T54S/T52S, X=1-3, Y=1-60; for SIP-T48G/T48S/T46G/T46S, X=1-6, Y=1-40; for SIP-T29G/T27P/T27G, X=1-6, Y=1-40 (Ext key 21 cannot be configured).

#### To configure phone lock via web user interface:

1. Click on **Features->Phone Lock**.
2. Select the desired value from the pull-down list of **Phone Lock Enable**.
3. Select the desired value from the pull-down list of **Phone Lock Type**.
4. Enter the unlock PIN in the **Phone Unlock PIN(0~15 Digit)** field.
5. Enter the desired time in the **Auto Lock(0~3600s)** field.

6. Click **Confirm** to accept the change.

#### To configure the emergency number via web user interface:

1. Click on **Features->Phone Lock**.
2. Enter the emergency number in the **Emergency** field.

For multiple emergency numbers, enter a comma between every two numbers.

3. Click **Confirm**.

**To configure a phone lock DSS key via web user interface:**

1. Click on **Dsskey->Line Key** (or **Programmable Key**).
2. In the desired DSS key field, select **Phone Lock** from the pull-down list of **Type**.
3. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

4. Click **Confirm** to accept the change.

**To configure the type of phone lock via phone user interface:**

1. Press **Menu->Basic->Phone Lock**.
2. Enter the unlock PIN (default PIN: 123) in the **Unlock PIN** field.
3. Press or , or the **Switch** soft key to select the desired value from the **Lock Enable** field.
4. Press or , or the **Switch** soft key to select the desired value from the **Lock Type** field.
5. Enter the desired interval of automatic phone lock in the **Auto Lock** field.
6. Press the **Save** soft key to accept the change.

**To change the unlock PIN via phone user interface:**

1. Press **Menu->Basic->Change PIN**.
2. Enter the current unlock PIN in the **Old PIN** field.
3. Enter the new unlock PIN in the **New PIN** field.

4. Enter the new unlock PIN again in the **Confirm PIN** field.
5. Press the **Save** soft key to accept the change.

**To configure a phone lock DSS key via phone user interface:**

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press ◀ or ▶ , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◀ or ▶ , or the **Switch** soft key to select **Phone Lock** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Press the **Save** soft key to accept the change.

## Transport Layer Security (TLS)

TLS is a commonly-used protocol for providing communications privacy and managing the security of message transmission, allowing IP phones to communicate with other remote parties and connect to the HTTPS URL for provisioning in a way that is designed to prevent eavesdropping and tampering.

TLS protocol is composed of two layers: TLS Record Protocol and TLS Handshake Protocol. The TLS Record Protocol completes the actual data transmission and ensures the integrity and privacy of the data. The TLS Handshake Protocol allows the server and client to authenticate each other and negotiate an encryption algorithm and cryptographic keys before data is exchanged.

The TLS protocol uses asymmetric encryption for authentication of key exchange, symmetric encryption for confidentiality, and message authentication codes for integrity.

- **Symmetric encryption:** For symmetric encryption, the encryption key and the corresponding decryption key can be told by each other. In most cases, the encryption key is the same as the decryption key.
- **Asymmetric encryption:** For asymmetric encryption, each user has a pair of cryptographic keys – a public encryption key and a private decryption key. The information encrypted by the public key can only be decrypted by the corresponding private key and vice versa. Usually, the receiver keeps its private key. The public key is known by the sender, so the sender sends the information encrypted by the known public key, and then the receiver uses the private key to decrypt it.

IP phones support TLS version 1.0, 1.1 and 1.2. A cipher suite is a named combination of authentication, encryption, and message authentication code (MAC) algorithms used to negotiate the security settings for a network connection using the TLS/SSL network protocol. IP phones support the following cipher suites:

- DHE-RSA-AES256-SHA
- DHE-DSS-AES256-SHA
- AES256-SHA

- EDH-RSA-DES-CBC3-SHA
- EDH-DSS-DES-CBC3-SHA
- DES-CBC3-SHA
- DES-CBC3-MD5
- DHE-RSA-AES128-SHA
- DHE-DSS-AES128-SHA
- AES128-SHA
- RC2-CBC-MD5
- IDEA-CBC-SHA
- DHE-DSS-RC4-SHA
- RC4-SHA
- RC4-MD5
- RC4-64-MD5
- EXP1024-DHE-DSS-DES-CBC-SHA
- EXP1024-DES-CBC-SHA
- EDH-RSA-DES-CBC-SHA
- EDH-DSS-DES-CBC-SHA
- DES-CBC-SHA
- DES-CBC-MD5
- EXP1024-DHE-DSS-RC4-SHA
- EXP1024-RC4-SHA
- EXP1024-RC4-MD5
- EXP-EDH-RSA-DES-CBC-SHA
- EXP-EDH-DSS-DES-CBC-SHA
- EXP-DES-CBC-SHA
- EXP-RC2-CBC-MD5
- EXP-RC4-MD5

The following figure illustrates the TLS messages exchanged between the IP phone and TLS server to establish an encrypted communication channel:

No.	Time	Source	Destination	Protocol	Info
1	0.000000	192.168.3.86	192.168.0.230	SSLv3	Client Hello
2	0.021345	192.168.0.230	192.168.3.86	SSLv3	Server Hello, Certificate, Server Key Exchange, Server Hello Done
3	0.954947	192.168.3.86	192.168.0.230	SSLv3	Client Key Exchange, Change Cipher Spec, Encrypted Handshake Message
4	0.970099	192.168.0.230	192.168.3.86	SSLv3	Change Cipher Spec, Encrypted Handshake Message
5	1.012295	192.168.3.86	192.168.0.230	SSLv3	Application data, Application data
6	1.013562	192.168.0.230	192.168.3.86	SSLv3	Application data
7	1.013667	192.168.0.230	192.168.3.86	SSLv3	Application data

# Frame 13: 652 bytes on wire (5216 bits), 652 bytes captured (5216 bits)  
 # Ethernet II, Src: Vmware\_72:c9:2e (00:0c:29:72:c9:2e), Dst: Xiamenye\_11:12:b7 (00:15:65:11:12:b7)  
 # Internet Protocol, Src: 192.168.0.230 (192.168.0.230), Dst: 192.168.3.86 (192.168.3.86)  
 # Transmission Control Protocol, Src Port: https (443), Dst Port: nmserver (2244), Seq: 1482, Ack: 437, Len: 586  
 # Secure Socket Layer

**Step1:** IP phone sends "Client Hello" message proposing SSL options.

**Step2:** Server responds with "Server Hello" message selecting the SSL options, sends its public key information in "Server Key Exchange" message and concludes its part of the negotiation with "Server Hello Done" message.

**Step3:** IP phone sends session key information (encrypted by server's public key) in the "Client Key Exchange" message.

**Step4:** Server sends "Change Cipher Spec" message to activate the negotiated options for all future messages it will send.

IP phones can encrypt SIP with TLS, which is called SIPS. When TLS is enabled for an account, the SIP message of this account will be encrypted, and a lock icon appears on the LCD screen after the successful TLS negotiation.

## Certificates

The IP phone can serve as a TLS client or a TLS server. The TLS requires the following security certificates to perform the TLS handshake:

- **Trusted Certificate:** When the IP phone requests a TLS connection with a server, the IP phone should verify the certificate sent by the server to decide whether it is trusted based on the trusted certificates list. The IP phone has 77 built-in trusted certificates. You can upload 10 custom certificates at most. The format of the trusted certificate files must be \*.pem, \*.cer, \*.crt and \*.der and the maximum file size is 5MB. For more information on 77 trusted certificates, refer to [Appendix C: Trusted Certificates](#) on page 874.
- **Server Certificate:** When clients request a TLS connection with the IP phone, the IP phone sends the server certificate to the clients for authentication. The IP phone has two types of built-in server certificates: a unique server certificate and a generic server certificate. You can only upload one server certificate to the IP phone. The old server certificate will be overridden by the new one. The format of the server certificate files must be \*.pem and \*.cer and the maximum file size is 5MB.
  - **A unique server certificate:** It is unique to an IP phone (based on the MAC address) and issued by the Yealink Certificate Authority (CA).
  - **A generic server certificate:** It issued by the Yealink Certificate Authority (CA). Only if no unique certificate exists, the IP phone may send a generic certificate for authentication.

The IP phone can authenticate the server certificate based on the trusted certificates list. The trusted certificates list and the server certificates list contain the default and custom certificates. You can specify the type of certificates the IP phone accepts: default certificates, custom certificates or all certificates.

Common Name Validation feature enables the IP phone to mandatorily validate the common name of the certificate sent by the connecting server. And Security verification rules are compliant with RFC 2818.

### Note

In TLS feature, we use the terms trusted and server certificate. These are also known as CA and device certificates.

Resetting the IP phone to factory defaults will delete custom certificates by default. But this feature is configurable by the parameter "static.phone\_setting.reserve\_certs\_enable" using the configuration file.

## Procedure

Configuration changes can be performed using the following methods.

Central Provisioning (Configuration Files)	<MAC>.cfg	Configure TLS on a per-line basis. <b>Parameter:</b> account.X.sip_server.Y.transport_type
	<y0000000000xx>.cfg	Configure the TLS version. <b>Parameter:</b> static.security.default_ssl_method
		Configure trusted certificates feature. <b>Parameters:</b> static.security.trust_certificates static.security.ca_cert static.security.cn_validation
		Configure server certificates feature. <b>Parameter:</b> static.security.dev_cert
		Upload the trusted certificates. <b>Parameter:</b> static.trusted_certificates.url
		Delete all uploaded trusted certificates. <b>Parameter:</b> static.trusted_certificates.delete
		Upload the server certificates. <b>Parameter:</b> static.server_certificates.url
		Delete all uploaded server certificates. <b>Parameter:</b> static.server_certificates.delete
		Configure the custom certificates. <b>Parameter:</b> static.phone_setting.reserve_certs_enable
Web User Interface		
Configure TLS on a per-line basis. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=accou		

	nt-register&q=load&acc=0
	<p>Configure trusted certificates feature.</p> <p>Upload the trusted certificates.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=trusted-cert&amp;q=load</p>
	<p>Configure server certificates feature.</p> <p>Upload the server certificates.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=server-cert&amp;q=load</p>

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>account.X.sip_server.Y.transport_type<sup>[1]</sup></b> (Y ranges from 1 to 2)	<b>0, 1, 2 or 3</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the transport method the IP phone uses to communicate with the SIP server for a specific account.</p> <p><b>0</b>-UDP  <b>1</b>-TCP  <b>2</b>-TLS  <b>3</b>-DNS-NAPTR</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Register-&gt;SIP Server Y-&gt;Transport</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.security.default_ssl_method</b>	<b>0, 3, 4 or 5</b>	<b>3</b>
<p><b>Description:</b></p> <p>Configures the TLS version the IP phone uses to negotiate with the provisioning server when using TLS transport method to download the boot file and configuration files from the provisioning server.</p> <p><b>0</b>-use TLS 1.0 to negotiate with the provisioning server</p> <p><b>3</b>-use SSL V23, and let the IP phone and provisioning server negotiate the method. The IP phone will use TLS 1.2 to negotiate with the provisioning server first, if it is not compatible,</p>		



Parameters	Permitted Values	Default
<p>the IP phone will use TLS 1.1 to negotiate with the provisioning server...</p> <p><b>4</b>-use TLS 1.1 to negotiate with the provisioning server</p> <p><b>5</b>-use TLS 1.2 to negotiate with the provisioning server</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.security.trust_certificates</b>	<b>0 or 1</b>	<b>1</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to only trust the server certificates in the Trusted Certificates list.</p> <p><b>0</b>-Disabled, the IP phone will trust the server no matter whether the certificate sent by the server is valid or not.</p> <p><b>1</b>-Enabled, the IP phone will authenticate the server certificate based on the trusted certificates list. Only when the authentication succeeds, the IP phone will trust the server.</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Security-&gt;Trusted Certificates-&gt;Only Accept Trusted Certificates</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.security.ca_cert</b>	<b>0, 1 or 2</b>	<b>2</b>
<p><b>Description:</b></p> <p>Configures the type of certificates in the Trusted Certificates list for the IP phone to authenticate for TLS connection.</p> <p><b>0</b>-Default Certificates</p> <p><b>1</b>-Custom Certificates</p> <p><b>2</b>-All Certificates</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Security-&gt;Trusted Certificates-&gt;CA Certificates</p> <p><b>Phone User Interface:</b></p>		

Parameters	Permitted Values	Default
None		
<b>static.security.cn_validation</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to mandatorily validate the CommonName or SubjectAltName of the certificate sent by the server.</p> <p><b>0</b>-Disabled  <b>1</b>-Enabled</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Security-&gt;Trusted Certificates-&gt;Common Name Validation</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.security.dev_cert</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures the type of the device certificates for the IP phone to send for TLS authentication.</p> <p><b>0</b>-Default Certificates  <b>1</b>-Custom Certificates</p> <p><b>Note:</b> If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Security-&gt;Server Certificates-&gt;Device Certificates</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.trusted_certificates.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the access URL of the custom trusted certificate used to authenticate the connecting server.</p> <p><b>Example:</b></p> <p>static.trusted_certificates.url = http://192.168.1.20/tc.crt</p>		

Parameters	Permitted Values	Default
<p><b>Note:</b> The certificate you want to upload must be in *.pem, *.crt, *.cer or *.der format.</p> <p><b>Web User Interface:</b> Security-&gt;Trusted Certificates-&gt;Load Trusted Certificates File</p> <p><b>Phone User Interface:</b> None</p>		
<b>static.trusted_certificates.delete</b>	<b>http://localhost/all</b>	<b>Blank</b>
<p><b>Description:</b> Deletes all uploaded trusted certificates.</p> <p><b>Example:</b> static.trusted_certificates.delete = http://localhost/all</p> <p><b>Web User Interface:</b> None</p> <p><b>Phone User Interface:</b> None</p>		
<b>static.server_certificates.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<p><b>Description:</b> Configures the access URL of the server certificate the IP phone sends for authentication.</p> <p><b>Example:</b> static.server_certificates.url = http://192.168.1.20/ca.pem</p> <p><b>Note:</b> The certificate you want to upload must be in *.pem or *.cer format.</p> <p><b>Web User Interface:</b> Security-&gt;Server Certificates-&gt;Load Server Certificates File</p> <p><b>Phone User Interface:</b> None</p>		
<b>static.server_certificates.delete</b>	<b>http://localhost/all</b>	<b>Blank</b>
<p><b>Description:</b> Deletes all uploaded server certificates.</p> <p><b>Example:</b> static.server_certificates.delete = http://localhost/all</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
None <b>Phone User Interface:</b> None		
<b>static.phone_setting.reserve_certs_enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to reserve custom certificates after it is reset to factory defaults. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

<sup>[1]</sup>X is the account ID. For SIP-T54S/T48G/T48S/T46G/T46S/T29G, X=1-16; for SIP-T52S/T42G/T42S, X=1-12; for SIP-T41P/T41S/T27P/T27G, X=1-6; for SIP-T40P/T40G/T23P/T23G, X=1-3; for SIP-T21(P) E2, X=1-2; for SIP-T19(P) E2, X=1.

#### To configure TLS on a per-line basis via web user interface:

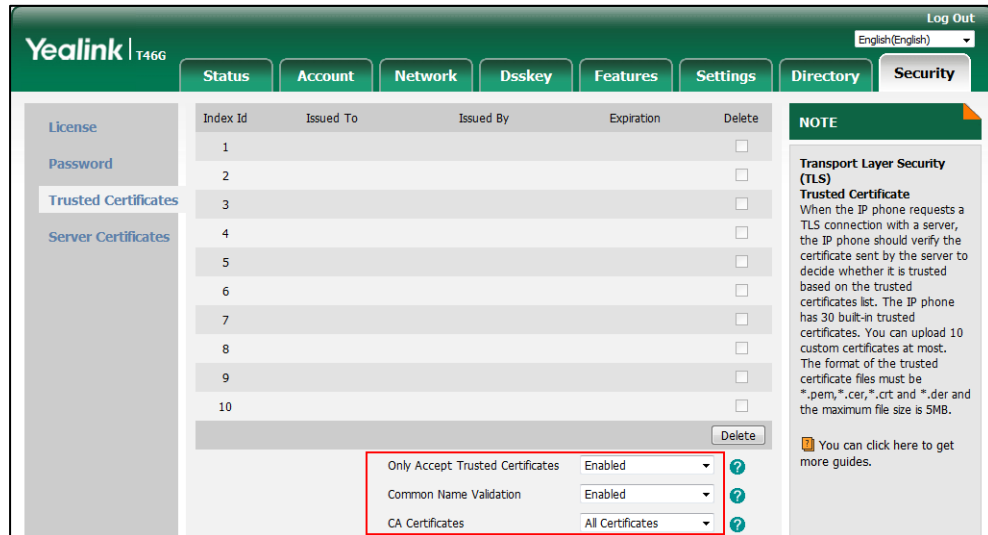
1. Click on **Account->Register**.
2. Select the desired account from the pull-down list of **Account**.
3. Select **TLS** from the pull-down list of **Transport**.

The screenshot displays the Yealink T46G web interface. The 'Account' tab is active, showing configuration for 'Account1'. The 'Transport' dropdown menu is highlighted with a red box, indicating the selection of 'TLS'. The interface includes a sidebar with 'Register', 'Basic', 'Codec', and 'Advanced' sections. A 'NOTE' section on the right provides additional information about account registration and NAT traversal.

4. Click **Confirm** to accept the change.

To configure the trusted certificates via web user interface:

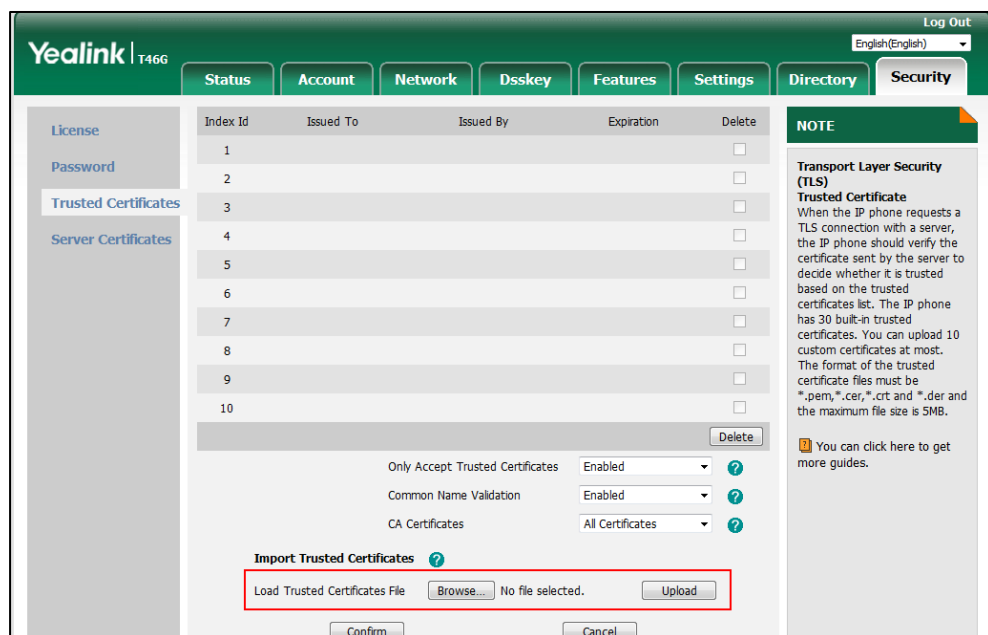
1. Click on **Security->Trusted Certificates**.
2. Select the desired values from the pull-down lists of **Only Accept Trusted Certificates**, **Common Name Validation** and **CA Certificates**.



3. Click **Confirm** to accept the change.  
A dialog box pops up to prompt that the settings will take effect after a reboot.
4. Click **OK** to reboot the phone.

To upload a trusted certificate via web user interface:

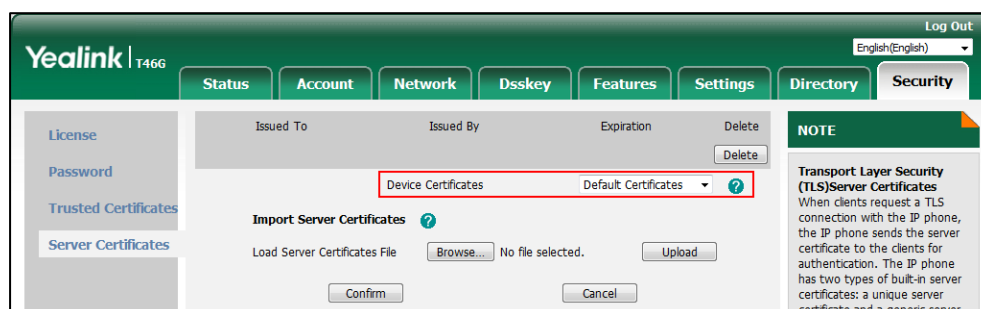
1. Click on **Security->Trusted Certificates**.
2. Click **Browse** to select the certificate (\*.pem, \*.crt, \*.cer or \*.der) from your local system.



3. Click **Upload** to upload the certificate.

**To configure the server certificates via web user interface:**

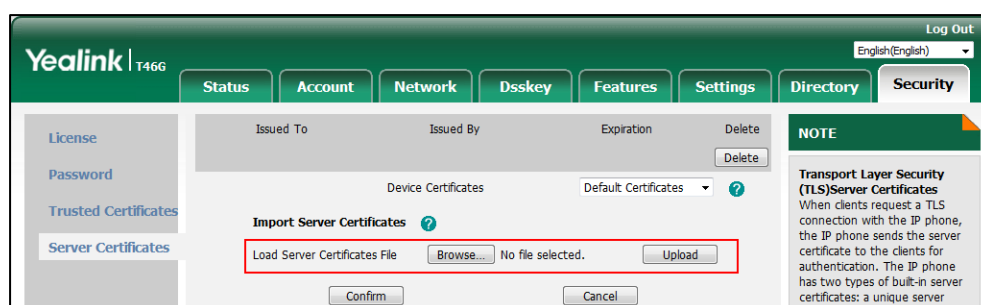
1. Click on **Security**->**Server Certificates**.
2. Select the desired value from the pull-down list of **Device Certificates**.



3. Click **Confirm** to accept the change.

**To upload a server certificate via web user interface:**

1. Click on **Security**->**Server Certificates**.
2. Click **Browse** to select the certificate (\*.pem and \*.cer) from your local system.



3. Click **Upload** to upload the certificate.

## Secure Real-Time Transport Protocol (SRTP)

Secure Real-Time Transport Protocol (SRTP) encrypts the RTP during VoIP phone calls to avoid interception and eavesdropping. The parties participating in the call must enable SRTP feature simultaneously. When this feature is enabled on both phones, the type of encryption to use for the session is negotiated between the IP phones. This negotiation process is compliant with [RFC 4568](#).

When a user places a call on the enabled SRTP phone, the IP phone sends an INVITE message with the RTP encryption algorithm to the destination phone. As described in [RFC 3711](#), RTP streams may be encrypted using an AES (Advanced Encryption Standard) algorithm.

Example of the RTP encryption algorithm carried in the SDP of the INVITE message:

```
m=audio 11780 RTP/SAVP 0 8 18 9 101
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:NzFINTUwZDk2OGVIOTc3YzNkYTkwZWVhMTM1YWFj
```

```

a=crypto:2 AES_CM_128_HMAC_SHA1_32
inline:NzkyM2FjNzQ2ZDgxYjg0MzQwMGVmMGUxMzdmNWFm
a=crypto:3 F8_128_HMAC_SHA1_80 inline:NDliMWIzZGE1ZTAwZjA5ZGFhNjQ5YmEANTMzYzA0
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:9 G722/8000
a=fmtp:101 0-15
a=rtpmap:101 telephone-event/8000
a=ptime:20
a=sendrecv

```

The callee receives the INVITE message with the RTP encryption algorithm, and then answers the call by responding with a 200 OK message which carries the negotiated RTP encryption algorithm.

Example of the RTP encryption algorithm carried in the SDP of the 200 OK message:

```

m=audio 11780 RTP/SAVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:NGY4OGViMDYzZjQzYTNiOTNkOWRiYzRiMjM0Yzcy
a=sendrecv
a=ptime:20
a=fmtp:101 0-15

```

SRTP is configurable on a per-line basis. When SRTP is enabled on both IP phones, RTP streams will be encrypted, and a lock icon appears on the LCD screen of each IP phone after successful negotiation.

#### Note

If you enable SRTP, then you should also enable TLS. This ensures the security of SRTP encryption. For more information on TLS, refer to [Transport Layer Security \(TLS\)](#) on page 802.

## Procedure

SRTP can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<MAC>.cfg	Configure SRTP feature on a per-line basis. <b>Parameter:</b> account.X.srtp_encryption
------------------------------------------------------	-----------	-----------------------------------------------------------------------------------------------

<b>Web User Interface</b>	<p>Configure SRTP feature on a per-line basis.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=account-adv&amp;q=load&amp;acc=0</p>
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**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>account.X.srtp_encryption<sup>[1]</sup></b>	<b>0, 1 or 2</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures whether to use voice encryption service for a specific account.</p> <p><b>0</b>-Disabled, the IP phone will not use voice encryption service.</p> <p><b>1</b>-Optional, the IP phone will negotiate with the other IP phone what type of encryption to utilize for the session.</p> <p><b>2</b>-Compulsory, the IP phone is forced to use SRTP during a call.</p> <p><b>Web User Interface:</b></p> <p>Account-&gt;Advanced-&gt;RTP Encryption(SRTP)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure SRTP feature via web user interface:**

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.



3. Select the desired value from the pull-down list of **RTP Encryption(SRTP)**.

The screenshot shows the Yealink T46G web interface with the 'Account' tab selected. The 'RTP Encryption(SRTP)' dropdown menu is highlighted with a red box, showing 'Optional' as the selected value. Other fields include Keep Alive Type, Keep Alive Interval, RPort, Subscribe Period, PTime, BLF List URI, BLF List Pickup Code, BLF List Barge In Code, BLF List Retrieve Call Parked Code, Shared Line, Call Pull Feature Access Code, Dialog Info Call Pickup, VQ RTPC-XR Collector Name, VQ RTPC-XR Collector Address, and VQ RTPC-XR Collector Port. A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

4. Click **Confirm** to accept the change.

## Encrypting and Decrypting Files

Yealink IP phones support downloading encrypted files from the server and encrypting files before/when uploading them to the server. You can encrypt the following files:

- **Configuration files:** MAC-Oriented CFG file (<MAC>.cfg), Common CFG file (y0000000000xx.cfg), MAC-local CFG file (<MAC>-local.cfg) or other custom CFG files (for example, sip.cfg, account.cfg)
- **Contact Files:** <MAC>-contact.xml

To encrypt/decrypt files, you may have to configure an AES key.

## Configuration Parameters

### Procedure

Configuration changes can be performed using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure whether to only download and resolve the encrypted files. <b>Parameter:</b> static.auto_provision.update_file_mode
--------------------------------------------------	---------------------	------------------------------------------------------------------------------------------------------------------------------------

		Configure the decryption method. <b>Parameter:</b> static.auto_provision.aes_key_in_file
		Configure AES keys. <b>Parameters:</b> static.auto_provision.aes_key_16.com static.auto_provision.aes_key_16.mac
		Specify if the contact file is encrypted when it is uploaded from the phone to the server. <b>Parameter:</b> static.auto_provision.encryption.directory
		Specify if the MAC-local CFG file is encrypted when it is uploaded from the phone to the server. <b>Parameter:</b> static.auto_provision.encryption.config
		<b>Web User Interface</b>
<b>Web User Interface</b>		Configure AES keys. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=settings-autop&q=load
<b>Phone User Interface</b>		Configure AES keys.

