

WT-3202 VoIP Gateway User's Manual

<Version: V1.1 (A)>

Congratulations on your purchase of the product. Please read the manual carefully to ensure your gateway work in best status.

Security and Notes:

- ◆ Don't use it in chemical plant, gas station or near the exploder place.
- ◆ Don't use it near the equipment easy to be interfered by wireless.
- ◆ Avoid using it in too high or low temperature, and avoid exposing the product in sun or high-humidity condition.

Note: The specifications and information regarding the products in this manual are subject to change without notice. We reserve the right of improving the product without informing users and ultimate interpretation right of its performance.

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<http://www.windtower.com.cn>

1. Revision information

Revision History					
Versi	Note	Author	Reviewed by	Approved by	Effective date

2. Terminology

Terminology	
Name	Note

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1 Overview

The popularization of the Internet drives the rapid development of a wide variety of IP-based applications. The IP telephone technology has become the major means for operators to develop voice services now. Especially, IP technology becomes the core of the next generation network (NGN), so the IP-based voice technology will keep soaring speeding the future and become the No. 1 choice of new operators in exploring services.

As an Integrated Access Device, the upstream port of the IAD can be directly connected to the IP network and its downstream port can be connected to multiple ordinary telephone sets, to provide basic accesses for POTS users. This user port gateway can support two telephone lines and connect multiple telephone sets. At the same time, this device is capable of Internet gateway and can access data stream, such as Email or Internet information. It is also applicable for small-size enterprises and IP telephone bars.

2 Packing

The IAD is packed with color chassis. Upon receiving the product, please confirm whether the fittings are complete. The packing box contains a set of IAD, 1 piece of RJ45 network cable, one IAD power adapter and user's manual.

3 Safety Instructions

To ensure your safety and safe use of this product, please pay attention to the following items:

- Follow the instructions in the user's manual.
- Keep the device far away from chemicals and regent.
- Store/use the equipment in dry and well-ventilated environment.
- Never open the chassis lest the device is short-circuited or damaged.

4 Introduction to IAD

The IAD works with the most popular LINUX embedded operation system and has special CPU and DSP compression algorithms, featuring universal functions and applicable to a wide variety of needs.

Basic features:

- One 10/100 BASE-T WAN port, used to connect broadband data network
- One 10/100 BASE-T LAN port, bridged to the WAN port.
- 2 analog loops starts the FXS interface (RJ-11), used to connect 2 telephones
- Supporting DHCP Client or static IP address allocation plan
- Supporting 802.1Q VLAN and VLAN Tag
- Mute compression and comfort tone generation technology ensure clear conversation quality.
- Self-adaptive jitter cache ensures smooth voice function
- Lost-packet compensation guarantee mechanism provides a better voice quality.
- Built-in Internet gateway function
- Supporting NAT (Network Address Translation) and NAPT
- Supporting DHCP Server, used for the dynamic address allocation plan of LAN devices
- Built-in PPPoE client, used for broadband access user authentication
- Easily-configured Console port
- Supporting remote configuration of Web mode and remote software downloading/upgrading

5 Performance Indices

Description of Product Model	
IAD 2 FXS	2-port IP voice gateway, SIP protocol
Physical Specifications	
Size	113mm (L) × 90mm (W) × 28mm (H)
Power supply	AC/DC power adapter, 12V DC DC input: 12V DC/1A
Power consumption	< 10W
Weight	About 0.15 kg

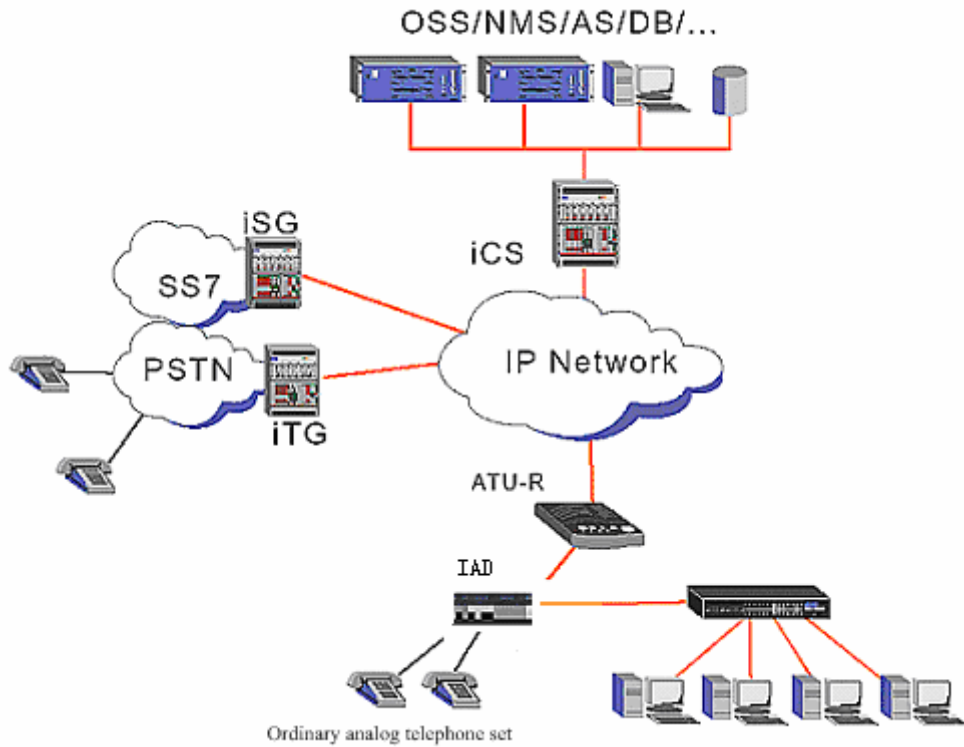
Reliability	System availability > 99.999%, MTBF > 100,000 hours, MTTR < 5 min
Ambient requirements	
Working temperature	0°C ~ 50°C
Storage temperature	-10°C ~ 50°C
Relative humidity	5% ~ 95%, non-condensing
Technical Specifications for the interface attribute gateway	
Supporting MGCP call control protocol	
Mute processing/four wave processing	
RTP/RTCP voice channel	
Voice compression algorithm G.729, G.723 and G.726	
Analog voice port (FXS), 2 ports	
Signal format: DTMF	
Echo suppression: G.165/G.168	
DTMF signal detection/generation	
Compatible to the Internet protocols, such as TCP/IP, UDP, ARP, TFTP and ICMP	
Supporting SNMP Version II	
Compatible to IEEE 802.3 10BASE-TX Ethernet	
Compatible to IEEE 802.3u 100BASE-TX fast Ethernet	

