



3300IP VoIP Phone | Users Manual



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Function

- Provide a Backup SIP Server
- Support NAT (Network Address Translation), Firewall
- Support DHCP for phone to accept IP via WAN port or assign IP address via LAN port
- Support PPPoE (used while connecting ADSL, cable modem)
- Update the firmware and/or configuration file program through HTTP,FTP and TFTP
- Check the dynamic voice; Soft the noise; Buffer technique of voice
- Hold Function
- Hotline Function (dial a specific number as soon as going off-hook, or set a delay before dialing)
- Speed-dial
- Call-forward, Three-way conference call
- Caller ID display
- DND (Do Not Disturb), Black List, Limit List
- Auto-answer.
- Set through standard Web Browser
- Remote Management Function
- Classification management for common user's password and superuser's password.

Standard and Protocols

- IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- PPPoE
- DHCP Client and Server
- Support G.711a/u,G729, G7231 5.3/6.3 audio Codec
- SIP RFC3261, RFC 2543
- Support IAX2
- TCP/IP: Internet transfer and control protocol (Transport Control Protocol/Internet Protocol)
- RTP: Real-time Transport Protocol
- RTCP: Real-time Control Protocol
- VAD/CNG save bandwidth (Voice Activity Detection/
- Telnet: Internet's remote login protocol
- DNS: Domain Name Server
- TFTP: Trivial File Transfer Protocol
- Introduction
- This is the user manual for the 3300IP. Some configuration is necessary before using the 3300IP phone. This manual will illustrate how to set up the phone through keyboard and web User Interface (UI).

1.1 Overview of Hardware

1.1.1

The two RJ-45 network interface support the 10/100M Ethernet. The default WAN interface is a DHCP Client server. User connect the WAN interface to ADSL or switch, Lan is web-bridge mode, and bridged the LAN and WAN into the same network. You can use the administrator's user name "admin" and password "admin" to login and set.

1.1.2

Only the WAN interface supports the POE.

1.2 Overview of Software

Network Protocol		
<ul style="list-style-type: none"> SIP v1(RFC2543) V2(RFC3261) IP/TCP/UDP/RTP/RTCP IP/ICMP/ARP/RARP/SNTP 	<ul style="list-style-type: none"> TFTP Client/DHCP Client/PPPOE Client Telnet/HTTP Server DNS Clients 	
Codec		
<ul style="list-style-type: none"> G.711: 64K bit/s(PCM) G.723.1: 63k/5.3k bit/s G.726: 16k/24k/32k/40k bit/s(ADPCM) 	<ul style="list-style-type: none"> G.729A: 8k bit/s(CS-ACELP) G.729B: adds VAD & CNG to G.729 	
Voice Quality		
<ul style="list-style-type: none"> VAD: Voice activity detection CNG: Comfortable noise generator LEC: Line echo canceller 	<ul style="list-style-type: none"> Packet Loss Compensation Adaptive Jitter Buffer 	
Call Function		
<ul style="list-style-type: none"> Call Hold Call Waiting Call Forward 3-way conference 		
Tone		
<ul style="list-style-type: none"> Ring Tone Ring Back Tone 	<ul style="list-style-type: none"> Dial Tone Busy Tone 	
Phone Function		
<ul style="list-style-type: none"> Volume Adjustment Speed dial key Phonebook 		
IP Assignment		
<ul style="list-style-type: none"> IP (Static IP) DHCP PPPoE 		
Security		
<ul style="list-style-type: none"> HTTP 1.1 basic/digest authentication Web setup MD5 for SIP authentication (RFC2069/RFC2617) 		
QoS		
<ul style="list-style-type: none"> QoS field 		
NAT Traversal		
<ul style="list-style-type: none"> STUN 		
Configuration		
<ul style="list-style-type: none"> Web Browser Console/Telnet Keypad 		
DTMF		
<ul style="list-style-type: none"> DTMF RELAY DTMF RFC 2833 DTMF SIP Info 		
SIP Server		
<ul style="list-style-type: none"> Support two SIP server working at the same time Provide a Backup SIP Server 		
Firmware Upgrade		
<ul style="list-style-type: none"> TFTP 	<ul style="list-style-type: none"> HTTP 	<ul style="list-style-type: none"> FTP

2. Handset keys for 3300IP

2.1 Function table of keystroke

Name	Status	Function/Display
Hold	Call	Hold on
Store (submerged key)	Press beneath faceplate	To enter storage mode for speed dialing
Mute	Press	Mute voicepath
MWI - TouchLite	Flashing or Not	Message Waiting Indication and Speed Dial
Redial	Call/Dialing	Call the number dialed last time
Spkr		Speaker
Volume +	Call	Increase the volume
Volume -	Call	Decrease the volume
M1 ~ M10	On-hook	10 speed dial numbers
1	Dialing	"1"
2	Dialing	"2"
3	Dialing	"3"
4	Dialing	"4"
5	Dialing	"5"
6	Dialing	"6"
7	Dialing	"7"
8	Dialing	"8"
9	Dialing	"9"
0	Dialing	"0"
*	Dialing	"*"
#	Dialing	It can be regarded as the first number being dialed out or the end mark for ending number

2.1.1 Store numbers:

Press "STORE" key (submerged under faceplate) to enter into the setting state of storing keys, and input the number to store, then press the corresponding speed dial key to store the numbers.