**Details of Configuration Parameters:**

Parameters	Permitted Values	Default
<b>static.auto_provision.update_file_mode</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> <p>Enables or disables the IP phone only to download the encrypted files.</p> <p><b>0</b>-Disabled, the IP phone will download the configuration files (for example, sip.cfg, account.cfg, &lt;MAC&gt;-local.cfg) and &lt;MAC&gt;-contact.xml file from the server during auto provisioning no matter whether the files are encrypted or not. And then resolve these files and update settings onto the IP phone system.</p> <p><b>1</b>-Enabled, the IP phone will only download the encrypted configuration files (for example, sip.cfg, account.cfg, &lt;MAC&gt;-local.cfg) or &lt;MAC&gt;-contact.xml file from the server during auto provisioning, and then resolve these files and update settings onto the IP phone system.</p>		

Parameters	Permitted Values	Default
<b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>static.auto_provision.aes_key_in_file</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to decrypt configuration files using the encrypted AES keys. <b>0</b> -Disabled, the IP phone will decrypt the encrypted configuration files using plaintext AES keys configured on the IP phone. <b>1</b> -Enabled, the IP phone will download <xx_Security>.enc files (for example, <sip_Security>.enc, <account_Security>.enc) during auto provisioning, and then decrypts these files into the plaintext keys (for example, key2, key3) respectively using the phone built-in key (for example, key1). The IP phone then decrypts the encrypted configuration files using corresponding key (for example, key2, key3). <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>static.auto_provision.aes_key_16.com</b>	<b>16 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the plaintext AES key for encrypting/decrypting the Common CFG/Custom CFG file. The valid characters contain: 0 ~ 9, A ~ Z, a ~ z and the following special characters are also supported: # \$ % * + , - . : = ? @ [ ] ^ _ { } ~. <b>Example:</b> static.auto_provision.aes_key_16.com = 0123456789abcdef <b>Note:</b> For decrypting, it works only if the value of the parameter "static.auto_provision.aes_key_in_file" is set to 0 (Disabled). If the downloaded MAC-Oriented file is encrypted and the parameter "static.auto_provision.aes_key_16.mac" is left blank, the IP phone will try to encrypt/decrypt the MAC-Oriented file using the AES key configured by the parameter "static.auto_provision.aes_key_16.com". <b>Web User Interface:</b> Settings->Auto Provision->Common AES Key		

Parameters	Permitted Values	Default
<b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Auto Provision->Common		
<b>static.auto_provision.aes_key_16.mac</b>	<b>16 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the plaintext AES key for encrypting/decrypting the MAC-Oriented files (<MAC>.cfg, <MAC>-local.cfg and <MAC>-contact.xml). The valid characters contain: 0 ~ 9, A ~ Z, a ~ z and the following special characters are also supported: # \$ % * + , - . : = ? @ [ ] ^ _ { } ~. <b>Example:</b> static.auto_provision.aes_key_16.mac = 0123456789abmins <b>Note:</b> For decrypting, it works only if the value of the parameter "static.auto_provision.aes_key_in_file" is set to 0 (Disabled). If the downloaded MAC-Oriented file is encrypted and the parameter "static.auto_provision.aes_key_16.mac" is left blank, the IP phone will try to encrypt/decrypt the MAC-Oriented file using the AES key configured by the parameter "static.auto_provision.aes_key_16.com". <b>Web User Interface:</b> Settings->Auto Provision->MAC-Oriented AES Key <b>Phone User Interface:</b> Menu->Advanced (default password: admin) ->Auto Provision->MAC-Oriented AES		
<b>static.auto_provision.encryption.directory</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to encrypt <MAC>-contact.xml file using the plaintext AES key. <b>0</b> -Disabled, the contact file will be uploaded unencrypted and will replace the one (encrypted or unencrypted) stored on the server if you have configured to back up the contacts to the server by the parameter "static.auto_provision.local_contact.backup.enable". <b>1</b> -Enabled, the contact file will be encrypted uploaded and will replace the one (encrypted or unencrypted) stored on the server if you have configured to back up the contacts to the server by the parameter "static.auto_provision.local_contact.backup.enable". The plaintext AES key is configured by the parameter "static.auto_provision.aes_key_16.mac". <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		

Parameters	Permitted Values	Default
<b>static.auto_provision.encryption.call_log</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to encrypt &lt;MAC&gt;-calllog.xml file using the plaintext AES key.</p> <p><b>0</b>-Disabled, the call log file will be uploaded unencrypted and will replace the one (encrypted or unencrypted) stored on the server if you have configured to back up the call log to the server by the parameter "static.auto_provision.local_calllog.backup.enable".</p> <p><b>1</b>-Enabled, the call log file will be encrypted uploaded and will replace the one (encrypted or unencrypted) stored on the server if you have configured to back up the call log to the server by the parameter "static.auto_provision.local_calllog.backup.enable". The plaintext AES key is configured by the parameter "static.auto_provision.aes_key_16.mac".</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.encryption.config</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to encrypt &lt;MAC&gt;-local.cfg file using the plaintext AES key.</p> <p><b>0</b>-Disabled, the MAC-local CFG file will be uploaded unencrypted and will replace the one (encrypted or unencrypted) stored on the server if you have configured to back up the MAC-local CFG file to the server by the parameter "static.auto_provision.custom.sync".</p> <p><b>1</b>-Enabled, the MAC-local CFG file will be uploaded encrypted and will replace the one (encrypted or unencrypted) stored on the server if you have configured to back up the MAC-local CFG file to the server by the parameter "static.auto_provision.custom.sync". The plaintext AES key is configured by the parameter "static.auto_provision.aes_key_16.mac".</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To configure AES keys via web user interface:**

1. Click on **Settings->Auto Provision**.
2. Enter the values in the **Common AES Key** and **MAC-Oriented AES Key** fields.

AES keys must be 16 characters and the supported characters contain: 0-9, A-Z, a-z and the following special characters are also supported: # \$ % \* + , - . : = ? @ [ ] ^ \_ { } ~.

3. Click **Confirm** to accept the change.

#### To configure AES keys via phone user interface:

1. Press **Menu**->**Advanced** (default password: admin) ->**Auto Provision**.
2. Enter the values in the **Common** and **MAC-Oriented AES** fields.

AES keys must be 16 characters and the supported characters contain: 0-9, A-Z, a-z and the following special characters are also supported: # \$ % \* + , - . : = ? @ [ ] ^ \_ { } ~.

3. Press the **Save** soft key to accept the change.

## Encrypting and Decrypting Configuration Files

Encrypted configuration files can be downloaded from the provisioning server to protect against unauthorized access and tampering of sensitive information (for example, login passwords, registration information).

Yealink provides three encryption tools:

- Config\_Encrypt\_Tool.exe (via graphical tool for Windows platform)
- Config\_Encrypt.exe (via DOS command line for Windows platform)
- yealinkencrypt (for Linux platform)

The encryption tools encrypt plaintext configuration files (for example, account.cfg, <y0000000000xx>.cfg, <MAC>.cfg) (one by one or in batch) using 16-character symmetric keys (the same or different keys for configuration files) and generate encrypted configuration files with the same file name as before.

#### Note

You can also configure the <MAC>-local.cfg files to be automatically encrypted using 16-character symmetric keys when uploading to the server (by setting the value of the parameter "static.auto\_provision.encryption.config" to 1).

These tools also encrypt the plaintext 16-character symmetric keys using a fixed key, which is the same as the one built in the IP phone, and generate new files named as <xx\_Security>.enc (xx is the name of the configuration file, for example, y000000000028\_Security.enc for y000000000028.cfg file, account\_Security.enc for account.cfg). This tool generates another new file named as Aeskey.txt to store the plaintext 16-character symmetric keys for each configuration file.

For security reasons, administrator should upload encrypted configuration files, <xx\_Security>.enc files to the root directory of the provisioning server. During auto provisioning, the IP phone requests to download the boot file first and then download the referenced configuration files. For more information on boot file, refer to [Boot Files](#) on page 99. For example, the IP phone downloads account.cfg file and it is encrypted. The IP phone will request to download <account\_Security>.enc file (if enabled) and decrypt it into the plaintext key (for example, key2) using the built-in key (for example, key1). Then the IP phone decrypts account.cfg file using key2. After decryption, the IP phone resolves configuration files and updates configuration settings onto the IP phone system.

The following describes how to use use "Config\_Encrypt\_Tool.exe" to encrypt the account.cfg file. For more information on the other two encryption tools, refer to [Yealink Configuration Encryption Tool User Guide](#).

The way the IP phone processes other configuration files is the same to that of the account.cfg file.

## Procedure to Encrypt Configuration Files

### To encrypt the account.cfg file:

1. Double click "Config\_Encrypt\_Tool.exe" to start the application tool.

The screenshot of the main page is shown as below:



When you start the application tool, a file folder named "Encrypted" is created automatically in the directory where the application tool is located.

2. Click **Browse** to locate configuration file(s) (for example, account.cfg) from your local system in the **Select File(s)** field.

To select multiple configuration files, you can select the first file and then press and hold

the **Ctrl** key and select other files.

- (Optional.) Click **Browse** to locate the target directory from your local system in the **Target Directory** field.

The tool uses the file folder "Encrypted" as the target directory by default.

- (Optional.) Mark the desired radio box in the **AES Model** field.

If you mark the **Manual** radio box, you can enter an AES key in the **AES KEY** field or click **Re-Generate** to generate an AES key in the **AES KEY** field. The configuration file(s) will be encrypted using the AES key in the **AES KEY** field.

If you mark the **Auto Generate** radio box, the configuration file(s) will be encrypted using random AES key. The AES keys of configuration files are different.

#### Note

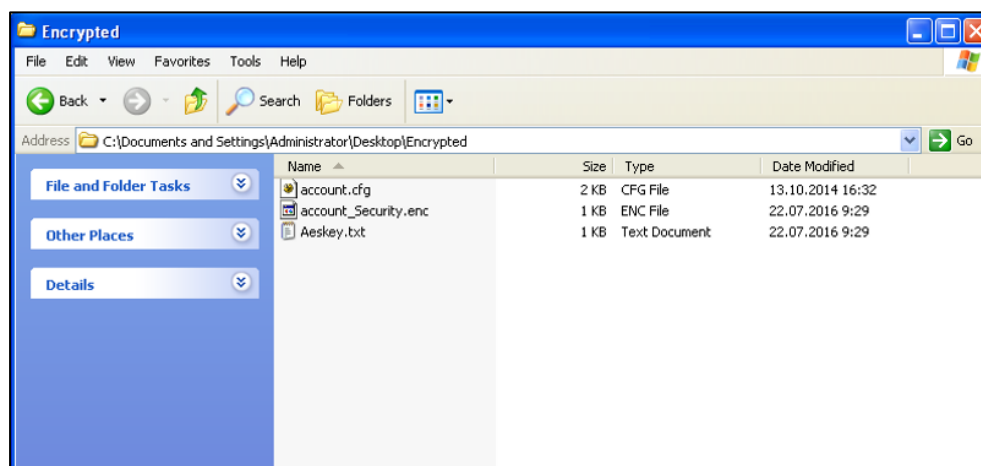
AES keys must be 16 characters and the supported characters contain: 0 ~ 9, A ~ Z, a ~ z and the following special characters are also supported: # \$ % \* + , - . : = ? @ [ ] ^ \_ { } ~.

- Click **Encrypt** to encrypt the configuration file(s).



- Click **OK**.

The target directory will be automatically opened. You can find the encrypted CFG file(s), encrypted key file(s) and an Aeskey.txt file storing plaintext AES key(s).





## Encrypting and Decrypting Contact Files

Encrypted contact files can be used to protect against unauthorized access and tampering of private information (for example, contact number). It is helpful for protecting trade secrets.

You can configure the contact files to be automatically encrypted using 16-character symmetric keys (configured by the parameter "static.auto\_provision.aes\_key\_16.mac") when uploading to the server (by setting the value of the parameter "static.auto\_provision.encryption.directory" to 1). The encrypted contact files have the same file names as before. The encrypted contact files can be downloaded from the server and decrypted using 16-character symmetric keys during auto provisioning. If the parameter "static.auto\_provision.aes\_key\_16.mac" is left blank, the value of the parameter "static.auto\_provision.aes\_key\_16.com" will be used.

If the downloaded contact files is encrypted, the IP phone will try to decrypt <MAC>-contact.xml file using the plaintext AES key. After decryption, the IP phone resolves contact files and updates contact information onto the IP phone system.

## Incoming Signaling Validation

Yealink IP phones support the following three optional levels of security for validating incoming network signaling:

- **Source IP address validation:** ensure request is received from an IP address of a server belonging to the set of target SIP servers.
- **Digest authentication:** challenge requests with digest authentication using the local credentials for the associated registered account.
- **Source IP address validation and digest authentication:** apply both of the above methods.

### Procedure

Incoming signaling validation can be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cf g	Configure incoming signaling validation. <b>Parameter:</b> sip.request_validation.source.list sip.request_validation.digest.list sip.request_validation.digest.realm sip.request_validation.event
------------------------------------------------------	-------------------------	------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

**Details of the Configuration Parameters:**

Parameters	Permitted Values	Default
<b>sip.request_validation.source.list</b>	<b>A valid string</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the name of the request method for which source IP address validation will be applied.</p> <p>It is used to ensure the request that is received from the IP address of a SIP server.</p> <p><b>Example:</b></p> <p>sip.request_validation.source.list = INVITE, NOTIFY</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>sip.request_validation.digest.list</b>	<b>A valid string</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the name of the request method for which digest authentication will be applied.</p> <p>It is used to challenge requests with digest authentication that use the local credentials for the associated registered account.</p> <p><b>Example:</b></p> <p>sip.request_validation.digest.list = INVITE, SUBSCRIBE</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>sip.request_validation.digest.realm</b>	<b>A valid string</b>	<b>YealinkS PIP</b>
<p><b>Description:</b></p> <p>Configures the string used for authentication parameter Realm when performing the digest authentication.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

Parameters	Permitted Values	Default
<b>sip.request_validation.event</b>	<b>A valid string</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures which events specified within the Event header of SUBSCRIBE or NOTIFY request should be validated when performing the digest authentication.</p> <p>If it is left blank, all events will be validated.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		



# Troubleshooting

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This chapter provides an administrator with general information for troubleshooting some common problems that he (or she) may encounter while using IP phones.

## Troubleshooting Methods

IP phones can provide feedback in a variety of forms such as log files, packets, status indicators and so on, which can help an administrator more easily find the system problem and fix it.

The following are helpful for better understanding and resolving the working status of the IP phone.

- [Viewing Log Files](#)
- [Capturing Packets](#)
- [Enabling Watch Dog Feature](#)
- [Getting Information from Status Indicators](#)
- [Analyzing Configuration File](#)
- [Exporting All the Diagnostic Files](#)
- [Ping and Traceroute](#)

## Viewing Log Files

If your IP phone encounters some problems, commonly the local log files or syslog files are needed.

You can configure the phone to log events locally. There are two types of local log files: <MAC>-boot.log (for example, 0015659188f2-boot.log) and <MAC>-sys.log (for example, 0015659188f2-sys.log). These two local log files can be exported via web user interface separately. You can configure the IP phone to periodically upload the local log files to the provisioning server (only support an FTP/TFTP as the provisioning server) or the specific server (if configured), avoiding the local log loss. You can specify the severity level of the log to be reported to the <MAC>-sys.log file. The default local log level is 3.

You can also configure the IP phone to send syslog messages to a syslog server in real time. You can specify the severity level of the syslog to be sent to a syslog server. The default system log level is 3.

## Local Logging

### Procedure

Local logging can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>. cfg	Configure local logging feature. <b>Parameter:</b> static.local_log.enable
		Configure the severity level of the logs to be reported to the <MAC>-sys.log file. <b>Parameter:</b> static.local_log.level
		Configure the maximum size of the log files to be stored on the phone. <b>Parameter:</b> static.local_log.max_file_size
		Configure the maximum size of the local log files to be stored on the server. <b>Parameter:</b> static.auto_provision.local_log.backup.append.max_file_size
		Configure the IP phone to upload local log files to the server. <b>Parameter:</b> static.auto_provision.local_log.backup.enable
		Configure the period of the local log files uploads to the server. <b>Parameter:</b> static.auto_provision.local_log.backup.upload_period
		Configure the behavior when local log files on the server reach the maximum size. <b>Parameter:</b> static.auto_provision.local_log.backup.append.limit_mode
		Configure whether the local log files on the server are overwritten or appended.

		<b>Parameter:</b>  static.auto_provision.local_log.backup.append
		Configure the waiting time before the phone uploads the <MAC>-boot.log file to the server after bootup.  <b>Parameter:</b>  static.auto_provision.local_log.backup.bootlog.upload_wait_time
		Configure the upload path of the local log files.  <b>Parameter:</b>  static.auto_provision.local_log.backup.path
<b>Web User Interface</b>		Configure local logging feature.  Configure the severity level of the logs to be reported to the <MAC>-sys.log file.  Configure the maximum size of the log files to be stored on the phone.  <b>Navigate to:</b>  http://<phoneIPAddress>/servlet?p=settings-config&q=load

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.local_log.enable</b>	<b>0 or 1</b>	<b>1</b>
<b>Description:</b> Enables or disables the IP phone to record log to the log files locally. <b>0</b> -Disabled, the IP phone will stop recording log to the log files (<MAC>-boot.log and <MAC>-sys.log) locally. The log files recorded before are still kept on the phone. <b>1</b> -Enabled, the IP phone will continue to record log to the log files (<MAC>-boot.log and <MAC>-sys.log) locally. You can upload the local log files to the provisioning server or a specific server or export them to the local system. <b>Note:</b> We recommend you not to disable this feature. <b>Web User Interface:</b> Settings->Configuration->Enable Local Log <b>Phone User Interface:</b>		

Parameters	Permitted Values	Default
None		
<b>static.local_log.level</b>	<b>Integer from 0 to 6</b>	<b>3</b>
<p><b>Description:</b></p> <p>Configures the lowest level of local log information to be reported to the &lt;MAC&gt;-sys.log file.</p> <p>When you choose a log level, you are including all events of an equal or higher severity level and excluding events of a lower severity level. The logging level you choose determines the lowest severity of events to log.</p> <p><b>0</b>-system is unusable  <b>1</b>-action must be taken immediately  <b>2</b>-critical condition  <b>3</b>-error conditions  <b>4</b>-warning conditions  <b>5</b>-normal but significant condition  <b>6</b>-informational</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Configuration-&gt;Local Log Level</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.local_log.max_file_size</b>	<b>Refer to the following content</b>	<b>Refer to the following content</b>
<p><b>Description:</b></p> <p>Configures the maximum size (in KB) of the log files (&lt;MAC&gt;-boot.log and &lt;MAC&gt;-sys.log) to be stored on the IP phone.</p> <p>When this size is about to be exceeded,</p> <p>(1) If the local log files are configured to be uploaded to the server by the parameter "static.auto_provision.local_log.backup.enable", the IP phone will clear all the local log files on the phone once successfully backing up.</p> <p>(2) If the value of the parameter "static.auto_provision.local_log.backup.enable" is set to 0 (Disabled), the IP phone will erase half of the logs from the oldest log information on the phone.</p> <p><b>For SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T29G/T27P/T27G IP phones:</b></p> <p>The default value: 1024</p>		



Parameters	Permitted Values	Default
<p>The permitted values: Integer from 1024 to 2048</p> <p><b>For SIP-T40P/T40G/T23P/T23G/T21(P) E2/T19(P) E2:</b></p> <p>The default value: 256</p> <p>The permitted values: Integer from 256 to 1024</p> <p><b>Example:</b></p> <p>static.local_log.max_file_size = 1024</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Configuration-&gt;Max Log File Size (1024-2048KB)</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.local_log.backup.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to upload the local log files (&lt;MAC&gt;-boot.log and &lt;MAC&gt;-sys.log) to the provisioning server or a specific server.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, the IP phone will upload the local log files to the provisioning server or the specific server to back up these files when one of the following happens:</p> <ul style="list-style-type: none"> <li>- Auto provisioning is triggered;</li> <li>- The size of the local log files reaches maximum configured by the parameter "static.local_log.max_file_size";</li> <li>- It's time to upload local log files according to the upload period configured by the parameter "static.auto_provision.local_log.backup.upload_period".</li> </ul> <p><b>Note:</b> The upload path is configured by the parameter "static.auto_provision.local_log.backup.path".</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.local_log.backup.upload_period</b>	<b>Integer from 30 to 86400</b>	<b>30</b>
<p><b>Description:</b></p> <p>Configures the period (in seconds) of the local log files (&lt;MAC&gt;-boot.log and &lt;MAC&gt;-sys.log) uploads to the provisioning server or a specific server.</p>		

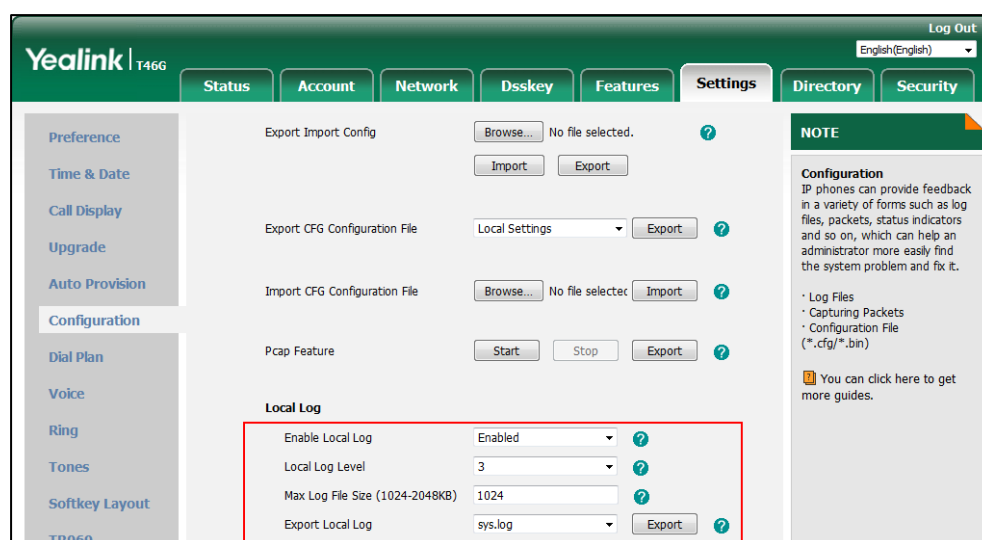
Parameters	Permitted Values	Default
<p><b>Example:</b></p> <p>static.auto_provision.local_log.backup.upload_period = 60</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.local_log.backup.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.local_log.backup.path</b>	<b>URL within 1024 characters</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the upload path of the local log files (&lt;MAC&gt;-boot.log and &lt;MAC&gt;-sys.log). If you leave it blank, the IP phone will upload the local log files to the provisioning server. If you configure a relative URL (for example, /upload), the IP phone will upload the local log files by extracting the root directory from the access URL of the provisioning server. If you configure an absolute URL with protocol (for example, tftp), the IP phone will upload the local log files using the desired protocol. If no protocol, the IP phone will use the same protocol with auto provisioning for uploading files.</p> <p><b>Example:</b></p> <p>static.auto_provision.local_log.backup.path = tftp://10.3.6.133/upload/</p> <p><b>Note:</b> It works only if the value of the parameter "static.auto_provision.local_log.backup.enable" is set to 1 (Enabled).</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.auto_provision.local_log.backup.append</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Configures whether uploaded local log files (&lt;MAC&gt;-boot.log and &lt;MAC&gt;-sys.log) overwrite existing files or are appended to existing files.</p> <p><b>0-Overwrite</b></p>		

Parameters	Permitted Values	Default
<b>1-Append</b> (not applicable to TFTP Server) <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>static.auto_provision.local_log.backup.append.limit_mode</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Configures the behavior when local log files (<MAC>-boot.log and <MAC>-sys.log) on the provisioning server or a specific server reach the maximum file size. <b>0-Append Delete</b> , the server will delete the old log and the IP phone will continue to uploading log. <b>1-Append Stop</b> , the IP phone will stop uploading log. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>static.auto_provision.local_log.backup.append.max_file_size</b>	<b>Integer from 200 to 65535</b>	<b>1024</b>
<b>Description:</b> Configures the maximum size (in KB) of the local log files (<MAC>-boot.log and <MAC>-sys.log) to be stored on the provisioning server or a specific server. <b>Example:</b> static.auto_provision.local_log.backup.append.max_file_size = 1025 <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>static.auto_provision.local_log.backup.bootlog.upload_wait_time</b>	<b>Integer from 1 to 86400</b>	<b>120</b>
<b>Description:</b>		

Parameters	Permitted Values	Default
<p>Configures the waiting time (in seconds) before the phone uploads the &lt;MAC&gt;-boot.log file to the provisioning server or a specific server after startup.</p> <p><b>Example:</b></p> <p>static.auto_provision.local_log.backup.bootlog.upload_wait_time = 121</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To export the system log to a local PC via web user interface:**

1. Click on **Settings->Configuration**.
2. Select **Enabled** from the pull-down list of **Enable Local Log**.
3. Select **6** from the pull-down list of **Local Log Level**.  
The default local log level is "3".
4. Enter the limit size of the log files in the **Max Log File Size (1024-2048KB)** field.
5. Select **sys.log** from the pull-down list of **Export Local Log**.
6. Click **Confirm** to accept the change.



7. Reproduce the issue.
8. Click **Export** to open the file download window, and then save the file to your local system.

**To export the boot log to a local PC via web user interface:**

1. Click on **Settings->Configuration**.
2. Select **Enabled** from the pull-down list of **Enable Local Log**.
3. Select **boot.log** from the pull-down list of **Export Local Log**.

4. Click **Confirm** to accept the change.
5. Click **Export** to open the file download window, and then save the file to your local system.

**To view the log files on your local system:**

The <MAC>-boot.log file can only log the last reboot events.

The following figure shows a portion of a <MAC>-boot.log (for example, 00156574b150-boot.log):

```

1 Jan 1 00:00:24 syslogd started: BusyBox v1.10.3
2 Jan 1 00:00:25 sys [655]: ANY <0+emerg > sys log :type=1,time=0,E=3,W=4,N=5,I=6,D=7
3 Jan 1 00:00:25 sys [655]: ANY <0+emerg > ANY =3
4 Jan 1 00:00:25 sys [655]: ANY <0+emerg > Version :7.2.0.10 for release
5 Jan 1 00:00:25 sys [655]: ANY <0+emerg > Built-at :Apr 20 2016,11:32:02
6 May 26 00:00:02 Log [706]: ANY <0+emerg > Log log :sys=1,cons=1,time=0,E=3,W=4,N=5,I=6,D=7
7 May 26 00:00:02 Log [706]: ANY <0+emerg > ETLL=3
8 May 26 00:00:02 auto[706]: ANY <0+emerg > autoServer log :type=1,time=0,E=3,W=4,N=5,I=6,D=7
9 May 26 00:00:02 auto[706]: ANY <0+emerg > ANY =3
10 May 26 00:00:02 auto[706]: ANY <0+emerg > Version :6.1.0.8 for release
11 May 26 00:00:02 auto[706]: ANY <0+emerg > Built-at :May 25 2016,10:26:42
12 May 26 00:00:02 sys [706]: ANY <0+emerg > sys log :type=1,time=0,E=3,W=4,N=5,I=6,D=7
13 May 26 00:00:02 sys [706]: ANY <0+emerg > LSYS=3
14 May 26 00:00:02 ATP [706]: ANY <0+emerg > ATP log :type=1,time=0,E=3,W=4,N=5,I=6,D=7
15 May 26 00:00:02 ATP [706]: ANY <0+emerg > ANY =3
16 May 26 00:00:05 sys [835]: ANY <0+emerg > sys log :type=1,time=0,E=3,W=4,N=5,I=6,D=7
17 May 26 00:00:05 sys [835]: ANY <0+emerg > LSYS=3
18 May 26 00:00:05 sua [835]: ANY <0+emerg > sua log :type=1,time=0,E=3,W=4,N=5,I=6,D=7
19 May 26 00:00:05 sua [835]: ANY <0+emerg > ANY =5
20 May 26 00:00:05 sua [835]: ANY <0+emerg > ANY =3
21 May 26 00:00:06 Log [884]: ANY <0+emerg > Log log :sys=1,cons=0,time=0,E=3,W=4,N=5,I=6,D=7
22 May 26 00:00:06 Log [884]: ANY <0+emerg > ANY =5
23 May 26 00:00:07 ipv[887]: ANY <0+emerg > 807.194.980:ipv[887] log :type=1,time=1,E=3,W=4,N=5,I=6,D=7
24 May 26 00:00:07 ipv[887]: ANY <0+emerg > 807.196.179:Version :1.0.0.8 for release
25 May 26 00:00:07 ipv[887]: ANY <0+emerg > 807.197.104:Built-at :Feb 29 2016,14:11:35
26 May 26 00:00:07 ipv[887]: ANY <0+emerg > 807.198.138:ANY =4
27 May 26 00:00:07 sys [887]: ANY <0+emerg > sys log :type=1,time=0,E=3,W=4,N=5,I=6,D=7
28 May 26 00:00:07 sys [887]: ANY <0+emerg > LSYS=3
29 May 26 00:00:08 TR9 [897]: ANY <0+emerg > TR9 log :sys=1,cons=0,time=0,E=3,W=4,N=5,I=6,D=7

```

The <MAC>-boot.log file is forced to report the logs with all severity levels.

The following figure shows a portion of a <MAC>-sys.log (for example, 00156574b150-sys.log):

```

1 May 31 09:02:05 Log [884]: DSSK<3+error > get page:ExpIndex error![255]
2 May 31 09:02:37 Log [884]: DSSK<3+error > get page:ExpIndex error![255]
3 May 31 09:03:16 Log [884]: DSSK<3+error > get page:ExpIndex error![255]
4 May 31 09:03:27 Log [884]: DSSK<3+error > get page:ExpIndex error![255]
5 May 31 09:03:41 Log [884]: DSSK<3+error > get page:ExpIndex error![255]
6 May 31 09:03:47 Log [884]: DSSK<3+error > get page:ExpIndex error![255]
7 May 31 19:28:18 sys [1076]: ANY <0+emerg > sys log :type=1,time=0,E=3,W=4,N=5,I=6,D=7
8 May 31 19:28:18 sys [1076]: ANY <0+emerg > LSYS=3
9 Jun 1 02:33:52 Log [884]: DSSK<3+error > get page:ExpIndex error![255]
10 Jun 1 07:28:17 sys [1111]: ANY <0+emerg > sys log :type=1,time=0,E=3,W=4,N=5,I=6,D=7
11 Jun 1 07:28:17 sys [1111]: ANY <0+emerg > LSYS=3
12 Jun 1 11:34:57 sua [835]: SUB <3+error > [000] BLF Can't find js by sid(0)
13 Jun 1 11:34:57 sua [835]: SUB <3+error > [000] BLF Can't find js by sid(0)
14 [ web ]
15 step = 2

```

The <MAC>-sys.log file reports the logs with a configured severity level and the higher. For example, if you have configured the severity level of the log to be reported to the <MAC>-sys.log file to 4, then the log with a severity level of 0 to 4 will all be reported.

