



iCallDroid spot User Manual



Address: F/3, Building 127, Jindi Industrial Zone, Futian District, Shenzhen, Guangdong, China, 518048

Tel: +86-755-82535461, 82535095, 82535362

Fax: +86-755-83823074

Business Contact: sales@openvox.cn

Technical Support: support@openvox.com.cn

Business Hours: 09:00-18:00(GMT+8) from Monday to Friday

URL: www.openvox.cn

Menu

| | |
|-----------------------------|----|
| Getting Started | 4 |
| Introduction | 4 |
| Packing list | 5 |
| Specification..... | 5 |
| Hardware Setup | 6 |
| First Login to Wizard | 7 |
| One touch to deploy* | 14 |
| Dashboard..... | 14 |
| Wifi Router..... | 15 |
| Networks..... | 15 |
| WAN/LAN/TimeZone | 15 |
| Wireless | 16 |
| Dynamic DNS | 18 |
| VPN | 19 |
| Utilities..... | 20 |
| DHCP Server | 20 |
| Wireless Mac Filter | 21 |
| Port Forward | 22 |
| Firewall/UPNP/DMZ..... | 23 |
| QoS..... | 24 |
| Disk and Sharing | 25 |
| System..... | 28 |
| Real Status | 29 |
| Call Details Report..... | 29 |
| Logs View | 29 |
| Admin User | 30 |
| Reboot | 30 |
| Reset Factory | 31 |
| VoIP..... | 31 |
| PBX General | 31 |
| Extensions..... | 32 |
| Line Provider | 38 |
| Advanced | 42 |
| Outbound Routes..... | 42 |
| Inbound Routes..... | 45 |
| Conference..... | 47 |
| Queues..... | 48 |
| Simple IVR..... | 49 |
| Time Frames..... | 51 |
| PBX More | 52 |
| Sound Files..... | 53 |

| | |
|----------------|----|
| Option | 54 |
| Glossary | 58 |

Getting Started

Thank you for purchasing iCallDroid spot (WiFi Router+ IP PBX).It is one cost-efficient yet easy-to-configure IP PBX in the market today.

Administrating a VoIP system can be a daunting task for administrators unfamiliar with VoIP.

This guide is designed to help you plan and configure iCallDroid spot Voice over IP (VoIP) deployments.

Introduction



The iCallDroid spot (300M WiFi Router+ IP PBX) is the ideal system for small businesses and home offices requiring a pint-sized yet powerful on-premise wifi router IP PBX. It supports up to 4 concurrent calls with G.729 codec, but it might be less under high loading network transmission because of more CPU consumption.

The compact solid-state device supports 16 extensions and offers a wide range of IP PBX telephony features.

Keeping up with the demands of sustainability, the iCallDroid spot is based on a low-power, high performance MIPS processor, providing the complicated communication features including the hardest HD communication protocol, complete router features and QoS (Ensure the voice quality in a case the bandwidth is not enough). Meanwhile, the feature of one touch to deploy the phones makes the configuration of phones easy and enjoyable thing.

Packing list

1 unit iCallDroid spot

1 Piece Power Supply (12V,1A)

1 piece of 2-meter Network cables

1 Piece 8G MLC USB

Specification

MIPS Processor

64MB RAM/16MB FLASH

1xRJ45 10/100MB Ethernet port WAN

1xRJ45 Debug port

Button: Reset Button, One Touch to auto deploy

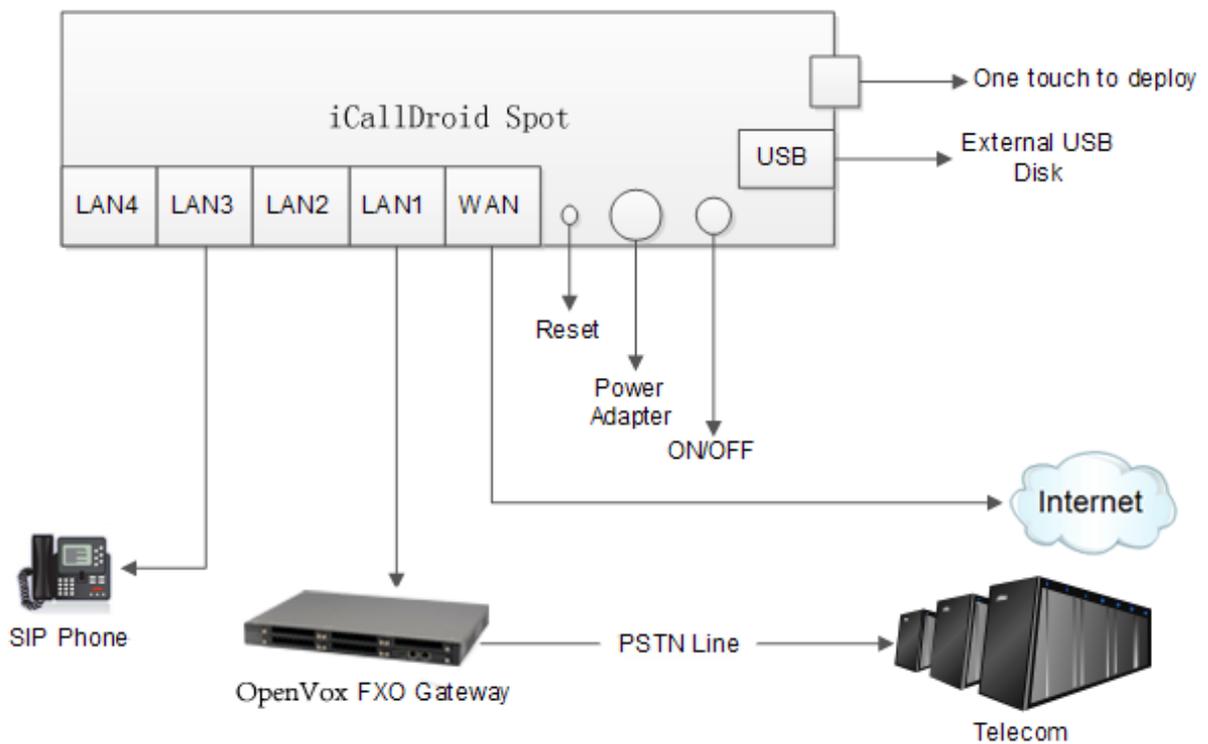
Power adapter: AC 100~240V input and DC 12V/1A output

Power consumption: 1.2-2.0W

Operating humidity: 10~95%

Operating temperature: 0~45°C

Hardware Setup



You may check the above picture to configurate.

Step 1: Connect the LAN port of iCallDroid spot with your corporate IP network. Before you connect the iCallDroid spot to the network, please check if your network can work normally.

Step 2: Plug in and open your browser to visit the web address: [Http://192.168.1.1](http://192.168.1.1).

Make sure your PC IP address is 192.168.1.XXX.

(If you use IE6 and above, the prefix address <http://> can't be left out)

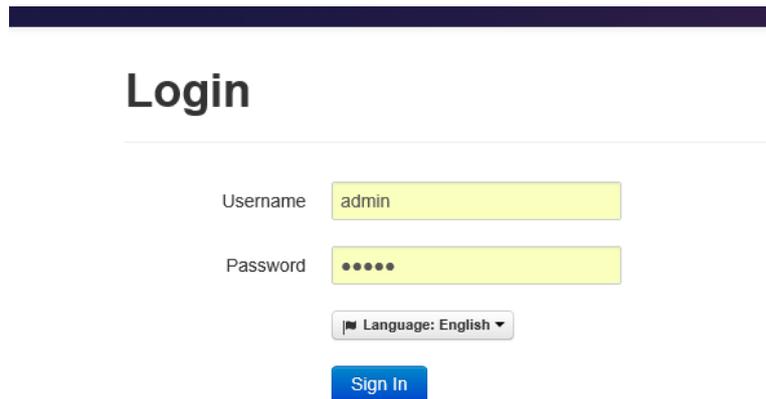
Now we access to the Wizard page.

Username: admin (By default)

Password: admin (By default)

Or you can find the WiFi SSID : WiFi Router and log in. The visit the web : [Http://192.168.1.1](http://192.168.1.1)

You can also choose the web language.



Login

Username

Password

Language: English ▾

[Sign In](#)

First Login to Wizard

This is your first time to log in, it will show Wizard Processing. It is simple and brief to deploy.

In most cases, the default settings can be used for the rest of the configuration.

Wizard Processing Intro

1 2 3 4 5 6 7

Thank you for choosing our product. It is the first time to start the system, we will help you to do the fast configuration to make it work. If need to set up by yourself just [click the Abort](#) to access the normal interface and you can set the Wizard again when need later.

On this page, you can select or cancel the Wizard projects:

- WAN / Time Zone
- Wireless
- Extensions
- Trunk
- PBX

[Abort](#) [← Prev](#) [Next →](#)

If you want to quit the Wizard, just Abort it. And if you want to access Wizard, just click the Wizard.

The screenshot shows the 'Welcome To The Unified Communications System' dashboard. The left sidebar contains navigation options: Wizard, Dashboard (selected), Logout, Networks, Utilities, System, PBX General, Advanced, and PBX More. The main content area is divided into several panels:

- WAN:** MAC 00:86:30:4A:01:89, Protocol static, IP Addr 172.16.0.210, Netmask 255.255.0.0, Gateway 172.16.0.254, DNS 172.16.0.254
- LAN:** MAC 00:86:30:4A:01:88, IP Addr 192.168.1.1, Netmask 255.255.0.0
- WIFI:** MAC 00:86:30:4A:01:8A, SSID WifiRouterPBX_4A018A, Channel auto, Encryption No Encryption
- VPN:** (Empty panel)
- SYS:** Firmware 5.5.4796.0, Model 0110, Serial ID 00010418000013, Memory 70% used, Timezone UTC
- PBX:** Devices (View Real Status), SIP Port (TCP/UDP) 0.0.0.0:6620, Concurrent 4 calls, Settings: 16 sip extensions, 8 feature extensions, 8 trunks, 16 call routes, 4 conferences, 10 queues, 10 simpleivr.

You can click Prev to return and Next to do the following steps.

On the Wizard page, you can set the WAN IP and Time Zone, Wireless, Extension, line provider and PBX.

You can select which part need to set then do the Next.