6 Networking Mode

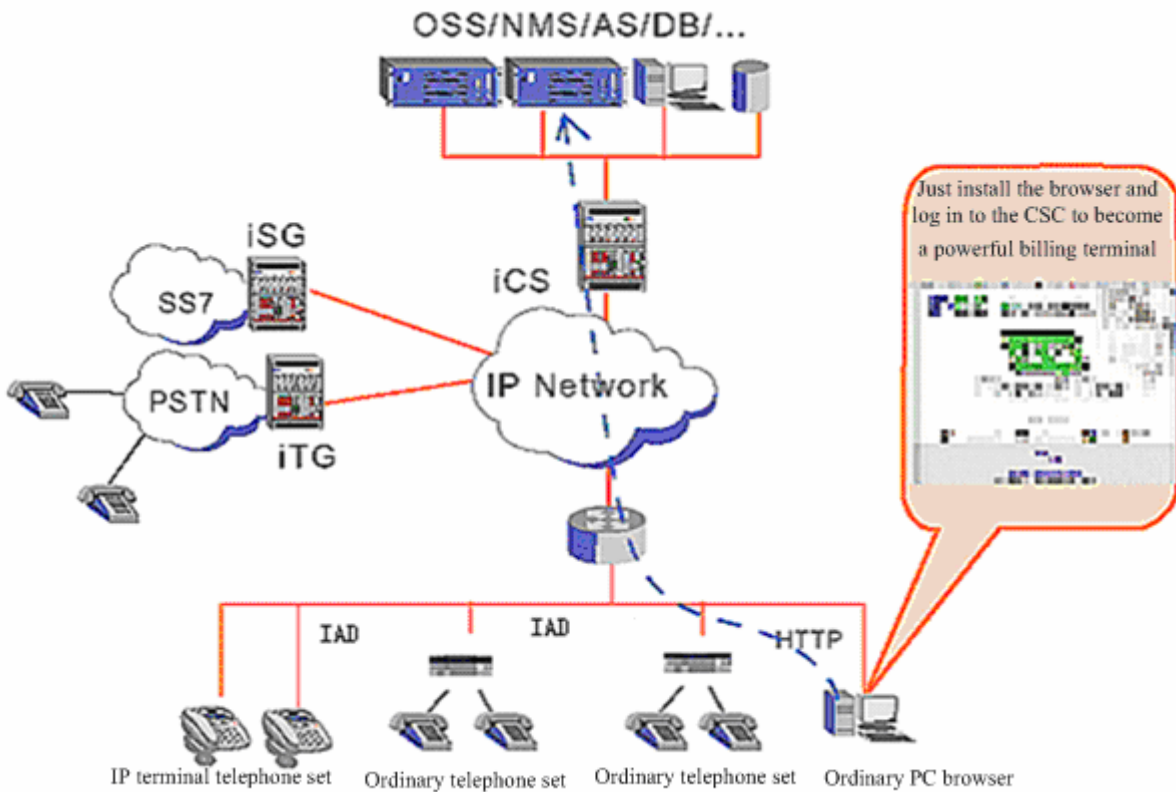
6.1 Typical Applications

The IAD user gateway integrates the Internet gateway and VoIP gateway into a box. Small-size enterprises can use the 10/100Mbps LAN interface to connect local PC with uplink connected to DSL Modem or Cable Modem. They can also use category-5 cables to connect the ISP switch (as shown in Fig. 1). Two FXS can be connected to two ordinary analog telephone sets to provide conversation based on IP network.

The IAD user gateway also is the ideal access equipment for small and medium scale of telephone bars (as shown in Fig.2). The FXS port can be connected to two ordinary telephone sets. The uplink can be connected to the ITSP (Internet Telephony Service Provider) network through an Ethernet interface connected to the small switch or router.

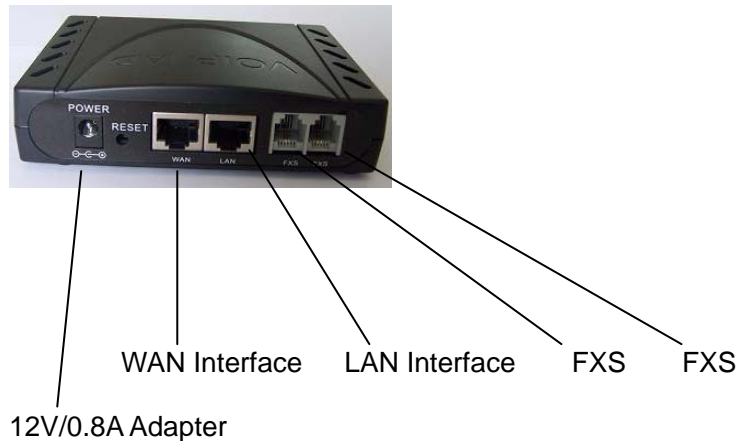


Internet + VOIP System Architecture for Small-size Enterprise/Branch (Copper Cable DSL)



Operator IP Telephone Bar System Architecture (Category-5 Cable)

7 Appearance Description



8 Configuration Description

The IAD provides two ways to modify WEB parameters: through LAN interface and WAN interface. Below describes how to enter the WEB interface configuration parameter (refer to Chapter 9 *WEB configuration interface description*) through LAN interface (refer to Section 8.1) or WAN interface (refer to 8.2).

8.1 LAN Configuration Environment

- Configure the "TCP/IP Protocol" of PC according to Fig. 1 with the default IP addresses of PC and IAD LAN in the same network segment.
- Use straight-through cables in the figure.
- After configuration, input the IAD default IP address in IE address bar. Each IAD will be allocated with an initial Lan IP address before delivery, assumed to be 192.168.0.200.

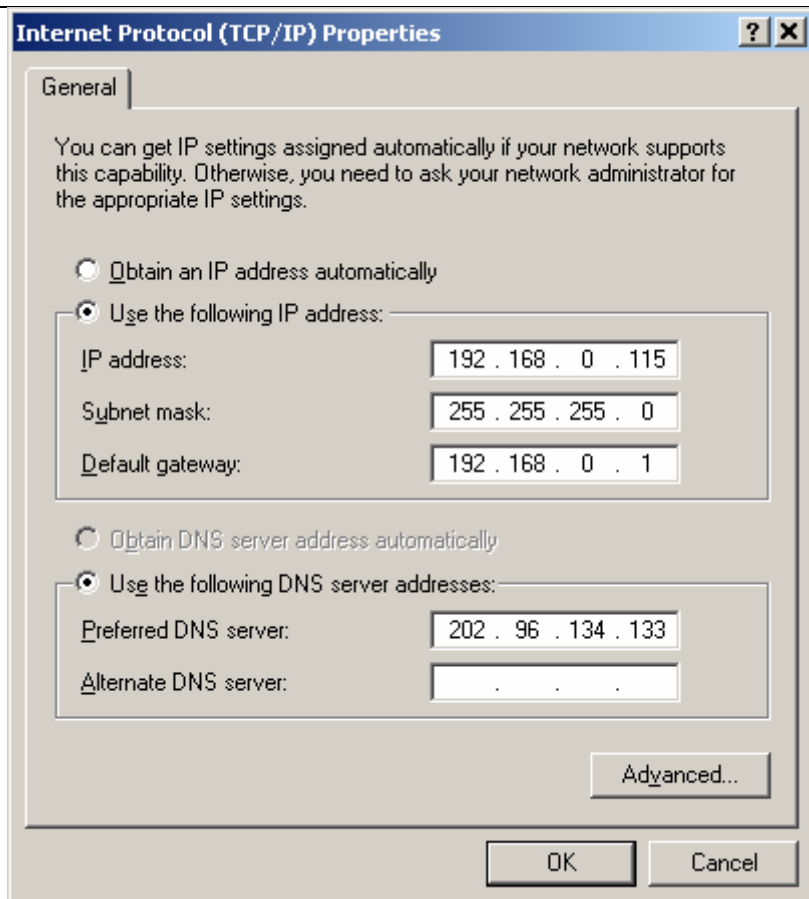


Fig. 1

8.2 WAN Configuration Environment

- Configure the "TCP/IP Protocol" of PC according to Fig. 3 with the PC and WAN interface in the same network segment.
- Configure the device according to Fig. 4.
- Use straight-through cables in the figure.
- After configuration, input the IAD default IP address in IE address bar. Each IAD will be allocated with an initial Wan IP address before delivery, assumed to be 192.168.1.200.