2.1.2 Call volume control

Press "Volume +" to increase the volume, and "Volume -" to decrease the volume.

2.1.3 Hold Function

Hold the current call line.

2.1.4 Redial Function

Redial the number dialed last time.

2.1.5 MWI – Message Waiting Indication

The Red TouchLite key serves as an indicator that messages are present as well as a speed dial key to dial in and retrieve those messages. If messages are present, press the flashing red MWI key to go off-hook and dial the programmed voicemail number. First, program the key in the web UI.

2.1.6 Speed Dial Keys

The 3300IP is configured with either 5 (MWD5) or 10 (MWD) speed dial keys, programmable through the web UI under Advanced -> Memory Key.

3. Web browser User Interface to Configure Phone

Insert one end of net wire to interface of network card of computer, then insert the other end to LAN interface of the phone, and set the computer IP on the same net with the phone IP or make it gets IP automatically. Then open the IE, input the phone IP address in the address field, you will enter web setting page of phone 3300IP.

3.1 Login:

The default user name and password are admin/admin and guest/guest.

Username:

Password:

3.2 Current state

This page layout shows the work state of VoIP phone. The network part shows the connection state of WAN interface and LAN interface and the network setting; the work state of Public SIP service of VoIP part, and here you can see the registration and whether registered to the server or not. The Phone Number part shows the telephone numbers in Private SIP server and Public SIP server.

Current Status

Network			
WAN		LAN	
Connect Mode	DHCP	IP Address	192.168.10.1
MAC Address	00:0e:5c:ca:73:32	DHCP Server	ON
IP Address	192.168.0.40		
Gateway			
Phone Number			
SIP LINE 1	ztiondai@60.12.174.226:5060	Registered	
SIP LINE 2	@:5060	Unapplied	

Version: VOIP PHONE v1.7.81.05 Feb 28 2008 10:23:22

3.3 Network

3.3.1 Wan Config

WAN port network setting page.

Support static IP, dynamic obtain IP and PPPoE.

WAN Configuration

Wan Status	
Active IP	192.168.0.10
Current Netmask	255.255.255.0
Current Gateway	192.168.0.1
MAC Address	00:19:13:00:5b:ae
Get MAC Time	20080728

WAN Setting

Static DHCP PPPoE

APPLY

Configure Static IP

WAN Setting

Static DHCP PPPoE

Static IP Address: 192.168.1.179

Netmask: 255.255.255.0

Gateway: 192.168.1.1

DNS Domain:

Primary DNS: 202.96.134.133

Alter DNS: 202.96.128.68

APPLY

- Enable Static;
- Set 3300IP's IP address in the IP Address;
- Set netmask in the Netmask field;
- Set router IP address in the Gateway;
- DNS Domain:
- Set local DNS server in the Preferred DNS and

the Alternate DNS.

Configure to dynamic obtain IP

Enable DHCP;

If there is DHCP server in your local network, 3300IP will automatically obtain WAN port network information from your DHCP server.

Configure PPPoE:

WAN Setting

Static DHCP PPPoE

PPPoE Server: ANY

Username: user123

Password: *****

APPLY

- Enable PPPoE
- PPPoE server: Enter "ANY" if no specified from your ITSP.
- Enter PPPoE username and pin in the username and password.

3300IP will automatically obtain WAN port network information from your ITSP if PPPoE setting and the setup are correct. **Notice:** If user accesses the IP phone through WAN port. He/She should use the new IP address to access the IP phone when the WAN port address was changed.

3.3.2 LAN Config

LAN IP Netmask: Set the IP Netmask for the LAN

DHCP Server: Enable DHCP service in LAN port; after user changed LAN IP, phone will automatically modify DHCP Lease Table and save the configure according to IP and netmask, DHCP server configure won't take effect unless you reboot the device.

NAT: Enable NAT.

Bridge Mode: Enable this option to switch to bridge mode. IP phone won't assign IP for its LAN port in bridge mode and its LAN and WAN port will be in the same network. (This setting won't take effect unless you save the config and reboot the device)

LAN Configuration

Lan Set

LAN IP: 192.168.10.1

Netmask: 255.255.255.0

DHCP Service:

NAT:

Bridge Mode:

APPLY

3.4 VoIP

3.4.1 SIP Config

Setting page of public SIP server:

Register Server Addr:

Register address of public SIP server

Register Server Port:

Register port of public SIP server, default port is 5060

Register Username:

Username of your SIP account (Always the same as the phone number)

Register Password: Password of your SIP account.

Proxy Server Addr: IP address of proxy SIP server (SIP provider always use the same IP for register server and proxy server, in this case you don't need to configure the proxy server information.)

Proxy Server Port: Signal port of SIP proxy

Proxy Username: proxy server username

Proxy Password: proxy server password

Domain Realm: SIP domain, enter the sip domain if any, otherwise 3300IP will use the proxy server address as sip domain.

Local SIP port: Local SIP register port, default 5060

Phone Number: Phone number of your SIP account

Enable Register: Enable/Disable SIP register.3300IP won't send register info to SIP server if disable register.

Enable Message Waiting: The configuration allows/forbids Message Waiting.