(You can also follow all the default settings and confirm)

Next is the WAN and Time Zone :

Set how to connect to the net. To set timezone to your local.

Protocol STATIC IP
 DHCP
 PPPOE

IP Address

Netmask

Gateway

DNS 1

DNS 2

Time Zone

You can choose the Protocol Static IP, DHCP, PPPoE.

Static IP

Using a static IP address is the most reliable way to ensure your server IP address does not change. To find an IP address that is not in use on your network and will not be used for another client by the DHCP server or used by some other devices.

DHCP (Dynamic Host Configuration Protocol, DHCP)

PPPoE (Point to Point Protocol over Ethernet)

Fill out the account information from your telecom operator and **Next**.

Set how to connect to the net, To set timezone to your local.

Protocol
 STATIC IP
 DHCP
 PPPOE

Username

Password

Service

Dns Mode

Dns IP:

Connect Mode

Time Zone

Time Zone

You can set the local time here and it is important for generating accurate call reports for the system. And Time Frame will also analysis the system time to switch to the proper IVR.

If you select the incorrect time zone, or you move to a different time zone later, you can change it in the Wizard or in the Network-WAN/LAN/Time Zone.

Time Zone

Now we came to the Wireless page:

Set to enable wireless.

Enable Wifi Yes No

SSID

Hide SSID

Encryption ▼

Crypto ▼

Key

Here you can set the wireless option.

Enable Wifi and set the SSID (Service Set Identifier)

According to the tip in the black, choose your encryption, crypto and key.

Now we come to the Extension page.

The system has already auto generated 16 SIP extension by default.

(iCallDroid spot supports max 16 SIP extensions.)

You can set the extension number, password and the caller name.

If want to set the extensions quantity as you like, just amend this in **Extension** table.

Wizard Processing Extensions

1 2 3 4 5 6 7

Create the extension, you can go directly to the next step or modify the extension number or password.

| | | | | | |
|---------|-----|-----------|----------|--------------|--------------|
| Number: | 800 | Password: | 66876233 | Caller Name: | default: 800 |
| Number: | 801 | Password: | 94391230 | Caller Name: | default: 801 |
| Number: | 802 | Password: | 75369629 | Caller Name: | default: 802 |
| Number: | 803 | Password: | 42917292 | Caller Name: | default: 803 |
| Number: | 804 | Password: | 19885287 | Caller Name: | default: 804 |
| Number: | 805 | Password: | 10632983 | Caller Name: | default: 805 |
| Number: | 806 | Password: | 29721975 | Caller Name: | default: 806 |
| Number: | 807 | Password: | 47024526 | Caller Name: | default: 807 |
| Number: | 808 | Password: | 14680209 | Caller Name: | default: 808 |
| Number: | 809 | Password: | 40814707 | Caller Name: | default: 809 |

Next we come to Trunk Setting.

Wizard Processing Trunk

1 2 3 4 5 6 7

Set to connect to ITSP(Internet Telephony Service Provider), you can go directly to the next step to skip this area.

Name

Provider Host

Provider Port

Account

Password

Set a SIP Trunk here. Fill out the Provider Host, account and password from your Internet Telephony Service Provider then one SIP Trunk will be built.

iCallDroid spot supports maximum 8 SIP trunks. If you want to build more, just do it in the **Line Provider**.

Next is PBX setting :

Wizard Processing PBX

1 2 3 4 5 6 7

Set PBX routes and settings.

Abort Prev Next

Outbound Calls Default outbound will auto choose ther Trunk which you set the default outbound line.

Incoming Calls When the call comes in, the system will ring simultaneously all the numbers, unless there is a number picked up the phone.
 Playback a voice menu, and wait user input number, or press 0 to number 800.

Conference Number

Abort Prev Next

Here you can set how the system will deal with when make outbound calls or receive the inbound calls.

Outbound calls: Auto-select the Trunk which you set it as the default outbound line.

Incoming calls: Ring all the numbers when the call comes in or just into an IVR.

If want to set other routes, you can do it in Outbound Routes and Inbound Routes after the Wizard.

You can also set the conference number here. By default is 300.

Next well done! Just confirm to process.

Wizard Processing Final

1 2 3 4 5 6 7

Thanks for your patience. All is ready now. Please click 'Confirm to Process' to start. Please don't power off or reboot the device when in the process. If it failed, restart the device and login web to do the factory reset. Click 'Abort' to stop the process.

Abort Prev Confirm to Process

Then the iCallDroid spot begin to configure. Now enjoy the HTML5 interface.

It will auto-restart the PBX and if you can't see the page you can refresh it. If you amended the static IP address, re-visit the changed IP. Re-log in, you will see the dashboard of PBX iCallDroid spot.

Wizard Processing Working, Please be patient and do not power off the device!



OK. This is the first time to access the iCallDroid spot.

One touch to deploy*

Dashboard

You can see the status of WAN, LAN, VPN, Wifi, System and PBX in the dashboard.

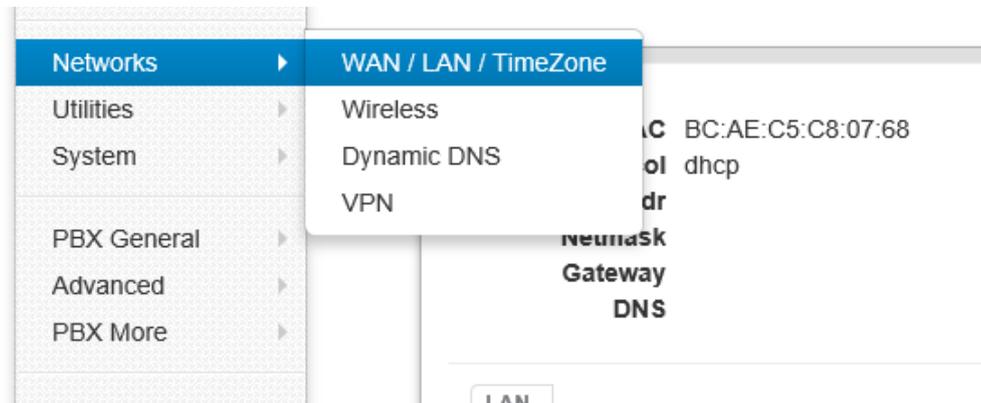
Usually you will find the information you need here.

The dashboard is titled 'Welcome To The Unified Communications System' and is logged in as 'admin'. It features a sidebar menu with options like Wizard, Dashboard, Logout, Networks, Utilities, System, PBX General, Advanced, and PBX More. The main content area is divided into several panels:

- WAN:**
 - MAC: 00:86:30:4A:01:89
 - Protocol: static
 - IP Addr: 172.16.0.210
 - Netmask: 255.255.0.0
 - Gateway: 172.16.0.254
 - DNS: 172.16.0.254
- LAN:**
 - MAC: 00:86:30:4A:01:88
 - IP Addr: 192.168.1.1
 - Netmask: 255.255.255.0
- WIFI:**
 - MAC: 00:86:30:4A:01:8A
 - SSID: WifiRouterPBX_4A018A
 - Channel: auto
 - Encryption: No Encryption
- VPN:** (Empty panel)
- SYS:**
 - Firmware: 5.5.4796.0
 - Model: 0110
 - Serial ID: 00010418000013
 - Memory: 68% used
 - Timezone: UTC
- PBX:**
 - Devices: [View Real Status](#)
 - SIP Port (TCP/UDP): 0.0.0.0:6620
 - Concurrent: 4 calls
 - Settings:
 - 16 sip extensions
 - 8 feature extensions
 - 8 trunks
 - 16 call routes
 - 4 conferences
 - 10 queues
 - 10 simplivr's

Wifi Router

Networks

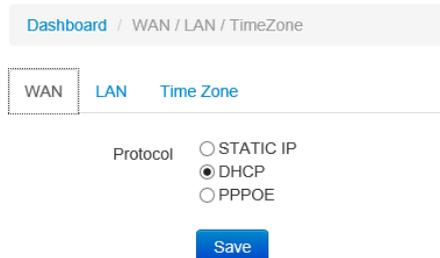


Here as the same as the setting in the Wizard.

You can change the WAN Port here and also Time Zone.

WAN/LAN/TimeZone

WAN / LAN / TimeZone



Here you can set the WAN port.

WAN / LAN / TimeZone

Dashboard / WAN / LAN / TimeZone

WAN LAN Time Zone

IP Address

Netmask

Here you can set the LAN port.

WAN / LAN / TimeZone

Dashboard / WAN / LAN / TimeZone

WAN LAN Time Zone

Time Zone

Here you can set the time Zone.

Wireless

Here is the wireless setting. Just as it in the Wizard.



The general setting is the same as it in the Wizard.

Wireless

[Dashboard](#) / [Wireless](#)

[General](#) [Expert](#)

Enable Wifi Yes No

SSID

Hide SSID

Encryption ▼

[Save](#)

You will have this option as in Expert below:

Channel, HT Bandwidth, Wireless Mode, Tx Power, WMM (Wi-Fi Multimedia)

WMM(Wi-Fi Multimedia): is a Wi-Fi Alliance interoperability certification, based on the IEEE 802.11e standard. It provides basic Quality of service (QoS) features to IEEE 802.11 networks. WMM prioritizes traffic according to four Access Categories (AC) - voice, video, best effort, and background. However, it does not provide guaranteed throughput. It is suitable for simple applications that require QoS, such as Voice over IP (VoIP) on Wi-Fi phones.

Wireless

Dashboard / Wireless

General Expert

Channel ▼

HT Bandwidth 20Mhz Only 20Mhz/40Mhz Auto 40Mhz Only

Wireless Mode ▼

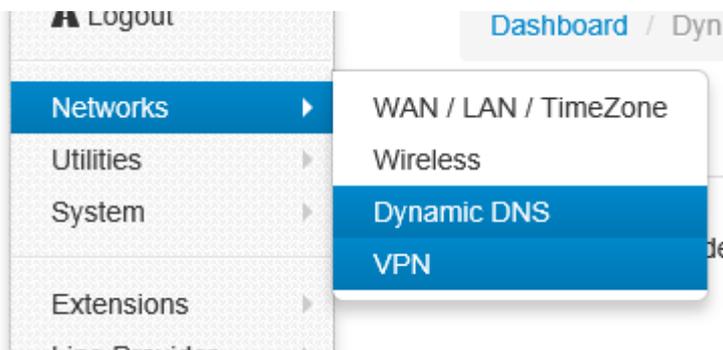
Tx Power %

WMM Disable Enable

Dynamic DNS

Here you can set the Dynamic DNS.