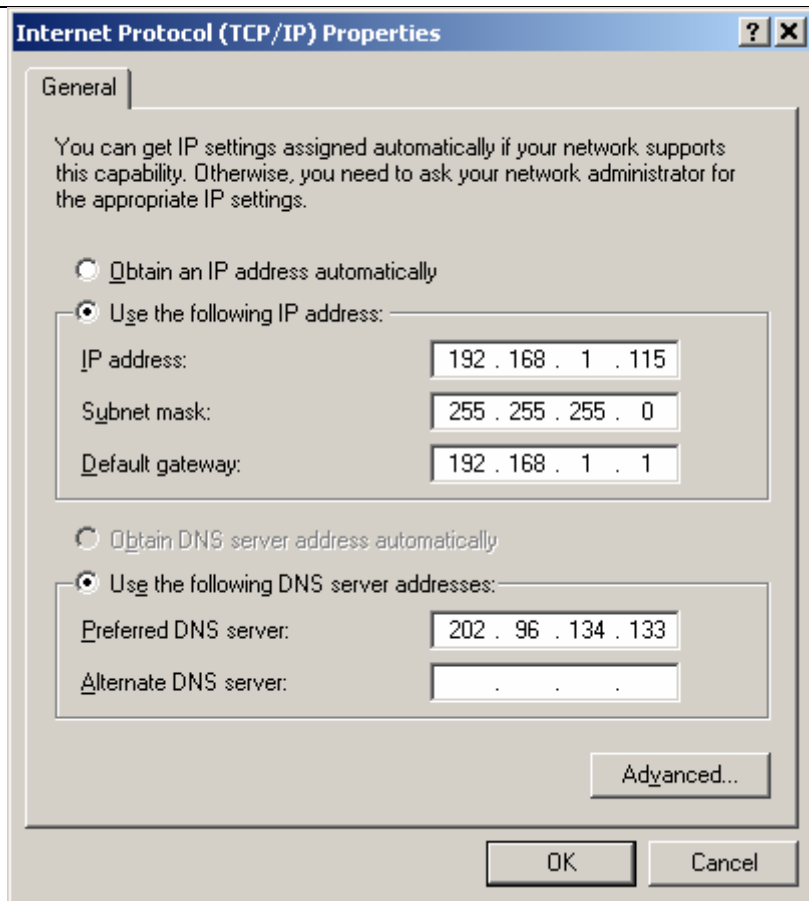


Fig. 3

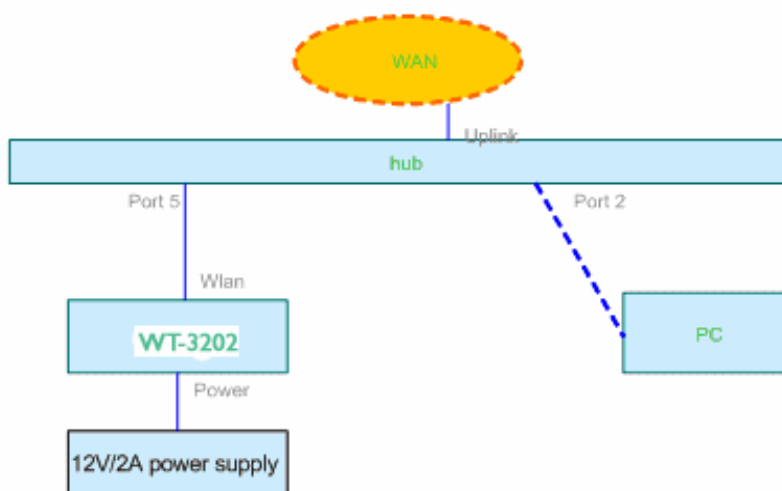


Fig. 4

9 Configuration in WEB Mode

9.1 LOGIN WEB

Through IAD's Wan or Lan IP to login web,Default Admin Username:**admin** ,Default Admin Password:**888888** ,Default User Username:**user** ,Default User Password:**888888**.

When login as admin user, will see like figure 7:

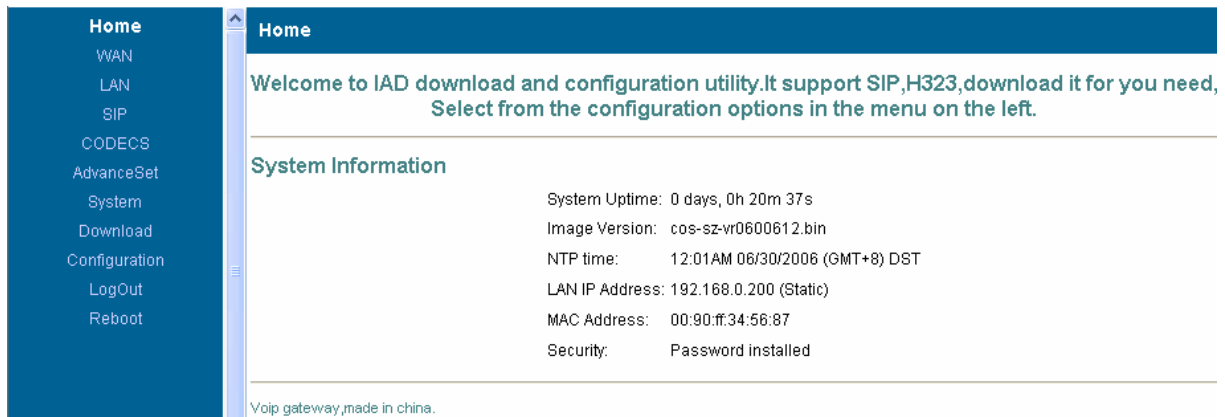
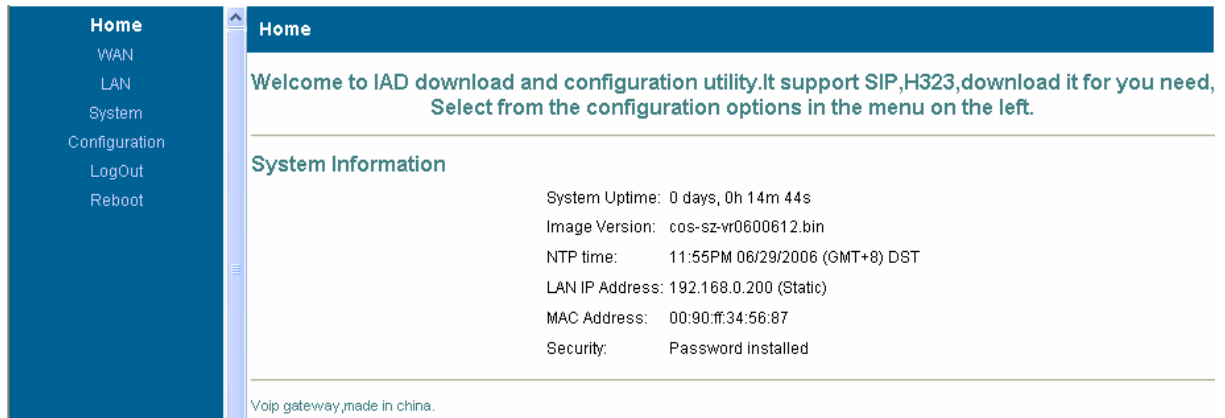
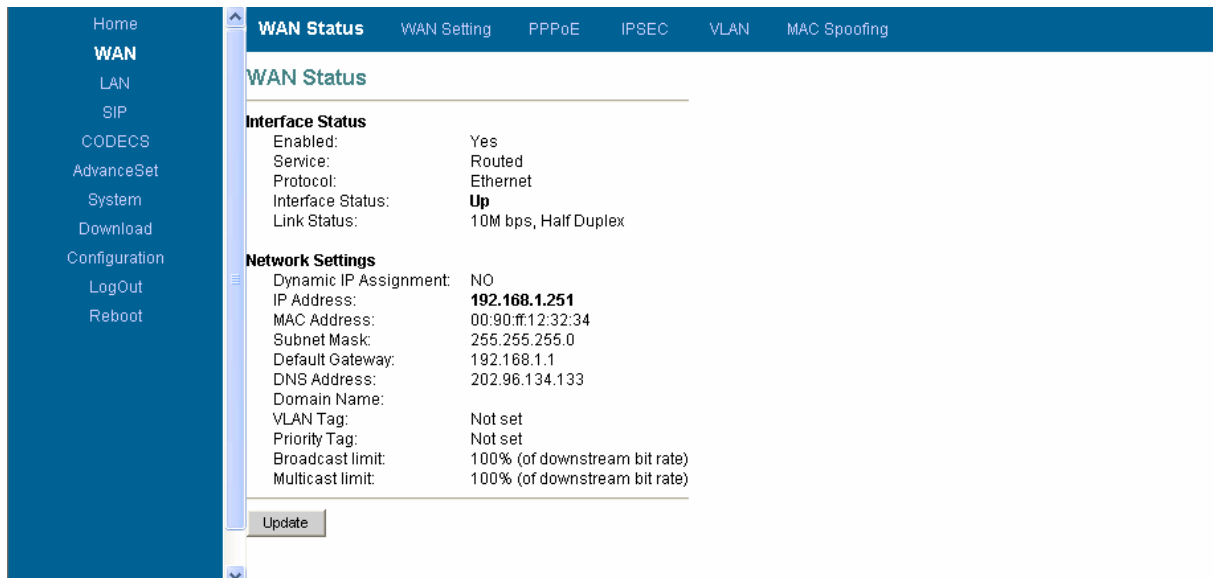