Advanced SIP Setting

Register Expire Time: register expire time, default is 60 seconds. 3300IP will auto configure this expire time to the server recommended setting if it is different from the SIP server.

Call Forward: Please refer to Value_add_service for detail. No answer: If no answer, it will forward to appointed phone.

Always: The caller always forward to the appointed phone.

Forward Photo Number: call the forwarded phone number.

Detect Interval Time: Co-work with the Auto Detect Server, if Auto Detect Server is enable, 3300IP will periodically detect if the SIP server is available according this setting.

User Agent:

Encrypt Key: The particular service system decrypts of the key, matching with the server Type usage, the key provide by the particular service system supplier, default is empty

Server Type: The particular service system supplier carries out the sign and speeches to encrypt, default is common

DTMF Mode: DTM signal sending mode: support RFC2833, DTMF_RELAY (inband audio) and SIP info

RFC Protocol Edition: Current 3300IP SIP version. Set to RFC 2543 if the gate need to communicate to devices (such as CISCO5300) using the SIP 1.0. Default is RFC 3261.

3.4.2 Iax2 Config

Setting page of public IAX server:

IAX Server Addr: Register address of public IAX server

IAX Server Port: Register port of public IAX server, default port is 4569

Account Name: Username of your SIP account (Always the same as the phone number)

Account Password: Password of your IAX account.

Local port: Signal port of local, default port is 4569

Phone Number: Phone number of your IAX account

Voice mail number: If the IAX support voice mail, but your username of the voice mail is letters which you can not input with the ATA, then you use the number to stand for your username

Voice mail text: if IAX support voice mail, config the domain name of your mail box here.

Echo test number: If the platform support echo test, and the number is test form, the config the test number to replace the text format The echo test is to test the working status of terminals and platform

Echo test text: echo test number in text format

Refresh time: IAX refresh time

Enable Register: enable or disable register

Enable G.729: Using G.729 speech coding mandatory consultations

IAX2 (Default Protocol): Set IAX 2 as the default protocol, if not the system will choose SIP as default

3.5 Advance

3.5.1 DHCP Server

DHCP server manage page.

User may trace and modify DHCP server information in this page.

DHCP Lease Table: display the IP—MAC corresponding table that the server distributed.

Lease Table Name: Lease table name.

Start IP: Start IP of lease table.

End IP: End IP of lease table. Network device connecting to the 3300IP LAN port can dynamic obtain

the IP in the range between start IP and end IP.

Lease Time: DHCP server lease time.

Netmask: Netmask of lease table.

Gateway: Default gateway of lease table

DNS: default DNS server of lease table.

DNS Relay: enable DNS relay function.

User may use below setting to add a new lease table.

Notice: This setting won't take effect unless you save the config and reboot the device

3.5.2 NAT

Advance NAT setting. Maximum 10 items for TCP and UDP port mapping.

DHCP Lease Table: Show IP—MAC corresponding table assigned by DHCP server.

IPSec ALG: Enable/Disable IPSec ALG;

FTP ALG: Enable/Disable FTP ALG;

PPTP ALG: Enable/Disable PPTP ALG;

Transfer Type: Transfer type using port mapping.

Inside IP: LAN device IP for port mapping.

Inside Port: LAN device port for port mapping.

Outside Port: WAN port for port mapping.

Click **Add** to add new port mapping item and **Delete** to delete current port mapping item.

DMZ Config:

3.5.3 STUN

This page is used to set the private sip server, stun server, and back up sip server information. STUN Server setting: SIP STUN is used to realize SIP penetrates through NAT, when the phone configures IP and port of STUN server (default is 3478) and select Enable SIP Stun, common SIP server can be used to realize the phone to penetrate through NAT. In this way, If you have common SIP proxy and STUN server parked public network, it is all right, but STUN only support three NAT ways: FULL CONE, restricted, port restricted;

STUN Server Addr: configure stun server address;

STUN Server Port: configure stun server port default 3478

STUN Effect Time: stun detect NAT type circle, unit: minute.

Local SIP Port: The SIP port of this phone

Load: Load the choices of SIP line.

Use Stun: Stun. Set the Stun that allows/forbids use user setting.

3.5.4 Net Service

HTTP Port: configure HTTP transfer port; default is 80. User may change this port to enhance system's security. When this port is changed, please use <http://xxx.xxx.xxx.xxx:xxxx/> to reconnect.

Telnet Port: configure telnet transfer port, default is 23.

RTP Initial Port: RTP initial port.

RTP Port Quantity: Maximum RTP port quantity, default is 200

Notice:

Settings in this page won't take effect unless save and reboot the device.

If you need to change telnet port or HTTP port, please use the port greater than 1024, because ports under 1024 is system remain ports.

HTTP service if HTTP is set to 0.

3.5.5 Firewall settings

Firewall setting page: User may set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices to access the internet.

Access list support two type limits: input_access limit or output_access limit. Each type supports 10 items maximum.

3300IP firewall filter is base WAN port. So the source address or input destination address should be WAN port IP address.

Configuration:

In_access enable enable in_access rule

Out_access enable enable out_access rule

Input/Output: current adding is input rule or output rule.

Deny/Permit: current adding is deny rule or permit rule.

Protocol Type: protocol using in this rule: TCP/IP/ICMP/UDP.

Port Range: port range if this rule

Src Addr: source address. It is can be single IP address or network address.