Dynamic DNS or DDNS is a method of updating, in real time, a Domain Name System (DNS) to point to a changing IP address on the Internet. This is used to provide a persistent domain name for a resource that may change location on the network.



Dynamic DNS

Dashboard / Dynamic DNS

General

Provider

Here you can choose two free dynamic domain web to apply the account.

Dynamic DNS

Dashboard / Dynamic DNS

General

Provider

Username

Password

Domain

Check Time (min)

VPN

Virtual Private Network



VPN

Dashboard / VPN

General

| | |
|-------------------------------------|---|
| VPN Mode | <input type="text" value="Disable"/> <input checked="" type="text" value="Permission LAN"/> <input type="text" value="PBX Only"/> |
| Protocol | PPTP |
| Server | <input type="text"/> |
| Username | <input type="text" value="admin"/> |
| Password | <input type="password"/> |
| <input type="button" value="Save"/> | |

If you choose Disable that means the VPN doesn't work.

Permission LAN means all the data from the LAN will go via VPN.

PBX Only means only the PBX data (VoIP) will go via VPN.

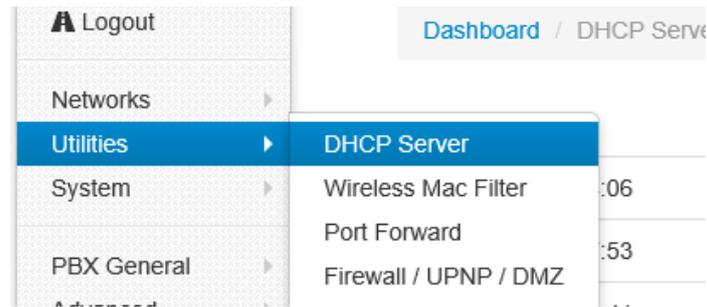
Utilities

DHCP Server

You can check the devices in the network here and also set the server.

You can enable DHCP server or not.

You can set where Client IP Start from and Max clients.



DHCP Server

[Dashboard](#) / [DHCP Server](#)

| Lease Expiry | MAC | IP Address | Device Name |
|---|--------------|------------|-------------|
| Enable DHCP <input checked="" type="radio"/> Yes <input type="radio"/> No | | | |
| Client IP Start | 172.16.0.210 | 100 | |
| Max Clients | 150 | | |
| <input type="button" value="Save"/> | | | |

DHCP Server

[Dashboard](#) / [DHCP Server](#)

| Lease Expiry | MAC | IP Address | Device Name |
|---------------------|-------------------|--------------|--------------------------|
| 1970-01-01 23:04:06 | 00:37:6d:2a:bf:bb | 172.16.0.123 | * |
| 1970-01-01 22:37:53 | c4:6a:b7:ef:d8:8d | 172.16.0.129 | android-ee14438474b638c7 |
| 1970-01-01 20:12:41 | bc:77:37:66:ab:02 | 172.16.0.164 | Apple-PC |

| | | | |
|---|--------------|-----|--|
| Enable DHCP <input checked="" type="radio"/> Yes <input type="radio"/> No | | | |
| Client IP Start | 172.16.0.188 | 100 | |
| Max Clients | 150 | | |
| <input type="button" value="Save"/> | | | |

Wireless Mac Filter

You can set the white list and black list to manage the devices which want to visit the network.

All the devices in white list are permitted to visit the network.

While if in the black list, no device can visit the network.



Wireless Mac Filter

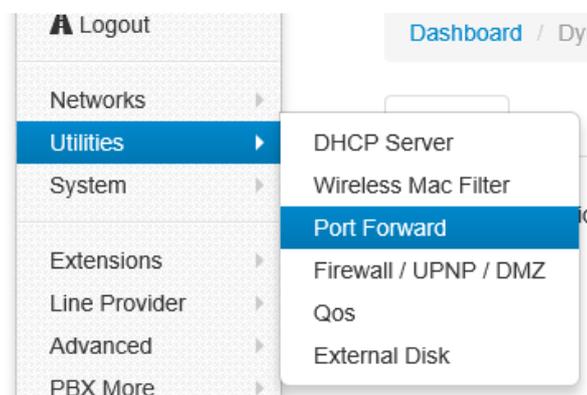
Dashboard / Wireless Mac Filter

Control Disabled White list Black list

| MAC Address | |
|-------------|----------------------|
| 1 | <input type="text"/> |
| 2 | <input type="text"/> |
| 3 | <input type="text"/> |
| 4 | <input type="text"/> |
| 5 | <input type="text"/> |
| 6 | <input type="text"/> |
| 7 | <input type="text"/> |

Port Forward

Set the IP address, Mac, port from the source and destination, choose the protocol and enable.



Port Forward

Dashboard / Port Forward

Port Forward

[Save](#)

| | Source | Destination | Protocol |
|---------------------------------|---|--|------------------------------|
| <input type="checkbox"/> Enable | Zone: All <input type="text"/> IP: <input type="text"/> MAC: <input type="text"/> PORT: <input type="text"/> | Zone: All <input type="text"/> IP: <input type="text"/> PORT: <input type="text"/> | TCP+UDP <input type="text"/> |
| <input type="checkbox"/> Enable | Zone: All <input type="text"/> IP: <input type="text"/> MAC: <input type="text"/> PORT: <input type="text"/> | Zone: All <input type="text"/> IP: <input type="text"/> PORT: <input type="text"/> | TCP+UDP <input type="text"/> |

Firewall/UPNP/DMZ

UPN (Universal Plug and Play)

DMZ (Demilitarized zone)

Here you can set the DMZ PC IP, if enable the UPNP&NAT-PMP and WAN Ping response, WAN web access, WAN Ftp and WAN PBX.

Firewall

Dashboard / Firewall

General [Access Rules](#)

DMZ PC IP

UPNP & NAT-PMP Disabled Enable

WAN Ping Response Yes No

WAN Web Access Yes No

WAN Ftp Access Yes No

WAN Pbx Access Yes, port is No

Firewall

Dashboard / Firewall

General Access Rules

Save

| | Source | Destination | Protocol | Action |
|---------------------------------|--|--|------------------------------|-----------------------------|
| <input type="checkbox"/> Enable | All <input type="text"/> <input type="text"/> <input type="text"/> | All <input type="text"/> <input type="text"/> <input type="text"/> | TCP+UDP <input type="text"/> | Accept <input type="text"/> |
| <input type="checkbox"/> Enable | All <input type="text"/> <input type="text"/> <input type="text"/> | All <input type="text"/> <input type="text"/> <input type="text"/> | TCP+UDP <input type="text"/> | Accept <input type="text"/> |

Here you can set the Access Rules.

QoS

Quality of service. The system enables QoS by default.

It ensures the voice quality in a case the bandwidth is not enough.

You can also set the WAN download/upload bandwidth.

Qos

Dashboard / Qos

General

Enable WAN Qos YES NO

WAN Download Bandwidth kBit/s

WAN Upload Bandwidth kBit/s

Save

Disk and Sharing

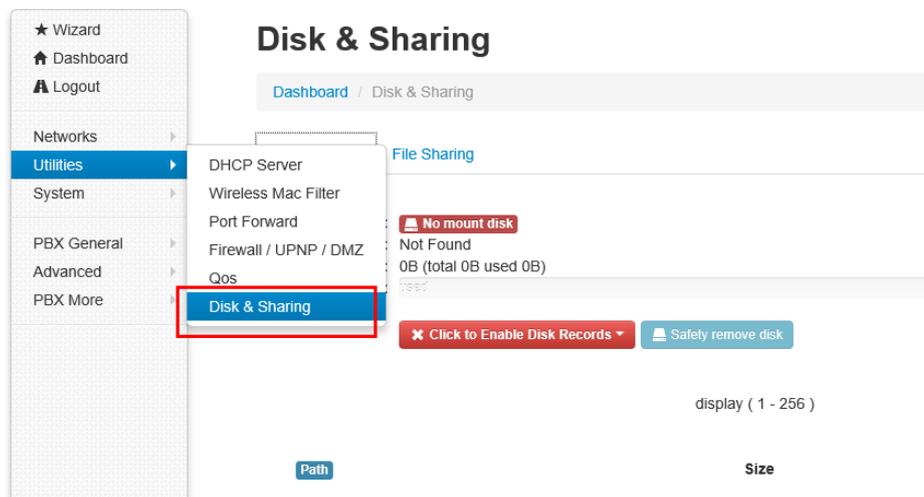
Here you can insert a USB device to record the calls and voicemail.

And also set the file sharing.

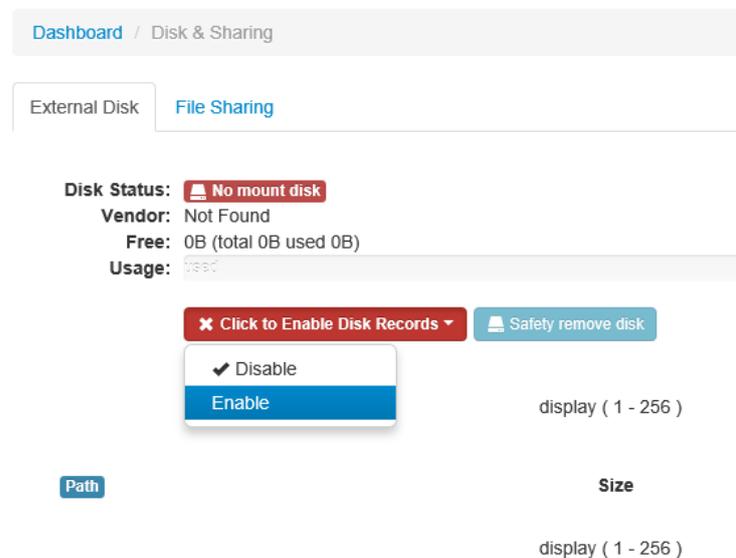
It supports FAT32, EXT4 or based the MLC USB disk.