Fig. 7

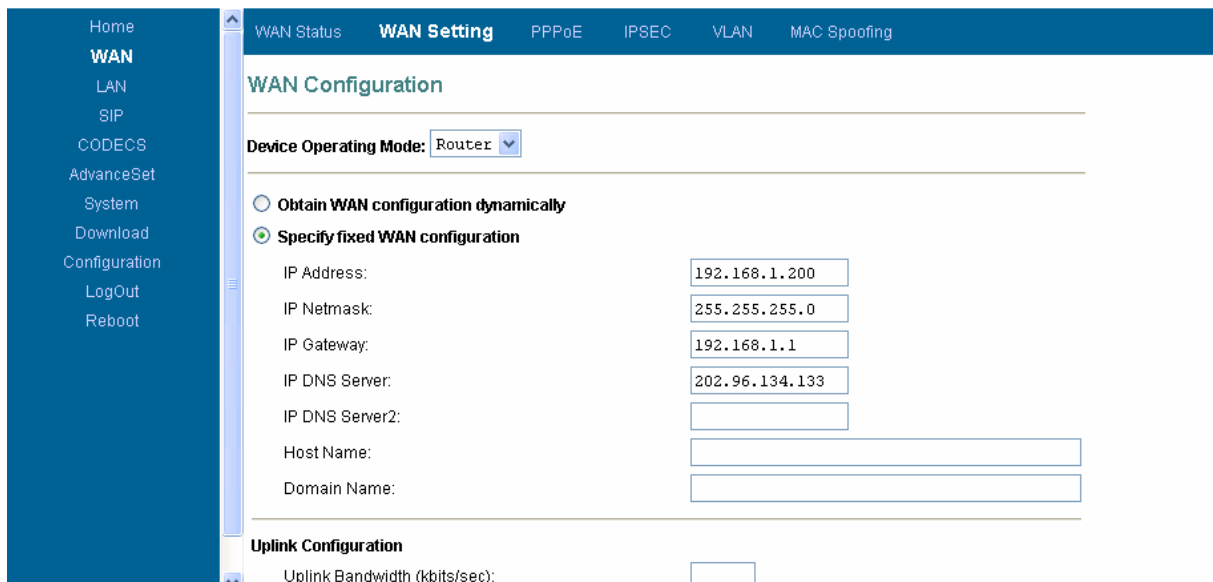
When login as user,will see like figure 8:



9.2 WAN Configuration



This page is the first page displayed when the device's web pages are accessed. It shows how long the device has been running since its last reboot, the IP address the device is currently using, whether or not the device is password protected, and also displays the main application and downloader application firmware versions. In addition, MAC address of the WAN port, and serial number of the device, if it has one, are also displayed in this page.



This page and its sub-pages are available on device that supports routing/bridging and allows viewing and configuration of the WAN interface status/settings. The default WAN interface IP address is set to 192.168.1.200. Please note that any actions/modifications which alter the topology of the Ethernet Bridge will result in the spanning tree protocol to relearn. During this relearn period (may take up to one minute), HTTP access will be unavailable on all bridged

interfaces (including the LAN interface). On this sub-page you can specify whether the WAN interface is to use PPPoE, and configure the PPPoE client. First, select whether PPPoE is enabled or disabled. If enabled, enter the username and password required for the login (authentication) process.

Setting the idle-timeout will result in the PPP connection being torn down if the client detects inactivity on the connection over the specified timeout period. Leaving this field blank will result in the connection being permanently up (i.e. without timeout). Setting the echo-timeout will result in the PPP connection to send the echo request to server in specific time and the echo-count is used for disconnecting this PPP connection if the device does not receive the response of echo request.

If the PPPoE server (service provider) requires any special Service Name or AC Name to be set, you can specify these tags here.

Press "Save PPPoE Settings" to save and apply the WAN interface PPPoE client settings. Any new settings will take effect immediately.

For the setting of MAC Spoofing, if two IADs are configured with the same MAC address, they cannot establish conversation.

9.3 LAN Configuration

This page allows configuration of the device's local network settings. Sub-pages are available for the following on the left of page: the LAN interface status, configuration of the LAN interface settings, and configuration of DHCP server settings, and Port Forwarding (NAPT) Settings.

The screenshot shows a web interface for LAN configuration. On the left is a vertical navigation menu with the following items: Home, WAN, LAN (highlighted), SIP, CODECS, AdvanceSet, System, Download, Configuration, LogOut, and Reboot. The main content area has a top navigation bar with tabs: LAN Status, LAN Setting (selected), DHCP, Routing, Port Forwarding, and VLAN. Below the tabs is the title "LAN Configuration". The settings are organized into sections: "Network Settings" with "IP Address:" (192.168.0.200) and "Subnet Mask:" (255.255.255.0); "Multicast Limits" with "Broadcast limit:" (100 % of Ethernet connection bitrate) and "Multicast limit:" (100 % of Ethernet connection bitrate); and "PHY Speed mode:" (AUTO Negotiation). At the bottom is a "Save LAN Settings" button.

This sub-page allows the user to configure the private LAN interface settings. Assign an IP address to the LAN Ethernet port. This IP address is also the default router address for the devices on the private LAN. The default LAN interface IP address is set to 192.168.0.200. Enter the subnet mask for the private LAN. If you wish to set the broadcast and multicast limits for the bridge/router, enter these values as percentages of the LAN interface Ethernet bit rate. Leaving these values blank will imply values of 100%.

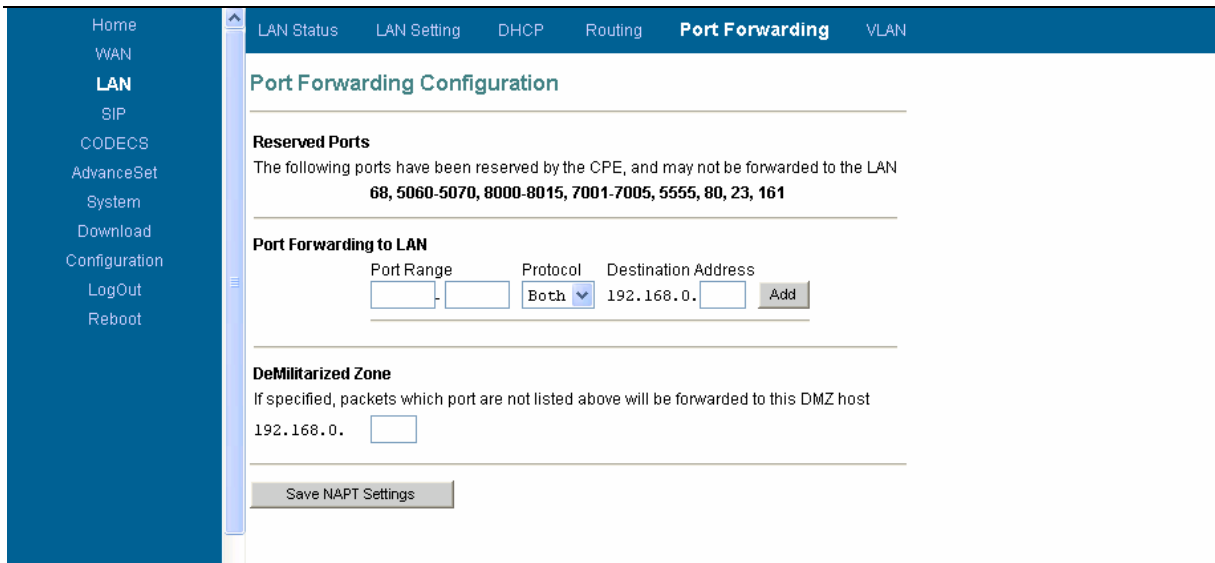
Press "Save LAN Settings" to save and apply the LAN interface settings. Any new settings will take effect immediately.

This sub-page allows configuration of the private LAN DHCP server. The DHCP server settings do not apply if the device is operating in bridge only mode.

Specify whether the device's internal DHCP server feature is enabled or disabled. Also indicate the IP address range to use for DHCP assignments to the LAN. Specify the domain name (optional) that is provided to LAN clients via DHCP. Two optional static DNS server IP addresses can be entered that will be provided to LAN clients via DHCP. These are in addition to the DNS servers automatically provided by WAN connections. Press "Save DHCP Settings" to save and apply the DHCP server settings. Any new settings will take effect immediately.

In addition, up to 8 static DHCP address assignments can be configured. To add a static IP assignment, enter the LAN device's name (must be unique in the private network) and/or MAC address. Specify the static IP to be assigned and press the "Add" button. To remove an existing entry, press the "Remove" button next to the entry to be removed.