You can verify whether you got the correct log through the following key fields:

- <0+emerg>
- <1+alert>
- <2+crit>
- <3+error>
- <4+warning>
- <5+notice>
- <6+info>

## Syslog

### Procedure

Syslog can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure syslog feature. <b>Parameter:</b> static.syslog.enable
		Configure syslog server and port. <b>Parameters:</b> static.syslog.server static.syslog.server_port
		Configure the transport protocol that the IP phone uses to upload log to the syslog server. <b>Parameter:</b> static.syslog.transport_type
		Configure the lowest severity level of the logs to be displayed in the syslog. <b>Parameter:</b> static.syslog.level
		Configure the facility that generates the log messages. <b>Parameter:</b> static.syslog.facility
		Configure the IP phone to prepend the MAC address to the log messages uploaded to the syslog server. <b>Parameter:</b>

		static.syslog.prepend_mac_address.enable
<b>Web User Interface</b>		<p>Configure syslog feature.</p> <p>Configure syslog server and port.</p> <p>Configure the transport protocol that the IP phone uses to upload log to the syslog server.</p> <p>Configure the lowest severity level of the logs to be displayed in the syslog.</p> <p>Configure the facility that generates the log messages.</p> <p>Configure the IP phone to prepend the MAC address to the log messages uploaded to the syslog server.</p> <p><b>Navigate to:</b></p> <p>http://&lt;phoneIPAddress&gt;/servlet?p=settings-config&amp;q=load</p>

### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.syslog.enable</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to upload log messages to the syslog server in real time.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Configuration-&gt;Enable Syslog</p> <p><b>Phone User Interface:</b></p> <p>None</p>		
<b>static.syslog.server</b>	<b>IP address or domain name</b>	<b>Blank</b>
<p><b>Description:</b></p> <p>Configures the IP address or domain name of the syslog server.</p> <p><b>Example:</b></p> <p>static.syslog.server = 192.168.1.100</p> <p><b>Web User Interface:</b></p>		

Parameters	Permitted Values	Default
Settings->Configuration->Syslog Server <b>Phone User Interface:</b> None		
<b>static.syslog.server_port</b>	<b>Integer from 1 to 65535</b>	<b>514</b>
<b>Description:</b> Configures the port of the syslog server. <b>Example:</b> static.syslog.port = 515 <b>Web User Interface:</b> Settings->Configuration->Port <b>Phone User Interface:</b> None		
<b>static.syslog.transport_type</b>	<b>0, 1 or 2</b>	<b>0</b>
<b>Description:</b> Configures the transport protocol that the IP phone uses when uploading log messages to the syslog server. <b>0</b> -UDP <b>1</b> -TCP <b>2</b> -TLS <b>Web User Interface:</b> Settings->Configuration->Syslog Transport Type <b>Phone User Interface:</b> None		
<b>static.syslog.level</b>	<b>Integer from 0 to 6</b>	<b>3</b>
<b>Description:</b> Configures the lowest level of syslog information that displays in the syslog. When you choose a log level, you are including all events of an equal or higher severity level and excluding events of a lower severity level. The logging level you choose determines the lowest severity of events to log.		



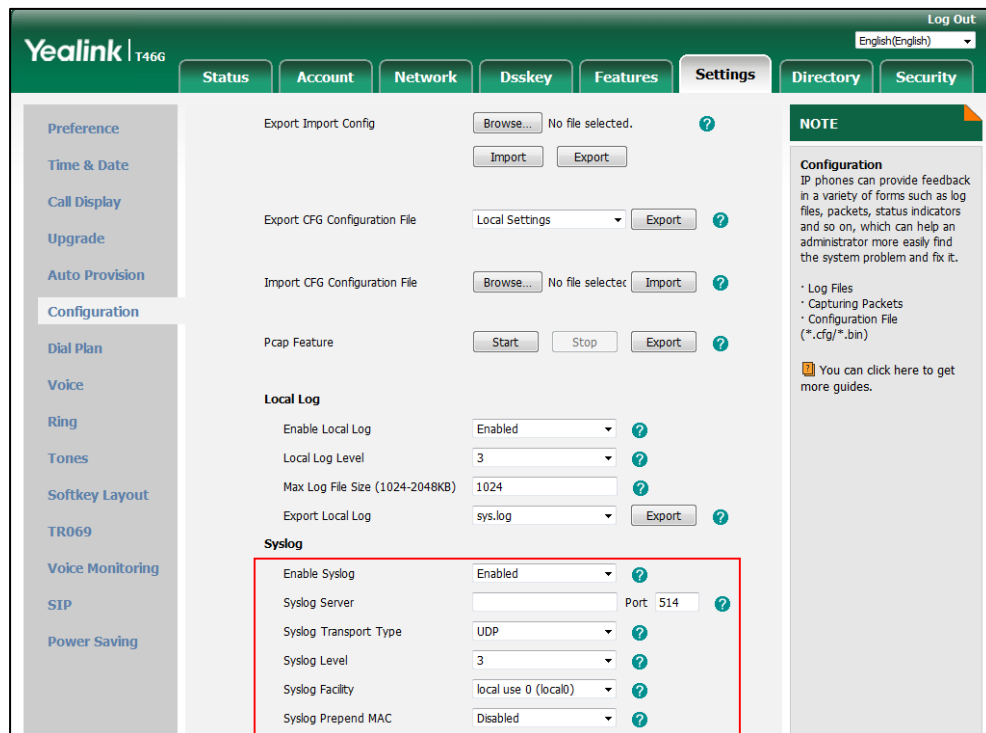
Parameters	Permitted Values	Default
<b>0</b> -Emergency: system is unusable <b>1</b> -Alert: action must be taken immediately <b>2</b> -Critical: critical conditions <b>3</b> -Critical: error conditions <b>4</b> -Warning: warning conditions <b>5</b> -Warning: normal but significant condition <b>6</b> -Informational: informational messages <b>Web User Interface:</b> Settings->Configuration->Syslog Level <b>Phone User Interface:</b> None		
<b>static.syslog.facility</b>	<b>Integer from 0 or 23</b>	<b>16</b>
<b>Description:</b> Configures the facility that generates the log messages. <b>0</b> -kernel messages <b>1</b> -user-level messages <b>2</b> -mail system <b>3</b> -system daemons <b>4</b> -security/authorization messages (note 1) <b>5</b> -messages generated internally by syslogd <b>6</b> -line printer subsystem <b>7</b> -network news subsystem <b>8</b> -UUCP subsystem <b>9</b> -clock daemon (note 2) <b>10</b> -security/authorization messages (note 1) <b>11</b> -FTP daemon <b>12</b> -NTP subsystem <b>13</b> -log audit (note 1) <b>14</b> -log alert (note 1) <b>15</b> -clock daemon (note 2) <b>16</b> -local use 0 (local0) <b>17</b> -local use 1 (local1)		

Parameters	Permitted Values	Default
<b>18</b> -local use 2 (local2) <b>19</b> -local use 3 (local3) <b>20</b> -local use 4 (local4) <b>21</b> -local use 5 (local5) <b>22</b> -local use 6 (local6) <b>23</b> -local use 7 (local7) <b>Note:</b> For more information, refer to <a href="#">RFC 3164</a> . <b>Web User Interface:</b> Settings->Configuration->Syslog Facility <b>Phone User Interface:</b> None		
<b>static.syslog.prepend_mac_address.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the IP phone to prepend the MAC address to the log messages uploaded to the syslog server. <b>0</b> -Disabled <b>1</b> -Enabled <b>Web User Interface:</b> Settings->Configuration->Syslog Prepend MAC <b>Phone User Interface:</b> None		

**To configure the phone to upload the system log to a syslog server via web user interface:**

1. Click on **Settings->Configuration**.
2. Select the desired value from the pull-down list of **Enable Syslog**.
3. Enter the syslog server address in the **Syslog Server** field.
4. Enter the syslog server port in the **Port** field.
5. Select the desired transport type from the pull-down list of **Syslog Transport Type**.
6. Select the desired log level from the pull-down list of **Syslog Level**.
7. Select the desired facility from the pull-down list of **Syslog Facility**.

- Select the desired value from the pull-down list of **Syslog Prepend MAC**.



- Click **Confirm** to accept the change.

#### To view the syslog messages on your syslog server:

You can view the syslog file in the desired folder on the syslog server. The location of the folder may differ from the syslog server. For more information, refer to the network resources.

The following figure shows a portion of the syslog:

Jun 02 08:42:17	10.2.20.160	local0.notice	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: APP <5+notice> [SIP] dtmf_payload :101
Jun 02 08:42:17	10.2.20.160	local0.notice	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: APP <5+notice> [SIP] version :0
Jun 02 08:42:17	10.2.20.160	local0.notice	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: APP <5+notice> [SIP] call channels info
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000] cb_nict_kill_transaction (id=88)
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000] m=audio 7150 RTP/AVP 9 0 8 18 101
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000] Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REGISTER, SUBSCRIBE, NOTIFY,
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000] CSeq: 4 INVITE
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000] Call-ID: ZWQ3MWM5ZDgwZDMyMmZjY2lkN2YyMzQ1NTJiNWl5Nzg.
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000] From: <sip:101@10.2.1.43:5060>tag=4086693836
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000] To: '102' <sip:102@10.2.1.43:5060>tag=8d378436
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000] Contact: <sip:102@10.2.1.43:5060>
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000] Via: SIP/2.0/UDP 10.2.20.160:5060;branch=z9hG4bK2209216298
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000] SIP/2.0 200 OK
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000]
Jun 02 08:42:17	10.2.20.160	local0.notice	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <5+notice> [000] Message rcv: (from src=10.2.1.43:5060 len=808)
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: SIP <6+notice> [SIP] match line:101 host:10.2.1.43
Jun 02 08:42:17	10.2.20.160	local0.notice	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: NET <5+notice> [255] <<<== UDP socket 10.2.1.43:5060: read 808 bytes
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: SUA <6+info > [000] ****eCore event(0x0010)ECORE_CALL_PROCEEDING ****
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000]
Jun 02 08:42:17	10.2.20.160	local0.info	Jun 2 00:42:48	[00:15:65:74:b1:50]	sua [845]: DLG <6+info > [000]

## Capturing Packets

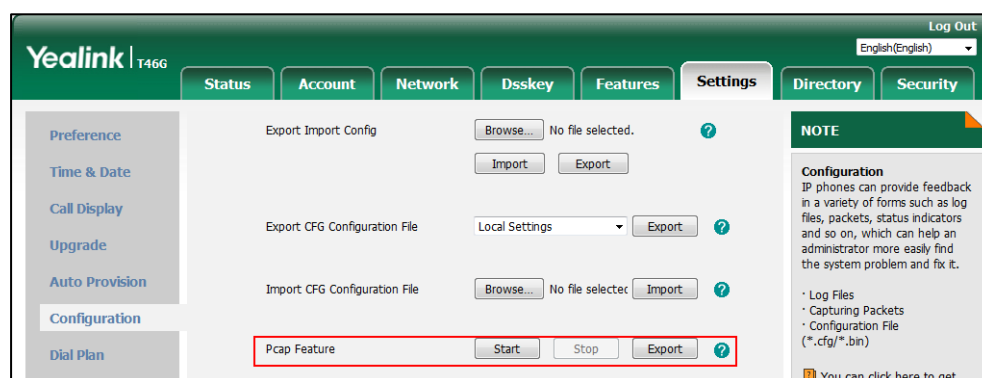
You can capture packet in two ways: capturing the packets via web user interface or using the Ethernet software. You can analyze the packet captured for troubleshooting purpose.

## Capturing the Packets via Web User Interface

For Yealink IP phones, you can export the packets file to the local system and analyze it.

**To capture packets via web user interface:**

1. Click on **Settings->Configuration**.
2. Click **Start** in the **Pcap Feature** field to start capturing signal traffic.
3. Reproduce the issue to get stack traces.
4. Click **Stop** in the **Pcap Feature** field to stop capturing.
5. Click **Export** to open the file download window, and then save the file to your local system.



## Capturing the Packets Using the Ethernet Software

### Receiving data packets from the HUB

Connect the Internet port of the IP phone and the PC to the same HUB, and then use Sniffer, Ethereal or Wireshark software to capture the signal traffic.

### Receiving data packets from PC port

Connect the Internet port of the IP phone to the Internet and the PC port of the IP phone to a PC. Before capturing the signal traffic, make sure the data packets can be received from the Internet port to the PC port.

### Procedure

Span to PC port can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure span to PC port. <b>Parameter:</b> static.network.span_to_pc_port
<b>Web User Interface</b>		Configure span to PC port. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p

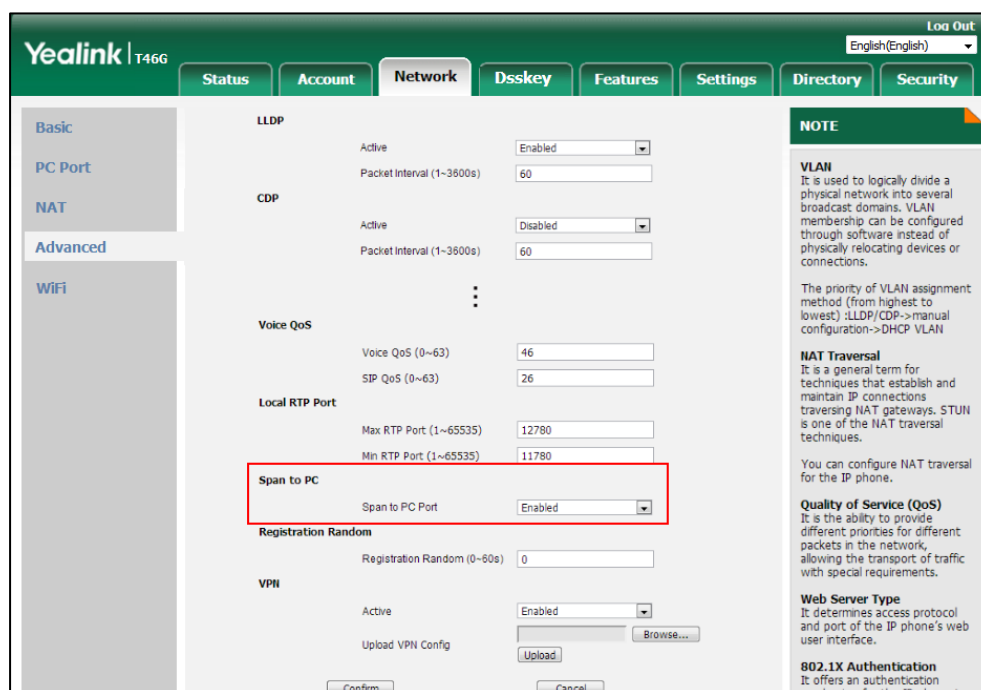
=network-adv&amp;q=load

**Details of the Configuration Parameter:**

Parameter	Permitted Values	Default
<b>static.network.span_to_pc_port</b>	<b>0 or 1</b>	<b>0</b>
<p><b>Description:</b></p> <p>Enables or disables the IP phone to span data packets received from the Internet port to the PC port.</p> <p><b>0</b>-Disabled</p> <p><b>1</b>-Enabled, all data packets from Internet port can be received by PC port.</p> <p><b>Note:</b> It works only if the value of the parameter "static.network.pc_port.enable" is set to 1 (Auto Negotiation). If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Network-&gt;Advanced-&gt;Span to PC-&gt;Span to PC Port</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

**To enable span to PC port via web user interface:**

1. Click on **Network->Advanced**.
2. Select **Enabled** from the pull-down list of **Span to PC Port**.



3. Click **Confirm** to accept the change.

A dialog box pops up to prompt that the settings will take effect after a reboot.

4. Click **OK** to reboot the phone.

Then you can use Sniffer, Ethereal or Wireshark software to capture the signal traffic.

## Enabling Watch Dog Feature

The IP phone provides a troubleshooting feature called "Watch Dog", which helps you monitor the IP phone status and provides the ability to get stack traces from the last time the IP phone failed. If Watch Dog feature is enabled, the IP phone will automatically reboot when it detects a fatal failure. This feature can be configured using the configuration files or via web user interface.

### Procedure

Watch Dog can be configured using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure Watch Dog feature. <b>Parameter:</b> static.watch_dog.enable
<b>Web User Interface</b>		Configure Watch Dog feature. <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=settings-preference&q=load

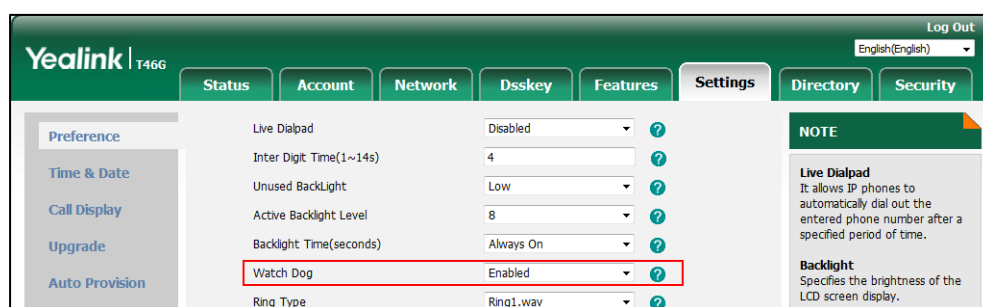
### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
static.watch_dog.enable	0 or 1	1
<b>Description:</b> Enables or disables the Watch Dog feature. <b>0</b> -Disabled <b>1</b> -Enabled, the IP phone will reboot automatically when the system is broken down. <b>Web User Interface:</b> Settings->Preference->Watch Dog <b>Phone User Interface:</b> None		

### To configure watch dog feature via web user interface:

1. Click on **Settings->Preference**.

2. Select the desired value from the pull-down list of **Watch Dog**.




3. Click **Confirm** to accept the change.

## Getting Information from Status Indicators

Status indicators may consist of the power LED, MESSAGE key LED, line key indicator, headset key indicator and the on-screen icon.

The following shows two examples of obtaining the IP phone information from status indicators on SIP-T46G IP phones:

- If a LINK failure of the IP phone is detected, a prompting message "Network unavailable" and the icon  will appear on the LCD screen.
- If a call is muted, the MUTE key LED illuminates.

For more information on the icons, refer to [Appendix G: Reading Icons](#) on page 899.

## Analyzing Configuration Files

Wrong configurations may have an impact on your phone use. You can export configuration file(s) to check the current configuration of the IP phone and troubleshoot if necessary. You can also import configuration files for a quick and easy configuration.

Six types of configuration files can be exported to your local system:

- config.bin
- <MAC>-all.cfg
- <MAC>-local.cfg
- <MAC>-static.cfg
- <MAC>-non-static.cfg
- <MAC>-config.cfg

We recommend you to edit the exported CFG file instead of the BIN file to change the phone's current settings if your phone is running firmware version 73 or later. For more information on configuration files, refer to [Configuration Files](#) on page 101.

## BIN Configuration Files

The config.bin file is an encrypted file. For more information on config.bin file, contact your Yealink reseller.

### Procedure

Configuration changes can be performed using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Specify the access URL for the custom configuration files.  <b>Parameter:</b> static.configuration.url
<b>Web User Interface</b>		Export or import the custom configuration files.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=settings-config&q=load

### Details of the Configuration Parameter:

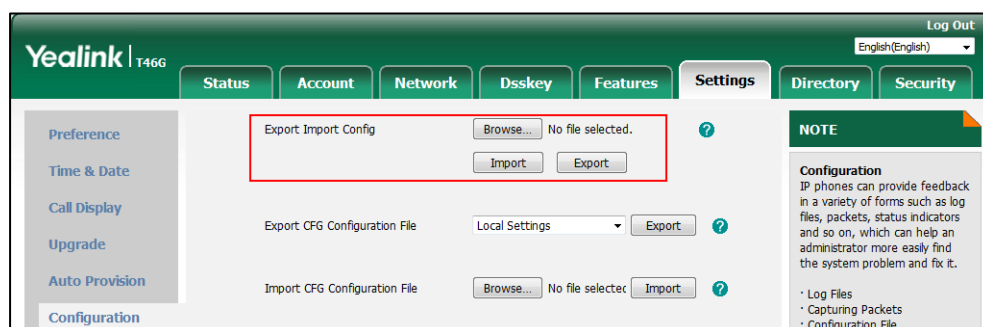
Parameter	Permitted Values	Default
static.configuration.url	URL within 511 characters	Blank
<b>Description:</b> Configures the access URL for the custom configuration files.  <b>Note:</b> The file format of custom configuration file must be *.bin. If you change this parameter, the IP phone will reboot to make the change take effect.  <b>Web User Interface:</b> Settings->Configuration->Export or Import Configuration  <b>Phone User Interface:</b> None		

### To export BIN configuration files via web user interface:

1. Click on **Settings->Configuration**.

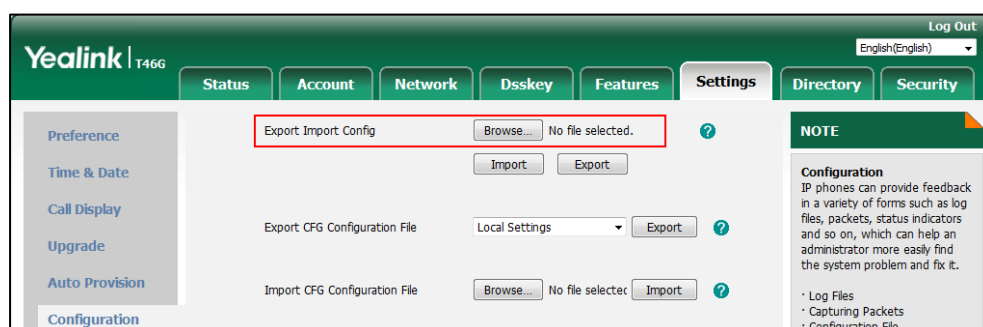


2. In the **Export or Import Configuration** block, click **Export** to open the file download window, and then save the file to your local system.



**To import a BIN configuration file via web user interface:**

1. Click on **Settings->Configuration**.
2. In the **Export or Import Configuration** block, click **Browse** to locate a BIN configuration file from your local system.



3. Click **Import** to import the configuration file.

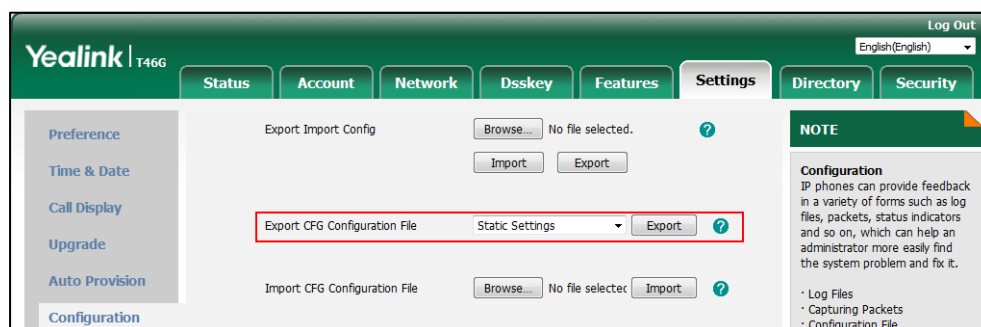
## CFG Configuration Files

Five CFG configuration files can be exported:

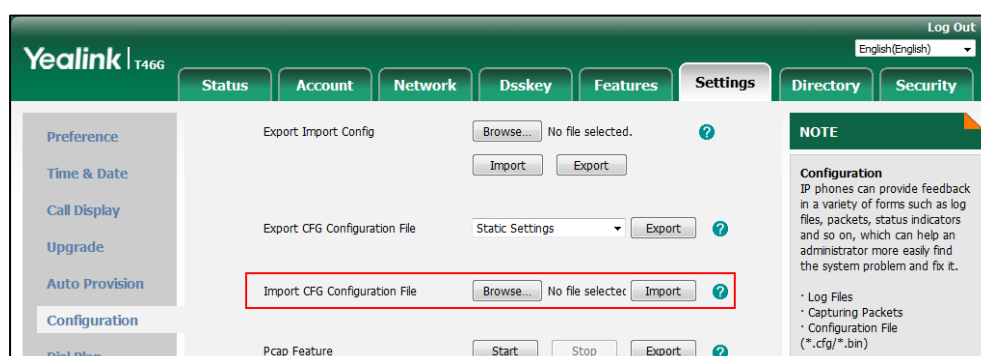
- **<MAC>-local.cfg**: It contains changes associated with non-static settings made via phone user interface and web user interface. It can be exported only if the value of the parameter "static.auto\_provision.custom.protect" is set to 1.
- **<MAC>-all.cfg**: It contains all changes made via phone user interface, web user interface and using configuration files.
- **<MAC>-static.cfg**: It contains all changes associated with static settings (for example, network settings) made via phone user interface, web user interface and using configuration files.
- **<MAC>-non-static.cfg**: It contains all changes associated with non-static settings made via phone user interface, web user interface and using configuration files.
- **<MAC>-config.cfg**: It contains changes associated with non-static settings made using configuration files. It can be exported only if the value of the parameter "static.auto\_provision.custom.protect" is set to 1.

**To export CFG configuration files via web user interface:**

1. Click on **Settings->Configuration**.
2. Select the desired CFG configuration file from the pull-down list of **Export CFG Configuration File**.
3. Click **Export** to open file download window, and then save the file to your local system.

**To import CFG configuration files via web user interface:**

1. Click on **Settings->Configuration**.
2. In the **Import CFG Configuration File** block, click **Browse** to locate a CFG configuration file from your local system.



3. Click **Import** to import the configuration file.

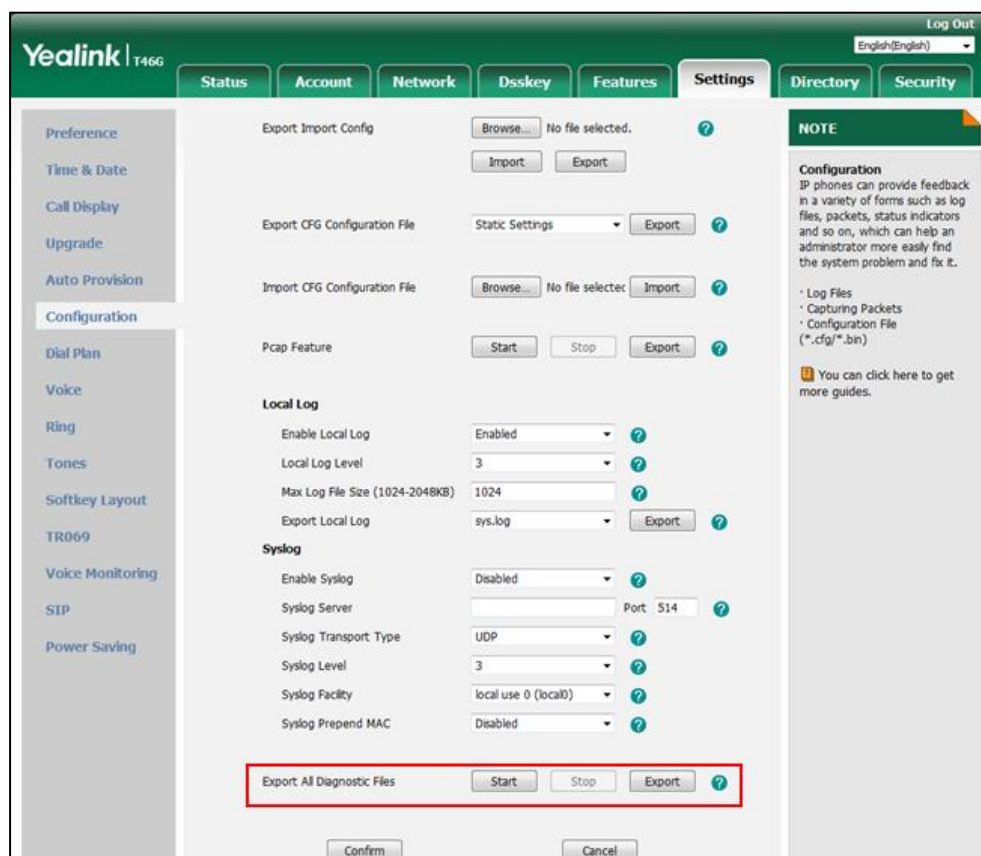
## Exporting All the Diagnostic Files

Yealink IP phones support three types of diagnostic files (including Pcap trace, log files (boot.log and sys.log) and BIN configuration files) to help analyze your problem. You can export these files at a time and troubleshoot if necessary. The file format of exported diagnostic file is \*.tar.

**To export all diagnostic files via web user interface:**

1. Click on **Settings->Configuration**.
2. Click **Start** in the **Export All Diagnostic Files** field to begin capturing signal traffic.  
The system log level will be automatically set to 6.
3. Reproduce the issue.

4. Click **Stop** in the **Export All Diagnostic Files** field to stop the capture.  
The system log level will be reset to 3.
5. Click **Export** to open file download window, and then save the diagnostic file to your local system.



A diagnostic file named **allconfig.tgz** is successfully exported to your local system.

#### Note

If the issue cannot be reproduced, just directly click **Export** to export all diagnostic files.

#### To view the diagnostic file on your local system:

1. Extract the combined diagnostic files to your local system.
2. Open the folder you extracted to and identify the files you will view.

You can select to export the Pcap trace, log files (boot.log and sys.log) and BIN configuration files respectively.

For more information, refer to [Capturing Packets](#) on page 841, [Viewing Log Files](#) on page 827 and [BIN Configuration Files](#) on page 846.

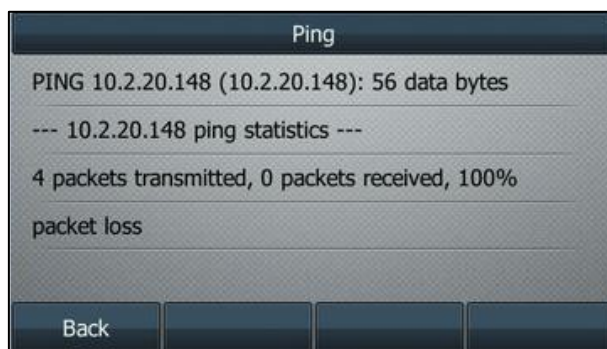
## Ping and Traceroute

You can use ping and traceroute diagnostics for troubleshooting network connectivity problems via phone user interface.

**To capture ping packets via phone user interface:**

1. Press **Menu->Features->Diagnostics->Network->Ping**.
2. Enter an IP address or a URL address (for example, 10.2.20.148) in the **Ping IP or URL** field.
3. Press the **Start** soft key to start a ping.

The LCD screen displays as below:



4. Press or to view more network status information or press the **Back** soft key to back to the previous screen.

You can use the ping statistics information to check the network connection situation.

The following table shows the network connection situation and ping statistics:

Ping Statistics	Network Connection Situation
4 packets transmitted, 4 packets received, 0% packet loss	The network connection between you and the target is strong.
4 packets transmitted, 1, 2 or 3 packets received, 75%, 50% or 25% packet loss	The network connection between you and the target is unstable or faulty.
4 packets transmitted, 0 packets received, 100% packet loss	The network connection between you and the target is unreachable.