Dest Addr: destination address. It is can be IP address or network address.

Src Mask: source address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID

Des Mask: Destination address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID

3.5.6 VLAN Configuration

3300IP phone implement QoS based on 802.1p. The QoS is used to mark the network communication priority in the data link/MAC sub-layer. 3300IP will sort the packets using the QoS and sends it to the destination.

VLAN Enable: If enable the VLAN service, the second layer will realize separate voice, signal and data transmission. To realize separate voice and data transmission by dispose for IP precedence of ToS area of voice transmission. To reach upper layer switch or router have priority to transfer voice transmission. (The prerequisite is the upper layer switch or router has to

identify ToS area.)

VLAN ID: Dispose VLAN ID is adding a Tag header after realize enable the VLAN function. The realized voice packets transfer at the same VLAN. The prerequisite is it must the same as VLAN of upper switch. The value range are 1~4094.

DiffServ Enable: If enable the VLAN service, it indicates use DSCP mode to realize three layers QoS. This moment, the DSCP of SIP signals which between 3300IP Phone and MGC. It will use Class Selector 5 (The value is 0xA0). And the DSCP of mediums information (In RTP packets) would be used the values of DiffServ Value field.

DiffServ Value: The value range:

0x28, 0x30, 0x38, 0x48, 0x50, 0x58, 0x68, 0x70, 0x78, 0x88, 0x90, 0x98, 0xb8. default is 0xb8 ,0xb8 stands for best fast transmission; 28-38 is guarantee for the transmission priority for the 1st rank , 48-58 is guarantee for the transmission priority for the 2nd rank, 68-78 is guarantee for the transmission priority for the 3rd rank, 88-98 is guarantee for the transmission priority for the 4th rank.

802. IP Priority: The priority of 802.ip

3.5.7 Digital Map

Digit map is a set of rules to determine when the user has finished dialing.

3300IP support below digital map:

Digital Map is based on some rules to judge when user end their dialing and send the number to the server. 3300IP support following digital map:

- End With "#": Use # as the end of dialing.
- Fixed Length: When the length of the dialing match, the call will be sent.
- Timeout: Specify the timeout of the last dial digit. The call will be sent after timeout
- Prefix: User define digital map
-

[] represents the range of digit, can be a range such as [1-4], or use comma such as [1,3,5], or use a list such as [234]

x represents any one digit between 0~9

Tn represents the last digit timeout. n represents

the time from 0~9 second, it is necessary. Tn must be the last two digits in the entry. If Tn is not included in the entry, we use T0 as default, it means system will send the number immediately if the number matches the entry.

Example:

8[2-8]xxxxx All number from 8200000 to 8899999 will be sent immediately.

955xx5 digits numbers begin with 9 will be sent immediately.

10060 Number 10060 will be sent will be immediately
22xxxxx T1 7 digits numbers begin with 22 will be sent after one second

39[3,9]xxxx, 7 digits numbers begin with 393 or 399 will be sent immediately.

3.5.8 Call Service Settings

User configure the value add service such as hotline, call forward, call transfer, call waiting, 3-way conference call, auto-answer, etc in this page

Hotline: configure hotline number. 3300IP immediately dials this number after hook-off if it is set.

Auto Answer: Enable/disable auto answer function.

No Disturb: DND, do not disturb, enable this option to refuse any calls.

Ban Outgoing: Enable this to ban outgoing calls.

Enable Call Transfer: Please refer to Value_add_service for detail.

Enable Call Waiting: Enable/disable Call Waiting
Enable Three Way Call: Please refer to Value_add_service for detail.

Accept Any Call: If this option is disabled, 3300IP refuse the incoming call when the called number is different from 3300IP's phone number.

No Answer Time: no answer call forward time setting.

Black List: incoming call in these phone numbers will be refused.

Limit List: outgoing calls with these phone numbers will be refused

3.5.9 Memory Key Programming

This page layout shows the number setting of Speedial keys. There are several special functions that can be programmed into Memory Keys, such as **Transfer**, **Transfer-Silent**, and **Call-Park/Retrieve**.

3.5.9.1 Memory Key Special Programming – Transfer, Transfer Silent and 1-second Pause

Transfer occurs when there is an active/live call in progress. If no call is active, the 'T' is ignored, and the memory key will just call the programmed number (1107). In the settings below, **Memory 1** (T1107) will transfer to extension 1107 when pressed. The user will hear the DTMF tones as they are transferred to 1107.

Memory Key Setting	
Memory 1	T1107
Memory 2	TS1107

In the example of **Memory 2**, (TS1107) will transfer “silently” to extension 1107. ‘Silent’ means the user will not hear the DTMF tones.

In **Memory 3**, the transfer to 1107 will occur, after the connection is established, a pause will happen then digits are sent. P=1-second pause, so PPP = 3 seconds, then the digits (12345) are sent.

Memory Key Setting

Memory 1	T1107
Memory 2	TS1107
Memory 3	T1107PPP12345

3.5.9.2 Call Park and Retrieve

Call Park/Retrieve allows the phone’s user to send a call to a ‘Park’ location (setup within the PBX), where another user may then ‘Retrieve’ the call from a different phone.