To view the current active DHCP client binding table (the internal list of devices for which the server has provided an IP address lease), press "View DHCP Table". A popup window will appear displaying the list of bindings. Press "Update" to update the binding information. Press "Close" to close the binding table window. To clear the device's internal DHCP client binding table, press "Clear client binding table". Be aware that doing this will destroy the DHCP server's knowledge of the LAN clients it has provisioned, and may result in client problems when a client tries to renew its lease.



This page allows the user to customize the devices port forwarding feature. The port forwarding feature does not apply if the device is operating in bridge only mode.

Port forwarding provides WAN access to the internal LAN, by specifying that traffic over certain ports are to be directed at particular LAN hosts. This feature is available only in router mode.

Up to 8 ports forwarding entries ("Pinholes") can be configured. To add a port forwarding entry, configure the Port Range to be forwarded, the Protocol to be forward (TCP, UDP or both), and destination LAN IP Address, press the "Add" button to add the entry. To remove an existing entry, click the "Remove" button next to that entry you wish to remove.

Note that certain port numbers are reserved by the CPE for its own internal use. These ports may not be used for port forwarding to the LAN. Ports which may be reserved by the CPE include those used by VoIP call signaling, RTP packets, HTTP and SNMP. All reserved (unavailable) ports will be displayed on this page.

9.4 SIP Configuration

Home
WAN
LAN
SIP
CODECS
AdvanceSet
System
Download
Configuration
LogOut
Reboot

Server Extensions User 1 Sip Book OOB Signalling ToS/DiffServ Tone/Ring VLAN

(Current Server: sip.ez-gk.net : 5060 ; Domain: sip.ez-gk.net) (Current Server: : 0 ; Domain:)

Address: (IP or FQDN) Address: (IP or FQDN)

Port: Port:

Domain Name: Domain Name:

Send Registration Request with Expire Time Send Registration Request with Expire Time

Outbound Proxy IP: (IP or FQDN) Outbound Proxy IP: (IP or FQDN)

Outbound Proxy Port: Outbound Proxy Port:

RTP Port Number Setting(5000~65535) ~

NAT Traversal Settings

NONE

UPnP Control Point

STUN Server IP: (IP or FQDN) STUN Server Port:

Leaving a setting blank will force the unit to use the information obtained via DHCP and/or DNS

This page allows configuration of the SIP server and endpoint settings.

Enter the address and port value of the SIP server. The address may be an IP address or the name of the server. If no SIP server address is entered, the device will attempt to provision a SIP server using a DNS query. For this to be successful, ensure that the DNS settings on the device include a DNS server address which is configured with the SIP server address and will respond to the query, and the appropriate domain name of the network.

If you wish to specify a special SIP domain name, you may enter the domain name here. If no domain name is entered, the SIP domain name will be set to that of the network (i.e. that which is obtained via DHCP, or specified on the LAN settings page, section 2.3.2).

The currently provisioned SIP Server and Domain are displayed beside "SIP Server Settings" for informational purposes.

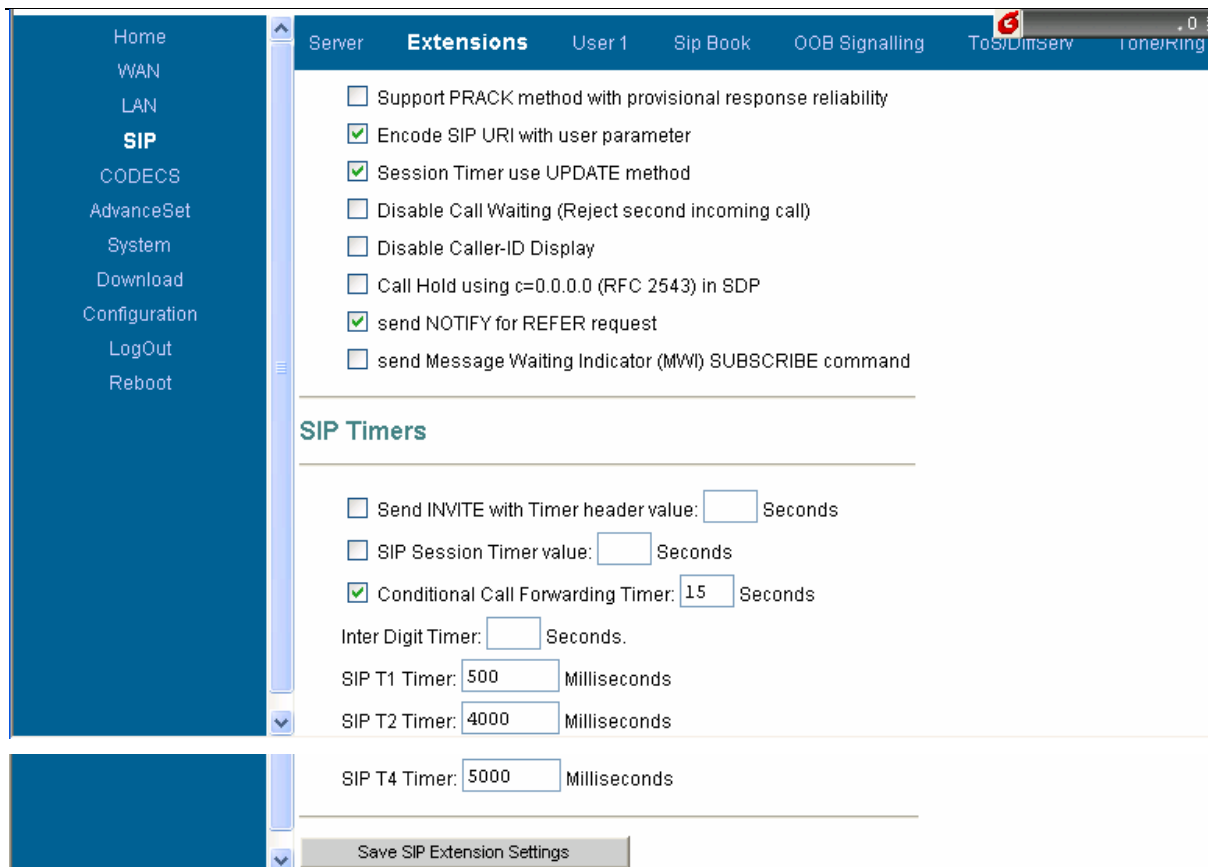
Select whether or not to send a Registration Request to the SIP server by checking the box next to "Send Registration Request".

For the endpoint, set the dial plan to be used by all lines (refer to "Appendix D" for details on the dialplan representation), and select the transport method to be used for SIP signaling (either UDP or TCP).

For each line on the endpoint (NOTE: The IP Phone has a single line), enter the Line Phone Number, Caller-ID Name, signaling port value, authentication Username and Password, and select if AEC is to be performed on this line.

Press "Save SIP Settings" to save the new values.

Extensions



This page allows specification of the SIP signaling stack behavior under certain scenarios.

If you wish for the SIP stack to implement reliable transmission of provisional responses according to RFC 3262 (using the PRACK method), check the option "Support PRACK method with provisional response reliability".

If you wish for the SIP stack to include the user parameter "user=phone" in the SIP URI header(s), check the option "Encode SIP URI with user parameter".

If you wish for the SIP stack to send INVITE messages with the "Timer" header field present, check the option "Send INVITE with Timer header value" and enter the Timer header value.

If you wish for the SIP stack to implement a session timer according to "draft-sip-session-timer", select the option "SIP Session Timer value", and enter the session time-out value.

Press "Save SIP Extension Settings" to save the new values.

Line 1	Phone Number	CallerID Name	Port	AEC On	User Name	Password	Register State
Primary Server	<input type="text"/>	<input type="text"/>	5060	ON	<input type="text"/>	<input type="text"/>	DOWN
Secondary Server	<input type="text"/>	<input type="text"/>	5060	ON	<input type="text"/>	<input type="text"/>	<input type="text"/>

Line1 Gain Control

Input Gain Control (-12 ~ 18)db: db

Output Gain Control (-12 ~ 18)db: db

Save SIP User Settings

This sub-page and the next User2 sub-page allow the user to configure the device with phone number, caller ID, username and password specified by the service provider.