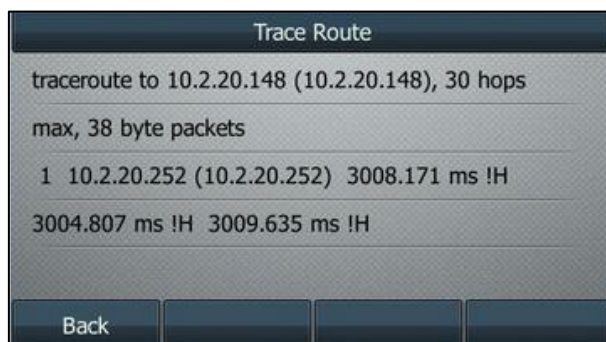
If you find that the network connection between you and the target is unreachable, you can trace route of the target to see where the problem is.

**To trace route packets via phone user interface:**

1. Press **Menu->Features->Diagnostics->Network->Trace Route**.
2. Enter an IP address or a URL address (for example, 10.2.20.148) in the **Tracert IP or URL** field.

3. Press the **Start** soft key to start tracing the route.

The LCD screen displays as below:



It shows the target IP or URL, the maximum number of hops that will be allowed, and the size of the packets being sent.

4. Press or to view more network status information or press the **Back** soft key to back to the previous screen.

A traceroute can end with one of several error indications indicating why the traceroute cannot proceed. The following table shows the possible error messages:

Error Message	Meaning
* * *	Target unreachable.
!H	Host unreachable.
!N	Network unreachable.

## Troubleshooting Solutions

This section describes solutions to common issues that may occur while using the IP phone. Upon encountering a scenario not listed in this section, contact your Yealink reseller for further support.

### IP Address Issues

#### Why doesn't the IP phone get an IP address?

Do one of the following:

If your phone connects to the wired network:

- Ensure that the Ethernet cable is plugged into the Internet port on the IP phone and the Ethernet cable is not loose.
- Ensure that the Ethernet cable is not damaged.
- Ensure that the IP address and related network parameters are set correctly.
- Ensure that your network switch or hub is operational.

- Ensure that the Wi-Fi feature is disabled.

If your phone connects to the wireless network:

- If the network is secure, ensure the entered password is right.
- Ensure your gateway/router enables the wireless network feature.

## How to solve the IP conflict problem?

Do one of the following:

- Reset another available IP address for the IP phone.
- Check network configuration via phone user interface at the path **Menu->Advanced** (default password: admin) ->**Network->WAN Port->IPv4** (or **IPv6**). If the Static IP is selected, select DHCP instead.

## Is there a specific format in configuring IPv6 on Yealink IP phones?

### Scenario 1:

If the IP phone obtains the IPv6 address, the format of the URL to access the web user interface is "*[IPv6 address]*" or "*http(s)://[IPv6 address]*". For example, if the IPv6 address of your phone is "fe80::204:13ff:fe30:10e", you can enter the URL (for example, "[fe80::204:13ff:fe30:10e]" or "http(s)://[fe80::204:13ff:fe30:10e]") in the address bar of a web browser on your PC to access the web user interface.

### Scenario 2:

Yealink IP phones support using FTP, TFTP, HTTP and HTTPS protocols to download configuration files or resource files. You can use one of these protocols for provisioning.

When provisioning your IP phone obtaining an IPv6 address, the provisioning server should support IPv6 and the format of the access URL of the provisioning server can be "*tftp://[IPv6 address or domain name]*". For example, if the provisioning server address is "2001:250:1801::1", the access URL of the provisioning server can be "tftp://[2001:250:1801::1]". For more information on provisioning, refer to [Yealink\\_SIP\\_IP\\_Phones\\_Auto\\_Provisioning\\_Guide\\_V81](#).

## Time and Date Issues

### Why doesn't the IP phone display time and date correctly?

Check if the IP phone is configured to obtain the time and date from the NTP server automatically. If your phone is unable to access the NTP server, configure the time and date manually.

## Display Issues

### Why is the LCD screen blank?

Do one of the following:

- Ensure that the IP phone is properly plugged into a functional AC outlet.
- Ensure that the IP phone is plugged into a socket controlled by a switch that is on.
- If the IP phone is plugged into a power strip, try plugging it directly into a wall outlet.
- If your phone is PoE powered, ensure that you are using a PoE-compliant switch or hub.

### Why does the IP phone display “No Service”?

The LCD screen prompts “No Service” message when there is no available SIP account on the IP phone.

Do one of the following:

- Ensure that an account is actively registered on the IP phone at the path **Menu->Status->More->Accounts**.
- Ensure that the SIP account parameters have been configured correctly.

## Phone Book Issues

### What is the difference between a remote phone book and a local phone book?

A remote phone book is placed on a server, while a local phone book is placed on the IP phone flash. A remote phone book can be used by everyone that can access the server, while a local phone book can only be used by a specific phone. A remote phone book is always used as a central phone book for a company; each employee can load it to obtain the real-time data from the same server.

## Audio Issues

### How to increase or decrease the volume?

Press the volume key to increase or decrease the ringer volume when the IP phone is idle or ringing, or to adjust the volume of engaged audio device (handset, speakerphone or headset) when there is an active call in progress.

### Why do I get poor sound quality during a call?

If you have poor sound quality/acoustics like intermittent voice, low volume, echo or other

noises, the possible reasons could be:

- Users are seated too far out of recommended microphone range and sound faint, or are seated too close to sensitive microphones and cause echo.
- Intermittent voice is mainly caused by packet loss, due to network congestion, and jitter, due to message recombination of transmission or receiving equipment (for example, timeout handling, retransmission mechanism, buffer under run).
- Noisy equipment, such as a computer or a fan, may cause voice interference. Turn off any noisy equipment.
- Line issues can also cause this problem; disconnect the old line and redial the call to ensure another line may provide better connection.

### **Why is there no sound when the other party picks up the call?**

If the caller and receiver cannot hear anything - there is no sound at all when the other party picks up the call, the possible reason could be: the phone cannot send the real-time transport protocol (RTP) streams, in which audio data is transmitted, to the connected call.

Try to disable the 180 ring workaround feature. For more information, refer to [180 Ring Workaround](#) on page 382.

### **Why does the IP phone play the local ringback tone instead of media when placing a long distance number without plus 0?**

Ensure that the 180 ring workaround feature is disabled. For more information, refer to [180 Ring Workaround](#) on page 382.

## **Wi-Fi and Bluetooth Issues**

### **Why is the wireless signal strength low?**

Ensure the IP phone and your gateway/ router are within the working range and there is no obvious interference (walls, doors, and so on.) between them.

### **Why can't I connect the Bluetooth device with the IP phone all the time?**

Try to delete the registration information of the Bluetooth device on both IP phone and Bluetooth device, and then pair and connect it again. Contact Yealink field application engineer and your Bluetooth device manufacturer for more information.

### **Why the Bluetooth headset affects IP phone's voice quality?**

You may not experience the best voice quality if you use a Bluetooth headset while the 2.4 GHz band is enabled or while you are in an environment with many other Bluetooth devices. This possible loss in voice quality is due to inherent limitations with Bluetooth technology.



## Firmware and Upgrading Issues

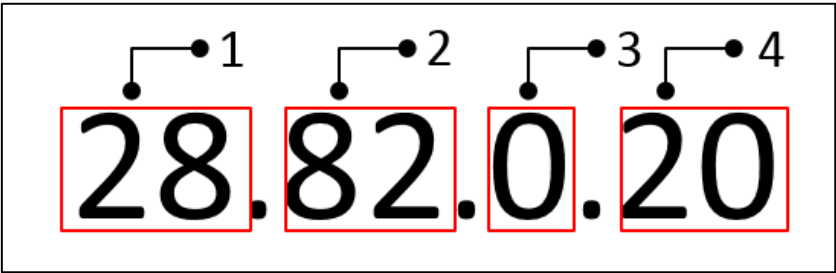
### Why doesn't the IP phone upgrade firmware successfully?

Do one of the following:

- Ensure that the target firmware is not the same as the current firmware.
- Ensure that the target firmware is applicable to the IP phone model.
- Ensure that the current or the target firmware is not protected.
- Ensure that the power is on and the network is available in the process of upgrading.
- Ensure that the web browser is not closed or refreshed when upgrading firmware via web user interface.

### How can I verify the firmware generation and version of the phone?

Press the **OK/√** key when the IP phone is idle to check the firmware version. For example:  
28.82.0.20.



	Item	Description
1	28	<p>Firmware ID.</p> <p>The firmware ID for each IP phone model is:</p> <ul style="list-style-type: none"><li>• 70: SIP-T54S/T52S</li><li>• 35: SIP-T48G</li><li>• 28: SIP-T46G</li><li>• 66: SIP-T48S/T46S/T42S/T41S</li><li>• 29: SIP-T42G</li><li>• 36: SIP-T41P</li><li>• 54: SIP-T40P</li><li>• 76: SIP-T40G</li><li>• 46: SIP-T29G</li><li>• 45: SIP-T27P</li><li>• 69: SIP-T27G</li></ul>

	Item	Description
		<ul style="list-style-type: none"> <li>• 44: SIP-T23P/G</li> <li>• 52: SIP-T21(P) E2</li> <li>• 53: SIP-T19(P) E2</li> </ul>
2	82	Firmware generation. <b>Note:</b> The larger it is, the newer the firmware generation is.
3	0	A fixed number.
4	20	Firmware version. <b>Note:</b> With the same firmware generation, the larger it is, the newer the firmware version is.

## Why doesn't the IP phone update the configuration?

Do one of the following:

- Ensure that the configuration is set correctly.
- Reboot the phone. Some configurations require a reboot to take effect.
- Ensure that the configuration is applicable to the IP phone model.
- The configuration may depend on support from a server.

## Provisioning Issues

### What is auto provisioning?

Auto provisioning refers to the update of IP phones, including update on configuration parameters, local phone book, firmware and so on. You can use auto provisioning on a single phone, but it makes more sense in mass deployment.

### What is PnP?

Plug and Play (PnP) is a method for IP phones to acquire the provisioning server address. With PnP enabled, the IP phone broadcasts the PnP SUBSCRIBE message to obtain a provisioning server address during startup. Any SIP server recognizing the message will respond with the preconfigured provisioning server address, so the IP phone will be able to download the CFG files from the provisioning server. PnP depends on support from a SIP server.

## System Log Issues

### Why can't I export the system log to a provisioning server (FTP/TFTP server)?

Do one of the following:

- Ensure that the FTP/TFTP server is downloaded and installed on your local system.
- Ensure that you have configured the FTP/TFTP server address correctly via web user interface on your IP phone.
- Reboot the phone. The configurations require a reboot to take effect.

### Why can't I export the system log to a syslog server?

Do one of the following:

- Ensure that the syslog server supports saving the syslog files exported from IP phone.
- Ensure that you have configured the syslog server address correctly via web user interface on your IP phone.
- Reboot the phone. The configurations require a reboot to take effect.

## Resetting Issues

Generally, some common issues may occur while using the IP phone. You can reset your phone to factory configurations after you have tried all troubleshooting suggestions but do not solve the problem. Resetting the phone to factory configurations clears the flash parameters, removes log files, user data, and cached data, and resets the administrator password to admin. All custom settings will be overwritten after resetting.

Five ways to reset the phone:

- **Reset local settings:** All configurations saved in the <MAC>-local.cfg file on the IP phone will be reset. Changes associated with non-static settings made via web user interface and phone user interface are saved in the <MAC>-local.cfg file.
- **Reset non-static settings:** All non-static settings on the phone will be reset. After resetting the non-static settings, the IP phone will perform the auto provisioning process immediately.
- **Reset static settings:** All static settings on the phone will be reset.
- **Reset userdata & local config:** All the local cache data (for example, userdata, history, directory) will be cleared. And all configurations saved in the <MAC>-local.cfg configuration file on the IP phone will be reset.
- **Reset to factory:** All configurations on the phone will be reset.

You can reset the IP phone to default factory configurations. The default factory configurations

are the settings that reside on the IP phone after it has left the factory. You can also reset the IP phone to custom factory configurations if required. The custom factory configurations are the settings that defined by the user to keep some custom settings after resetting. You have to import the custom factory configuration files in advance.

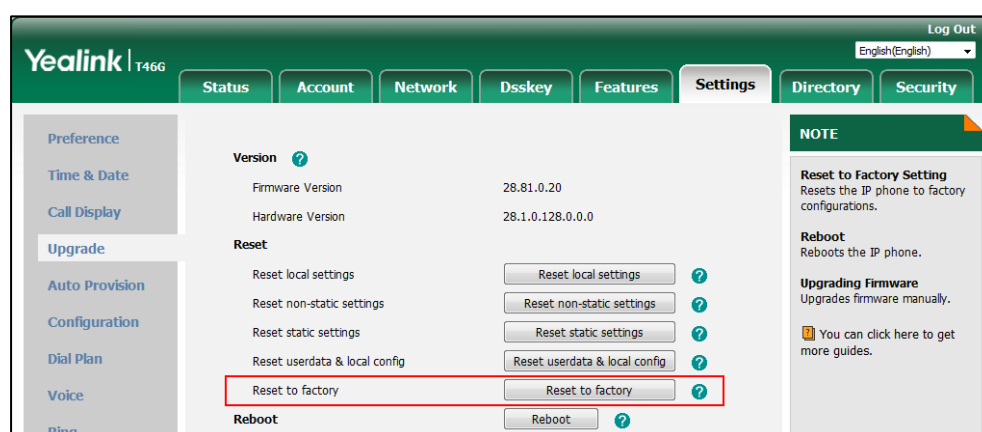
#### Note

The **Reset local settings/Reset non-static settings/Reset static settings/Reset userdata & local config** option on the web user interface appears only if the value of the parameter "static.auto\_provision.custom.protect" is set to 1.

## How to reset the IP phone to default factory configurations?

To reset the IP phone via web user interface:

1. Click on **Settings->Upgrade**.
2. Click **Reset to factory** in the **Reset to factory** field.



The web user interface prompts the message "Do you want to reset to factory?".

3. Click **OK** to confirm the resetting.

The IP phone will be reset to factory successfully after startup.

#### Note

Reset of your phone may take a few minutes. Do not power off until the phone starts up successfully.

## How to reset the IP phone to custom factory configurations?

### Procedure

Configuration changes can be performed using the following methods.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cf g	Configure the Custom Factory Configuration feature.  <b>Parameter:</b> static.features.custom_factory_config.enable
		Configure the access URL of the custom factory configuration files.  <b>Parameter:</b> static.custom_factory_configuration.url
<b>Web User Interface</b>		Configure the access URL of the custom factory configuration files.  <b>Navigate to:</b> http://<phoneIPAddress>/servlet?p=settings-custom_factory_configuration&q=load

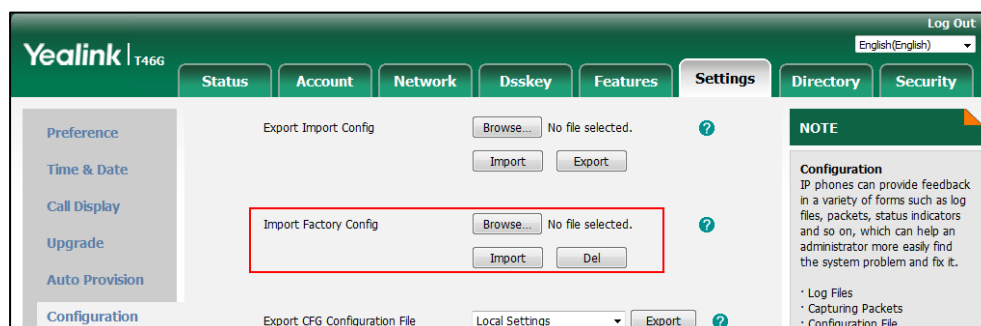
### Details of Configuration Parameters:

Parameters	Permitted Values	Default
<b>static.features.custom_factory_config.enable</b>	<b>0 or 1</b>	<b>0</b>
<b>Description:</b> Enables or disables the Custom Factory Configuration feature. <b>0</b> -Disabled <b>1</b> -Enabled, <b>Import Factory Configuration</b> item will be displayed on the IP phone's web user interface at the path <b>Settings-&gt;Configuration</b> . You can import a custom factory configuration file or delete the user-defined factory configuration via web user interface. <b>Web User Interface:</b> None <b>Phone User Interface:</b> None		
<b>static.custom_factory_configuration.url</b>	<b>URL within 511 characters</b>	<b>Blank</b>
<b>Description:</b> Configures the access URL of the custom factory configuration files.		

Parameters	Permitted Values	Default
<p><b>Note:</b> It works only if the value of the parameter “static.features.custom_factory_config.enable” is set to 1 (Enabled) and the file format of custom factory configuration file must be *.bin. If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p><b>Web User Interface:</b></p> <p>Settings-&gt;Configuration-&gt;Import Factory Config</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

### To import the custom factory configuration files via web user interface:

1. Click on **Settings->Configuration**.
2. Click **Browse** to locate the custom factory configuration file from your local system.



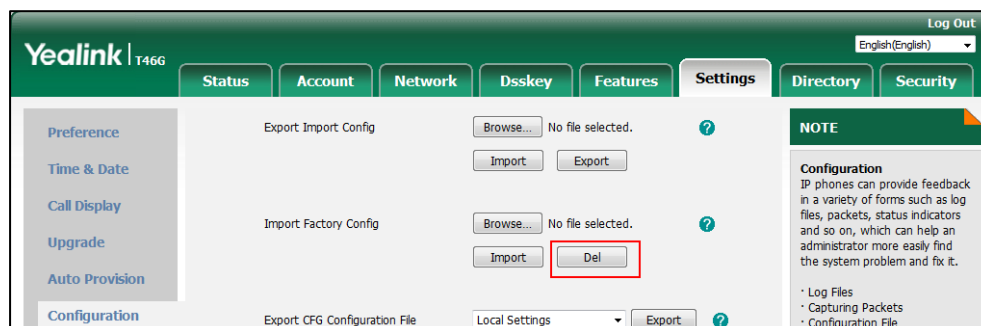
3. Click **Import**.

When the custom factory configuration file is imported successfully, you can reset the IP phone to custom factory configurations. For more information on how to reset to factory configuration via web user interface, refer to [How to reset the IP phone to default factory configurations?](#) on page 858.

You can delete the user-defined factory configurations via web user interface.

### To delete the custom factory configuration files via web user interface:

1. Click on **Settings->Configuration**.
2. Click **Del** in the **Import Factory Config** field.



The web user interface prompts the message "Are you sure delete user-defined factory configuration?".

3. Click **OK** to delete the custom factory configuration files.

The imported custom factory file will be deleted. The IP phone will be reset to default factory configurations after resetting.

## Rebooting Issues

### How to reboot the IP phone remotely?

You can reboot the IP phones remotely using a SIP NOTIFY message with "Event: check-sync" header. Whether the IP phone reboots or not depends on the value of the parameter "sip.notify\_reboot\_enable". If the value is set to 1, or the value is set to 0 and the header of the SIP NOTIFY message contains an additional string "reboot=true", the IP phone will reboot immediately.

The NOTIFY message is formed as shown:

```
NOTIFY sip:<user>@<dsthost> SIP/2.0
To: sip:<user>@<dsthost>
From: sip:sipsak@<srchost>
CSeq: 10 NOTIFY
Call-ID: 1234@<srchost>
Event: check-sync;reboot=true
```

### Procedure

Changes can only be configured using the configuration file.

<b>Central Provisioning (Configuration File)</b>	<y0000000000xx>.cfg	Configure the IP phone behavior when receiving a SIP NOTIFY message which contains the header "Event: check-sync".  <b>Parameter:</b> sip.notify_reboot_enable
------------------------------------------------------	---------------------	-------------------------------------------------------------------------------------------------------------------------------------------------------------------------

### Details of the Configuration Parameter:

Parameter	Permitted Values	Default
sip.notify_reboot_enable	0, 1 or 2	1
<b>Description:</b>		

Parameter	Permitted Values	Default
<p>Configure the IP phone behavior when receiving a SIP NOTIFY message which contains the header "Event: check-sync".</p> <p><b>0</b>-The IP phone will reboot only if the SIP NOTIFY message contains an additional string "reboot=true".</p> <p><b>1</b>-The IP phone will be forced to reboot.</p> <p><b>2</b>-The IP phone will ignore the SIP NOTIFY message.</p> <p><b>Web User Interface:</b></p> <p>None</p> <p><b>Phone User Interface:</b></p> <p>None</p>		

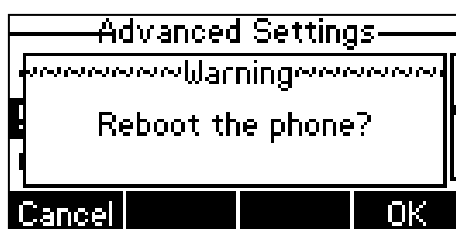
## How to reboot the IP phone via web/phone user interface?

You can reboot your IP phone via web/phone user interface.

### To reboot the phone via phone user interface:

1. Press **Menu**->**Advanced** (default password: admin) ->**Reboot**.

The LCD screen prompts the following warning:

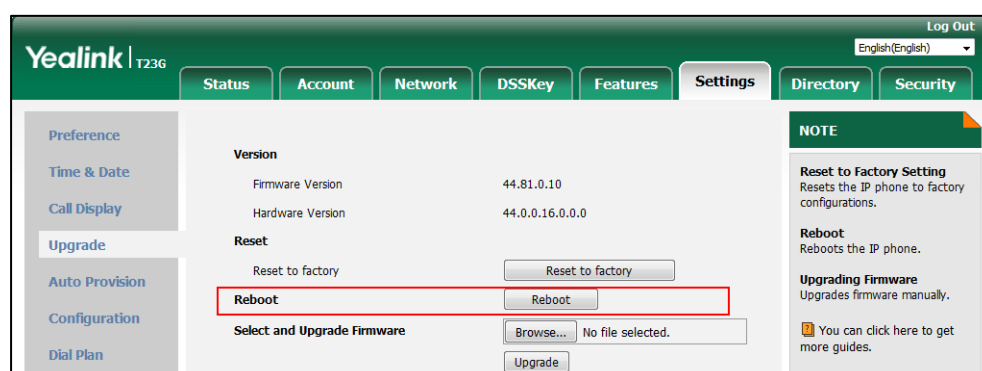


2. Press the **OK** soft key to reboot the phone.

The phone begins rebooting. Any reboot of the phone may take a few minutes.

### To reboot the phone via web user interface:

1. Click on **Settings**->**Upgrade**.
2. Click **Reboot** to reboot the IP phone.





The phone begins rebooting. Any reboot of the phone may take a few minutes.

## Protocols and Ports Issues

### What communication protocols and ports do Yealink IP phones support?

Source Device	Source IP	Source Port	Destination Device	Destination IP	Destination Port (Listening port)	Protocol	Description of destination port
IP phones	IP address of IP phones	2~65535	IP phone or voice gateway	IP address of IP phone or voice gateway	Determined by destination device.	UDP	RTP protocol port, it is used to send or receive audio stream.
		1024~65535	SIP Server	IP address of SIP server	Determined by destination device.	UDP/TCP	SIP protocol port, it is used for signaling interaction with SIP server.
		1024~65535	TR-069 Server	IP address of TR-069 server	Determined by destination device.	TCP	TR-069 protocol port, it is used to communicate with TR-069server.
		1024~65535	File server	IP address of file server	Determined by destination device.	TCP	HTTP protocol port, it is used to download file.
		1024~65535	Remote phone book server	IP address of remote phone book server	Determined by destination device.	TCP	HTTP protocol port, it is used to access the remote phone book.
		1024~65535	AA	IP address of AA	Determined by destination device.	TCP	HTTP protocol port, it is used for AA communication.
		68	DHCP Server	IP address of DHCP server	67	UDP	DHCP protocol port, it is used to obtain IP address from DHCP server.

Source Device	Source IP	Source Port	Destination Device	Destination IP	Destination Port (Listening port)	Protocol	Description of destination port
		1024~65535	LDAP Server	IP address of LDAP server	Determined by destination device.	TCP	LDAP protocol port, it is used to obtain the contact information from LDAP server.
		1024~65535	NTP Server	IP address of NTP server	123	UDP	NTP protocol port, it is used to synchronize time from NTP time server.
		1024~65535	Syslog Server	IP address of syslog server	514	UDP	Syslog protocol port, it is used for IP phones to upload syslog information to syslog server.
		1024~65535	PNP Server	IP address of PNP server (Default value: 224.0.1.75)	5059	UDP/TCP	Protocol port, it is used to obtain the URL of updating file from PNP server.
			Multipaging	Multipaging	65000 65001		
PC	IP address of PC	Determined by the destination	IP phones	IP address of IP phones	1~65535	TCP	HTTP port (default value: 80)
					1~65535	TCP	HTTP port (default value: 443)
SIP Server	IP address of SIP Server				1024~65534	UDP/TCP	SIP protocol port, it is used for signaling interaction with SIP server.
IP phone of voice gateway	IP address of IP phone or voice				2~65535	UDP	RTP protocol port, it is used by destination device to send or receive audio stream.

Source Device	Source IP	Source Port	Destination Device	Destination IP	Destination Port (Listening port)	Protocol	Description of destination port
	gateway	device.					
TR-069 Server	IP address of TR-069 Server				1024~65535	TCP	TR-069 protocol port, it is used to communicate with TR-069server.

## Password Issues

### How to restore the administrator password?

Factory reset can restore the original password. All custom settings will be overwritten after reset.

## Logo Issues

### Why does the IP phone use DOB format logo file instead of popular BMP, JPG and so on?

The IP phone only uses logo file in DOB format, as the DOB format file has a high compression ratio (the size of the uncompressed file compared to that of the compressed file) and can be stored in smaller space. Tools for converting BMP format to DOB format are available. For more information, refer to [Customizing a Logo Template File](#) on page 248.

## Power and Startup Issues

### What will happen if I connect both PoE cable and power adapter? Which has the higher priority?

IP phones use the PoE preferentially.

### Why does the IP phone have no power?

If no lights appear on the IP phone when it is powered up, do one of the following:

- Reboot your IP phone.
- Replace the power adapter.

### Why is the LCD screen black?

If the power indicator LED is on, the keypad is usable but the LCD screen is black, please reboot your IP phone.

### Why does the IP phone always display the Yealink logo?

If your IP phone does not boot, check if the provisioning server is accessible on the network and a valid software firmware and valid configuration files are available. Try to use recovery mode to get your phone ready. For more information on recovery mode, refer to *Recovery Mode on Yealink IP phones*.

## Other Issues

### How do I find the basic information of the IP phone?

Tap **Menu**->**Status** when the IP phone is idle to check the basic information (for example, IP address, MAC address and firmware version).

### What is the difference among user name, register name and display name?

Both user name and register name are defined by the server. User name identifies the account, while register name matched with a password is for authentication purposes. Display name is the caller ID that will be displayed on the callee's phone LCD screen. Server configurations may override the local ones.

### What do "on code" and "off code" mean?

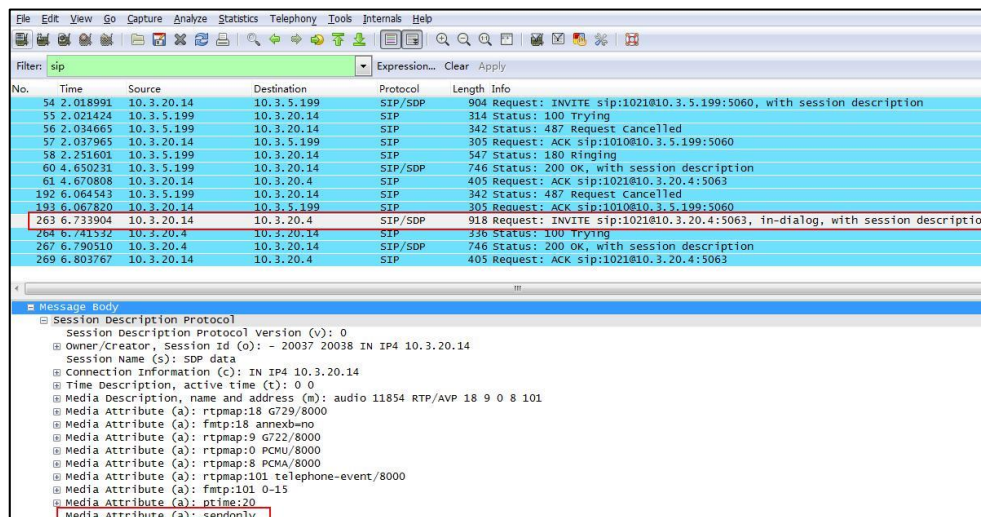
They are codes that the IP phone sends to the server when a certain action takes place. On code is used to activate a feature on the server side, while off code is used to deactivate a feature on the server side.

For example, if you set the Always Forward on code to be \*78 (may vary on different servers), and the target number to be 201. When you enable Always Forward on the IP phone, the IP phone sends \*78201 to the server, and then the server will enable Always Forward feature on the server side, hence being able to get the right status of the extension.

For anonymous call/anonymous call rejection feature, the phone will send either the on code or off code to the server according to the value of Send Anonymous Code/Send Rejection Code. For more information, refer to [Anonymous Call](#) on page 357 and [Anonymous Call Rejection](#) on page 361.

## What is the difference between enabling and disabling the RFC 2543 Hold feature?

Capturing packets after you enable the RFC 2543 Hold feature. SDP media direction attributes (such as a=sendonly) per RFC 2543 is used in the INVITE message when placing a call on hold.