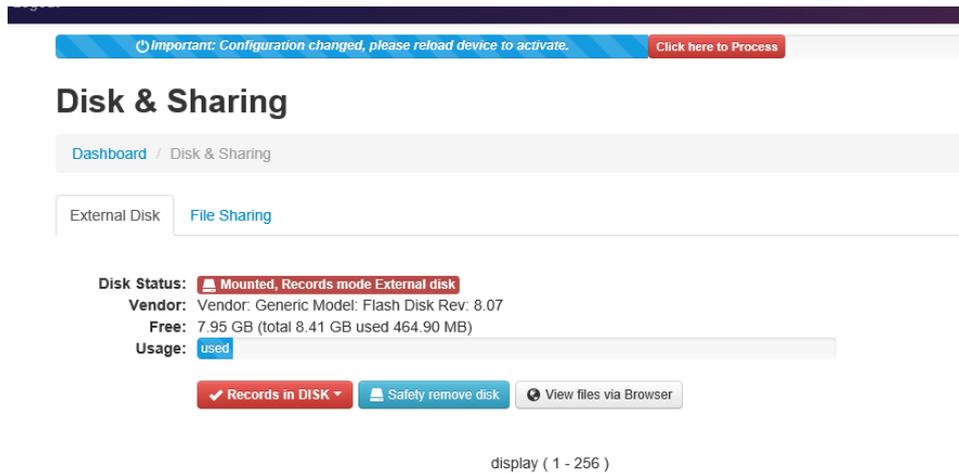
We recommend using the USB device with power adaptor alone.

Insert one USB. Please remember to click the button “Enbale” to record in the Disk.

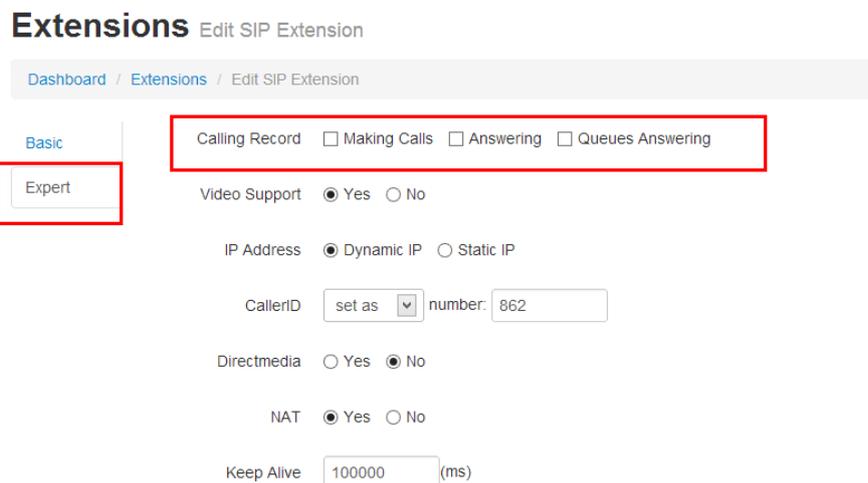
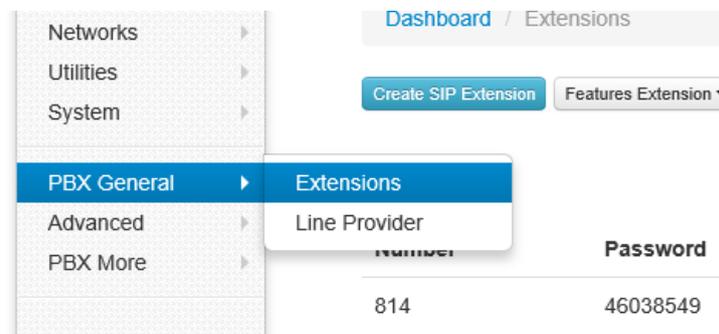


Disk & Sharing





Now you can find the **call recording option** in the Extensions-Extension-Expert. Recording file will be saving on USB external disk memory only.



You can also check the PBX data-Extensions to find the record file.

You can download and delete the files.

Or you can check them in FTP as below.

Disk Status: Mounted, Records mode External disk
Vendor: Vendor: PNY Model: USB 2.0 FD Rev: 1100
Free: 6.91 GB (total 8.14 GB used 1.24 GB)
Usage: used 15.21%

Records mode external disk ▾ Pbxdata FTP Manage safety remove disk

display (1 - 256)

| Path pbxdata/extension/863/20130416 | Size | Modify |
|-------------------------------------|--------|---------------------|
| Back to .. | | |
| 20130416054005_863_862.WAV | 0.00 B | 2013-04-16 05:40:04 |
| 20130416054044_863_862.WAV | 0.00 B | 2013-04-16 05:40:44 |
| 20130416054235_863_862.WAV | 0.00 B | 2013-04-16 05:42:34 |
| 20130416054330_862_863.WAV | 0.00 B | 2013-04-16 05:43:30 |
| VM_20130416054343_862_863.WAV | 1.43 K | 2013-04-16 05:43:44 |
| 20130416055023_862_863.WAV | 0.00 B | 2013-04-16 05:50:22 |

Disk & Sharing

[Dashboard](#) / [Disk & Sharing](#)

[External Disk](#) [File Sharing](#)

Disk Status: Mounted, Records mode External disk
Vendor: Vendor: Generic Model: Flash Disk Rev: 8.07
Free: 7.95 GB (total 8.41 GB used 464.90 MB)
Usage: used

Records in DISK ▾ Safety remove disk View files via Browser

display (1 - 256)

| Path | Size |
|---|---|
| pbxdata/ | remove |

When you want to extract the USB, close all the files related to USB before click the safety remove disk and wait around 1-3 seconds. Then it will be ok as below.

External Disk

[Dashboard](#) / [External Disk](#)

Disk Status: Error: Records mode External Disk, But no mount disk!
Vendor: Not Found
Free: 0B (total 0B used 0B)
Usage: used

Records mode external disk ▾ safety remove disk

display (1 - 256)

File Sharing

You can treat the iCallDroid spot as a file sharing server now.

Disk & Sharing

Dashboard / Disk & Sharing

External Disk **File Sharing**

FTP File Remote Sharing

IP

Port

Username

Password

View via Browser

Save

Use the FTP software to connect the Server and you can enjoy the file sharing.

The password is empty by default. You can log in the FTP anonymously.

IP, Port and the username can't be amended.

System

Panel Dashboard Logout

★ Wizard
🏠 Dashboard
👤 Logout

Networks
Utilities
System
PBX General
Advanced
PBX More

Real Status

Dashboard / Real Status

PBX Local Number

| | Number |
|--|--------|
| | 801 |
| | 802 |
| | 803 |

Real Status
Call Detail Report
Logs View
Admin User
Reboot
Reset Factory

Real Status

Real Status

[Dashboard](#) / Real Status

| PBX Local Number | | | | PBX Line Provider | | | |
|--------------------|---------|------|---------|-------------------|------|------|---------|
| Extensions | | | | Trunk Register | | | |
| | Number | Type | Address | Refresh | Type | Host | Account |
| UNKNOWN | 801 | sip | | | | | |
| UNKNOWN | 802 | sip | | | | | |
| Unmonitored | 803 | sip | | | | | |
| Other Resource | | | | Trunk Connect | | | |
| Number | Section | Type | | Trunk | Type | Host | Account |

Here you can see the PBX extension status, SIP Trunk status and other Resource in the later.

Call Details Report

Here you will find the call detail report.

Call Detail Report

[Dashboard](#) / Call Detail Report

| Account | Source | Destination | Calldate | Duration/Answer | Status |
|---------|--------|-------------|----------|-----------------|--------|
| | | | | | |

(1 - 60)

(1 - 60)

Logs View

You can find the system logs here. It is very convenient to check any problem.

Click to Refresh, it will show the latest log.

Logs View

Dashboard / Logs View

Click to Refresh

```

Jan 1 00:00:32 localhost user.notice ifup: Enabling Router Solicitations on loopback (lo)
Jan 1 00:00:33 localhost user.info firewall: adding wan (eth2.2) to zone wan
Jan 1 00:00:33 localhost user.notice miniupnpd: adding firewall rules for eth2.2 to zone wan
Jan 1 00:00:38 localhost user.notice dnsmasq: DNS rebinding protection is active, will discard upstream RFC1918 responses!
Jan 1 00:00:38 localhost user.notice dnsmasq: Allowing 127.0.0.0/8 responses
Jan 1 00:00:39 localhost user.notice dnsmasq: found already running DHCP-server on interface 'br-lan' refusing to start, use 'option force 1' to override
Jan 1 00:00:39 localhost daemon.info dnsmasq[2372]: started, version 2.62 cachesize 150
Jan 1 00:00:39 localhost daemon.info dnsmasq[2372]: compile time options: IPv6 GNU-getopt no-DBus no-i18n no-IDN DHCP no-DHCPv6 no-Lua TFTP no-contrack
Jan 1 00:00:39 localhost daemon.info dnsmasq[2372]: using local addresses only for domain lan
Jan 1 00:00:39 localhost daemon.info dnsmasq[2372]: reading /tmp/resolv.conf.auto

```

Admin User

You can amend the sign-in password here.

Admin User

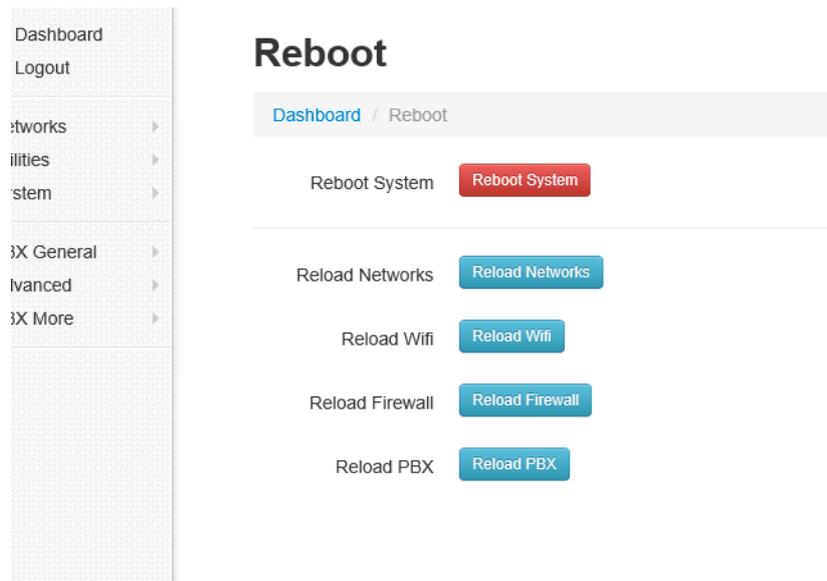
Dashboard / Admin User

| | |
|------------------|---|
| Current Password | <input type="text" value="current password"/> |
| New Password | <input type="text" value="new password"/> |
| Retry New | <input type="text" value="retry password"/> |
| | <input type="button" value="Save"/> |

Reboot

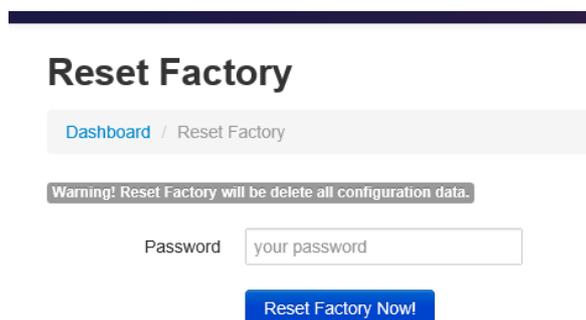
You can reboot the system here.