Line 2	Phone Number	CallerID Name	Port	AEC On	User Name	Password	Register State
Primary Server	900508	900508	5060	ON	900508	*****	UP
Secondary Server	<input type="text"/>	<input type="text"/>	5060	ON	<input type="text"/>	<input type="text"/>	<input type="text"/>

Line2 Gain Control

Input Gain Control (-12 ~ 18)db: db

Output Gain Control (-12 ~ 18)db: db

Save SIP User Settings

RTP Telephone Event Configuration

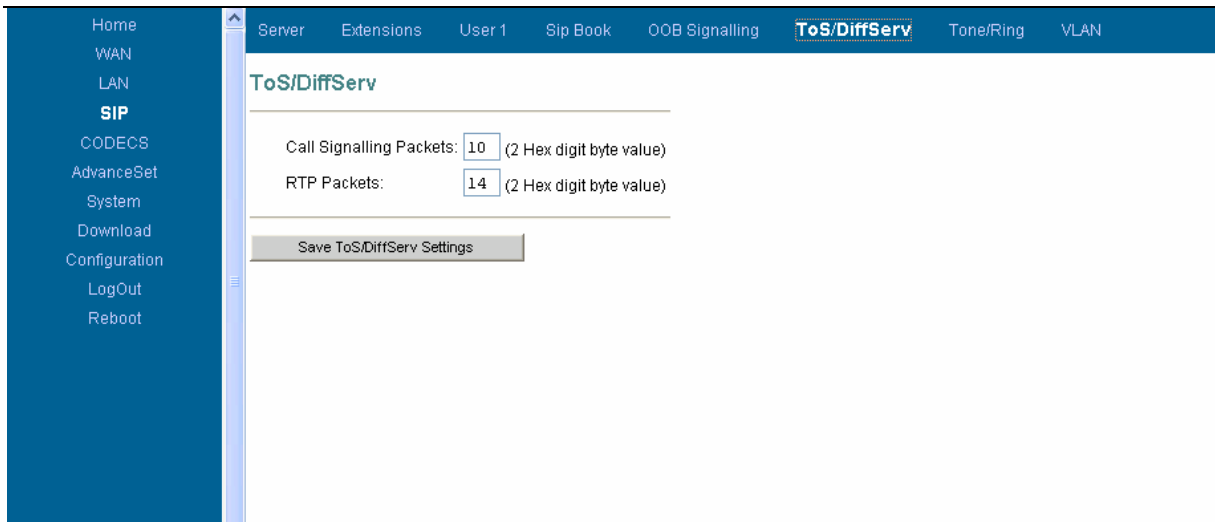
Send DTMF Events:

RFC2833 signalling using payload value:

Regenerate OOB DTMF tone

Save OOB Settings

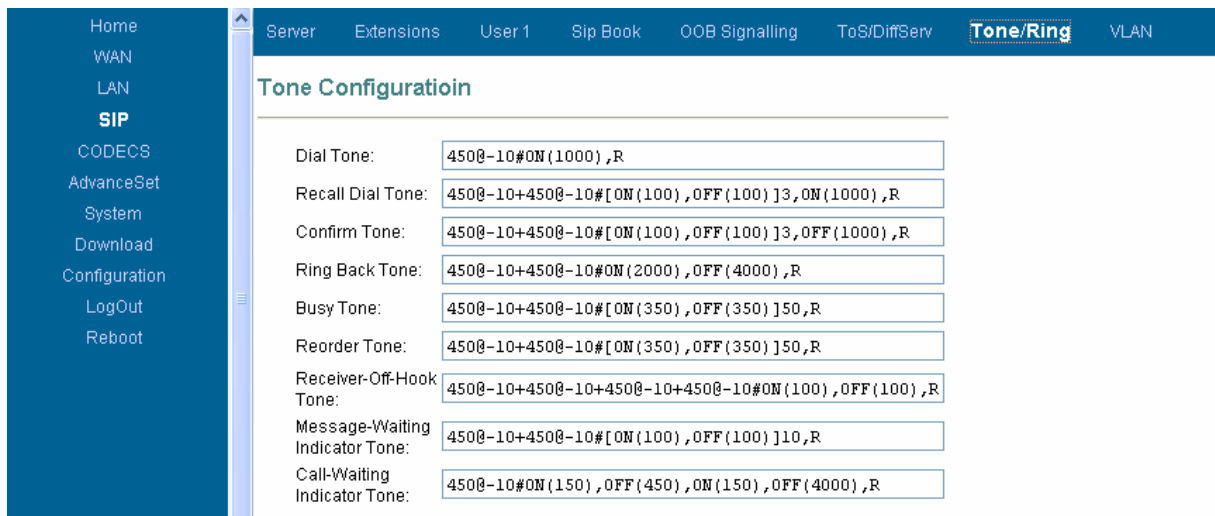
This sub-page allows configuration of the out-of-band signaling options for SIP. Select whether OOB telephone event signaling is to be done using the SIP INFO message, or to be done via RFC2833 RTP signaling



This sub-page is used to configure the Type-of-Service/Diffserv byte values which are to be used in the IP header of all transmitted SIP signaling packets and RTP packets. The ToS/DiffServ byte values are entered as two-digit hexadecimal values. If no special ToS/DiffServ value is to be used for a particular traffic type, enter "00" or leave the setting empty.

Press "Save ToS/DiffServ Settings" to save these new settings.

Tone



This sub-page is used to configure Tones which applies in order to acknowledge users.

Dial Tone: The tone you hear when you pick up handset

Recall Dial Tone: The tone when you hold callee and prepare to make another call.

Confirm Tone: The tone after you've set up some service, like DND (Do Not Disturb), Call Forwarding, etc.

Ring Back Tone: The audible ringing you hear before callee picks up and answers your call.

Busy Tone: The tone indicates the number you dialed is in busy now.

Reorder Tone: The tone you hear if you dial an invalid number or the call is not available.

Receiver-Off-Hook Tone: The tone to alert you to place the handset on-hook.

Message-Waiting-Indicator Tone: The tone to notify you to call for message box.

Call-Waiting-Indicator Tone: The tone to make you aware of the second incoming call while you're in conversations.

Ring

The screenshot shows a web interface titled "Ring Configuratioin". It contains two input fields for ring cadences. The first field is labeled "Default Ring:" and contains the text "ON(1000),OFF(2000),R". The second field is labeled "Call-Waiting Reminder Ring:" and contains the text "ON(125),OFF(625),ON(2000),OFF(2875),R". Below these fields is a button labeled "Save Tone and Ring Settings".

This sub-page is used to configure Ring Cadences required by Rings, Call-Waiting-Indicator, and Distinctive Ring features.

1. Ring Configuration:

Default Ring: Default ring cadence when the phone rings.

Call-Waiting Reminder Ring: Ring cadence of Call-Waiting Reminder Ring.

9.5 CODECS Setting

The screenshot shows the 'CODECS' configuration page. On the left is a navigation menu with options: Home, WAN, LAN, SIP, CODECS (selected), AdvanceSet, System, Download, Configuration, LogOut, and Reboot. The main content area is titled 'CODECS' and contains the following settings:

Selected	Silence Suppression
<input type="checkbox"/> G711U	ON
<input type="checkbox"/> G711A	ON
<input type="checkbox"/> G723	ON
<input type="checkbox"/> G726	ON
<input checked="" type="checkbox"/> G729	ON

Below the table, the 'Packetization' is set to 20ms. The 'Jitter Buffer' section has three options:

- Adaptive Jitter Buffer: 100ms (maximum playout delay in milliseconds)
- Fixed Jitter Buffer: 40ms (fixed playout delay in milliseconds)
- Automatically switch to Fixed Jitter Buffer upon fax/modem tone detection

A 'Save CODEC Configuration' button is located at the bottom of the configuration area.

If the device is running one of the four VoIP applications, this page is available for configuring the audio CODEC parameters, as well as the Jitter Buffer settings for the CODEC decoders.

Enter which CODECs are to be supported.

Select which complex codec is to be supported. Due to memory limitations, it is not possible to select G723 and G729 complex codec at the same time.

Select the packetization period to be used for each selected CODEC.

Select whether Silence Suppression is to be supported for each CODEC.

The Jitter Buffer settings apply to all active CODEC decoders. You may choose between an adaptive jitter buffer and a fixed jitter buffer. For an adaptive jitter buffer, choose the maximum allowable playout delay (in milliseconds). For a fixed jitter buffer, choose the fixed playout delay (in milliseconds).