Filter: sip

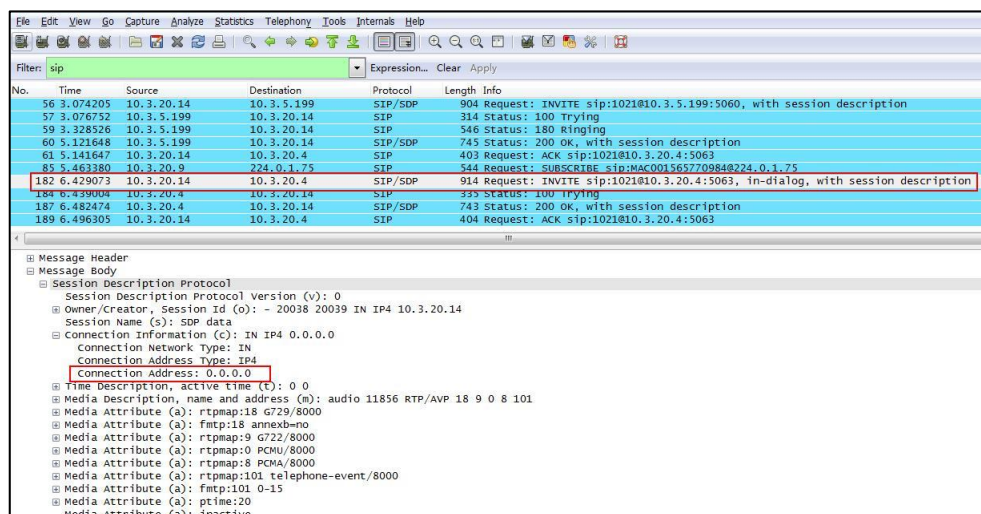
No.	Time	Source	Destination	Protocol	Length	Info
54	2.018991	10.3.20.14	10.3.5.199	SIP/SDP	904	Request: INVITE sip:1021@10.3.5.199:5060, with session description
55	2.021424	10.3.5.199	10.3.20.14	SIP	314	Status: 100 Trying
56	2.034665	10.3.5.199	10.3.20.14	SIP	342	Status: 487 Request cancelled
57	2.037965	10.3.20.14	10.3.5.199	SIP	305	Request: ACK sip:1010@10.3.5.199:5060
58	2.251601	10.3.5.199	10.3.20.14	SIP	547	Status: 180 Ringing
60	4.650231	10.3.5.199	10.3.20.14	SIP/SDP	746	Status: 200 OK, with session description
61	4.670808	10.3.20.14	10.3.20.4	SIP	405	Request: ACK sip:1021@10.3.20.4:5063
192	6.064543	10.3.5.199	10.3.20.14	SIP	342	Status: 487 Request cancelled
193	6.067820	10.3.20.14	10.3.5.199	SIP	305	Request: ACK sip:1010@10.3.5.199:5060
263	6.733904	10.3.20.14	10.3.20.4	SIP/SDP	918	Request: INVITE sip:1021@10.3.20.4:5063, in-dialog, with session description
264	6.741332	10.3.20.4	10.3.20.14	SIP	336	Status: 100 Trying
267	6.790510	10.3.20.4	10.3.20.14	SIP/SDP	746	Status: 200 OK, with session description
269	6.803767	10.3.20.14	10.3.20.4	SIP	405	Request: ACK sip:1021@10.3.20.4:5063

Message Body

```

Session Description Protocol
  Session Description Protocol Version (v): 0
  Owner/Creator, Session Id (o): - 20037 20038 IN IP4 10.3.20.14
  Session Name (s): SDP data
  Connection Information (c): IN IP4 10.3.20.14
  Time Description, active time (t): 0 0
  Media Description, name and address (m): audio 11854 RTP/AVP 18 9 0 8 101
  Media Attribute (a): rtpmap:18 G729/8000
  Media Attribute (a): fmtp:18 annex=no
  Media Attribute (a): rtpmap:9 G722/8000
  Media Attribute (a): rtpmap:0 PCMU/8000
  Media Attribute (a): rtpmap:8 PCMA/8000
  Media Attribute (a): rtpmap:101 telephone-event/8000
  Media Attribute (a): fmtp:101 0-15
  Media Attribute (a): pt=101
  Media Attribute (a): sendonly
  
```

Capturing packets after you disable the RFC 2543 Hold feature. SDP media connection address c=0.0.0.0 per RFC 3264 is used in the INVITE message when placing a call on hold.



Filter: sip

No.	Time	Source	Destination	Protocol	Length	Info
36	3.074205	10.3.20.14	10.3.5.199	SIP/SDP	904	Request: INVITE sip:1021@10.3.5.199:5060, with session description
57	3.076732	10.3.5.199	10.3.20.14	SIP	314	Status: 100 Trying
59	3.328526	10.3.5.199	10.3.20.14	SIP	546	Status: 180 Ringing
60	5.121648	10.3.5.199	10.3.20.14	SIP/SDP	745	Status: 200 OK, with session description
61	5.141647	10.3.20.14	10.3.20.4	SIP	403	Request: ACK sip:1021@10.3.20.4:5063
85	5.463380	10.3.20.9	224.0.0.1, 75	SIP	544	Request: SUBSCRIBE sip:MAC001365770984@224.0.0.1, 75
182	6.429073	10.3.20.14	10.3.20.4	SIP/SDP	914	Request: INVITE sip:1021@10.3.20.4:5063, in-dialog, with session description
184	6.439004	10.3.20.4	10.3.20.14	SIP	335	Status: 100 Trying
187	6.482474	10.3.20.4	10.3.20.14	SIP/SDP	743	Status: 200 OK, with session description
189	6.496305	10.3.20.14	10.3.20.4	SIP	404	Request: ACK sip:1021@10.3.20.4:5063

Message Header

Message Body

```

Session Description Protocol
  Session Description Protocol Version (v): 0
  Owner/Creator, Session Id (o): - 20038 20039 IN IP4 10.3.20.14
  Session Name (s): SDP data
  Connection Information (c): IN IP4 0.0.0.0
  Connection Network Type: IN
  Connection Address Type: IP4
  Connection Address: 0.0.0.0
  Time Description, active time (t): 0 0
  Media Description, name and address (m): audio 11856 RTP/AVP 18 9 0 8 101
  Media Attribute (a): rtpmap:18 G729/8000
  Media Attribute (a): fmtp:18 annex=no
  Media Attribute (a): rtpmap:9 G722/8000
  Media Attribute (a): rtpmap:0 PCMU/8000
  Media Attribute (a): rtpmap:8 PCMA/8000
  Media Attribute (a): rtpmap:101 telephone-event/8000
  Media Attribute (a): fmtp:101 0-15
  Media Attribute (a): pt=101
  Media Attribute (a): inactive
  
```

For more information on RFC 2543 hold feature, refer to [Call Hold](#) on page 392. For more information on capturing packets, refer to [Capturing Packets](#) on page 841.

## What is the difference between Shared Call Appearance (SCA) and Bridge Lines Appearance (BLA)?

SCA and BLA are similar signaling methods that enable more than one phone to share a SIP line. The method you use varies with the SIP server you are using. In the configuration files, the relevant configuration parameters of SCA and BLA are different. The barge-in and call pull feature are not available with BLA; but they are available with SCA.

For more information, refer to [Shared Call Appearance \(SCA\)](#) on page 574 and [Bridge Lines Appearance \(BLA\)](#) on page 582 .



# Appendix

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## Appendix A: Glossary

**802.1x**--an IEEE Standard for port-based Network Access Control (PNAC). It is a part of the IEEE 802.1 group of networking protocols. It provides an authentication mechanism to devices wishing to attach to a LAN or WLAN.

**ACS** (Auto Configuration server)--responsible for auto-configuration of the Central Processing Element (CPE).

**Cryptographic Key**--a piece of variable data that is fed as input into a cryptographic algorithm to perform operations such as encryption and decryption, or signing and verification.

**DHCP** (Dynamic Host Configuration Protocol)--built on a client-server model, where designated DHCP server hosts allocate network addresses and deliver configuration parameters to dynamically configured hosts.

**DHCP Option**--can be configured for specific values and enabled for assignment and distribution to DHCP clients based on server, scope, class or client-specific levels.

**DNS** (Domain Name System)--a hierarchical distributed naming system for computers, services, or any resource connected to the Internet or a private network.

**EAP-MD5** (Extensible Authentication Protocol-Message Digest Algorithm 5)--only provides authentication of the EAP peer to the EAP server but not mutual authentication.

**EAP-TLS** (Extensible Authentication Protocol-Transport Layer Security) --provides for mutual authentication, integrity-protected cipher suite negotiation between two endpoints.

**PEAP-MSCHAPv2** (Protected Extensible Authentication Protocol-Microsoft Challenge Handshake Authentication Protocol version 2) --provides for mutual authentication, but does not require a client certificate on the IP phone.

**FAC** (Feature Access Code)--special patterns of characters that are dialed from a phone keypad to invoke particular features.

**HTTP** (Hypertext Transfer Protocol)--used to request and transmit data on the World Wide Web.

**HTTPS** (Hypertext Transfer Protocol over Secure Socket Layer)--a widely-used communications protocol for secure communication over a network.

**IEEE** (Institute of Electrical and Electronics Engineers)--a non-profit professional association headquartered in New York City that is dedicated to advancing technological innovation and excellence.

**LAN** (Local Area Network)--used to interconnects network devices in a limited area such as a

home, school, computer laboratory, or office building.

**MIB** (Management Information Base)--a virtual database used for managing the entities in a communications network.

**OID** (Object Identifier)--assigned to an individual object within a MIB.

**PnP** (Plug and Play)--a term used to describe the characteristic of a computer bus, or device specification, which facilitates the discovery of a hardware component in a system, without the need for physical device configuration, or user intervention in resolving resource conflicts.

**ROM** (Read-only Memory)--a class of storage medium used in computers and other electronic devices.

**RTP** (Real-time Transport Protocol)--provides end-to-end service for real-time data.

**TCP** (Transmission Control Protocol)--a transport layer protocol used by applications that require guaranteed delivery.

**UDP** (User Datagram Protocol)--a protocol offers non-guaranteed datagram delivery.

**URI** (Uniform Resource Identifier)--a compact sequence of characters that identifies an abstract or physical resource.

**URL** (Uniform Resource Locator)--specifies the address of an Internet resource.

**VLAN** (Virtual LAN)-- a group of hosts with a common set of requirements, which communicate as if they were attached to the same broadcast domain, regardless of their physical location.

**VoIP** (Voice over Internet Protocol)--a family of technologies used for the delivery of voice communications and multimedia sessions over IP networks.

**WLAN** (Wireless Local Area Network)--a type of local area network that uses high-frequency radio waves rather than wires to communicate between nodes.

**XML-RPC** (Remote Procedure Call Protocol)--which uses XML to encode its calls and HTTP as a transport mechanism.

## Appendix B: Time Zones

Time Zone	Time Zone Name
-11	Samoa
-10	United States-Hawaii-Aleutian, United States-Alaska-Aleutian
-9:30	French Polynesia
-9	United States-Alaska Time
-8	Canada(Vancouver,Whitehorse), Mexico(Tijuana,Mexicali), United States-Pacific Time
-7	Canada(Edmonton,Calgary), Mexico(Mazatlan,Chihuahua), United States-MST no DST, United States-Mountain Time
-6	Canada-Manitoba(Winnipeg), Chile(Easter Islands), Mexico(Mexico City,Acapulco), United States-Central Time, Guatemala, El Salvador, Honduras, Nicaragua, Costa Rica, Belize
-5	Bahamas(Nassau), Canada(Montreal,Ottawa,Quebec), Cuba(Havana), United States-Eastern Time, Peru
-4:30	Venezuela(Caracas)
-4	Canada(Halifax,Saint John), Chile(Santiago), Paraguay(Asuncion), United Kingdom-Bermuda(Bermuda), United Kingdom(Falkland Islands), Trinidad&Tobago
-3:30	Canada-New Foundland(St.Johns)
-3	Argentina(Buenos Aires), Brazil(DST), Brazil(no DST), Denmark-Greenland(Nuuk)
-2:30	Newfoundland and Labrador
-2	Brazil(no DST)
-1	Portugal(Azores)
0	Denmark-Faroe Islands(Torshavn), GMT, Greenland, Ireland(Dublin), Morocco, Portugal(Lisboa,Porto,Funchal), Spain-Canary Islands(Las Palmas), United Kingdom(London)
+1	Albania(Tirane), Austria(Vienna), Belgium(Brussels), Caicos, Chad, Croatia(Zagreb), Czech Republic(Prague), Denmark(Kopenhagen), France(Paris), Germany(Berlin), Hungary(Budapest), Italy(Rome), Luxembourg(Luxembourg), Macedonia(Skopje), Namibia(Windhoek), Netherlands(Amsterdam), Spain(Madrid)
+2	Estonia(Tallinn), Finland(Helsinki), Gaza Strip(Gaza), Greece(Athens), Israel(Tel Aviv), Jordan(Amman), Latvia(Riga), Lebanon(Beirut), Moldova(Kishinev), Romania(Bucharest), Russia(Kaliningrad), Syria(Damascus), Turkey(Ankara), Ukraine(Kyiv, Odessa)
+3	East Africa Time, Iraq(Baghdad), Russia(Moscow)
+3:30	Iran(Teheran)
+4	Armenia(Yerevan), Azerbaijan(Baku), Georgia(Tbilisi), Kazakhstan(Aktau), Russia(Samara)

Time Zone	Time Zone Name
+4:30	Afghanistan(Kabul)
+5	Kazakhstan(Aqtobe), Kyrgyzstan(Bishkek), Pakistan(Islamabad), Russia(Chelyabinsk)
+5:30	India(Calcutta)
+5:45	Nepal(Katmandu)
+6	Kazakhstan(Astana, Almaty), Russia(Novosibirsk,Omsk)
+6:30	Myanmar(Naypyitaw)
+7	Russia(Krasnoyarsk), Thailand(Bangkok)
+8	Australia(Perth), China(Beijing), Russia(Irkutsk, Ulan-Ude), Singapore(Singapore)
+8:45	Eucla
+9	Japan(Tokyo), Korea(Seoul), Russia(Yakutsk,Chita)
+9:30	Australia(Adelaide), Australia(Darwin)
+10	Australia(Brisbane), Australia(Hobart), Australia(Sydney,Melbourne,Canberra), Russia(Vladivostok)
+10:30	Australia(Lord Howe Islands)
+11	New Caledonia(Noumea), Russia(Srednekolymsk Time)
+11:30	Norfolk Island
+12	New Zealand(Wellington,Auckland), Russia(Kamchatka Time)
+12:45	New Zealand(Chatham Islands)
+13	Tonga(Nukualofa)
+13:30	Chatham Islands
+14	Kiribati

## Appendix C: Trusted Certificates

Yealink IP phones trust the following CAs by default:

- DigiCert High Assurance EV Root CA
- Deutsche Telekom Root CA 2
- Equifax Secure Certificate Authority
- Equifax Secure eBusiness CA-1
- Equifax Secure Global eBusiness CA-1
- GeoTrust Global CA
- GeoTrust Global CA2
- GeoTrust Primary Certification Authority
- GeoTrust Primary Certification Authority G2
- GeoTrust Universal CA
- GeoTrust Universal CA2
- Thawte Personal Freemail CA

- Thawte Premium Server CA
- Thawte Primary Root CA
- Thawte Primary Root CA - G2
- Thawte Primary Root CA - G3
- Thawte Server CA
- VeriSign Class 1 Public Primary Certification Authority
- VeriSign Class 1 Public Primary Certification Authority - G2
- VeriSign Class 1 Public Primary Certification Authority - G3
- VeriSign Class 2 Public Primary Certification Authority - G2
- VeriSign Class 2 Public Primary Certification Authority - G3
- VeriSign Class 3 Public Primary Certification Authority
- VeriSign Class 3 Public Primary Certification Authority - G2
- VeriSign Class 3 Public Primary Certification Authority - G3
- VeriSign Class 3 Public Primary Certification Authority - G4
- VeriSign Class 3 Public Primary Certification Authority - G5
- VeriSign Class 4 Public Primary Certification Authority - G2
- VeriSign Class 4 Public Primary Certification Authority - G3
- VeriSign Universal Root Certification Authority
- ISRG Root X1 (Let's Encrypt Authority X1 and Let's Encrypt Authority X2 certificates are signed by the root certificate ISRG Root X1.)
- Baltimore CyberTrust Root
- DST Root CA X3
- Verizon Public SureServer CA G14-SHA2
- AddTrust External CA Root
- Go Daddy Class 2 Certification Authority
- Class 2 Primary CA
- Cybertrust Public SureServer SV CA
- DigiCert Assured ID Root G2
- DigiCert Assured ID Root G3
- DigiCert Assured ID Root CA
- DigiCert Global Root G2
- DigiCert Global Root G3
- DigiCert Global Root CA
- DigiCert Trusted Root G4
- Entrust Root Certification Authority
- Entrust Root Certification Authority - G2
- Entrust.net Certification Authority (2048)

- GeoTrust Primary Certification Authority - G3
- GlobalSign Root CA
- GlobalSign
- Starfield Root Certificate Authority - G2
- TC TrustCenter Class 2 CA II
- TC TrustCenter Class 3 CA II
- TC TrustCenter Class 4 CA II
- TC TrustCenter Universal CA I
- TC TrustCenter Universal CA III
- Thawte Universal CA Root
- VeriSign Class 3 Secure Server CA - G2
- VeriSign Class 3 Secure Server CA - G3
- Thawte SSL CA
- StartCom Certification Authority
- StartCom Certification Authority G2
- Starfield Services Root Certificate Authority - G2
- RapidSSL CA
- Go Daddy Root Certificate Authority - G2
- Cybertrust Global Root
- COMODOSSLCA
- COMODO RSA Domain Validation Secure Server CA
- COMODO RSA Certification Authority
- AmazonRootCA4
- AmazonRootCA3
- AmazonRootCA2
- AmazonRootCA1
- Yealink Root CA
- Yealink Equipment Issuing CA
- SIP Core

**Note**

Yealink endeavors to maintain a built-in list of most common used CA Certificates. Due to memory constraints, we cannot ensure a complete set of certificates. If you are using a certificate from a commercial Certificate Authority not in the list above, you can send a request to your local distributor. At this point, you can upload your particular CA certificate into your phone. For more information on uploading custom CA certificate, refer to [Transport Layer Security \(TLS\)](#) on page 802.

## Appendix D: Configuring DSS Keys

This section provides the DSS key parameters you can configure on IP phones. DSS key consists of line key (line key is not applicable to SIP-T19(P) E2 IP phones), programable key and ext key (ext key is only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones).

The following table lists the number of DSS keys you can configure for each phone model:

Phone Model	Line Key	Programable Key	Ext Key
SIP-T54S	27	13	60
SIP-T52S	21	11	60
SIP-T48G/S	29	13	40
SIP-T46G/S	27	13	40
SIP-T42G/S	15	11	/
SIP-T41P/S	15	11	/
SIP-T40P/G	3	11	/
SIP-T29G	27	14	39
SIP-T27P/G	21	14	39
SIP-T23P/G	3	11	/
SIP-T21(P) E2	2	11	/
SIP-T19(P) E2	/	11	/

### Note

The programable key takes effect only if the IP phone is idle.

The ext key takes effect only if the expansion module is connected to the IP phone.

The following tables list relationship between the values of X in the following parameters and programmable keys for each phone model.

# X ranges from 1 to 14.

programmablekey.X.type =

programmablekey.X.line =

programmablekey.X.value =

programmablekey.X.xml\_phonebook =

programmablekey.X.history\_type =

programmablekey.X.pickup\_value =

# X ranges from 1 to 4.

programmablekey.X.label =

<b>Phone Model</b> <b>X</b>	<b>SIP-T19(P) E2</b>	<b>SIP-T23P/T23G/T21(P) E2</b>	<b>SIP-T29G/T27P/T27G</b>	<b>SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G</b>	<b>SIP-T54S/T48G/T48S/T46G/T46S</b>
1	SoftKey1	SoftKey1	SoftKey1	SoftKey1	SoftKey1
2	SoftKey2	SoftKey2	SoftKey2	SoftKey2	SoftKey2
3	SoftKey3	SoftKey3	SoftKey3	SoftKey3	SoftKey3
4	SoftKey4	SoftKey4	SoftKey4	SoftKey4	SoftKey4
5	Up	Up	Up	Up	Up
6	Down	Down	Down	Down	Down



<b>Phone Model</b> <b>X</b>	<b>SIP-T19(P) E2</b>	<b>SIP-T23P/T23G/T21(P) E2</b>	<b>SIP-T29G/T27P/T27G</b>	<b>SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G</b>	<b>SIP-T54S/T48G/T48S/T46G/T46S</b>
7	Left	Left	Left	Left	Left
8	Right	Right	Right	Right	Right
9	OK	OK	OK	OK	OK
10		Cancel	Cancel	Cancel	Cancel
11			CONF		
12			HOLD		HOLD
13	MUTE		MUTE	MUTE	MUTE
14	TRAN	TRAN	TRAN		TRAN

**Note** For SIP-T23P/T23G/T21(P) E2 IP phones, Cancel key can be configured only when the parameter "features.keep\_mute.enable" is set to 0 (Disabled).

For SIP-T54S/T52S/T48G/T48S/T46G/T46S/T42G/T42S/T41P/T41S/T40P/T40G/T29G/T27P/T27G/T19(P) E2 IP phones, Mute key can be configured only when the parameter "features.keep\_mute.enable" is set to 0 (Disabled).

DSS key can be assigned with various key features. The parameters of the DSS key are detailed in the following:

<b>Parameter-</b> linekey.X.type	<b>Configuration File</b> <y0000000000xx>.cfg
<b>Parameter-</b> programablekey.X.type	
<b>Parameter-</b> expansion_module.X.key.Y.type	
<b>Description</b>	<p>Configures key feature for the DSS key.</p> <p><b>For line keys</b> (not applicable to SIP-T19(P) E2 IP phones):</p> <p>X ranges from 1 to 29 (for SIP-T48G/S)</p> <p>X ranges from 1 to 27 (for SIP-T54S/T46G/T46S/T29G)</p> <p>X ranges from 1 to 15 (for SIP-T42G/T42S/T41P/T41S)</p> <p>X ranges from 1 to 21 (for SIP-T52S/T27P/T27G)</p> <p>X ranges from 1 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>X ranges from 1 to 2 (for SIP-T21(P) E2)</p> <p><b>For programable keys:</b></p> <p>X=1-10, 12-14 (for SIP-T54S/T48G/T48S/T46G/T46S)</p> <p>X=1-10, 13 (for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G)</p> <p>X=1-14 (for SIP-T29G/T27P/T27G)</p> <p>X=1-10, 14 (for SIP-T23P/T23G/T21(P) E2)</p> <p>X=1-9, 13, 14 (for SIP-T19(P) E2)</p> <p><b>For ext keys</b> (only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones):</p> <p><b>For SIP-T54S/T52S:</b></p> <p>X ranges from 1 to 3, Y ranges from 1 to 60.</p> <p><b>For SIP-T48G/T48S/T46G/T46S:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 40.</p> <p><b>For SIP-T29G/T27P/T27G:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 20, 22 to 40 (Ext key 21 cannot be configured).</p> <p><b>For line keys:</b></p>

	<p><b>Valid types are:</b></p> <p><b>0</b>-N/A</p> <p><b>1</b>-Conference</p> <p><b>2</b>-Forward</p> <p><b>3</b>-Transfer</p> <p><b>4</b>-Hold</p> <p><b>5</b>-DND</p> <p><b>7</b>-ReCall</p> <p><b>8</b>-SMS</p> <p><b>9</b>-Direct Pickup</p> <p><b>10</b>-Call Park</p> <p><b>11</b>-DTMF</p> <p><b>12</b>-Voice Mail</p> <p><b>13</b>-Speed Dial</p> <p><b>14</b>-Intercom</p> <p><b>15</b>-Line</p> <p><b>16</b>-BLF</p> <p><b>17</b>-URL</p> <p><b>18</b>-Group Listening</p> <p><b>20</b>-Private Hold</p> <p><b>22</b>-XML Group</p> <p><b>23</b>-Group Pickup</p> <p><b>24</b>-Multicast Paging (Paging)</p> <p><b>25</b>-Record</p> <p><b>27</b>-XML Browser</p> <p><b>34</b>-Hot Desking</p> <p><b>35</b>-URL Record</p> <p><b>38</b>-LDAP (only appear when the parameter "ldap.enable" is set to 1)</p> <p><b>39</b>-BLF List</p> <p><b>40</b>-Prefix</p> <p><b>41</b>-Zero Touch</p> <p><b>42</b>-ACD</p> <p><b>45</b>-Local Group</p> <p><b>50</b>-Phone Lock</p> <p><b>56</b>-Retrieve Park</p> <p><b>61</b>-Directory</p>
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	<p><b>66</b>-Paging List</p> <p><b>73</b>-Custom Key (only appear when the parameter "features.enhanced_dss_keys.enable" is set to 1)</p> <p><b>For programmable keys:</b></p> <p><b>Valid types are:</b></p> <p><b>0</b>-N/A</p> <p><b>2</b>-Forward</p> <p><b>5</b>-DND</p> <p><b>7</b>-ReCall</p> <p><b>8</b>-SMS</p> <p><b>9</b>-Direct Pickup</p> <p><b>13</b>-Speed Dial</p> <p><b>14</b>-Intercom</p> <p><b>22</b>-XML Group</p> <p><b>23</b>-Group Pickup</p> <p><b>24</b>-Multicast Paging (Paging)</p> <p><b>27</b>-XML Browser</p> <p><b>28</b>-History</p> <p><b>30</b>-Menu</p> <p><b>32</b>-New SMS</p> <p><b>33</b>-Status</p> <p><b>34</b>-Hot Desking</p> <p><b>38</b>-LDAP (only appear when the parameter "ldap.enable" is set to 1)</p> <p><b>40</b>-Prefix (only applicable to soft keys 1-4)</p> <p><b>41</b>-Zero Touch</p> <p><b>43</b>-Local Directory (Local Phonebook)</p> <p><b>45</b>-Local Group</p> <p><b>47</b>-XML Directory (XML Phonebook)</p> <p><b>50</b>-Phone Lock</p> <p><b>51</b>-Switch Account Up (not applicable to SIP-T19(P) E2 IP phones)</p> <p><b>52</b>-Switch Account Down (not applicable to SIP-T19(P) E2 IP phones)</p> <p><b>61</b>-Directory</p> <p><b>66</b>-Paging List</p> <p><b>73</b>-Custom Key (only appear when the parameter "features.enhanced_dss_keys.enable" is set to 1)</p>
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	<p><b>85-Favorite (Local Favorite)</b></p> <p><b>For ext keys:</b></p> <p><b>Valid types are:</b></p> <p><b>0-NA</b></p> <p><b>1-Conference</b></p> <p><b>2-Forward</b></p> <p><b>3-Transfer</b></p> <p><b>4-Hold</b></p> <p><b>5-DND</b></p> <p><b>7-ReCall</b></p> <p><b>8-SMS</b></p> <p><b>9-Direct Pickup</b></p> <p><b>10-Call Park</b></p> <p><b>11-DTMF</b></p> <p><b>12-Voice Mail</b></p> <p><b>13-Speed Dial</b></p> <p><b>14-Intercom</b></p> <p><b>15-Line</b></p> <p><b>16-BLF</b></p> <p><b>17-URL</b></p> <p><b>18-Group Listening</b></p> <p><b>20-Private Hold</b></p> <p><b>22-XML Group</b></p> <p><b>23-Group Pickup</b></p> <p><b>24-Multicast Paging (Paging)</b></p> <p><b>25-Record</b></p> <p><b>27-XML Browser</b></p> <p><b>34-Hot Desking</b></p> <p><b>35-URL Record</b></p> <p><b>37-Switch</b> (only applicable to ext key 1 for SIP-T29G/T27P/T27G IP phones)</p> <p><b>38-LDAP</b> (only appear when the parameter "ldap.enable" is set to 1)</p> <p><b>39-BLF List</b></p> <p><b>40-Prefix</b></p> <p><b>41-Zero Touch</b></p> <p><b>42-ACD</b></p>
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	<b>45</b> -Local Group <b>50</b> -Phone Lock <b>56</b> -Retrieve Park <b>61</b> -Directory <b>66</b> -Paging List <b>73</b> -Custom Key (only appear when the parameter "features.enhanced_dss_keys.enable" is set to 1)
<b>Format</b>	Integer
<b>Default Value</b>	<p><b>For line keys:</b></p> <p><b>For SIP-T48G/S IP phones:</b>  The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p><b>For SIP-T54S/T46G/T46S/T29G IP phones:</b>  The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p><b>For SIP-T52S IP phones:</b>  The default value of the line key 1-12 is 15, and the default value of the line key 13-21 is 0.</p> <p><b>For SIP-T42G/S IP phones:</b>  The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p><b>For SIP-T41P/S IP phones:</b>  The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p><b>For SIP-T27P/G IP phones:</b>  The default value of the line key 1-6 is 15, and the default value of the line key 7-21 is 0.</p> <p><b>For SIP-T40P/T40G/T23P/T23G/T21(P) E2 IP phones:</b>  The default value is 15.</p> <p><b>For programable keys:</b></p> <p><b>For SIP-T54S/T48G/T48S/T46G/T46S IP phones:</b>  When X=1, the default value is 28 (History).  When X=2, the default value is 61 (Directory).  When X=3, the default value is 5 (DND).  When X=4, the default value is 30 (Menu).  When X=5, the default value is 28 (History).  When X=6, the default value is 61 (Directory).  When X=7, the default value is 51 (Switch Account Up).</p>

	<p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p><b>For SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G IP phones:</b></p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p><b>For SIP-T29G/T27P/T27G IP phones:</b></p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=11, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p>
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	<p><b>For SIP-T23P/T23G/T21(P) E2 IP phones:</b></p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p><b>For SIP-T19(P) E2 IP phones:</b></p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 0 (NA).</p> <p>When X=8, the default value is 0 (NA).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p><b>For ext keys:</b></p> <p><b>For SIP-T54S/T52S IP phones:</b></p> <p>When Y=1-60, the default value is 0 (NA).</p> <p><b>For SIP-T48G/T48S/T46G/T46S IP phones:</b></p> <p>When Y=1-40, the default value is 0 (NA).</p> <p><b>For SIP-T29G/T27P/T27G IP phones:</b></p> <p>When Y=1, 21, the default value is 37 (Switch).</p> <p>When Y=2-20, 22-40, the default value is 0 (NA).</p>
<b>Range</b>	<p>Valid values are:</p> <p><b>0</b>-N/A</p> <p><b>1</b>-Conference</p> <p><b>2</b>-Forward</p>