Also you can reload Networks, Wifi, Firewall, PBX.



Reset Factory

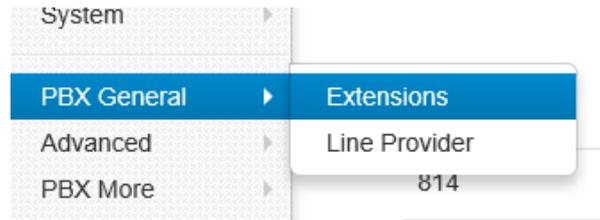
Factory Mode: Input your password to reset your device.(Admin by default)



VoIP

PBX General

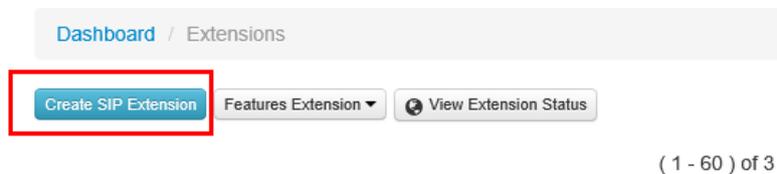
Here you can set the extensions and line provider.



Extensions

SIP (Session Initiation Protocol) [RFC 3261, 3262, 3263, 3264, and 3265]

Extensions



Create the SIP extensions: Set the number (not less than 3 digits), password (not less than 8 digits) and caller name if need.

No-Answer Option: When nobody answers after ringing. It will auto hang up, go to Voice Mail or Forward.(use Voice Mail need to insert the USB disk)

Extensions Edit SIP Extension

Dashboard / Extensions / Edit SIP Extension

Basic

Expert

Number

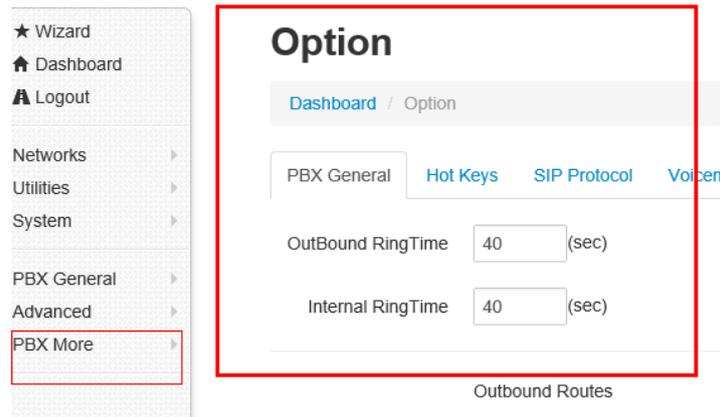
Password

CallerID Name

No-Answer Option Hangup Voicemail Forward

Forward Number

You can also set the ring time as below, after the ring time, it will execute the option.



Extension Expert part:

Basic

Expert

Video Support Yes No

IP Address Dynamic IP Static IP

CallerID

Directmedia Yes No

NAT Yes No

Keep Alive (ms)

DTMF Mode

Codec Priority

| | |
|----|-----------------------------------|
| 1. | <input type="text" value="GSM"/> |
| 2. | <input type="text" value="ALAW"/> |
| 3. | <input type="text" value="ULAW"/> |
| 4. | <input type="text" value="G729"/> |
| 5. | <input type="text" value="H264"/> |
| 6. | <input type="text" value="G722"/> |

For the Expert use, you can set the Video Support, IP address and so on

Video Support: Choose Yes or No.

IP address: Dynamic IP or Static IP. If you choose the Static IP, just fill your IP address and make sure it is the same with this device.

Call ID: You can use the extension number (Default) or use this number you set yourself.

CallerID
number: name:

Direct Media: The default is No.

If you choose yes, it will not transfer the voice data.

Nat: Please see the glossary

Keep Alive: Keep in touch with the contacted device.

DTMF Mode refers to the types of tones a phone can send and receive. Refer to your phone's user manual to find the type of DTMF tones used by your particular phone. Unless you are certain this setting needs to be changed, leave it at the default value, rfc2833.

DTMF Mode

Codec Priority: You can choose the codec priority here.

DTMF Mode

Codec Priority

| | | |
|----|-----------------------------------|----------------------------------|
| 1. | <input type="text" value="GSM"/> | <input type="button" value="v"/> |
| 2. | <input type="text" value="ALAW"/> | <input type="button" value="v"/> |
| 3. | <input type="text" value="ULAW"/> | <input type="button" value="v"/> |
| 4. | <input type="text" value="G729"/> | <input type="button" value="v"/> |
| 5. | <input type="text" value="H264"/> | <input type="button" value="v"/> |
| 6. | <input type="text" value="G722"/> | <input type="button" value="v"/> |



Important: Configuration changed, please reload device to activate. [Click here to Reload](#)

Extensions

Dashboard / Extensions

Create SIP extension Features Extension ▾

display (1 - 60) of 32

Press Create and Click the Reload in the Red Bar. One extension is well done. It will need the SIP account and password when you set SIP phone. So please remember the SIP account and password here.

If need to modify the setting or have a view, just click the edit button. To remove an extension from your system permanently, click the Delete button.

Click the Record, you can go to check the call record directly.

Extensions

Dashboard / Extensions

Create SIP Extension Features Extension ▾ View Extension Status

(1 - 60) of 3



| Number | Password | CallerID Name | Protocol | |
|--------|----------|---------------|----------|---|
| 803 | 803 | 803 | sip | records edit delete |
| 802 | 802 | 802 | sip | records edit delete |
| 801 | 801 | 801 | sip | records edit delete |

Follow Me

One of the feature extensions is Follow Me.

Extensions

Dashboard / Extensions

Create SIP extension Features Extension

Follow Me

Ring Group

If you have more than one number when a call comes in when the system will be one by one to try ring your numbers until you answer.

| Number | Password | E-Mail | Protocol | |
|--------|----------|--------|----------|---------------------|
| 831 | 10622155 | | sip | records edit delete |
| 830 | 52715081 | | sip | records edit delete |

If you have more than one number, when a call comes in, the system will ring your numbers one by one until someone answered.

You can set the detail in basic and expert. The expert is not available now.

You can set the order of the extensions number to ring.

Extensions Create FollowMe Extension

Dashboard / Extensions / Create FollowMe Extension

Basic

Expert

Number 8001

Password mypassword8001

Info_email yourname@yourdomain.com

Numbers

Up Down Remove

801
803
806
804

Select

800
802
805
807
808
809
810
811
812
813

Available numbers window.

Create

You can also set to forward to the external number as below.

Extensions Create FollowMe Extension

Dashboard / Extensions / Create FollowMe Extension

Basic

Number

Password

Internal Extensions

800
801
802
803
804

External Number

Ring Group

When the call comes in, all the extensions will ring simultaneously until someone to pick it up.

The setting is similar to Follow me.

Extensions Create RingGroup Extension

Dashboard / Extensions / Create RingGroup Extension

Basic

Number

Password

Numbers

801
802
803

And here you can also view the extension status.

Extensions

Dashboard / Extensions

(1 - 60) of 3

Click it and the page will turn to the real status page.

Real Status

Dashboard / Real Status

[Back to previous page](#)

PBX Local Number

Extensions

| | Number | Type | Address |
|-------------|--------|------|---------|
| UNKNOWN | 801 | sip | |
| UNKNOWN | 802 | sip | |
| Unmonitored | 803 | sip | |

Other Resource

| Number | Section | Type |
|--------|---------|------|
|--------|---------|------|

PBX Line Provider

Trunk Register

| Refresh | Type | Host | Account |
|---------|------|------|---------|
|---------|------|------|---------|

Trunk Connect

| Trunk | Type | Host | Account |
|-------|------|------|---------|
|-------|------|------|---------|

And click the back to previous page.

Real Status

Dashboard / Real Status

[Back to previous page](#)

PBX Local Number

Extensions

| | Number | Type | Address |
|-------------|--------|------|---------|
| UNKNOWN | 801 | sip | |
| UNKNOWN | 802 | sip | |
| Unmonitored | 803 | sip | |

Other Resource

| Number | Section | Type |
|--------|---------|------|
|--------|---------|------|

PBX Line Provider

Trunk Register

| Refresh | Type | Host | Account |
|---------|------|------|---------|
|---------|------|------|---------|

Trunk Connect

| Trunk | Type | Host | Account |
|-------|------|------|---------|
|-------|------|------|---------|

Line Provider

You can set the line provider here. Sip Register and Sip Direct.

Line Provider

Dashboard / Line Provider

[Add New Provider](#) [View Trunk Status](#)

Use SIP account and password to register to ITSP(Internet Telephony Services Provider).

| Name | Protocol | A |
|--------------|----------|---|
| SIP Register | | |
| SIP Direct | | |

After you finished the register information, remember to tick the Outbound Calls-Default outbound calls

SIP Register

SIP Register: Use a SIP account and password and Provider IP address to register to the ITSP.

You can also choose the system process rule when there is incoming call.

Default to answer: You can set the default configuration in the PBX More Option.