Finally, select whether or not a decoder should automatically switch from an adaptive jitter buffer to a fixed jitter buffer upon fax/modem tone detection. Adaptive jitter buffers are sometimes detrimental to fax transmission over G711 CODECs if they have to adapt too rapidly or too extensively due to inconsistent and widespread packet delays. In these adverse network conditions, a fixed jitter buffer provides superior performance when handling incoming fax transmissions over G711 CODECs.

Press "Save CODEC Settings" to save the new CODEC parameters.

9.6 System Setting

1. Set Security Password:

Click "System" item on the top menu.

Click "Security" on the left menu.

In Old password field, input old password , if you have it.

In New password field, input new password .

In confirm password field, input confirm password

Press "Change Password" button to save changes.

Timezone:

Find the current time from a list of cities.

Country Caller ID:

The caller ID can find out who's calling you and keep track of how often they call.

Users should set the country field according to their geographical location, otherwise the Caller ID function might not work properly.

1. Timezone setting:

Click "System" on the top menu.

Click "Localization" on the left menu.

In NTP Server field, enter a NTP server IP address . If you want to use the default NTP server, this field should be blank .

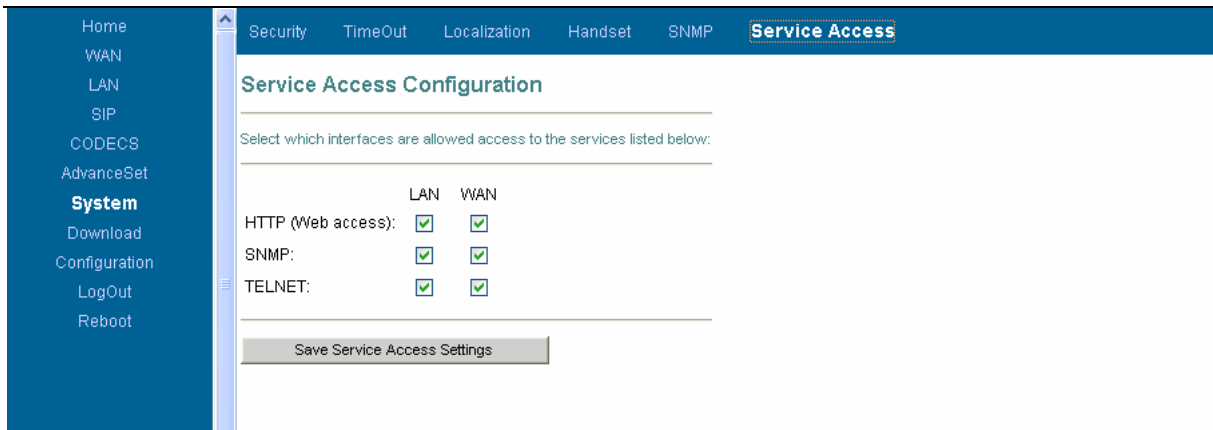
In Time Zone drop down menu, select one time zone .

In Adjust clock for daylight savings checkbox, if your country has daylight savings time, you can enable it

Press Save Localization Settings button , then system will redirect to the web page of reset.

This sub-page is used for configuring the device's SNMP manager. Configure the SNMP Trap Host IP address and community, the SNMP read and write community parameters, and the SNMP System Description and System Object ID parameters.

Press "Save SNMP Settings" to apply the new values. These settings will only take effect when the device is rebooted.



If the device is a multi-interface device (bridge/router), then this sub-page is available to enable/disable access to certain system level network services on the device's interfaces. From this page, an administrator can choose to allow or disallow HTTP, SNMP and Telnet access from devices on the LAN or WAN or both. Care must be taken while configuring the HTTP access, as once you have saved you will no longer be able to access the device's web pages if you have disabled HTTP on the interface you are currently accessing the device on.

Press "Save Service Access Settings" to save and apply the new access settings.

9.7 Download Setting

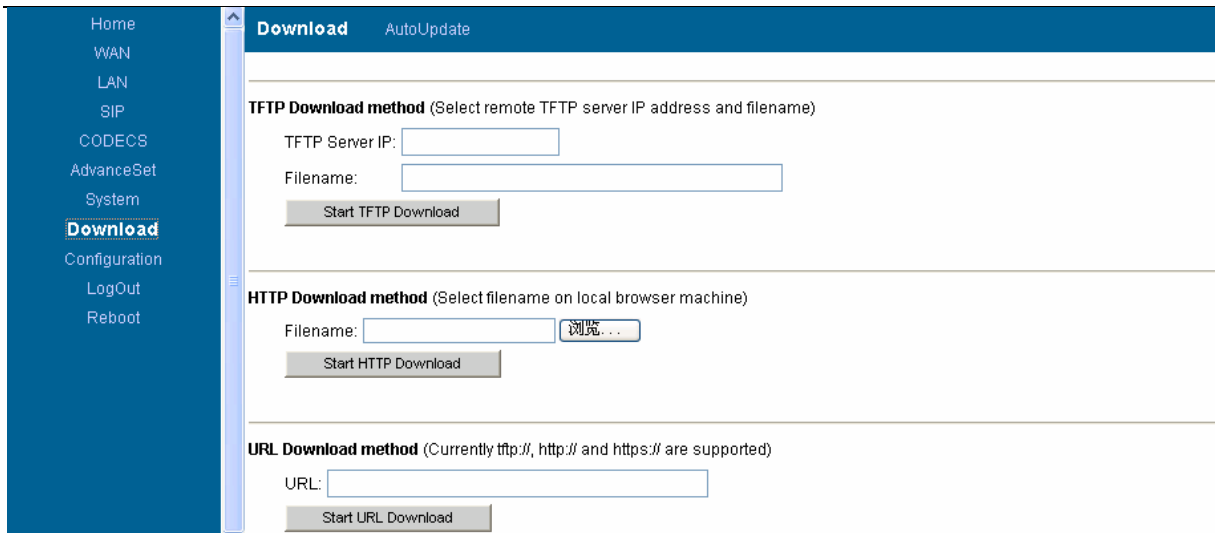
This page provides two options for downloading a new firmware application image to the device. If you wish to download the new firmware image using TFTP, enter the filename of the ROM image and enter the IP address of the TFTP server on which this file resides.

To initiate the TFTP download process, press "Start TFTP Download."

If the ROM image is stored on the same local machine you are using to access the device's web pages, you can choose to download the ROM file to the device using an HTTP post. Enter the filename of the ROM image or press "Browse" to help locate the file.

To initiate the HTTP download process, press "Start HTTP Download."

If the main application is executing at the time, the device will automatically reboot itself into the downloader mode and begin the download process. If the downloader application is executing at the time, the download process will begin. The download status will be displayed when the image download process is complete. Please refer to Section A "The Downloader Application" for more details on the download process.



HTTP Download method:

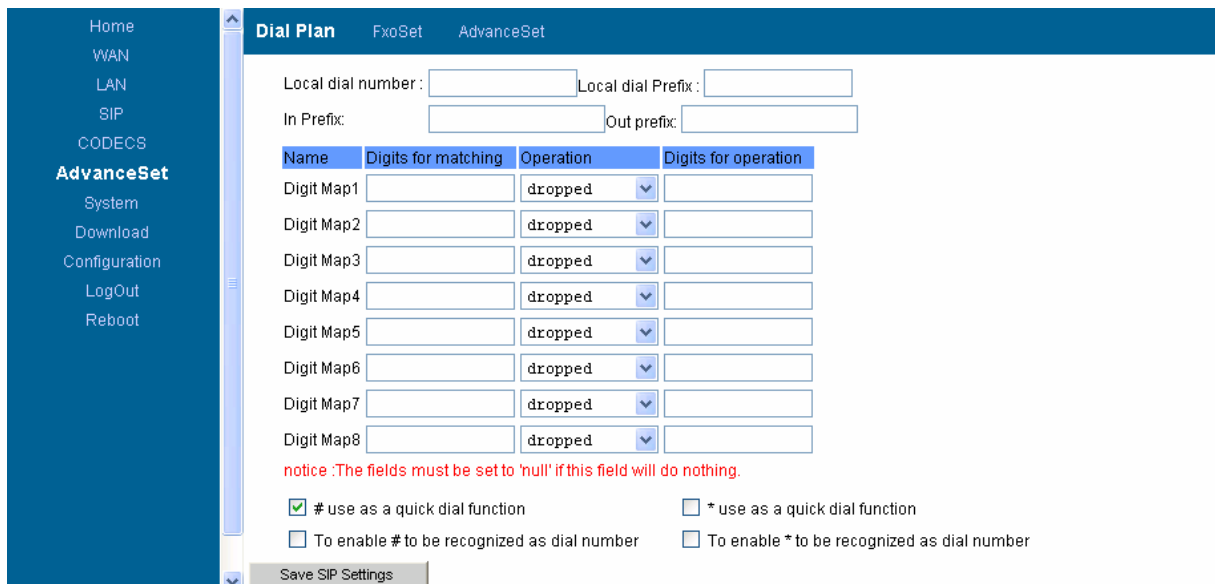
When using http to upgrade firmware, it will check firmware version before starting download process.