	<b>3-Transfer</b>
	<b>4-Hold</b>
	<b>5-DND</b>
	<b>7-ReCall</b>
	<b>8-SMS</b>
	<b>9-Direct Pickup</b>
	<b>10-Call Park</b>
	<b>11-DTMF</b>
	<b>12-Voice Mail</b>
	<b>13-Speed Dial</b>
	<b>14-Intercom</b>
	<b>15-Line</b>
	<b>16-BLF</b>
	<b>17-URL</b>
	<b>18-Group Listening</b>
	<b>20-Private Hold</b>
	<b>22-XML Group</b>
	<b>23-Group Pickup</b>
	<b>24-Multicast Paging (Paging)</b>
	<b>25-Record</b>
	<b>27-XML Browser</b>
	<b>28-History</b>
	<b>30-Menu</b>
	<b>32-New SMS</b>
	<b>33-Status</b>
	<b>34-Hot Desking</b>
	<b>35-URL Record</b>
	<b>37-Switch</b>
	<b>38-LDAP</b>
	<b>39-BLF List</b>
	<b>40-Prefix</b>
	<b>41-Zero Touch</b>
	<b>42-ACD</b>
	<b>43-Local Directory</b>
	<b>45-Local Group</b>
	<b>47-XML Directory</b>
	<b>50-Phone Lock</b>

	<b>51</b> -Switch Account Up <b>52</b> -Switch Account Down <b>56</b> -Retrieve Park <b>61</b> -Directory <b>66</b> -Paging List <b>73</b> -Custom Key <b>85</b> -Favorite (Local Favorite)
<b>Example</b>	linekey.1.type = 8

<b>Parameter-</b> linekey.X.line	<b>Configuration File</b> <y0000000000xx>.cfg
<b>Parameter-</b> programablekey.X.line	
<b>Parameter-</b> expansion_module.X.key.Y.line	
<b>Description</b>	<p>Configures the desired line to apply the key feature. (not applicable to SIP-T19(P) E2 IP phones)</p> <p><b>For line keys:</b></p> <p>X ranges from 1 to 29 (for SIP-T48G/S)</p> <p>X ranges from 1 to 27 (for SIP-T54S/T46G/T46S/T29G)</p> <p>X ranges from 1 to 15 (for SIP-T42G/T42S/T41P/T41S)</p> <p>X ranges from 1 to 21 (for SIP-T52S/T27P/T27G)</p> <p>X ranges from 1 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>X ranges from 1 to 2 (for SIP-T21(P) E2)</p> <p><b>For programable keys:</b></p> <p>X=1-10, 12-14 (for SIP-T54S/T48G/T48S/T46G/T46S)</p> <p>X=1-10, 13 (for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G)</p> <p>X=1-14 (for SIP-T29G/T27P/T27G)</p> <p>X=1-10, 14 (for SIP-T23P/T23G/T21(P) E2)</p> <p><b>For ext keys</b> (only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones):</p> <p><b>For SIP-T54S/T52S:</b></p> <p>X ranges from 1 to 3, Y ranges from 1 to 60.</p>

	<p><b>For SIP-T48G/T48S/T46G/T46S:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 40.</p> <p><b>For SIP-T29G/T27P/T27G:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 20, 22 to 40 (Ext key 21 cannot be configured).</p> <p>When assigning the following features, you do not need to configure this parameter:</p> <p><b>1-Conference</b></p> <p><b>2-Forward</b></p> <p><b>3-Transfer</b></p> <p><b>4-Hold</b></p> <p><b>5-DND</b></p> <p><b>7-ReCall</b></p> <p><b>8-SMS</b></p> <p><b>9-Direct Pickup</b></p> <p><b>11-DTMF</b></p> <p><b>17-URL</b></p> <p><b>18-Group Listening</b></p> <p><b>20-Private Hold</b></p> <p><b>22-XML Group</b></p> <p><b>24-Multicast Paging (Paging)</b></p> <p><b>25-Record</b></p> <p><b>27-XML Browser</b></p> <p><b>34-Hot Desking</b></p> <p><b>35-URL Record</b></p> <p><b>38-LDAP</b></p> <p><b>39-BLF List</b></p> <p><b>40-Prefix</b></p> <p><b>41-Zero Touch</b></p> <p><b>42-ACD</b></p> <p><b>45-Local Group</b></p> <p><b>50-Phone Lock</b></p> <p><b>61-Directory</b></p> <p><b>66-Paging List</b></p> <p><b>73-Custom Key</b></p>
<b>Format</b>	Integer
<b>Default Value</b>	For the programmable key and ext key, the default value

	<p>is not applicable.</p> <p>For the line key, when X=1, the default value is 1.</p> <p>When X=2, the default value is 2.</p> <p>When X=3 the default value is 3</p> <p>...</p> <p>When X=16 the default value is 16.</p>
<b>Range</b>	<p><b>Permitted Values:</b></p> <p>1 to 16 (for SIP-T54S/T48G/T48S/T46G/T46S/T29G)</p> <p>1 to 12 (for SIP-T52S/T42G/T42S)</p> <p>1 to 6 (for SIP-T41P/T41S/T27P/T27G)</p> <p>1 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>1 to 2 (for SIP-T21(P) E2)</p> <p>1-Line 1</p> <p>2-Line 2</p> <p>...</p> <p>16-Line 16</p>
<b>Example</b>	linekey.1.line = 2

<b>Parameter-</b> linekey.X.value	<p><b>Configuration File</b></p> <p>&lt;y0000000000xx&gt;.cfg</p>
<b>Parameter-</b> programablekey.X.value	
<b>Parameter-</b> expansion_module.X.key.Y.value	
<b>Description</b>	<p>Configures the value for some key features.</p> <p><b>For line keys</b> (not applicable to SIP-T19(P) E2 IP phones):</p> <p>X ranges from 1 to 29 (for SIP-T48G/S)</p> <p>X ranges from 1 to 27 (for SIP-T54S/T46G/T46S/T29G)</p> <p>X ranges from 1 to 15 (for SIP-T42G/T42S/T41P/T41S)</p> <p>X ranges from 1 to 21 (for SIP-T52S/T27P/T27G)</p> <p>X ranges from 1 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>X ranges from 1 to 2 (for SIP-T21(P) E2)</p>

	<p><b>For programable keys:</b></p> <p>X=1-10, 12-14 (for SIP-T54S/T48G/T48S/T46G/T46S)</p> <p>X=1-10, 13 (for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G)</p> <p>X=1-14 (for SIP-T29G/T27P/T27G)</p> <p>X=1-10, 14 (for SIP-T23P/T23G/T21(P) E2)</p> <p>X=1-9, 13, 14 (for SIP-T19(P) E2)</p> <p><b>For ext keys</b> (only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones):</p> <p><b>For SIP-T54S/T52S:</b></p> <p>X ranges from 1 to 3, Y ranges from 1 to 60.</p> <p><b>For SIP-T48G/T48S/T46G/T46S:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 40.</p> <p><b>For SIP-T29G/T27P/T27G:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 20, 22 to 40 (Ext key 21 cannot be configured).</p>
<b>Format</b>	String
<b>Default Value</b>	Blank
<b>Range</b>	String within 99 characters
<b>Example</b>	<p>When you assign the Speed Dial to the line key, this parameter is used to specify the number you want to dial out.</p> <p>linekey.1.value = 1001</p>

<b>Parameter-</b> linekey.X.label	<p><b>Configuration File</b></p> <p>&lt;y0000000000xx&gt;.cfg</p>
<b>Parameter-</b> programablekey.X.label	
<b>Parameter-</b> expansion_module.X.key.Y.label	
<b>Description</b>	<p>(Optional.) Configures the label displayed on the LCD screen for each line/soft/ext key.</p> <p>This is an optional configuration.</p> <p><b>For line keys</b> (not applicable to SIP-T19(P) E2 IP phones):</p>

	<p>X ranges from 1 to 29 (for SIP-T48G/S)</p> <p>X ranges from 1 to 27 (for SIP-T54S/T46G/T46S/T29G)</p> <p>X ranges from 1 to 15 (for SIP-T42G/T42S/T41P/T41S)</p> <p>X ranges from 1 to 21 (for SIP-T52S/T27P/T27G)</p> <p>X ranges from 1 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>X ranges from 1 to 2 (for SIP-T21(P) E2)</p> <p><b>For programmable keys:</b></p> <p>X ranges from 1 to 4.</p> <p><b>For ext keys</b> (only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones):</p> <p><b>For SIP-T54S/T52S:</b></p> <p>X ranges from 1 to 3, Y ranges from 1 to 60.</p> <p><b>For SIP-T48G/T48S/T46G/T46S:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 40.</p> <p><b>For SIP-T29G/T27P/T27G:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 20, 22 to 40 (Ext key 21 cannot be configured).</p>
<b>Format</b>	String
<b>Default Value</b>	Blank
<b>Range</b>	String within 99 characters
<b>Example</b>	linekey.1.label = Dir

<b>Parameter-</b> linekey.X.pickup_value	<p><b>Configuration File</b></p> <p>&lt;y0000000000xx&gt;.cfg</p>
<b>Parameter-</b> expansion_module.X.key.Y.pickup_value	
<b>Description</b>	<p>Configures the pickup code for BLF feature. (not applicable to SIP-T19(P) E2 IP phones)</p> <p>This parameter is only applicable to BLF feature.</p> <p><b>For line keys:</b></p> <p>X ranges from 1 to 29 (for SIP-T48G/S)</p> <p>X ranges from 1 to 27 (for SIP-T54S/T46G/T46S/T29G)</p> <p>X ranges from 1 to 15 (for</p>

	<p>SIP-T42G/T42S/T41P/T41S)</p> <p>X ranges from 1 to 21 (for SIP-T52S/T27P/T27G)</p> <p>X ranges from 1 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>X ranges from 1 to 2 (for SIP-T21(P) E2)</p> <p><b>For ext keys</b> (only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones):</p> <p><b>For SIP-T54S/T52S:</b></p> <p>X ranges from 1 to 3, Y ranges from 1 to 60.</p> <p><b>For SIP-T48G/T48S/T46G/T46S:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 40.</p> <p><b>For SIP-T29G/T27P/T27G:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 20, 22 to 40 (Ext key 21 cannot be configured).</p>
<b>Format</b>	String
<b>Default Value</b>	Blank
<b>Range</b>	String within 256 characters
<b>Example</b>	linekey.1.pickup_value = *88

<b>Parameter-</b> linekey.X.xml_phonebook	<p><b>Configuration File</b></p> <p>&lt;y0000000000xx&gt;.cfg</p>
<b>Parameter-</b> programablekey.X.xml_phonebook	
<b>Parameter-</b> expansion_module.X.key.Y.xml_phonebook	
<b>Description</b>	<p>Configures the desired group or remote phone book when multiple groups or remote phone books are configured on the IP phone.</p> <p>This parameter is only applicable to Local Group/XML Group features.</p> <p><b>For line keys</b> (not applicable to SIP-T19(P) E2 IP phones):</p> <p>X ranges from 1 to 29 (for SIP-T48G/S)</p>

	<p>X ranges from 1 to 27 (for SIP-T54S/T46G/T46S/T29G)</p> <p>X ranges from 1 to 15 (for SIP-T42G/T42S/T41P/T41S)</p> <p>X ranges from 1 to 21 (for SIP-T52S/T27P/T27G)</p> <p>X ranges from 1 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>X ranges from 1 to 2 (for SIP-T21(P) E2)</p> <p><b>For programmable keys:</b></p> <p>X=1-10, 12-14 (for SIP-T54S/T48G/T48S/T46G/T46S)</p> <p>X=1-10, 13 (for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G)</p> <p>X=1-14 (for SIP-T29G/T27P/T27G)</p> <p>X=1-10, 14 (for SIP-T23P/T23G/T21(P) E2)</p> <p>X=1-9, 13, 14 (for SIP-T19(P) E2)</p> <p><b>For ext keys</b> (only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones):</p> <p><b>For SIP-T54S/T52S:</b></p> <p>X ranges from 1 to 3, Y ranges from 1 to 60.</p> <p><b>For SIP-T48G/T48S/T46G/T46S:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 40.</p> <p><b>For SIP-T29G/T27P/T27G:</b></p> <p>X ranges from 1 to 6, Y ranges from 1 to 20, 22 to 40 (Ext key 21 cannot be configured).</p> <p>When the key feature is configured as Local Group, valid values are:</p> <p><b>0</b>-All contacts</p> <p><b>1</b>-First local group</p> <p>...</p> <p><b>5</b>-Fifth local group</p> <p>...</p> <p><b>48</b>-Forty-eighth local group</p> <p>When the key feature is configured as XML Group (remote phone book), valid values are:</p> <p><b>0</b>-First XML group</p> <p><b>1</b>-Second XML group</p> <p>...</p> <p><b>4</b>-Fifth XML group</p>
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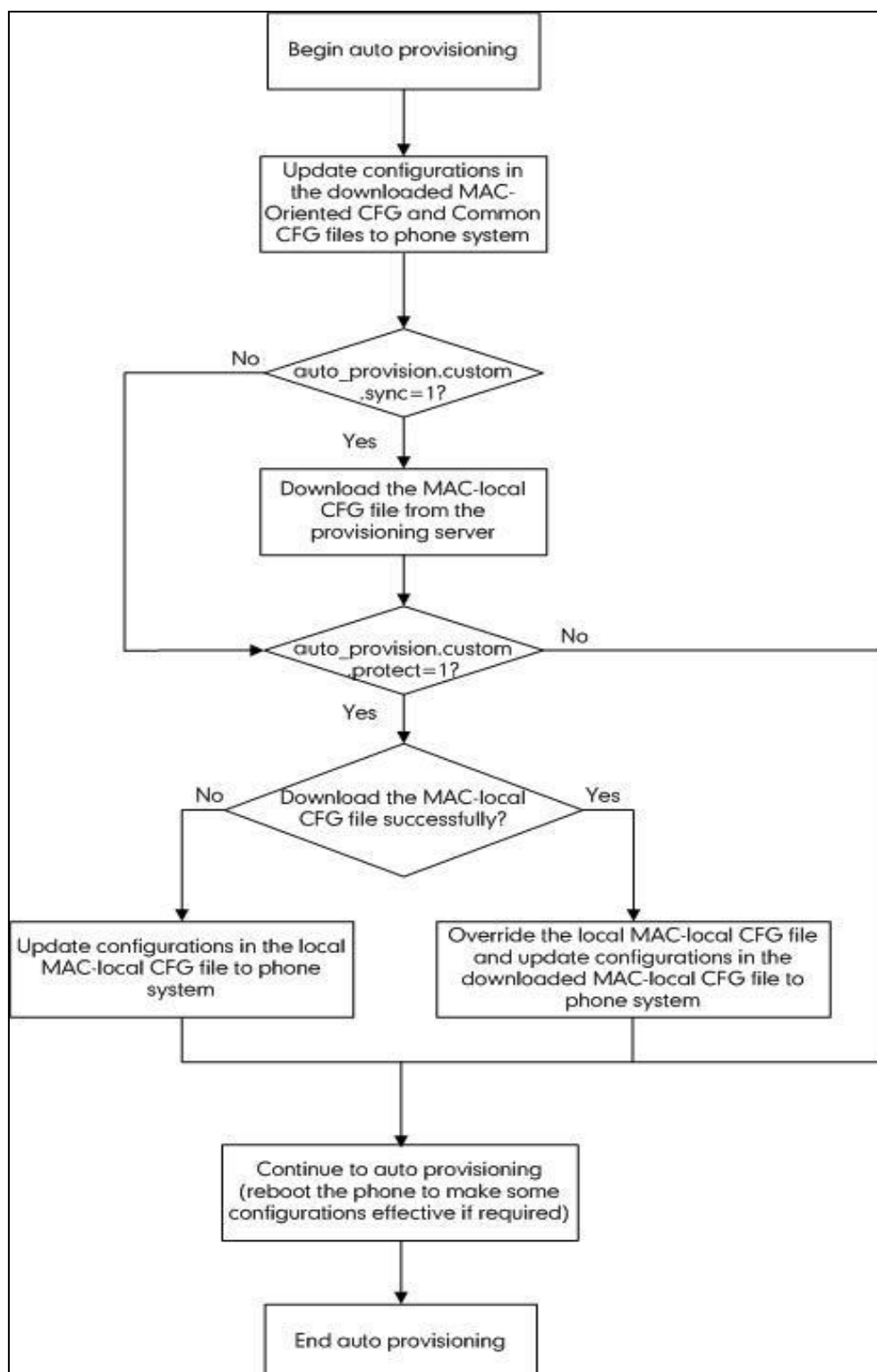
<b>Format</b>	Integer
<b>Default Value</b>	0
<b>Range</b>	0 to 48
<b>Example</b>	Configures the second remote phone book. linekey.1.xml_phonebook = 1

<b>Parameter-</b> linekey.X.extension	<b>Configuration File</b> <y0000000000xx>.cfg
<b>Parameter-</b> programmablekey.X.extension	
<b>Parameter-</b> expansion_module.X.key.Y.extension	
<b>Description</b>	<p>Configures the channel of multicast paging group. This parameter is only applicable to multicast paging features.</p> <p><b>For line keys</b> (not applicable to SIP-T19(P) E2 IP phones):</p> <p>X ranges from 1 to 29 (for SIP-T48G/S)</p> <p>X ranges from 1 to 27 (for SIP-T54S/T46G/T46S/T29G)</p> <p>X ranges from 1 to 15 (for SIP-T42G/T42S/T41P/T41S)</p> <p>X ranges from 1 to 21 (for SIP-T52S/T27P/T27G)</p> <p>X ranges from 1 to 3 (for SIP-T40P/T40G/T23P/T23G)</p> <p>X ranges from 1 to 2 (for SIP-T21(P) E2)</p> <p><b>For programmable keys:</b></p> <p>X=1-10, 12-14 (for SIP-T54S/T48G/T48S/T46G/T46S)</p> <p>X=1-10, 13 (for SIP-T52S/T42G/T42S/T41P/T41S/T40P/T40G)</p> <p>X=1-14 (for SIP-T29G/T27P/T27G)</p> <p>X=1-10, 14 (for SIP-T23P/T23G/T21(P) E2)</p> <p>X=1-9, 13, 14 (for SIP-T19(P) E2)</p> <p><b>For ext keys</b> (only applicable to SIP-T54S/T52S/T48G/T48S/T46G/T46S/T29G/T27P/T27G IP phones):</p> <p><b>For SIP-T54S/T52S:</b></p>

<b>Parameter-</b> linekey.X.extension	<b>Configuration File</b> <y0000000000xx>.cfg
<b>Parameter-</b> programmablekey.X.extension	
<b>Parameter-</b> expansion_module.X.key.Y.extension	
	X ranges from 1 to 3, Y ranges from 1 to 60. <b>For SIP-T48G/T48S/T46G/T46S:</b> X ranges from 1 to 6, Y ranges from 1 to 40. <b>For SIP-T29G/T27P/T27G:</b> X ranges from 1 to 6, Y ranges from 1 to 20, 22 to 40 (Ext key 21 cannot be configured).
<b>Format</b>	Integer
<b>Default Value</b>	0
<b>Range</b>	0 to 31
<b>Example</b>	Configures the channel of the second multicast paging group. linekey.2.extension = 2

## Appendix E: Auto Provisioning Flowchart (Keep User Personalized Configuration Settings)

The following shows auto provisioning flowchart for Yealink IP phones when a user wishes to keep user personalized configuration settings.



## Appendix F: Static Settings

You may need to know the differences between the parameters started with "static." and other common parameters:















































































- All static settings have no priority. They take effect no matter what method (web user interface or phone user interface or configuration files) you are using for provisioning.
- All static settings are never be saved to <MAC>-local.cfg file.
- All static settings are not affected by the overwrite mode. That is, the actual values will not be changed even if you delete the parameters associated with static settings, or you clear the values of the parameters associated with static settings in the configuration file.

























































































































For more information on static settings, refer to the Static Settings sheet in










































































































[\*Yealink\\_SIP-T2\\_Series\\_T19\\_T4\\_Series\\_T5\\_Series\\_Phones\\_Description of Configuration Parameters in CFG File\\_V82.xlsx\*](#).

## Appendix G: Reading Icons

Icons associated with different features may appear on the LCD screen. The following table provides a description for each icon on IP phones.




































































T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
													Network is unavailable
													Private line registers successfully
													Registration failed
	 (Flashing)	 (Flashing)											Registering
													Hands-free (speakerphone) mode
													Handset mode























































T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
													Headset mode
													Voice Mail
					/	/							Text Message
					AA	AA							Auto Answer
					DND	DND							Do Not Disturb
													Call Forward
													Call Hold
													Call Mute
					/	/							Keep Mute
													Ringer volume is 0

T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
													Phone Lock
													Received Calls
													Placed Calls
													Missed Calls
													Forwarded Calls
						/	/			/	/	/	Recording starts successfully (USB recording)
						/	/			/	/	/	Recording is paused (USB recording)
												/	Recording box is full (DSSKey recording)
												/	A call cannot be recorded (DSSKey)



































































T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
													recording)
												/	Recording starts successfully (DSSKey recording)
												/	Recording cannot be started (DSSKey recording)
												/	Recording cannot be stopped (DSSKey recording)
					VPN	VPN	VPN		VPN	VPN	VPN	VPN	VPN is enabled
					/	/	/			/	/	/	Bluetooth mode is on
					/	/	/			/	/	/	Bluetooth headset is both paired and connected
/		/		/	/	/	/	/	/	/	/	/	Bluetooth-Enabled mobile phone is both paired and connected
































































T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
					/	/	/			/	/	/	Wi-Fi connection is successful
					/	/	/			/	/	/	Wi-Fi connection failed
	/	/			/	/	/		/	/	/	/	Conference
													The default contact icon
					/	/	/		/	/	/	/	The default caller photo
					/	/	/		/	/	/	/	Line key type is Line (line is seized)
					/	/	/		/	/	/	/	Line key type is Speed Dial
					/	/	/		/	/	/	/	Line key type is BLF/BLF List (BLF/BLF list idle state)
					/	/	/		/	/	/	/	Line key type is BLF/BLF List (BLF/BLF list ringing state)
					/	/	/		/	/	/	/	Line key type is BLF/BLF List (BLF/BLF list talking state)





















T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
					/	/	/		/	/	/	/	Line key type is BLF (BLF hold state)
					/	/	/		/	/	/	/	Line key type is BLF/BLF List (BLF/BLF list callout state)
					/	/	/		/	/	/	/	Line key type is BLF/BLF List (BLF/BLF list failed state)
					/	/	/		/	/	/	/	Line key type is BLF List (BLF list call park state)
					/	/	/		/	/	/	/	Line key type is Voice Mail
					/	/	/		/	/	/	/	Line key type is Direct Pickup
					/	/	/		/	/	/	/	Line key type is Group Pickup
					/	/	/		/	/	/	/	Line key type is Call Park (park successfully/call park idle state)
					/	/	/		/	/	/	/	Line key type is Call Park (call park

T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
													ringing state)
					/	/	/		/	/	/	/	Park failed
					/	/	/		/	/	/	/	Line key type is Retrieve Park
					/	/	/		/	/	/	/	Line key type is Intercom (intercom idle state)
					/	/	/		/	/	/	/	Line key type is Intercom (intercom ringing state)
	 Talking	 Talking			/	/	/		/	/	/	/	Line key type is Intercom (intercom talking state)
	 Callout	 Callout			/	/	/		/	/	/	/	Line key type is Intercom (intercom callout state)
					/	/	/		/	/	/	/	Line key type is Intercom (intercom failed state)
					/	/	/		/	/	/	/	Line key type is DTMF/Prefix

T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
					/	/	/		/	/	/	/	Line key type is Local Group/XML Group/LDAP
					/	/	/		/	/	/	/	Line key type is XML Browser
					/	/	/		/	/	/	/	Line key type is Conference
					/	/	/		/	/	/	/	Line key type is Forward
					/	/	/		/	/	/	/	Line key type is Transfer
					/	/	/		/	/	/	/	Line key type is Hold/Private Hold
					/	/	/		/	/	/	/	Line key type is DND
					/	/	/		/	/	/	/	Line key type is Recall
					/	/	/		/	/	/	/	Line key type is SMS
					/	/	/		/	/	/	/	Line key type is Record/URL Record
					/	/	/		/	/	/	/	Line key type is Record/URL Record

T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
													(recording starts successfully)
					/	/	/		/	/	/	/	Line key type is Multicast Paging/Group Listening/Paging List
					/	/	/		/	/	/	/	Line key type is Hot Desking
					/	/	/		/	/	/	/	Line key type is ACD
					/	/	/		/	/	/	/	Line key type is Zero Touch
					/	/	/		/	/	/	/	Line key type is URL
					/	/	/		/	/	/	/	Line key type is Phone Lock
					/	/	/		/	/	/	/	Line key type is Directory
					/	/	/		/	/	/	/	Line key type is Custom Key
													The ACD state is available

T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
													The ACD state is unavailable
													The ACD state is Wrap up
													Log out of the ACD system
													The shared line/bridged line is idle
					/	/	/		/	/	/	/	The shared line receives ring-back tone
					/	/	/		/	/	/	/	The shared line receives an incoming call
					/	/	/		/	/	/	/	The shared line is in conversation
					/	/	/		/	/	/	/	The shared line conversation is placed on public hold
					/	/	/			/	/	/	USB flash drive is detecting

T54S/ T52S	T48S	T48G	T46S	T46G	T42S/ T41S	T42G/ T41P	T40P /G	T29G	T27G	T27P	T23P/T23 G/T21(P) E2	T19(P) E2	Description
(Flashing)	(Flashing)	(Flashing)	(Flashing)	(Flashing)				(Flashing)					
					/	/	/			/	/	/	USB flash drive is detected
													High Definition Voice

## Appendix H: SIP (Session Initiation Protocol)

This section describes how Yealink IP phones comply with the IETF definition of SIP as described in [RFC 3261](#).

This section contains compliance information in the following:

- [RFC and Internet Draft Support](#)
- [SIP Request](#)
- [SIP Header](#)
- [SIP Responses](#)
- [SIP Session Description Protocol \(SDP\) Usage](#)

### RFC and Internet Draft Support

The following RFC's and Internet drafts are supported:

- RFC 1321—The MD5 Message-Digest Algorithm
- RFC 1889—RTP Media control
- RFC 2112—Multipart MIME
- RFC 2327—SDP: Session Description Protocol
- RFC 2387—The MIME Multipart/Related Content-type
- RFC 2543—SIP: Session Initiation Protocol
- RFC 2617—Http Authentication: Basic and Digest access authentication
- RFC 2782—A DNS RR for specifying the location of services (DNS SRV)
- RFC 2806—URLs for Telephone Calls
- RFC 2833—RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 2915—The Naming Authority Pointer (NAPTR) DNS Resource Record
- RFC 2976—The SIP INFO Method
- RFC 3087—Control of Service Context using SIP Request-URI
- RFC 3261—SIP: Session Initiation Protocol (replacement for RFC 2543)
- RFC 3262—Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- RFC 3263—Session Initiation Protocol (SIP): Locating SIP Servers
- RFC 3264—An Offer/Answer Model with the Session Description Protocol (SDP)
- RFC 3265—Session Initiation Protocol (SIP) - Specific Event Notification
- RFC 3266—Support for IPv6 in Session Description Protocol (SDP)
- RFC 3310—HTTP Digest Authentication Using Authentication and Key Agreement (AKA)
- RFC 3311—The Session Initiation Protocol (SIP) UPDATE Method
- RFC 3312—Integration of Resource Management and SIP



- RFC 3313—Private SIP Extensions for Media Authorization
- RFC 3323—A Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC 3324—Requirements for Network Asserted Identity
- RFC 3325—SIP Asserted Identity
- RFC 3326—The Reason Header Field for the Session Initiation Protocol (SIP)
- RFC 3361—DHCP-for-IPv4 Option for SIP Servers
- RFC 3372—SIP for Telephones (SIP-T): Context and Architectures
- RFC 3398—ISUP to SIP Mapping
- RFC 3420—Internet Media Type message/sipfrag
- RFC 3428—Session Initiation Protocol (SIP) Extension for Instant Messaging
- RFC 3455—Private Header (P-Header) Extensions to the SIP for the 3GPP
- RFC 3486—Compressing the Session Initiation Protocol (SIP)
- RFC 3489—STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
- RFC 3515—The Session Initiation Protocol (SIP) Refer Method
- RFC 3550—RTP: Transport Protocol for Real-Time Applications
- RFC 3555—MIME Type Registration of RTP Payload Formats
- RFC 3581—An Extension to the SIP for Symmetric Response Routing
- RFC 3608—SIP Extension Header Field for Service Route Discovery During Registration
- RFC 3611—RTP Control Protocol Extended Reports (RTCP XR)
- RFC 3665—Session Initiation Protocol (SIP) Basic Call Flow Examples
- RFC 3666—SIP Public Switched Telephone Network (PSTN) Call Flows.
- RFC 3680—SIP Event Package for Registrations
- RFC 3702—Authentication, Authorization, and Accounting Requirements for the SIP
- RFC 3711—The Secure Real-time Transport Protocol (SRTP)
- RFC 3725—Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)
- RFC 3842—A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
- RFC 3856—A Presence Event Package for Session Initiation Protocol (SIP)
- RFC 3863—Presence Information Data Format
- RFC 3890—A Transport Independent Bandwidth Modifier for the SDP
- RFC 3891—The Session Initiation Protocol (SIP) "Replaces" Header
- RFC 3892—The Session Initiation Protocol (SIP) Referred-By Mechanism
- RFC 3959—The Early Session Disposition Type for SIP
- RFC 3960—Early Media and Ringing Tone Generation in SIP
- RFC 3966—The tel URI for telephone number
- RFC 3968—IANA Registry for SIP Header Field

- RFC 3969—IANA Registry for SIP URI
- RFC 4028—Session Timers in the Session Initiation Protocol (SIP)
- RFC 4083—3GPP Release 5 Requirements on SIP
- RFC 4235—An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)
- RFC 4244—An Extension to the SIP for Request History Information
- RFC 4317—Session Description Protocol (SDP) Offer/Answer Examples
- RFC 4353—A Framework for Conferencing with the SIP
- RFC 4458—SIP URIs for Applications such as Voicemail and Interactive Voice Response (IVR)
- RFC 4475—Session Initiation Protocol (SIP) Torture
- RFC 4485—Guidelines for Authors of Extensions to the SIP
- RFC 4504—SIP Telephony Device Requirements and Configuration
- RFC 4566—SDP: Session Description Protocol.
- RFC 4568—Session Description Protocol (SDP) Security Descriptions for Media Streams
- RFC 4575—A SIP Event Package for Conference State
- RFC 4579—SIP Call Control - Conferencing for User Agents
- RFC 4583—Session Description Protocol (SDP) Format for Binary Floor Control Protocol (BFCP) Streams
- RFC 4662—A SIP Event Notification Extension for Resource Lists
- RFC 4730—Event Package for KPML
- RFC 5009—P-Early-Media Header
- RFC 5079—Rejecting Anonymous Requests in SIP
- RFC 5359—Session Initiation Protocol Service Examples
- RFC 5589—Session Initiation Protocol (SIP) Call Control - Transfer
- RFC 5630—The Use of the SIPS URI Scheme in SIP
- RFC 5806—Diversion Indication in SIP
- RFC 5954—Essential Correction for IPv6 ABNF and URI Comparison in RFC 3261
- RFC 6026—Correct Transaction Handling for 2xx Responses to SIP INVITE Requests
- RFC 6141—Re-INVITE and Target-Refresh Request Handling in SIP
- draft-ietf-sip-cc-transfer-05.txt—SIP Call Control - Transfer
- draft-anil-sipping-bla-02.txt—Implementing Bridged Line Appearances (BLA) Using Session Initiation Protocol (SIP)
- draft-anil-sipping-bla-03.txt—Implementing Bridged Line Appearances (BLA) Using Session Initiation Protocol (SIP)
- draft-ietf-sip-privacy-00.txt—SIP Extensions for Caller Identity and Privacy, November
- draft-ietf-sip-privacy-04.txt—SIP Extensions for Network-Asserted Caller Identity and Privacy within Trusted Networks

- draft-levy-sip-diversion-08.txt–Diversion Indication in SIP
- draft-ietf-sipping-cc-conferencing-03.txt–SIP Call Control - Conferencing for User Agents
- draft-ietf-sipping-cc-conferencing-05.txt–Connection Reuse in the Session Initiation Protocol (SIP)
- draft-ietf-sipping-rtcp-summary-02.txt–Session Initiation Protocol Package for Voice Quality Reporting Event
- draft-ietf-sip-connect-reuse-06.txt–Connection Reuse in the Session Initiation Protocol (SIP)
- draft-ietf-bliss-shared-appearances-15.txt–Shared Appearances of a Session Initiation Protocol (SIP) Address of Record (AOR)

To find the applicable Request for Comments (RFC) document, go to <http://www.ietf.org/rfc.html> and enter the RFC number.