The screenshot displays the 'Option' configuration page in the iCallDroid web interface. The left sidebar shows a navigation menu with 'PBX More' highlighted. The main content area is titled 'Option' and contains several configuration sections:

- Outbound Routes:**
 - OutBound RingTime: 40 (sec)
 - Internal RingTime: 40 (sec)
 - Default Mode: [smart] Automatic try localnumb (dropdown)
 - Default Outbound: No Select (dropdown)
- Trunk Inbound Routes:**
 - Auto Find Extension: [disable] Do nothing. (dropdown)

Set DID number: you can set the DID number here. When you create the Queue or Conference and you can set the number here for the caller to go into.

Specific who to answer: Specific the certain extension to answer.

Finally you can tick the default outbound line or the outbound routes.

Call with prefix number: Set this number when you make a call with this prefix outbound,

It will auto go via this line. In a word, it is convenient for you to change the line when make calls between default line and another outbound line.

You can also view this setting in the Wizard.

Line Provider Add New SIP Register

[Dashboard](#) / [Line Provider](#) / Add New SIP Register

Basic

Expert

Name

Provider Host

Provider Port

Account

Password

Incoming Calls Default To Answer
 Set DID Number
 Specify who answer

Outbound Calls Default Outbound Line
 Outbound Routes
 Call with prefix number

[Add New](#)

Basic

Expert

Outbound force callerid

Default reg expiry (sec)

Allow callin Yes No

Progress Inband Outband

Keep alive (ms)

NAT Yes No

Video support Yes No

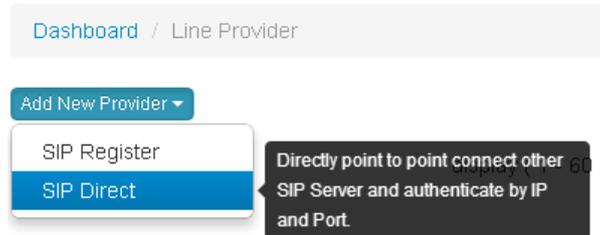
DTMF mode

Codec priority

| | |
|----|-----------------------------------|
| 1. | <input type="text" value="GSM"/> |
| 2. | <input type="text" value="ALAW"/> |
| 3. | <input type="text" value="ULAW"/> |
| 4. | <input type="text" value="G729"/> |
| 5. | <input type="text" value="H264"/> |
| 6. | <input type="text" value="G722"/> |

SIP Direct

Line Provider



You can create the SIP direct.

Here we use the 2 iCallDroid spot as example:

First check the SIP port : it is 6620.

Line Provider Add New SIP Direct

Dashboard / Line Provider / Add New SIP Direct

Basic Name

Expert Provider IP Address

Provider IP Port

Outbound Calls Default Outbound Line
 Outbound Routes
 Call with prefix number

[Add New](#)

Welcome To The Unified Communications System

WAN

- MAC 00:86:30:4A:02:3B
- Protocol static
- IP Addr 172.16.0.166
- Netmask 255.255.0.0
- Gateway 172.16.0.254
- DNS 8.8.8.8 172.16.0.254

SYS

- Firmware 5.5.3546.0
- Model 0101
- Serial ID 00010418000020
- Memory 58% used
- Timezone Asia/Shanghai

PBX

- SIP Port (TCP/UDP) 0.0.0.0:6620
- Concurrent 8 calls
- Settings 32 sip extensions
8 feature extensions
8 trunks
16 call routes
4 conferences
10 queues

Then input another IP address to create SIP direct. Make sure the port is the same and there is no two extensions with the same number. OK, done.

★ Wizard
 🏠 Dashboard
 👤 Logout

Networks
 Utilities
 System

PBX General
 Advanced
 PBX More

Extensions
 Line Provider

Line Provider Add New SIP Direct

Dashboard / Line Provider / Add New SIP Direct

Basic
 Expert

Name: PBX220 SIP Directly

Provider IP Address: 172.16.0.188

Provider IP Port: 5060

Outbound Calls:
 Default Outbound Line
 Outbound Routes
 Call with prefix number

Add New

Advanced

Outbound Routes

★ Wizard
 🏠 Dashboard
 👤 Logout

Networks
 Utilities
 System

PBX General
 Advanced
 PBX More

Outbound Routes
 Inbound Routes
 Conference
 Queues
 Simple IVR
 Time Frames

Outbound Routes

Dashboard / Outbound Routes

Create New Rule Default Rule: smart Default Outbound: Save Rules New order

Rule Note
 Outbound Routes for extensions dialing local / outside number.
 When dialing, the top-down one by one checking rules, if matched will be execute and end route.
 When dialing, not matched or no rule will try to execute 'Default Rule' settings.
 the order of rules by dragging and click 'Save Rules New order' to confirm.

Rule Name

Outbound Routes for extensions dialing local / outside number.

When dialing, the top-down one by one checking rules, if matched will be execute and end route.

When dialing, not matched or no rule will try to execute 'Default Rule' settings.

You can choose the default rule. In most cases, the default settings can be used for the rest of the configuration.

Default Rules:

[Smart] Automatically try local number, if not matched then try to select [efault Outbound] and call out.

[Local] Automatically try local number

[Disable] End the call

Outbound Routes

The screenshot shows the 'Outbound Routes' configuration page. At the top, there is a breadcrumb 'Dashboard / Outbound Routes'. Below it, there are three buttons: 'Create New Rule', 'Default Rule: disable', and 'Default Outbound:'. The 'Default Rule: disable' button is highlighted with a red box. A dropdown menu is open below it, showing three options: '[smart] Automically try local number, if not matched try to select 'Default outbound' and call out.', '[local] Automically try local number.', and '✓ [disable] End the call.'. Below the dropdown, there is a 'Save Rules New order' button. The page also contains a 'Rule Note' section with text explaining the routing logic and a note about modifying the order of rules.

Default Outbound: You can choose the default outbound line here.

The screenshot shows the 'Outbound Routes' configuration page. At the top, there is a breadcrumb 'Dashboard / Outbound Routes'. Below it, there are three buttons: 'Create New Rule', 'Default Rule: disable', and 'Default Outbound:'. The 'Default Outbound:' button is highlighted with a red box. Below the dropdown, there is a 'Save Rules New order' button. The page also contains a 'Rule Note' section with text explaining the routing logic and a note about modifying the order of rules.

Create New Rule first:

Now you can set the outbound routes.

You can refer the tips. Set the rules to match caller ID or match the called party ID (Callee), then set the relevant executive rules: we can format caller Id/ callee ID .If don't need, leave NULL.

Rule match Sets.

Match Caller Caller ID prefix is , and/or length is digits.

Match Called Party Callee ID prefix is , and/or length is digits.

If matched current rule, we can format callerid/callee within followed sets, if don't need, leave NULL.

Format Caller Trim digits from Caller ID, and/or add number in prefix, and/or append number with end.

Format Callee Trim digits from Callee ID, and/or add number in prefix, and/or append number with end.

Rule Name

Handling

[Create New](#)

Rules are different as many as you think.

Examples are in the routes manual.

After you create them, you can modify the order of rules by dragging and click 'Save Rules New order' to confirm. And you can also edit and delete the rules.

Outbound Routes

Dashboard / Outbound Routes

[Create New Rule](#)
Default Rule: disable ▾
Default Outbound: ▾
Save Rules New order

Rule Note
 Outbound Routes for extensions dialing local / outside number.
 When dialing, the top-down one by one checking rules, if matched will be execute and end route.
 When dialing, not matched or no rule will try to execute 'Default Rule' settings.
 You can modify the order of rules by dragging and click 'Save Rules New order' to confirm.

| Priority | Rule Name | Handling | |
|----------|-----------|-------------------|---|
| 1 | 89798 | Call Denied | edit delete |
| 2 | 5465 | Call Denied | edit delete |
| 3 | sdfasf | Call Local Number | edit delete |

Inbound Routes

Inbound Routes for Line provider dialing local number / transfer to other line provider.

Inbound Routes

Dashboard / Inbound Routes

[Create New Rule](#)
Auto Find Extension: autoivr ▾
Default Rule: disable ▾
Save Rules New order

Rule Note
 Inbound Routes for Line provider dialing local number / transfer to other line provider.
 When a call comes in, the system will prior process 'Auto Find Extension', next check the rules one by one from top to down, finally perform 'Default Rule'.
 You can modify the order of rules by dragging and click 'Save Rules New order' to confirm.

| Priority | Rule Name | From Line | Handling |
|----------|-----------|-----------|----------|
|----------|-----------|-----------|----------|

Create a new rule as below, you can create the new rules for the incoming calls.

Inbound Routes Create New Rule

Dashboard / Inbound Routes / Create New Rule

Rule match Sets.

Match From Line

Match Caller Caller ID prefix is and/or length is digits.

Match Called Party Callee ID prefix is and/or length is digits.

If matched current rule, we can format callerid/callee within followed sets, if don't need, leave NULL.

Format Caller Trim digits from Caller ID, and/or add number in prefix, and/or append number with end.

Format Callee Trim digits from Callee ID, and/or add number in prefix, and/or append number with end.

Rule Name

Handling

The rule setting is similar to outbound routes. You can refer the tips. Below is the diagram.

After you set several rules, you can edit or delete them.

You can also modify the order of rules by dragging and click 'Save Rules New order' to confirm.

Inbound Routes

Dashboard / Inbound Routes

Create New Rule Auto Find Extension: autoivr Default Rule: disable **Save Rules New order**

Rule Note
 Inbound Routes for Line provider dialing local number / transfer to other line provider.
 When a call comes in, the system will prior process 'Auto Find Extension', next check the rules one by one from top to down, finally perform 'Default Rule'.
 You can modify the order of rules by dragging and click 'Save Rules New order' to confirm.

| Priority | Rule Name | From Line | Handling | |
|----------|-----------|-------------------|----------------------------|-------------|
| 1 | feng | From: feng4 | Using Outbound Line: feng | edit delete |
| 2 | feng4 | Call Local Number | Using Outbound Line: feng4 | edit delete |

You can modify the order of rules by dragging and click 'Save Rules New order' to confirm.

When a call comes in, the system will prior process 'Auto Find Extension', next check the rules one by one from top to down, finally perform 'Default Rule'.