It is programmed as follows in two steps:

1. Under SIP Config => Advanced Set you will find Park Mode. Default is Off, and Park 1 is enabled or On.

Advanced SIP Setting

Register Expire Time	60	seconds
NAT Keep Alive Interval	60	seconds
User Agent	TMX SE2 V1.8.5-955	
Signal Key		
Media Key		
Local Port	5060	
Ring Type	Type 1	
Park Mode	Default	
Enable URI Convert	Default	
Enable Keep Authentication	<input type="checkbox"/>	
NAT Keep Alive	<input type="checkbox"/>	
Enable Via rport	<input checked="" type="checkbox"/>	
Enable PRACK	<input type="checkbox"/>	
Long Contact	<input type="checkbox"/>	
Dial Without Register	<input type="checkbox"/>	

2. Under Advanced => Memory Key, program the **Memory Key HdActive** to Transfer Silent the call to the Park Location (70 in the example below). Eg:TS70

Memory Key HdActive	TS70
Memory Key Hddle	TS71

Also program **Memory Key Hddle** to Retrieve from the Parked Location. In this case TS71. The Park and Retrieve numbers (70 and 71) are arbitrary, and subject to the programmer of the PBX. They are used here as an example only.

3.5.10 MMI Filter

MMI filter is used to make access limit to 3300IP phone. When MMI filter is enable. Only IP address within the start IP and end IP can access 3300IP phone.

3.5.1.1 Audio Settings

CODEC: select the prefer CODEC; support ulaw, alaw, G729 and G7231 5.3/6.3

Signal Standard: Signal standard for different area.

Handdown Time: hand down detect time.

Input Volume: Handset in volume.

Output Volume: Handset out volume.

Handfree Volume: Hand free volume

G729 Payload Length: G729 payload length

VAD: Enable/disable Voice Activity Detection

3.5.12 VPN

This page is VPN setting page, the IP phone support the VPN with UDP and L2TP protocol .The parameters is as below

VPN IP: After VPN registered successfully, VPN server will give an IP address to the terminal. If there is a IP address shown on terminal (except for 0.0.0.0), it means your VPN has registered UDP Tunnel

VPN Server Addr: register to the address of VPN server

VPN Server Port: register to the port of VPN server

Server Group ID: the group ID of UDP VPN

Server Area Code: the area code VPN server L2TP

VPN Server Addr: register to the address of VPN server

VPN User Name: L2TP VPN username

VPN Password: L2TP VPN password

UDPTunnel: use the UDP to visit VPN

L2TP: use the L2TP to visit VPN

Enable VPN: Enable the VPN server, you must choose UDP or L2TP type in advance

Notice: At the present, L2TP only support L2TP VPN server under Linux, UDP only support a private UDP VPN server.

3.6 Dial-Peer dial rule setting

Please refer to How to use dial rule for detail.

Number	Destination	Port	Mode	Alias	Suffix	Del Length
2T	255.255.255.255	5060	SIP	del	no suffix	1
3T	0.0.0.0	5060	SIP	del	no suffix	1
123	0.0.0.0	5060	SIP	all05332221015	no suffix	0
0T	0.0.0.0	5060	SIP	reg-85	no suffix	1
11	192.168.0.11	5060	SIP	no alias	no suffix	0

3.7 Config Manage

Save Config: save current settings.

Clear Config: restore to default settings.

Backup Config: Backup the config file, via point the right key of mouse → save target as... → will pop a save window, then type the config file name in the File name (the file type is text file)

Update Configuration: Update the current configuration through configuration files.

Notice: clear config in admin mode, all settings restores to factory default; clear config in guest modem, all settings except sip, advance sip restore to factory default.

3.8 Update Firmware

3.8.1 Update

Web Update:

Update the application or configuration files of the phone. The application document is .z format, and the configuration files are .cfg format.

Through clicking on the "browse" button to open the upgrade file or configuration file, then click on "Update" button. After the upgrade, 3300IP will automatically restart.

FTP Update:

Upload/download the configure file with FTP or TFTP server, or download firmware from FTP or TFTP server. Back up configure file to your FTP/TFTP server. configure use .cfg extension.

The Type includes two parts of config file export and config file import

Config file export: export the config file

Config file import: import the config file

3300IP phone support FTP and TFTP auto update, the gateway will auto obtain the configure file from your update server if configured. To obtain the original configure file, you can use the FTP/TFTP back up as

describe above. Configure file using module structure, user may remain the concerned modules and remove other modules. Put the configure file in the root directory of update server when finish editing.

3.8.2 Auto Update

Current Version: the system will display the current version number

Server Address: FTP/TFTP server address

Username: FTP server user name

Password: FTP server password

Config File Name: The name of configuration file

Config Encrypt Key: The encrypt key of confirmation file

Protocol Type: The protocol type that used for upgrading

Update Interval Time: The interval time that the terminals search for new configuration file.

Update Mode: auto provision mode; Disable: not auto update, Update after reboot: auto update after reboot, Update at time interval: auto update after a certain time

Configure file version was in the <<VOIP CONFIG FILE>> Version 1.0007 and <GLOBLE CONFIG MODULE> ConfFile Version
For instance:

Gateway original version is:
<<VOIP CONFIG FILE>>Version: 1.0000
<GLOBLE CONFIG MODULE> ConfFile Version: 6

User may edit the configure file version to:
<<VOIP CONFIG FILE>>Version: 1.0007
<GLOBLE CONFIG MODULE> ConfFile Version: 7

3.9 System Manage

3.9.1 Account Manage

Set web access account or keypad password of 3300IP.