## SIP Request

The following SIP request messages are supported:

Method	Supported	Notes
REGISTER	Yes	
INVITE	Yes	Yealink IP phones support mid-call changes such as placing a call on hold as signaled by a new INVITE that contains an existing Call-ID.
ACK	Yes	
CANCEL	Yes	
BYE	Yes	
OPTIONS	Yes	
SUBSCRIBE	Yes	
NOTIFY	Yes	
REFER	Yes	
PRACK	Yes	
INFO	Yes	
MESSAGE	Yes	
UPDATE	Yes	

Method	Supported	Notes
PUBLISH	Yes	

## SIP Header

The following SIP request headers are supported:

**Note** In the following table, a "Yes" in the Supported column means the header is sent and properly parsed.

Method	Supported	Notes
Accept	Yes	
Alert-Info	Yes	
Allow	Yes	
Allow-Events	Yes	
Authorization	Yes	
Call-ID	Yes	
Call-Info	Yes	
Contact	Yes	
Content-Length	Yes	
Content-Type	Yes	
CSeq	Yes	
Diversion	Yes	
History-Info	Yes	
Event	Yes	
Expires	Yes	
From	Yes	
Max-Forwards	Yes	
Min-SE	Yes	
P-Asserted-Identity	Yes	
P-Preferred-Identity	Yes	
Proxy-Authenticate	Yes	
Proxy-Authorization	Yes	

Method	Supported	Notes
RAck	Yes	
Record-Route	Yes	
Refer-To	Yes	
Referred-By	Yes	
Remote-Party-ID	Yes	
Replaces	Yes	
Require	Yes	
Route	Yes	
RSeq	Yes	
Session-Expires	Yes	
Subscription-State	Yes	
Supported	Yes	
To	Yes	
User-Agent	Yes	
Via	Yes	

## SIP Responses

The following SIP responses are supported:

### Note

In the following table, a “Yes” in the Supported column means the header is sent and properly parsed. The phone may not actually generate the response.

### 1xx Responses—Provisional

1xx Response	Supported	Notes
100 Trying	Yes	
180 Ringing	Yes	
181 Call Is Being Forwarded	Yes	
182 Queued	Yes	
183 Session Progress	Yes	

## 2xx Responses—Successful

2xx Response	Supported	Notes
200 OK	Yes	
202 Accepted	Yes	In REFER transfer.

## 3xx Responses—Redirection

3xx Response	Supported	Notes
300 Multiple Choices	Yes	
301 Moved Permanently	Yes	
302 Moved Temporarily	Yes	
305 Use Proxy	Yes	
380 Alternative Service	No	

## 4xx Responses—Request Failure

4xx Response	Supported	Notes
400 Bad Request	Yes	
401 Unauthorized	Yes	
402 Payment Required	Yes	
403 Forbidden	Yes	
404 Not Found	Yes	
405 Method Not Allowed	Yes	
406 Not Acceptable	No	
407 Proxy Authentication Required	Yes	
408 Request Timeout	Yes	
409 Conflict	No	
410 Gone	No	
411 Length Required	No	
413 Request Entity Too Large	No	
414 Request-URI Too Long	Yes	
415 Unsupported Media Type	Yes	

4xx Response	Supported	Notes
416 Unsupported URI Scheme	No	
420 Bad Extension	No	
421 Extension Required	No	
423 Interval Too Brief	Yes	
480 Temporarily Unavailable	Yes	
481 Call/Transaction Does Not Exist	Yes	
482 Loop Detected	Yes	
483 Too Many Hops	No	
484 Address Incomplete	Yes	
485 Ambiguous	No	
486 Busy Here	Yes	
487 Request Terminated	Yes	
488 Not Acceptable Here	Yes	
491 Request Pending	No	
493 Undecipherable	No	

### 5xx Responses—Server Failure

5xx Response	Supported	Notes
500 Server Internal Error	Yes	
501 Not Implemented	Yes	
502 Bad Gateway	No	
503 Service Unavailable	Yes	
504 Server Time-out	No	
505 Version Not Supported	No	
513 Message Too Large	No	

### 6xx Response—Global Failures

6xx Response	Supported	Notes
600 Busy Everywhere	Yes	

6xx Response	Supported	Notes
603 Decline	Yes	
604 Does Not Exist Anywhere	No	
606 Not Acceptable	No	

## SIP Session Description Protocol (SDP) Usage

SDP Headers	Supported
v–Session Description Protocol Version	Yes
o–Owner/Creator, Session Id	Yes
a–Media Attribute	Yes
c–Connection Information	Yes
b–Bandwidth Information	Yes
m–Media Description, name and address	Yes
s–Session Name	Yes
t–Time Description, active time	Yes

## Appendix I: SIP Call Flows

SIP uses six request methods:

INVITE–Indicates a user is being invited to participate in a call session.

ACK–Confirms that the client has received a final response to an INVITE request.

BYE–Terminates a call and can be sent by either the caller or the callee.

CANCEL–Cancels any pending searches but does not terminate a call that has already been accepted.

OPTIONS–Queries the capabilities of servers.

REGISTER–Registers the address listed in the To header field with a SIP server.

The following types of responses are used by SIP and generated by the IP phone or the SIP server:

SIP 1xx–Provisional Responses

SIP 2xx–Successful Responses

SIP 3xx–Redirection Responses

SIP 4xx–Request Failure Responses



SIP 5xx—Server Failure Responses

SIP 6xx—Global Failures Responses

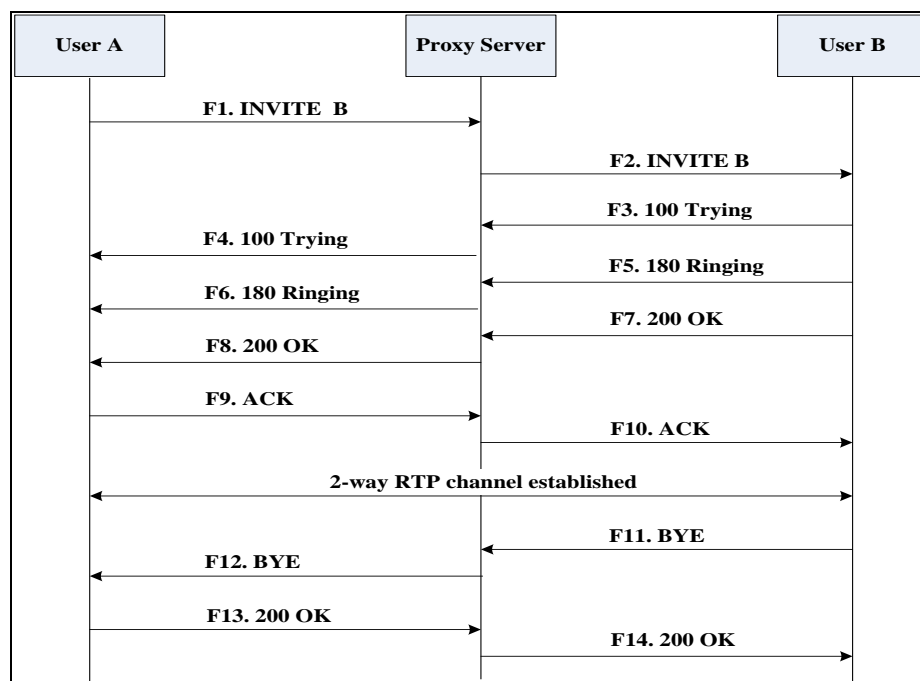
For more information on SIP Responses, refer to [SIP Responses](#) on page 915.

## Successful Call Setup and Disconnect

The following figure illustrates the scenario of a successful call. In this scenario, the two end users are User A and User B. User A and User B are located at Yealink SIP IP phones.

**The call flow scenario is as follows:**

1. User A calls User B.
2. User B answers the call.
3. User B hangs up.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>User A sends a SIP INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>The IP address of User B is inserted in the Request-URI field.</li> <li>User A is identified as the call session</li> </ul>

Step	Action	Description
		<p>initiator in the From field.</p> <ul style="list-style-type: none"> <li>A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>The transaction number within a single call leg is identified in the CSeq field.</li> <li>The media capability User A is ready to receive is specified.</li> <li>The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE–Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	100 Trying–User B to Proxy Server	User B sends a SIP 100 Trying response to the proxy server. The 100 Trying response indicates that the INVITE request has been received by User B.
F4	100 Trying–Proxy Server to User A	The proxy server forwards the SIP 100 Trying to User A to indicate that the INVITE request has been received by User B.
F5	180 Ringing–User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the User B is being alerted.
F6	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F7	200 OK– User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F8	200OK–Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.
F9	ACK–User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is

Step	Action	Description
		now active.
F10	ACK—Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F11	BYE—User B to Proxy Server	User B terminates the call session by sending a SIP BYE request to the proxy server. The BYE request indicates that User B wants to release the call.
F12	BYE—Proxy Server to User A	The proxy server forwards the SIP BYE request to User A to notify that User B wants to release the call.
F13	200 OK—User A to Proxy Server	User A sends a SIP 200 OK response to the proxy server. The 200 OK response indicates that User A has received the BYE request. The call session is now terminated.
F14	200 OK—Proxy Server to User B	The proxy server forwards the SIP 200 OK response to User B to indicate that User A has received the BYE request. The call session is now terminated.

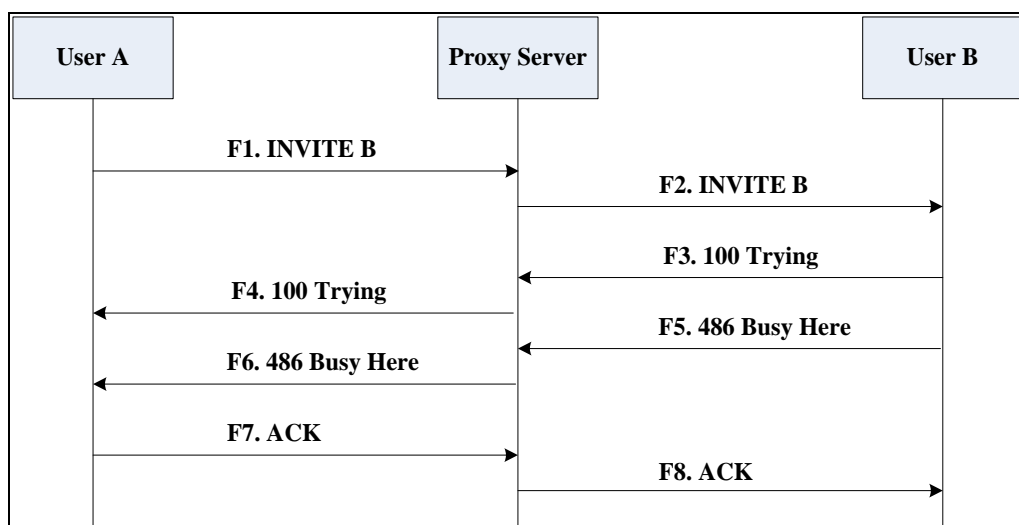
## Unsuccessful Call Setup—Called User is Busy

The following figure illustrates the scenario of an unsuccessful call caused by the called user's being busy. In this scenario, the two end users are User A and User B. User A and User B are located at Yealink SIP IP phones.

**The call flow scenario is as follows:**

1. User A calls User B.
2. User B is busy on the IP phone and unable or unwilling to take another call.

The call cannot be set up successfully.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>User A sends the INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>The IP address of User B is inserted in the Request-URI field.</li> <li>User A is identified as the call session initiator in the From field.</li> <li>A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>The transaction number within a single call leg is identified in the CSeq field.</li> <li>The media capability User A is ready to receive is specified.</li> <li>The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. Proxy server forwards the INVITE message to User B.
F3	100 Trying—User B to Proxy Server	User B sends a SIP 100 Trying response to the proxy server. The 100 Trying response

Step	Action	Description
		indicates that the INVITE request has been received by User B.
F4	100 Trying—Proxy Server to User A	The proxy server forwards the SIP 100 Trying to User A to indicate that the INVITE request has already been received.
F5	486 Busy Here—User B to Proxy Server	User B sends a SIP 486 Busy Here response to the proxy server. The 486 Busy Here response is a client error response indicating that User B is successfully connected but User B is busy on the IP phone and unable or unwilling to take the call.
F6	486 Busy Here—Proxy Server to User A	The proxy server forwards the 486 Busy Here response to notify User A that User B is busy.
F7	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The SIP ACK message indicates that User A has received the 486 Busy Here message.
F8	ACK—Proxy Server to User B	The proxy server forwards the SIP ACK to User B to indicate that the 486 Busy Here message has already been received.

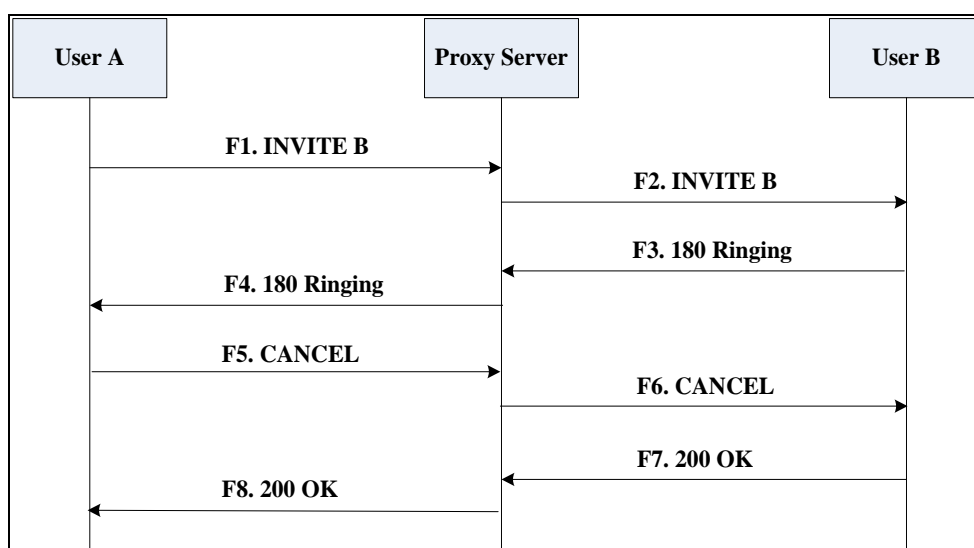
## Unsuccessful Call Setup—Called User Does Not Answer

The following figure illustrates the scenario of an unsuccessful call caused by the called user's no answering. In this scenario, the two end users are User A and User B. User A and User B are located at Yealink SIP IP phones.

**The call flow scenario is as follows:**

1. User A calls User B.
2. User B does not answer the call.
3. User A hangs up.

The call cannot be set up successfully.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>The IP address of User B is inserted in the Request-URI field.</li> <li>User A is identified as the call session initiator in the From field.</li> <li>A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>The transaction number within a single call leg is identified in the CSeq field.</li> <li>The media capability User A is ready to receive is specified.</li> <li>The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. Proxy server forwards the INVITE message to User B.
F3	180 Ringing—User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.

Step	Action	Description
F4	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	CANCEL–User A to Proxy Server	User A sends a SIP CANCEL request to the proxy server after not receiving an appropriate response within the time allocated in the INVITE request. The SIP CANCEL request indicates that User A wants to disconnect the call.
F6	CANCEL–Proxy Server to User B	The proxy server forwards the SIP CANCEL request to notify User B that User A wants to disconnect the call.
F7	200 OK–User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The SIP 200 OK response indicates that User B has received the CANCEL request.
F8	200 OK–Proxy Server to User A	The proxy server forwards the SIP 200 OK response to notify User A that the CANCEL request has been processed successfully.

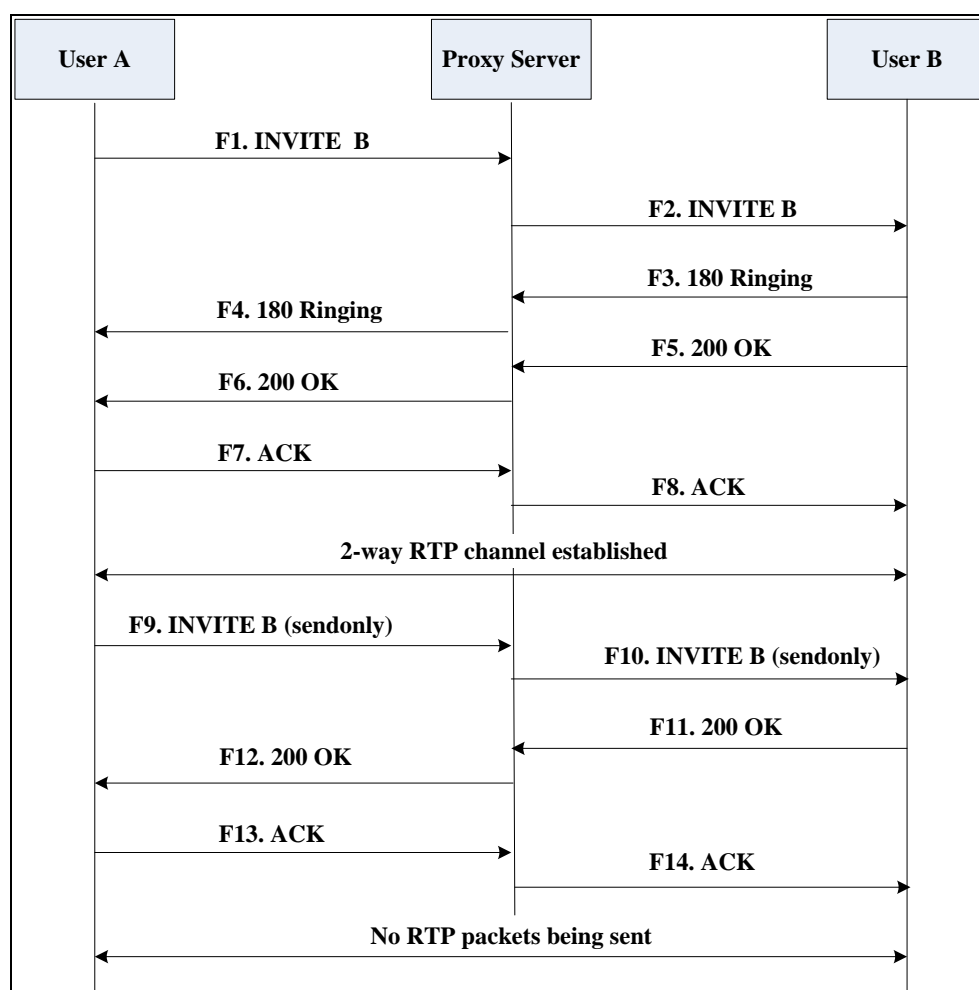
## Successful Call Setup and Call Hold

The following figure illustrates a successful call setup and call hold. In this scenario, the two end users are User A and User B. User A and User B are located at Yealink SIP IP phones.

**The call flow scenario is as follows:**

1. User A calls User B.
2. User B answers the call.

3. User A places User B on hold.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>The IP address of User B is inserted in the Request-URI field.</li> <li>User A is identified as the call session initiator in the From field.</li> <li>A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>The transaction number within a single call leg is identified in the CSeq field.</li> </ul>



Step	Action	Description
		<ul style="list-style-type: none"> <li>The media capability User A is ready to receive is specified.</li> <li>The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE–Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing–User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	200 OK–User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies the proxy server that the connection has been made.
F6	200 OK–Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.
F7	ACK–User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK–Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F9	INVITE–User A to Proxy Server	User A sends a mid-call INVITE request to the proxy server with new SDP session parameters, which are used to place the call on hold.
F10	INVITE–Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F11	200 OK–User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies

Step	Action	Description
		User A that the INVITE is successfully processed.
F12	200 OK–Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User B is successfully placed on hold.
F13	ACK–User A to Proxy Server	User A sends an ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F14	ACK–Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.

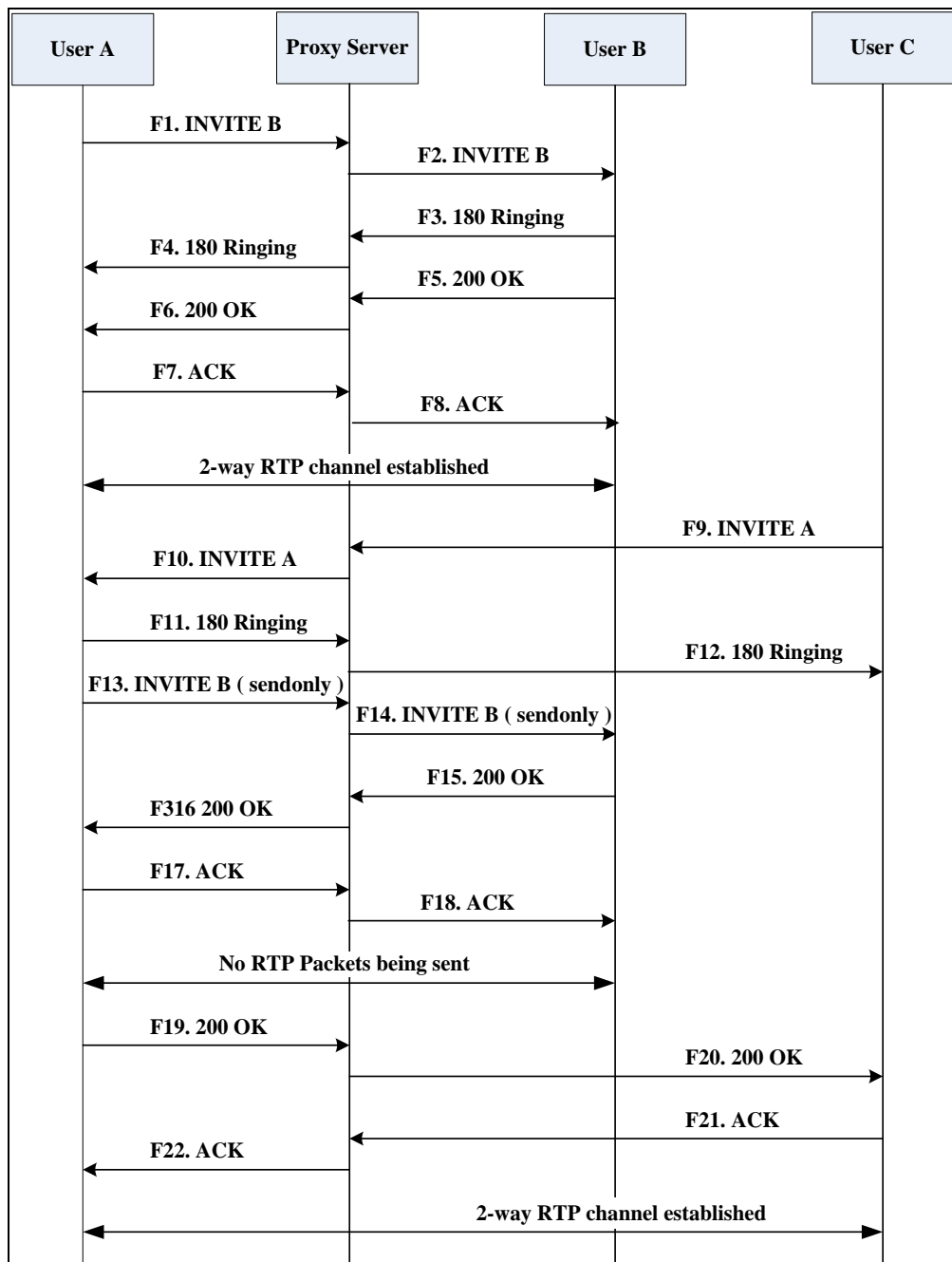
## Successful Call Setup and Call Waiting

The following figure illustrates a successful call between Yealink SIP IP phones in which two parties are in a call, one of the participants receives and answers an incoming call from a third party. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

**The call flow scenario is as follows:**

1. User A calls User B.
2. User B answers the call.
3. User C calls User B.

## 4. User B accepts the call from User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>The IP address of User B is inserted in the Request-URI field.</li> <li>User A is identified as the call session initiator in the From field.</li> </ul>

Step	Action	Description
		<ul style="list-style-type: none"> <li>A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>The transaction number within a single call leg is identified in the CSeq field.</li> <li>The media capability User A is ready to receive is specified.</li> <li>The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE–Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing–User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	200 OK–User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies proxy server that the connection has been made.
F6	200 OK–Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.
F7	ACK–User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK–Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F9	INVITE–User C to Proxy Server	User C sends a SIP INVITE message to the proxy server. The INVITE request is an invitation to User A to participate in a call

Step	Action	Description
		<p>session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>• The IP address of User A is inserted in the Request-URI field.</li> <li>• User C is identified as the call session initiator in the From field.</li> <li>• A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>• The transaction number within a single call leg is identified in the CSeq field.</li> <li>• The media capability User C is ready to receive is specified.</li> <li>• The port on which User A is prepared to receive the RTP data is specified.</li> </ul>
F10	INVITE–Proxy Server to User A	The proxy server maps the SIP URI in the To field to User A. The proxy server sends the INVITE message to User A.
F11	180 Ringing–User A to Proxy Server	User A sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F12	180 Ringing–Proxy Server to User C	The proxy server forwards the 180 Ringing response to User C. User C hears the ring-back tone indicating that User A is being alerted.
F13	INVITE–User A to Proxy Server	User A sends a mid-call INVITE request to the proxy server with new SDP session parameters, which are used to place the call on hold.
F14	INVITE–Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F15	200 OK–User B to Proxy Server	User B sends a 200 OK to the proxy server. The 200 OK response indicates that the INVITE was successfully processed.
F16	200 OK–Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User B is successfully placed on

Step	Action	Description
		hold.
F17	ACK–User A to Proxy Server	User A sends an ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F18	ACK–Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.
F19	200 OK–User A to Proxy Server	User A sends a 200 OK response to the proxy server. The 200 OK response notifies that the connection has been made.
F20	200 OK–Proxy Server User C	The proxy server forwards the 200 OK message to User C.
F21	ACK–User C to Proxy Server	User C sends a SIP ACK to the proxy server. The ACK confirms that User C has received the 200 OK response. The call session is now active.
F22	ACK–Proxy Server to User A	The proxy server forwards the SIP ACK to User A to confirm that User C has received the 200 OK response.