Inbound Routes

Dashboard / Inbound Routes

Create New Rule Auto Find Extension: autoivr Default Rule: disable Save Rules New order

Rule Note
 Inbound Routes for
 When a call comes
 You can modify the

- [disable] Ignore.
- [auto] Automatically find whois last answered and immediately transfer the call to that extension.
- ✓ [autoivr] Automatically find whois last answered and prompt IVR for the caller to choose to transfer the call to that extension or not.

| Priority | Rule Name | From Line | Handling |
|----------|-----------|-----------|----------|
|----------|-----------|-----------|----------|

Inbound Routes

Dashboard / Inbound Routes

Create New Rule Auto Find Extension: autoivr Default Rule: disable Save Rules New order

Rule Note
 Inbound Routes for Line provider dialing local number
 When a call comes in, the system will prior process
 You can modify the order of rules by dragging and clic

- [smart] Automic try localnumber, if no match try to call 310 as extension.
- [local] Automic try localnumber.
- ✓ [disable] End the call.

| Priority | Rule Name | From Line | Handling |
|----------|-----------|-----------|----------|
|----------|-----------|-----------|----------|

Conference

Set the conference number as you like and choose if you want to enable to announce the Join/leave by system or enable to playback the music when only one person in it.

Conference

Dashboard / Conference

Create Conference Room

display (1 - 60)

| Room Number | Announce Join/Leave | Music When Only Person |
|-------------|---------------------|------------------------|
| 300 | Enabled | Enabled |

edit delete

display (1 - 60)

Conference Create Conference room

[Dashboard](#) / [Conference](#) / Create Conference room

Room Number

Announce Join/Leave Disable
 Enable

One Person Playback Disable
 Music Enable

Create Conference room

After you set, you can make a call to the conference number via the extension.

Queues

Select this type if the extension will queue callers to speak with a representative. Call queues are often used to dial into a particular department or group; for example, the extension for the accounting department might be a call queue. You can customize your call queues in the extension setup process.

Queues

[Dashboard](#) / Queues

Create New Queue

[How does this work?](#)

display (1 - 60)



Queue Number

Remark

Timeout

display (1 - 60)



Now for example, we create a Queue, we add 802 and 800 into our queue.

Queues Create New Queue

Dashboard / Queues / Create New Queue

Basic Queue Number Exp: 302

Expert Remark

Numbers Up Down Remove

Select 801
802
803

Create New Queue

Queues Create New Queue

Dashboard / Queues / Create New Queue

Basic

Expert

Warning! Expert settings is only for professionals, if you do not know the meaning of the parameters do not modify.

Background Music Playback music to the caller
 Playback ring to the caller

Call Progress Every 20 (sec) announce busy voice, and if the caller waits for more than 180 (sec), it will transfer the call to the number exp: 8001

Service Strategy [ringall]: All extensions are rir

Member rings time 16 (sec)

When Pickup Direct Answer
 Announce Member's Number to the Caller

Create New Queue

You can also set the service strategy: Ring all, Radom, Memory.

You can also set the member rings time and when pick up, how you want to deal with.

Simple IVR

Simple IVR

Dashboard / Simple IVR

Create New IVR

How does this work?

| IVR Number | Remark | Playback |
|--------------------|--------|----------|
| display (1 - 60) | | ← ↑ → |
| display (1 - 60) | | ← ↑ → |

Simple IVR Create New IVR

[Dashboard](#) / [Simple IVR](#) / Create New IVR

Basic

Time Frames

Expert

IVR Number

Remark

Playback File

| | | | | |
|----------------|-------|-------------------------------------|-------------|---------------------------------------|
| Input Detected | Press | <input type="text" value="Exp: 0"/> | Transfer to | <input type="text" value="Exp: 302"/> |
| | Press | <input type="text" value="Exp: 1"/> | Transfer to | <input type="text" value="Exp: 302"/> |
| | Press | <input type="text" value="Exp: 2"/> | Transfer to | <input type="text" value="Exp: 302"/> |
| | Press | <input type="text" value="Exp: 3"/> | Transfer to | <input type="text" value="Exp: 302"/> |
| | Press | <input type="text" value="Exp: 4"/> | Transfer to | <input type="text" value="Exp: 302"/> |
| | Press | <input type="text" value="Exp: 5"/> | Transfer to | <input type="text" value="Exp: 302"/> |
| | Press | <input type="text" value="Exp: 6"/> | Transfer to | <input type="text" value="Exp: 302"/> |

Here we can set the IVR (Voice menu)

For example, we set one IVR number 333, choose the playback file-Welcome, and choose that press 0 to transfer to 800 and press 1 to transfer to 801. Create the new IVR.

Time Frame: You need to set **Time Frame** area then come back to choose.

| Level | Check Frame | Transfer to |
|-------|-------------------------------|---------------------------------------|
| 1 | <input type="text" value=""/> | <input type="text" value="exp: 801"/> |
| 2 | <input type="text" value=""/> | <input type="text" value="exp: 801"/> |
| 3 | <input type="text" value=""/> | <input type="text" value="exp: 801"/> |
| 4 | <input type="text" value=""/> | <input type="text" value="exp: 801"/> |
| 5 | <input type="text" value=""/> | <input type="text" value="exp: 801"/> |
| 6 | <input type="text" value=""/> | <input type="text" value="exp: 801"/> |
| 7 | <input type="text" value=""/> | <input type="text" value="exp: 801"/> |
| 8 | <input type="text" value=""/> | <input type="text" value="exp: 801"/> |

[Create New IVR](#)

Expert: You can set the mode when user input the invalid number.

Input Max Digit Len: For example, if you input 3 means if you press only one number when you access to the IVR, you need to press that number and # to call directly. Otherwise need to wait the Max Time.

Max Time: the waiting time to connect someone.

Input Retry: For example, if you input continuous two time wrong number, the system will automatically transfer to another number you input.

Simple IVR Create New IVR

Dashboard / Simple IVR / Create New IVR

Basic

Time Frames

Expert

Warning! Expert settings is only for professionals, if you do not know the meaning of the parameters do not modify.

Input Invalid Mode Invalid Playback
 Try Localnumber and Invalid Playback

Input Max Digit Len

Input MaxTime (sec)

Input Retry Max , Outride to transfer to

Create New IVR

After fill out the information you need, just to click create new IVR and reboot.

Time Frames

Time Frames

Dashboard / Time Frames

How does this work?

| Frame Name | Start Date | End Date | Start Day | End Day | Start Time | End Time | |
|---------------------------------------|----------------------|----------------------|----------------------|----------------------|----------------------|----------------------|------------------------------------|
| <input type="text" value="newframe"/> | <input type="text"/> | <input type="button" value="Add"/> |

In most cases, it is useful to create an IVR to run in the time of working days, holidays, nights and so on for the companies to handle with the business calls. This we called Time Frames.

As the above, we create one set working time and click “add”. Now we can come back to the simple IVR- time frame – to set it.

The IVR format is GSM file. You can upload the sounds file in **Sounds File**.

If all the parameters are zero, it means to ignore this time frame.

(Advanced-Simple IVR)

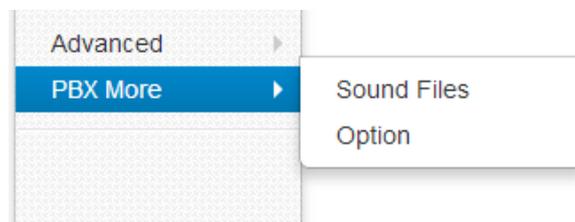
OK. When during the working time you set, all the incoming calls will transfer to 801.

Simple IVR Edit

[Dashboard](#) / [Simple IVR](#) / [Edit](#)

| | Level | Check Frame | Transfer to |
|-------------|-------|-------------------------------|-------------------------------|
| Basic | 1 | newframe <input type="text"/> | exp: 801 <input type="text"/> |
| Time Frames | 2 | <input type="text"/> | exp: 801 <input type="text"/> |
| Expert | 3 | <input type="text"/> | exp: 801 <input type="text"/> |
| | 4 | <input type="text"/> | exp: 801 <input type="text"/> |
| | 5 | <input type="text"/> | exp: 801 <input type="text"/> |
| | 6 | <input type="text"/> | exp: 801 <input type="text"/> |
| | 7 | <input type="text"/> | exp: 801 <input type="text"/> |
| | 8 | <input type="text"/> | exp: 801 <input type="text"/> |

PBX More



Sound Files

Sound Files Add New

Dashboard / Sound Files / Add New

File Name

File Extname

File Size

Upload File Not upload
 Web upload
 Recording through extension

By default, the file name is unmodified when you add a new sound.

It can be used as the Simple IVR's file.

File Ext name/ file size: Unmodified.

Upload file: you can choose the way to upload the file.

Web upload: Only support GSM file.

Upload File Not upload
 Web upload
 Recording through extension

Web load file **Only support GSM format files.**

When you input one extension number, it will record automatically.

Upload File Not upload
 Web upload
 Recording through extension

Recording extension

When you upload a sound file, you can listen, edit and delete.

Sound Files

[Dashboard](#) / [Sound Files](#)

[New File](#)

| Filename | Format | Size | |
|----------|--------|---------|--|
| welcome | gsm | 14.28KB | listen edit delete |

Option

PBX General:

Outbound and internal ring time: you can input the time of ring you like.

Extension dialing route: you can set the dialing route here as well as the Extension-Dialing

Route

Outbound Routes

Default Mode

Default Outbound

Trunk Dialing route: you can also find the setting in Line provider-Dial-in Route

Trunk Inbound Routes

Auto Find Extension ▼

Default Mode ▼

Default Extension

IVR Max Retry

IVR MAX TRY: For example, if we input 2 means the system will automatically quit when the IVR playback for twice.

IVR MAX RETRY Max Retry for all IVR menus.

Call Notification: When Make call to Extensions (SIP Ring Group Follow Me) or to Queues (Extensions Answered) or to IVR(Extensions Answered), PBX will send notification to url.

IVR Max Retry

Http Uri

Hot Keys

Option

Dashboard / Option

PBX General **Hot Keys** SIP Protocol Voicemail / FTP

Call Pickup + Callee extension number

Voicemail Playback

You can set the call pick up prefix number to pick the extension.

You can't set “#” as the prefix as it is used by the system.

Set the voice mail playback number to get your voice mail and listen.

Dial the number directly on your phone to listen.