3.9.2 Syslog config

Set the system log

Server IP: set the syslog server address

Server Port: set the syslog server port

MGR Log Level: set the MGR log level

SIP Log Level: set the SIP log level

IAX2 Log Level: set the IAX2 log level

Please click “**apply**” after setting

3.9.3 Phone Book

3.9.4 Time Set

This page layout is the setting of time of phone.

Server: type the IP address of time server

Timezone: select correct time zone in list box

Timeout: longest response time for SNTP

Daylight: Daylight saving time

SNTP: select SNTP server

12 Hours Format: select 12 hours format

Manual Config: The time setting

3.9.5 MMI SET

Set the greeting information on LCD.

3.9.6 Logout & Reboot

Logout: logout the Web entry.

Reboot Phone: logout the entry, and reboot the phone. When user to modify any config of the phone, it will take effect after being rebooted, you can enter into this layout and click "Reboot". And the phone will be rebooted automatically.

Note: Reboot IP phone, some setting needs to reboot to make it works. Please always save config before reboot, otherwise the setting will return to previous setting.

4. Operating Method for Dialing

4.1 How to dial IP Phone

You can make a call after being made a proper setting on your phone. Please confirm whether all the net wires are connected correctly.

If you want to make a call, you can make it after dialing the number and then pressing "#".

You can find IP address by the menu.

Modifying the IP address of the computer, and making it the same net with 3300IP.

Inputting the IP address of 3300IP in the browser, and then you can visit the setting layout of 3300IP after press the Enter key; super user account is admin/admin; common user account is guest/guest.

4.2 Set the phone being connected to server

4.2.1 Set the WAN interface

The connection ways of entering the Network→WAN Config layout phone of the net port: 3300IP could be connected to Internet by using the static IP, DHCP IP, or PPPoE dialing.

WAN Configuration

Wan Status

Active IP	192.168.0.39
Current Netmask	255.255.255.0
MAC Address	00.0e.6b:a7.06.4a
Current Gateway	
Mac Authenticating Code	Valid MAC

WAN Setting

Static DHCP PPPOE

Configure Static IP:

WAN Setting

Static DHCP PPPOE

Static IP Address	192.168.1.179
Netmask	255.255.255.0
Gateway	192.168.1.1
DNS Domain	
Primary DNS	202.96.134.133
Alter DNS	202.96.128.68

- choose static;
- fill in the IP address of 3300IP in the IP address;
- fill in the subnet mask in Netmask;
- fill in the router address or up Gateway address in the Gateway;
- fill in the local DNS server address in the Primary DNS and Alter DNS respectively.

Use the configure to dynamic obtain IP to get IP address:

- choose DHCP option.

Now, if the network has DHCP server, then 3300IP will get IP address, Netmask, Gateway, Primary DNS and Alter DNS from this DHCP server automatically.

Use PPPoE dialing to connect the Internet:

WAN Setting

Static DHCP PPPOE

PPPOE Server	ANY
Username	user123
Password	*****

- choose PPPoE option.
- please fill in the account and password which PPPoE have dialed in the PPPoE Username and Password.

So 3300IP could connect the Internet through PPPoE

dialing, and automatically get IP address, Netmask, Gateway, Primary DNS and Alter DNS and so on.

4.2.2 SIP setting:

SIP Configuration

SIP Line Select

SIP 1

Basic Setting

Register Status	Registered	Display Name	
Server Address	60.12.174.226	Proxy Server Address	
Server Port	5060	Proxy Server Port	
Account Name	jt10ndai	Proxy Username	
Password	*****	Proxy Password	
Phone Number	jt10ndai	Domain Realm	
Enable Register	<input checked="" type="checkbox"/>		

Enter into the VoIP → SIP Config to set the layout config and sip account information:

Register Server Addr: Register address of public SIP server

Register Server Port: Register port of public SIP server. default port is 5060

Register Username: Username of your SIP account (Always the same as the phone number)

Register Password: Password of your SIP account.

Phone Number: Phone number of your SIP account

- choose Enable Register;

You can dial VoIP phone when the WAN interface and IAX protocol are being set correctly.

4.2.3 IAX setting

IAX2 Configuration

IAX2

Register Status	Unregistered
IAX2 Server Addr	
IAX2 Server Port	4569
Account Name	
Account Password	
Phone Number	
Local Port	4569
Voice Mail Number	0
Voice Mail Text	mail
Echo Test Number	1
Echo Test Text	echo
Refresh Time	60 Seconds
Enable Register	<input type="checkbox"/>
Enable G.729	<input type="checkbox"/>
IAX2(Default Protocol)	<input type="checkbox"/>

IAX Server Addr: Register address of public IAX server

IAX Server Port: Register port of public IAX server, default port is 4569

Account Name: Username of your SIP account (Always the same as the phone number)

Account Password: Password of your IAX account.

Local port: Signal port of local, default port is 4569

Phone Number: Phone number of your IAX account

- choose Enable Register;
- if you use IAX account to make a call, please choose IAX (Default Protocol) , if you fail to choose it, then you can use SIP account to make a call again.
- if you use G..729 to arrange it ,please choose Enable G..729

You can dial VoIP phone when the WAN interface and IAX protocol.