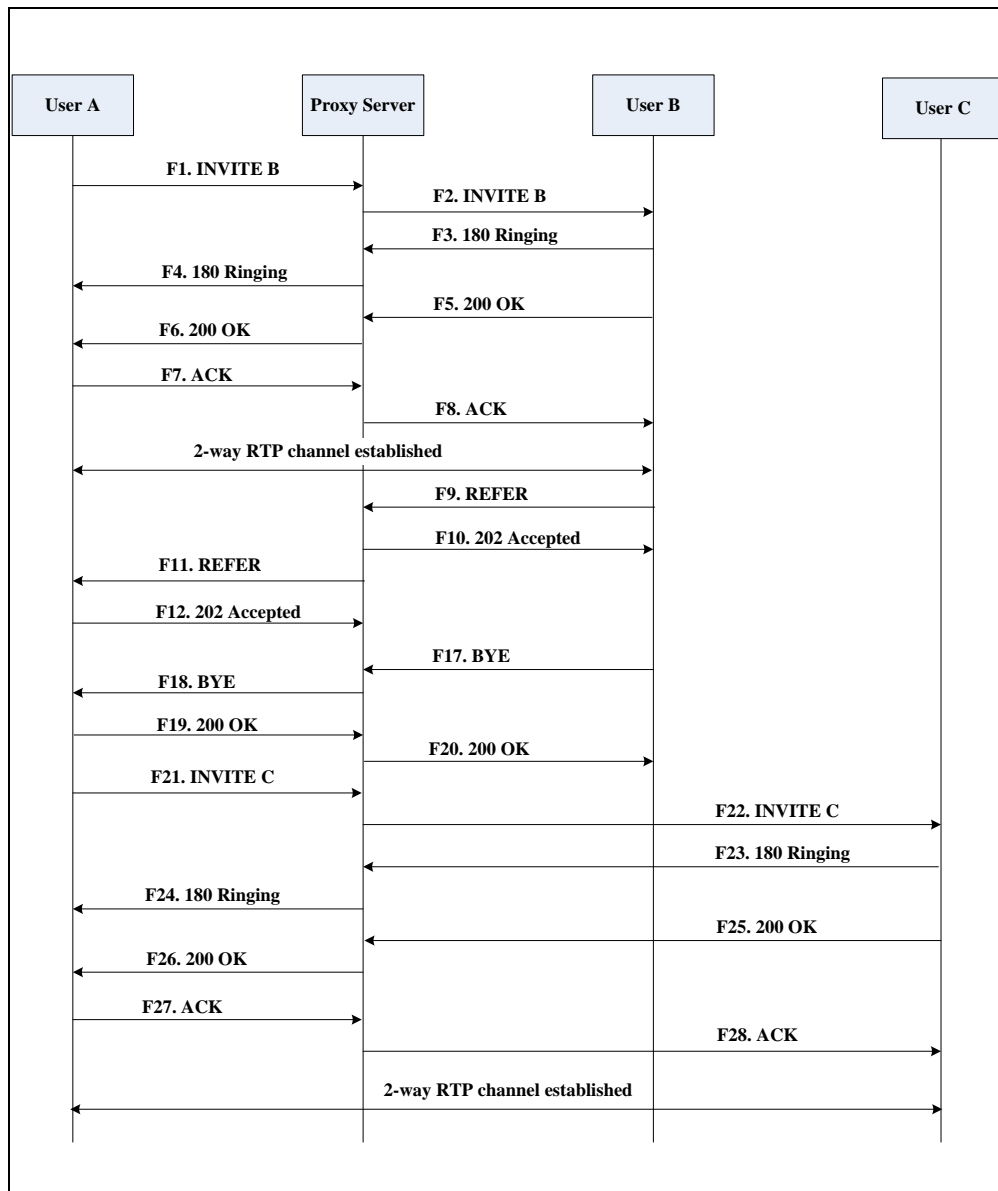
## Call Transfer without Consultation

The following figure illustrates a successful call between Yealink SIP IP phones in which two parties are in a call and then one of the parties transfers the call to a third party without consultation. This is called a blind transfer. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

The call flow scenario is as follows:

1. User A calls User B.
2. User B answers the call.
3. User B transfers the call to User C.
4. User C answers the call.

Call is established between User A and User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends an INVITE message to the proxy server. The INVITE request is an invitation to User B to participate in a call session.  In the INVITE request:

Step	Action	Description
		<ul style="list-style-type: none"> <li>The IP address of User B is inserted in the Request-URI field.</li> <li>User A is identified as the call session initiator in the From field.</li> <li>A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>The transaction number within a single call leg is identified in the CSeq field.</li> <li>The media capability User A is ready to receive is specified.</li> <li>The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE–Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing–User B to Proxy server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	200 OK–User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F6	200 OK–Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.
F7	ACK–User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK–Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.



Step	Action	Description
F9	REFER–User B to Proxy Server	User B sends a REFER message to the proxy server. User B performs a blind transfer of User A to User C.
F10	202 Accepted–Proxy Server to User B	The proxy server sends a SIP 202 Accept response to User B. The 202 Accepted response notifies User B that the proxy server has received the REFER message.
F11	REFER–Proxy Server to User A	The proxy server forwards the REFER message to User A.
F12	202 Accepted–User A to Proxy Server	User A sends a SIP 202 Accept response to the proxy server. The 202 Accepted response indicates that User A accepts the transfer.
F13	BYE–User B to Proxy Server	User B terminates the call session by sending a SIP BYE request to the proxy server. The BYE request indicates that User B wants to release the call.
F14	BYE–Proxy Server to User A	The proxy server forwards the BYE request to User A.
F15	200OK–User A to Proxy Server	User A sends a SIP 200 OK response to the proxy server. The 200 OK response confirms that User A has received the BYE request.
F16	200OK–Proxy Server to User B	The proxy server forwards the SIP 200 OK response to User B.
F17	INVITE–User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.
F18	INVITE–Proxy Server to User C	The proxy server maps the SIP URI in the To field to User C.
F19	180 Ringing–User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F20	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted

Step	Action	Description
F21	200OK–User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies the proxy server that the connection has been made.
F22	200OK–Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A.
F23	ACK– User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F24	ACK–Proxy Server to User C	The proxy server forwards the ACK message to User C. The ACK confirms that User A has received the 200 OK response. The call session is now active.

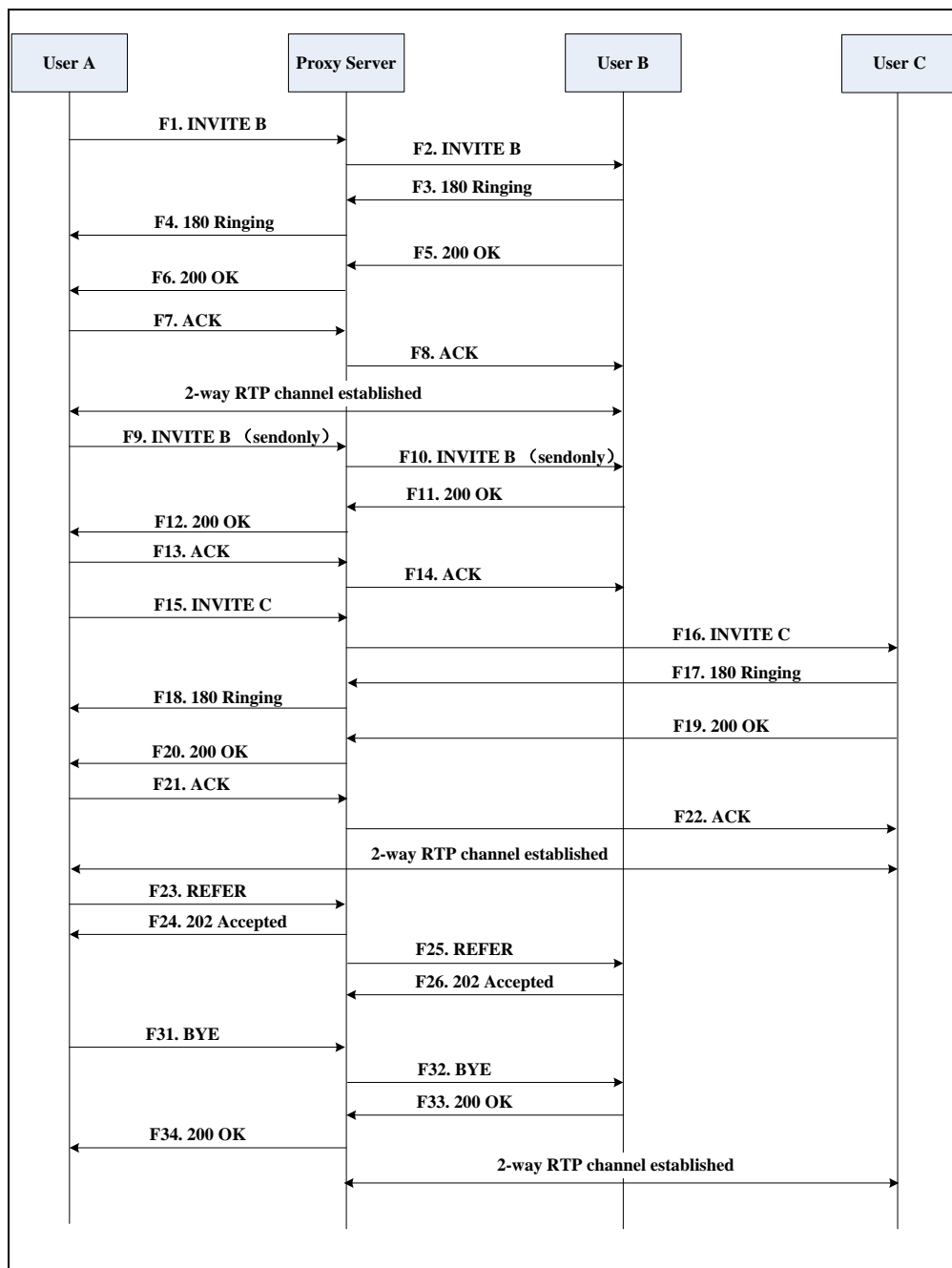
## Call Transfer with Consultation

The following figure illustrates a successful call between Yealink SIP IP phones in which two parties are in a call and then one of the parties transfers the call to the third party with consultation. This is called attended transfer. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

**The call flow scenario is as follows:**

1. User A calls User B.
2. User B answers the call.
3. User A calls User C.
4. User C answers the call.
5. User A transfers the call to User C.

Call is established between User B and User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>The IP address of User B is inserted in the Request-URI field.</li> <li>User A is identified as the call session</li> </ul>

Step	Action	Description
		<p>initiator in the From field.</p> <ul style="list-style-type: none"> <li>A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>The transaction number within a single call leg is identified in the CSeq field.</li> <li>The media capability User A is ready to receive is specified.</li> <li>The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE–Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing–User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	200 OK–User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F6	200 OK–Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.
F7	ACK–User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK–Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F9	INVITE–User A to Proxy Server	User A sends a mid-call INVITE request to the proxy server with new SDP session parameters, which are used to place the call

Step	Action	Description
		on hold.
F10	INVITE–Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F11	200 OK–User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the INVITE was successfully processed.
F12	200 OK–Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User B is successfully placed on hold.
F13	ACK–User A to Proxy Server	User A sends an ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F14	ACK–Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.
F15	INVITE–User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.
F16	INVITE–Proxy Server to User C	The proxy server maps the SIP URI in the To field to User C. The proxy server sends the INVITE request to User C.
F17	180 Ringing–User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F18	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.
F19	200OK–User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F20	200OK–Proxy Server to User A	The proxy server forwards the SIP 200 OK

Step	Action	Description
		response to User A. The 200 OK response notifies User A that the connection has been made.
F21	ACK— User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F22	ACK—Proxy Server to User C	The proxy server forwards the ACK message to User C. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F23	REFER—User A to Proxy Server	User A sends a REFER message to the proxy server. User A performs a transfer of User B to User C.
F24	202 Accepted—Proxy Server to User A	The proxy server sends a SIP 202 Accepted response to User A. The 202 Accepted response notifies User A that the proxy server has received the REFER message.
F25	REFER—Proxy Server to User B	The proxy server forwards the REFER message to User B.
F26	202 Accepted—User B to Proxy Server	User B sends a SIP 202 Accept response to the proxy server. The 202 Accepted response indicates that User B accepts the transfer.
F27	BYE—User A to Proxy Server	User A terminates the call session by sending a SIP BYE request to the proxy server. The BYE request indicates that User A wants to release the call.
F28	BYE—Proxy Server to User B	The proxy server forwards the BYE request to User B.
F29	200OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that User B has received the BYE request.
F30	200OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A.

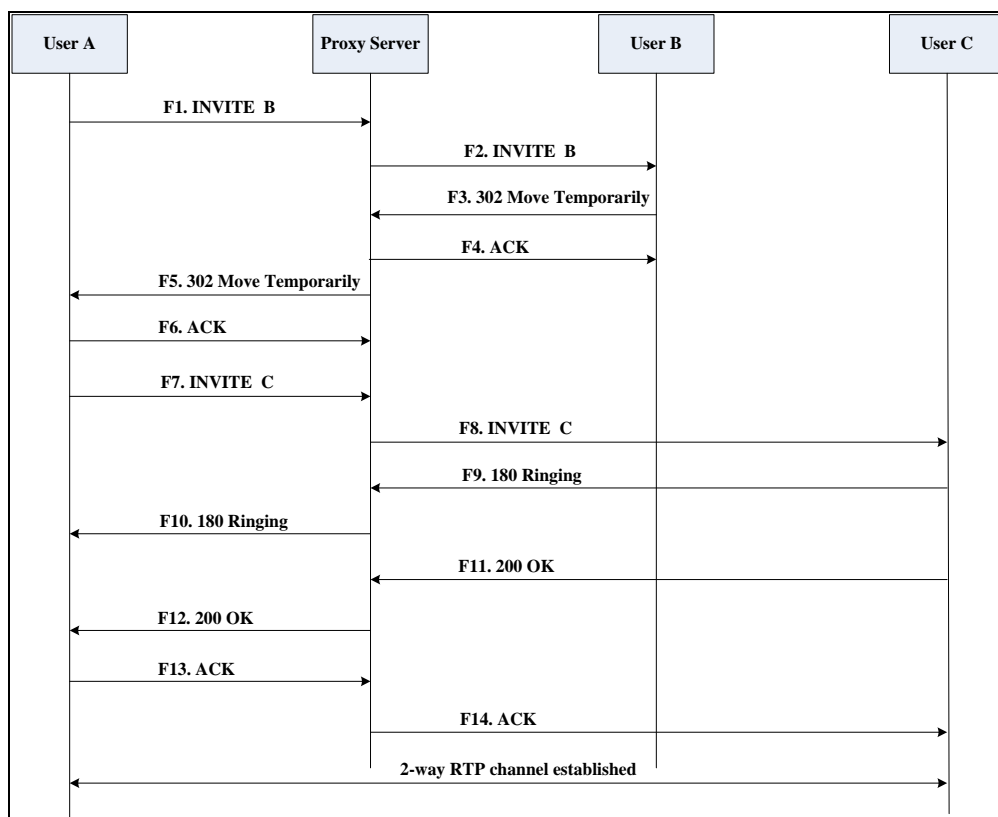
## Always Call Forward

The following figure illustrates successful call forwarding between Yealink SIP IP phones in which User B has enabled always call forward. The incoming call is immediately forwarded to User C when User A calls User B. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

**The call flow scenario is as follows:**

1. User B enables always call forward, and the destination number is User C.
2. User A calls User B.
3. User B forwards the incoming call to User C.
4. User C answers the call.

Call is established between User A and User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>• The IP address of the User B is inserted in the Request-URI field.</li> </ul>

Step	Action	Description
		<ul style="list-style-type: none"> <li>• User A is identified as the call session initiator in the From field.</li> <li>• A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>• The transaction number within a single call leg is identified in the CSeq field.</li> <li>• The media capability User A is ready to receive is specified.</li> <li>• The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE–Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	302 Move Temporarily–User B to Proxy Server	User B sends a SIP 302 Moved Temporarily message to the proxy server. The message indicates that User B is not available at SIP phone B. User B rewrites the contact-URI.
F4	ACK–Proxy Server to User B	The proxy server sends a SIP ACK to User B, the ACK message notifies User B that the proxy server has received the 302 Move Temporarily message.
F5	302 Move Temporarily–Proxy Server to User A	The proxy server forwards the 302 Moved Temporarily message to User A.
F6	ACK–User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK message notifies the proxy server that User A has received the 302 Move Temporarily message.
F7	INVITE–User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requested the call.
F8	INVITE–Proxy Server to User C	The proxy server maps the SIP URI in the To field to User C. The proxy server sends the SIP INVITE request to User C.
F9	180 Ringing–User C to Proxy	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response



Step	Action	Description
	Server	indicates that the user is being alerted.
F10	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.
F11	200OK–User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F12	200OK–Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A. The 200 OK response notifies User A that the connection has been made.
F13	ACK–User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F14	ACK–Proxy Server to User C	The proxy server forwards the ACK message to User C. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.

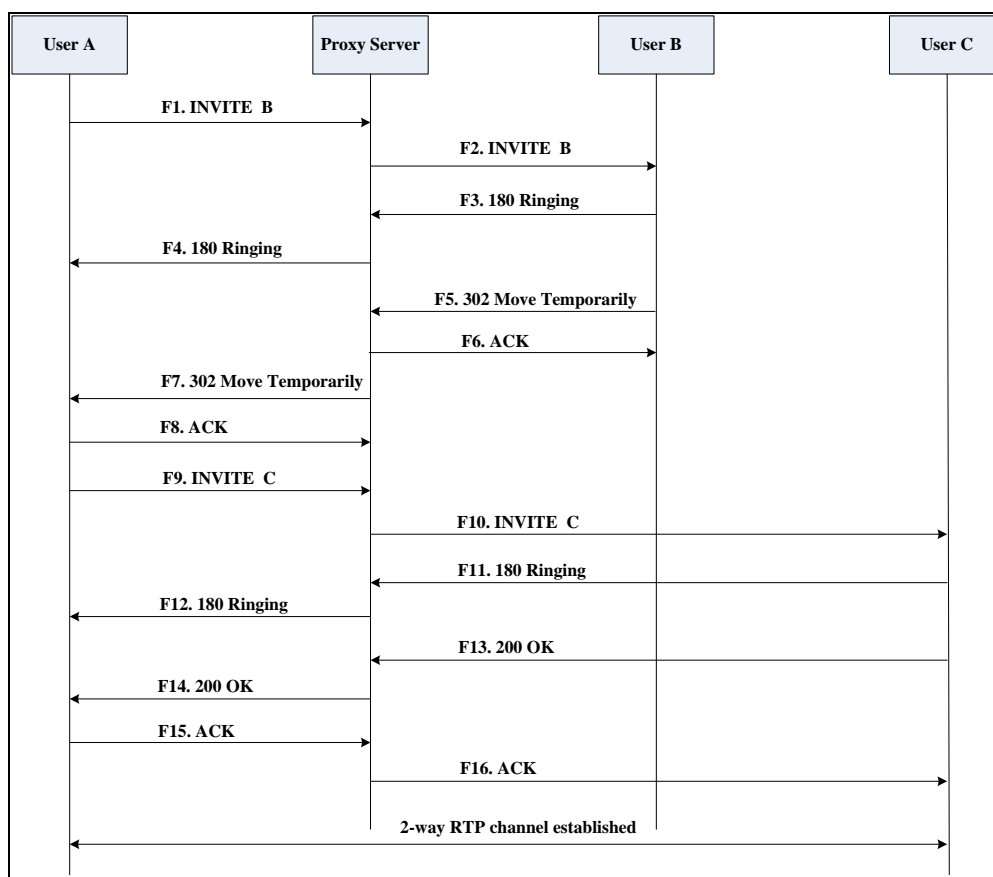
## Busy Call Forward

The following figure illustrates successful call forwarding between Yealink SIP IP phones in which User B has enabled busy call forward. The incoming call is forwarded to User C when User B is busy. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

**The call flow scenario is as follows:**

1. User B enables busy call forward, and the destination number is User C.
2. User A calls User B.
3. User B is busy.
4. User B forwards the incoming call to User C.
5. User C answers the call.

Call is established between User A and User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>User A sends the INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>The IP address of User B is inserted in the Request-URI field.</li> <li>User A is identified as the call session initiator in the From field.</li> <li>A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>The transaction number within a single call leg is identified in the CSeq field.</li> <li>The media capability User A is ready to receive is specified.</li> </ul>

Step	Action	Description
		<ul style="list-style-type: none"> <li>The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE–Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing–User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	302 Move Temporarily–User B to Proxy Server	User B sends a SIP 302 Moved Temporarily message to the proxy server. The message indicates that User B is not available at SIP phone B. User B rewrites the contact-URI.
F6	ACK–Proxy Server to User B	The proxy server sends a SIP ACK to User B, the ACK message notifies User B that the proxy server has received the ACK message.
F7	302 Move Temporarily–Proxy Server to User A	The proxy server forwards the 302 Moved Temporarily message to User A.
F8	ACK–User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK message notifies the proxy server that User A has received the ACK message.
F9	INVITE–User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.
F10	INVITE–Proxy Server to User C	The proxy server forwards the SIP INVITE request to User C.
F11	180 Ringing–User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F12	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.

Step	Action	Description
F13	200OK–User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F14	200OK–Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A.
F15	ACK– User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F16	ACK–Proxy Server to User C	The proxy server sends the ACK message to User C.

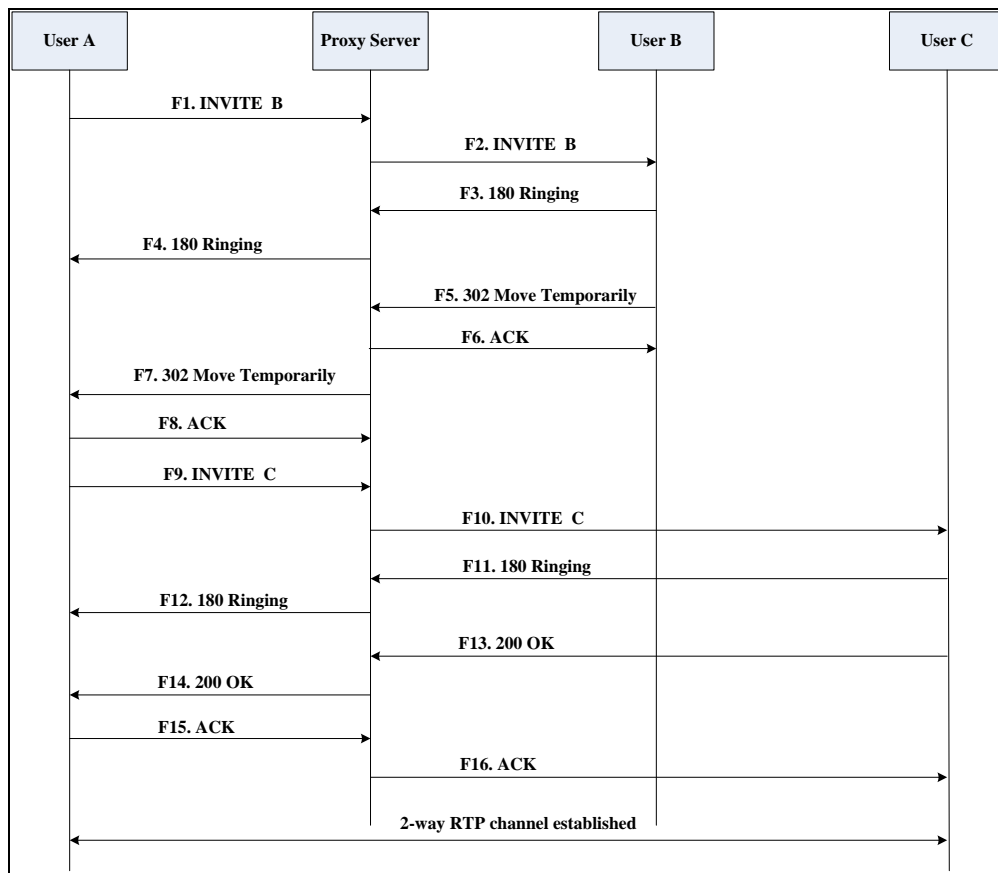
## No Answer Call Forward

The following figure illustrates successful call forwarding between Yealink SIP IP phones in which User B has enabled no answer call forward. The incoming call is forwarded to User C when User B does not answer the incoming call after a period of time. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

**The call flow scenario is as follows:**

1. User B enables no answer call forward, and the destination number is User C.
2. User A calls User B.
3. User B does not answer the incoming call.
4. User B forwards the incoming call to User C.
5. User C answers the call.

Call is established between User A and User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>User A sends the INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>The IP address of User B is inserted in the Request-URI field.</li> <li>User A is identified as the call session initiator in the From field.</li> <li>A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> <li>The transaction number within a single call leg is identified in the CSeq field.</li> <li>The media capability User A is ready to receive is specified.</li> </ul>

Step	Action	Description
		<ul style="list-style-type: none"> <li>The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE–Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing–User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	302 Move Temporarily–User B to Proxy Server	User B sends a SIP 302 Moved Temporarily message to the proxy server. The message indicates that User B is not available at SIP phone B. User B rewrites the contact-URI.
F6	ACK–Proxy Server to User B	The proxy server sends a SIP ACK to User B, the ACK message notifies User B that the proxy server has received the ACK message.
F7	302 Move Temporarily–Proxy Server to User A	The proxy server forwards the 302 Moved Temporarily message to User A.
F8	ACK–User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK message notifies the proxy server that User A has received the ACK message.
F9	INVITE–User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.
F10	INVITE–Proxy Server to User C	The proxy server forwards the SIP INVITE request to User C.
F11	180 Ringing–User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F12	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.

Step	Action	Description
F13	200OK–User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F14	200OK–Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A. The 200 OK response notifies User A that the connection has been made.
F15	ACK– User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F16	ACK–Proxy Server to User C	The proxy server sends the ACK message to User C. The ACK confirms that the proxy server has received the 200 OK response.

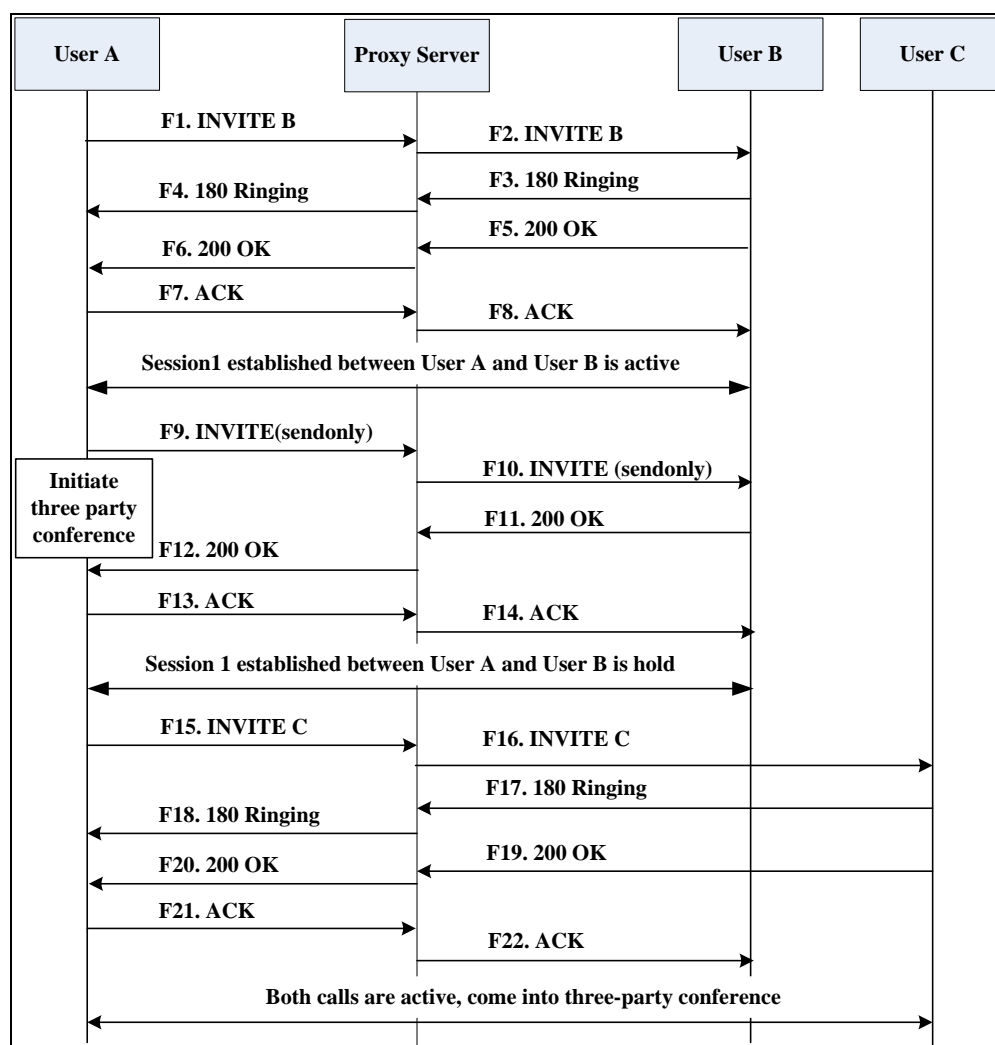
## Call Conference

The following figure illustrates successful three-way calling between Yealink IP phones in which User A mixes two RTP channels and therefore establishes a conference between User B and User C. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

**The call flow scenario is as follows:**

1. User A calls User B.
2. User B answers the call.
3. User A places User B on hold.
4. User A calls User C.
5. User C answers the call.

6. User A mixes the RTP channels and establishes a conference between User B and User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>User A sends the INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> <li>The IP address of User B is inserted in the Request-URI field.</li> <li>User A is identified as the call session initiator in the From field.</li> <li>A unique numeric identifier is assigned to the call and is inserted in the Call-ID field.</li> </ul>



Step	Action	Description
		<ul style="list-style-type: none"> <li>The transaction number within a single call leg is identified in the CSeq field.</li> <li>The media capability User A is ready to receive is specified.</li> <li>The port on which User B is prepared to receive the RTP data is specified.</li> </ul>
F2	INVITE–Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. Proxy server forwards the INVITE message to User B.
F3	180 Ringing–User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing–Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	200 OK–User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F6	200 OK–Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.
F7	ACK–User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK–Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F9	INVITE–User A to Proxy Server	User A sends a mid-call INVITE request to the proxy server with new SDP session parameters, which are used to place the call on hold.
F10	INVITE–Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.

Step	Action	Description
F11	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the INVITE is successfully processed.
F12	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User A that User B is successfully placed on hold.
F13	ACK—User A to Proxy Server	User A sends the ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F14	ACK—Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.
F15	INVITE—User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.
F16	INVITE—Proxy Server to User C	The proxy server maps the SIP URI in the To field to User C. The proxy server sends the SIP INVITE request to User C.
F17	180 Ringing—User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F18	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.
F19	200OK—User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F20	200OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A. The 200 OK response notifies User A that the connection has been made.

Step	Action	Description
F21	ACK– User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F22	ACK–Proxy Server to User C	The proxy server sends the ACK message to User C. The ACK confirms that the proxy server has received the 200 OK response.



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