SIP Protocol

PBX General Hot Keys **SIP Protocol** Voicemail / FTP

Anonymous Call In

TCP/UDP Bind Port

Max Register Expiry (sec)

Min Register Expiry (sec)

Default Register Expiry (sec)

Progress Mode

T.38 UDPTL

Enable Yes No

Anonymous Call In

UDP Bind Port: by default is 6020

Please note you should modify this port when you set the extension port is 5060.

Max Register Expiry

Min Register Expiry

Default Register Expiry

Progress Mode

T.38 UDPTL

You can also set the Jitter Buffer option. Default is enabling Jitter Buffer.

It will help to protect and keep the voice with good quality.

Jitter Buffer to enable or disable jitter buffer in sip protocol.

Enable Yes No

Force Receive Yes No

Max Length (ms)

Resync Threshold

Implementation Fixed Adaptive

Target Extra (ms)

SIP Behind NAT

This experimental feature is that when a PBX as a SIP server behind Nat, SIP Nats allows your SIP phone and IADs to register into PBX. And it needs the firewall and router's support.

PROXY IP:

SIP NAT in Experimental?

SIP Behind NAT Disabled
 External IP
 External DOMAIN(Dynamic Dns)

External Domain Refresh (sec)

Local Net Area

[Save](#)

This feature is experimental: When PBX as a SIP Server behind nat, SIP NAT allows remote sip phone or iad to register into PBX, it needs the firewall or router support.

Voicemail

Say Date time if choose yes, it will read the call record date.

Say caller ID: if choose yes, it will read the caller ID.

Option

[Dashboard](#) / [Option](#)

[PBX General](#) [Hot Keys](#) [SIP Protocol](#) [Voicemail](#)

Voicemail

Say Datetime Yes No

Say Callerid Yes No

[Save](#)

Glossary

ATA (Analog Telephony Adapter)

A device used to connect one or more standard analog telephones to a digital and/or non-standard telephone system such as a Voice Over IP based network.

DID (Direct Inward Dial)

A feature offered by telephone companies for use with their customers' private branch exchange (PBX) systems. In DID service, the telephone company provides one or more trunk lines to the customer for connection to the customer's PBX and allocates a range of telephone numbers to this line (or group of lines) and forwards all calls to such numbers via the trunk. As calls are presented to the PBX, the dialed destination number (DNIS) is transmitted, usually partially (e.g., last four digits), so that the PBX can route the call directly to the desired telephone extension within the organization without the need for an operator or attendant.

DID numbers are assigned to a communications gateway connected by a trunk to the public switched telephone network (PSTN) and the VoIP network. The gateway routes and translates calls between the two networks for the VoIP user. Calls originating in the VoIP network will appear to users on the PSTN as originating from one of the assigned DID numbers.

DNS (domain name system)

The Internet's name/address resolution service that translates alphabetic domain names into numeric IP addresses. For example, the domain name www.pbx.com might translate to 198.105.232.4. If a computer cannot access DNS, the user's web browser will not be able to find web sites and the user will not be able to receive or send email. The DNS system consists of three components: DNS data, name servers, and Internet protocols for getting the

data from the servers.

Domain name server

A computer that runs a program that converts a fully qualified domain name (FQDN) into its numeric

IP address and vice versa.

DTMF (Dual-Tone Multi-Frequency)

The signal that is generated when a user presses the touch keys of an ordinary telephone.

Also known as “Touchtone,” DTMF has essentially replaced pulse dialling. When a user presses touch keys, two tones of specific frequencies are generated (one from a high-frequency group and the other from a lowfrequency group), so it’s impossible for the voice to imitate the tones.

FTP (File Transfer Protocol)

A standard Internet protocol used to upload and download files between computers that are connected to the Internet. FTP uses the Internet’s TCP/IP protocols as does HTTP, which transfers displayable Web pages and related files, and SMTP, which transfers e-mail.

GSM (Global System for Mobile communication)

A wireless telephone standard in Europe and other parts of the world. GSM uses a variation of time division multiple access (TDMA), which is the most widely used of the three digital

wireless telephony technologies (TDMA, GSM, and CDMA). GSM digitizes and compresses data, then sends it down a channel with two other streams of user data, each in its own time slot. It operates at either the 900 MHz or 1800 MHz frequency band.

IP-PBX (Internet Protocol Private Branch Exchange)

A telephone switch (see “PBX”) located on a customer’s premises that utilize VoIP to manage and deliver calls.

ITSP (Internet Telephone Service Provider)

A company that offers an Internet data service for making telephone calls using VoIP. Most ITSPs use SIP, H.323, or IAX for transmitting telephone calls as IP data packets. Customers may use VoIP phones or traditional telephones with an analog telephony adapter (ATA).

ITU (International Telecommunication Union)

A telecommunications standards body that is guided by the United Nations. It was founded as the International Telegraph Union in Paris on May 17, 1865. The ITU acts as the global focal point for governments and the private sector in developing networks and services and is comprised of more than 185 countries and produces over 200 standards recommendations annually in the areas of information technology, consumer electronics, broadcasting, and multimedia communications.

IVR (Interactive Voice Response)

A telephone technology that allows a caller to respond to configured voice menus through voice and touch tone. The IVR system responds with pre-recorded audio to further direct callers on how to proceed.

LAN (Local Area Network)

A computer network covering a small physical area, like a home, office, or small group of buildings, such as a school, or an office park. LANs are connected primarily through Ethernet and can be connected to other LANs over any distance via telephone lines and radio waves. LANs have a high data transfer rate and are not very expensive to set up. See also "WAN."

MAC (Media Access Control) address

A hardware address that uniquely identifies most network adapters or network interface cards (NICs) by the manufacturer for identification. The manufacturer's registered identification number is usually part of the MAC address if it was assigned by the manufacturer. The MAC address is used by the Media Access Control protocol sub-layer of the Data-Link Layer (DLC) of telecommunication protocols.

MIPS (million instructions per second)

An old method for measuring a computer's speed and power and, by implication, for determining the amount of work a computer can do. It measures the approximate number of

machine instructions the computer can execute in 1 second (i.e., it measures CPU speed). Because there are so many variables with computer performance (e.g., varying amounts of time for different instructions, importance of I/O speed, etc.), MIPS ratings are not used that often anymore. However, a MIPS rating can give you a general idea of a computer's speed.

NAT (Network Address Translation or Network Address Translator)

The method for translating an IP address used within one network to a different IP address known within another network (one network is designated the *inside* network and the other is the *outside* network). NAT allows as a router, for example, to act as an agent between the public network (e.g., the Internet) and a private network (i.e., a local network), which means that a single, unique IP address can represent an entire group of computers.

PBX (Private Branch exchange)

A telephone exchange that serves a particular business or office, as opposed to one that is owned by a common carrier or telephone company and is used by many businesses or the general public. Users of the PBX share a certain number of outside lines for making telephone calls external to the PBX. PBXs have evolved over time, beginning as a manual switchboard or attendant console that was operated by a telephone operator (circuit switching) to the modern IP PBX. See also "IP PBX."

PSTN (Public Switched Telephone Network)

The network of the world's public circuit-switched telephone networks. Originally a network

of fixed line analog telephone systems, the PSTN is now almost entirely digital in its core and includes mobile as well as fixed (plain old telephone service, POTS) telephones. The PSTN is largely governed by technical standards created by the ITU-T, and uses E.163/E.164 telephone numbers for addressing.

Proxy Server

A server (a computer system or an application program) that acts as an intermediary for requests from clients seeking resources from other servers. The VoIP proxy server is used in a DMZ of a company's secure internal communication network and receives VoIP control messages and VoIP media streams.

Using the MAC address and source IP address contained in the control message, the proxy server pushes a policy change to the internal network's external firewall to open call control protocol ports and Real Time Protocol (RTP) ports only for packets from the source IP address. The VoIP proxy server hides the company's internal network address and directs incoming VoIP packets to an IP-PBX connected to the company's internal network.

RAM (Random Access Memory)

A form of computer data storage that allows stored data to be accessed in any order (i.e., "random access").

RAM is used by a computer's operating system, application programs, and currently used data, so that they can quickly be reached by the computer's processor. RAM is quickly readable and writeable compared to other kinds of computer storage (e.g., the hard disk,

floppy disk, and CD-ROM); However, data in RAM remains only as long as the computer is running. Once the computer has been turned off, RAM loses its data. When the computer is turned on again, the operating system and other files are once again loaded into RAM.

Router

A device for connecting one or more computers to other computers, networked devices, or to other networks. Compared to hubs and switches (which are also connecting types of devices), a router is the smartest and most complicated of the three. Routers can be programmed to understand and route the data its being asked to handle. Configuration is done through a user interface. Larger routers are capable of being programmed to communicate with other routers to determine the best method of getting network traffic from point A to point B. Hubs work at the data link and network layers (layers 2 and 3) of the OSI model.

SIP (Session Initiation Protocol) [RFC 3261, 3262, 3263, 3264, and 3265]

A signalling protocol for initiating and terminating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality (it is used mainly for voice and video calls over the Internet or data networks).

SIP Trunk

A service offered by an ITSP that allows businesses that have a PBX for their internal calls to use VoIP to go outside the enterprise network by using the same connection as the

Internet connection. Before SIP trunks can be deployed, a business must have a PBX with a SIP-enabled trunk side, an enterprise edge device that understands SIP, and an ITSP. See “ITSP.”

Soft-switch (software switch)

A term used to describe the software that is used to bridge a public switched telephone network (PSTN) and VoIP. This is done by separating the call control functions of a phone call from the media gateway (transport layer). The soft-switch is typically used to control connections at the junction point between circuit and packet networks.

UDP (User Datagram Protocol) [RFC 768]

A communications protocol that offers a limited amount of service when messages are exchanged between computers in a network that is using the Internet Protocol (IP). UDP merely performs IP traffic demultiplexing based on UDP port numbers, after which it provides a checksum that can be used by end systems to determine whether the datagrams received were corrupted by the network.

WAN (Wide Area Network)

A computer network that covers a broad area (e.g., any network that links across metropolitan, regional, or national boundaries). WANs are similar to the Internet in that they are not owned by a single organization. They exist under collective or distributed ownership and management. For WAN connectivity over the longer distances, ATM, frame relay, and

X.25 are used. Computers connected to

a WAN can be connected via the telephone system, leased lines, or satellites. WANs have a

lower data transfer rate when compared to LANs. See also "LAN."