Note: please choose Save Config in the Config Manage after setting the information, or the existing setting information will be failed after rebooting..

4.3 How to use the dial rule?

3300IP provide flexible dial rule, with different dial-rule configure, user can easily implement the following function:

- Replace, delete or add prefix of the dial number.
- Make direct IP to IP call
- Place the call to different servers according the prefix.

You can click "Add" to add a new dial rule. Below is the detail setting of the dial-rule:

Phone Number: The Number suit for this dial rule, can be set as full match or prefix match. Full match means that if the number user dialed is completely the same as this number, the call will use this dial-rule. Prefix match means that if prefix of the number that the user dials is the same as the prefix, the call will use this dial-rule, to distinguish from the full match case, you need to add "T" after the prefix number in the phone number setting.

Call Mode: support SIP.

Destination (optional): call destination, can be IP or domain. Default is 0.0.0.0, in this case the call will be routed to the Public SIP server. If you set the destination to 255.255.255.255, then the call will be routed to the private SIP server. Also you can key other address here to make direct IP calls

Port (optional): Configure the port of the destination, default is 5060 in SIP

Alias (optional): Set up the Alias. We support four Alias as below. Alias need to co-work with the Del Length:

add:xxx, add prefix to the phone number, can set to reduce the dial length.

all: xxx, replace the phone number with the xxx, can use as speed dial function.

del, delete the first N numbers. N is set in the Del Length

rep:xxx, replace the first N numbers. N is set call 86633-8215555, then you can set the phone number in the dial rule as 0633T, and set the Alias as rep:86633, and set the Del Length to 4. Then all calls begin with 0633 will be changed to 86633 xxxxxxxx.

Suffix (optional): Configure suffix, show no suffix if not set Instance

Dial-Peer

Number	Destination	Port	Mode	Alias	Suffix	Del Length
2T	255.255.255.255	5060	SIP	del	no suffix	1
3T	0.0.0.0	5060	SIP	del	no suffix	1
123	0.0.0.0	5060	SIP	all:06332221015	no suffix	0
0T	0.0.0.0	5060	SIP	rep:88	no suffix	1
11	192.168.0.11	5060	SIP	no alias	no suffix	0

Add Dial Peer

Phone Number:

Destination (optional):

Port(optional):

Alias(optional):

Call Mode:

Suffix(optional):

Delete Length (optional):

Dial Peer Option

2T rule: If the call starts with 2, the first 2 will be deleted, and the rest number will be sent to private SIP server.

3T rule: If the call starts with 3, the first 3 will be deleted, and the rest number will be sent to public SIP server.

123 rule: Dial 123 and will send 06332221015 to your server. Used as speed dial function.

0T rule: If the call starts with 0, the first 0 will be replaced by 86. Means that if you dial 06332221015 and AG-188 will send 866332221015 to your server.

11 rule: when you dial 11, the call will send to 192.168.0.11, suit for LAN application without set up a sip server.

Reference – Quick Keys

The TeleMatrix family of SIP phones has implemented an innovative way to determine various states of the SIP phone and the network it is attached to via the phone's dial-pad. By pressing the * key twice in succession followed by a series of numbers (generally matched to the command's corresponding letters.)

Many of these features are newly added in Firmware 1.8.0 and higher.

Show IP address - **47# (IP#)**

Description: Keying **47# causes the phone to read out the IP address currently assigned. If the phone has an LCD panel, it should be displayed there. If the phone has no display panel, or the address cannot be displayed, it should be read out audibly on the speaker (or handset if off hook.)

Show VLAN - **85# (VL#)**

Description: When entered respond with the Voice VLAN ID. If VLAN is disabled "Zero" to indicate "off" is spoken.

Set VLAN - **87# (87*password*VLANid#)**

Description: This allows the user to set the VLAN ID. The keypad password is also required. To set the VLAN ID when the password is 9382 and the VLAN ID is 30, enter "***87*123*301#". When a VLAN ID is set, the phone should set VLAN Enabled and the Voice VLAN ID. If necessary the LAN should be set to Bridge mode also.

Show Firmware Version - **39# (FW#)**

Description: When **39# is entered, the phone will display or audibly respond with the firmware version number such as "1.7.240.235".

Show TFTP Server Address - **83# (TF#)**

Description: When entered, the phone responds with the IP address of the TFTP server. This should be the TFTP server address being actually used. It might come from the configuration file, option 66, option 125 or option 43.

Re-program New ConfigID - **77*password*ConfigID# (RP*xxx*yyyy#)**

Description: After entering **77*, the user must enter the Keypad Password followed by a * then the ConfigID of the new config file, followed by #. If the password (in the existing config on the phone is 123, and the new config file is 1234.3300ip.txt), then the user would enter **77*123*1234#. The user should NOT be prompted for the password. The phone will then download the new config file 1234.3300ip.txt.

Reset phone to Factory Default – **33*<password># (DE*123#)**

Description: After entering **33*123#, the phone will reset to factory default, which implies that there is no programming information current, and the phone will prompt for a ConfigID at reboot.

Re-boot phone (without reset) – **72# (RB#)**

Description: After entering 72#, the phone will reboot

SIP TeleWorker Config File Download – **89*<tftp server IP address>*<ConfigID>#

Description: To download a config file for a phone remotely, this **TW code is used for designating a tftp server IP address and configID to download.

Reference – SIP Quick Start to Register a Single Phone

The Cetis, Inc SIP sets are SIP-only endpoints powered only by POE/802.3af and will acquire an IP address when met by a DHCP server. Begin by plugging your POE Ethernet cable into the WAN port. To retrieve an IP address, press * * 4(l) 7(P) #. The phone's IP address will be displayed on the LCD or spoken by audio through the speaker. Pointing your web browser to that IP address. You will be presented with a login screen.

Login = admin

Password = admin

This is the Default Current Status Page

Current Status

TELEMATRIX

Current Status
Network
VOIP
Advanced
Dial-peer
Config Manage
Update Firmware
System Manage

Network			
WAN		LAN	
Connect Mode	DHCP	IP Address	192.168.10.1
MAC Address	00:19:f3:01:2f:c2	DHCP Server	ON
IP Address	10.50.251.244		
Gateway	10.50.0.15		

Phone Number		
SIP LINE 1	1001@10.50.0.45:5060	Registered
SIP LINE 2	@:5060	Unapplied

Version: 9602SIP V1.0.90-603 Aug 18 2009 17:37:23
Firmware Version

The scope of this document is to provide the basic screenshots necessary to quickly configure the guest phone for basic registration and show the user where to program voicemail retrieval keys and speed dials. Our example SIP phone is programmed as extension 1001. The firmware version of the phone is circled in red.

Program your connection under the VoIP -> SIP Configuration Screen

SIP Configuration

TELEMATRIX

Current Status
Network
VOIP
Advanced
Dial-peer
Config Manage
Update Firmware
System Manage

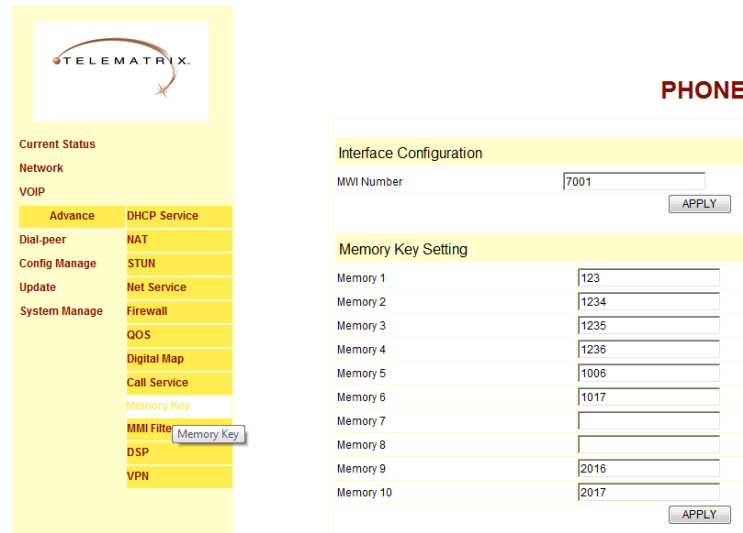
SIP Line Select
SIP 1 Load

Basic Setting			
Register Status	Registered	Display Name	
Server Address	10.50.0.45	Proxy Server Address	
Server Port	5060	Proxy Server Port	
Account Name	1001	Proxy Username	
Password	****	Proxy Password	
Phone Number	1001	Domain Realm	
Enable Register	<input checked="" type="checkbox"/>	Enable Message Waiting	<input checked="" type="checkbox"/>

APPLY
Advanced Set

Server address of the IP/PBX, extension number/account name, phone number and display name, enable register and enable message waiting indication are all programmed here.

Speed dial and Touchlite Message keys are programmed under Advance -> Memory Key



MWI Number is the field for programming the Red *Message Waiting Indicator* Touchlite.

In this example, it is programmed to dial number (7001) on the IP/PBX for retrieving voicemail messages.

Manual Programming - Programming the 3300IP via the LCD and Submerged Keys

There are 2 submerged keys (above the 10 speed dial keys/underneath the faceplate). The left (store) key is used to store speed dials as well as enter the internal menu to check status and program the phone.

1. **Speed Dial:** To program a speed dial, Press the store key... press the digits you want stored... press a speed dial key.

2. **Internal Menu:** To enter the phone's internal menu, hold down the left (store) key for three seconds. At that time, you will be prompted to enter the password (default password = 123) then press the other key (right key = flash) to enter the LCD navigation menu.



Vol up key navigates up
Vol Down key navigates down
Right/Flash key is Enter/Select
Left/Store key is Escape (back out of that menu option)

To determine the IP address assigned to the phone, Navigate to Config -> Network -> WAN -> Status using the keys and process outlined above.

Other useful documents are [Reference-IP DECT Handset Registration and De-Registration](#) & [Reference- 1.8.0 Quick Keys](#)

Reset to Factory Default

To reset the phone to the factory default settings, press “#” during the startup procedure (At power up you can see a black progress bar). The phone will enter into “post” mode, then input *#168 Then you will see “clearing conf” on the screen, next you see “conf reset”. Now you have reset to the default settings. Lastly, cycle power (remove and replace WAN power cable) to restart the phone with factory default settings.