In Filename field, press Browsing Button.

Press Start HTTP Download button to start downloading file.

If firmware version doesn't fit in with old version, it won't allow updating.

9.8 Advance Setting



- Dial Plan such as 12.1.1
- Digit Map 1: For the No prefixed with 13, 013 will substitute.

eg., when call 1368644111, the send no will be 01368644111

- Digit Map 2: For the No prefixed with 8, 07558 will substitute

But due to the setting of Digit Map 1 & Digit Map 2, this rule

exclude the No prefixed with 86 & 88.

eg., when call 81971911, the send no will be 075581971911

- Digit Map 3: For the No prefixed with 86, 86 will be dropped.

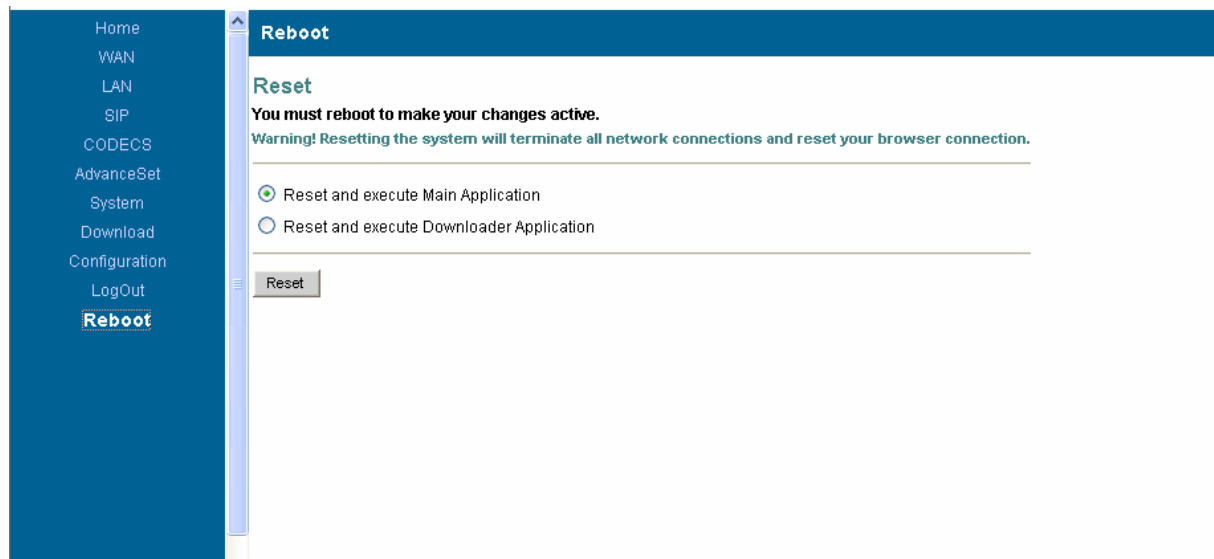
eg., when call 8607552647xxxx, the send no will be 07552647xxxx

- Digit Map 4: For the No prefixed with 88, no change

eg., when call 88011500, the send no will be 88011500

1. Radius billing Setting: Start billing software, Enable “Use Radius server Ip” check button, enter the IP address of target computer
2. Use FXS Hotline Number: Enable “Use FXS Hotline Number” check button, when you pick up telephone receiver, the HotLine No will be send automatically immediately.
3. PSTN Dialout Number: Lifeline setting
4. Encrypt setting: association server setting
5. Use Time Limit: Setting the expire date of account No
6. Polarity billing setting: Enable “Use Polarity billing” check button will Start Polarity billing function

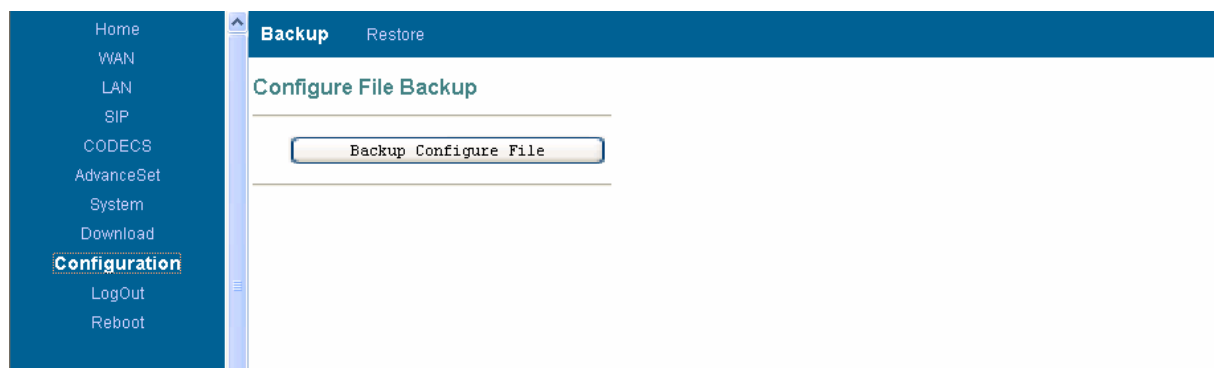
9.9 Reset Setting



This page provides options for resetting the device. Select whether you wish to reset the device and start executing the main (default) application, or whether you wish to reset the device and start executing the internal downloader application.

Press "Reset" to reset the device

9.10 Config Backup



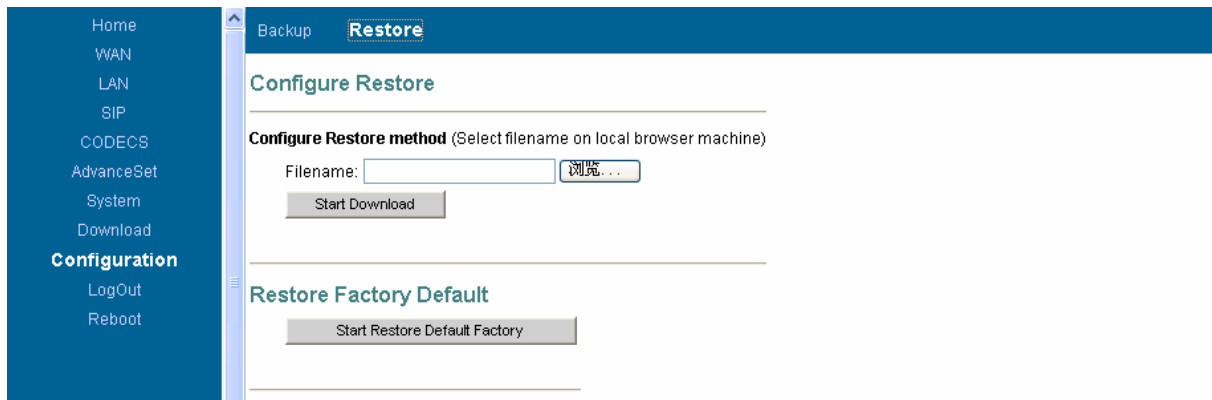
Backup configuration values of system settings to a file from the device.

Click "Configuration" item on the top menu.

Click "Backup" item on the left menu.

Press Backup Configure File button to save configuration file.

9.11 Config Restore



This page is used to restore configuration values of system settings from a previously saved configuration file, or default factory values that stored inside the device.

Restore configuration from a file.

Click "Configuration" item on the top menu.

Click "Restore" item on the left menu.

Press Browsing button to select file by backup from local machine.

Press Start Download button to process downloading file.

After downloading file is finished, the web system will redirect to restart device.

Restore default factory values form device.

Click "Configuration" item on the top menu.

Click "Restore" item on the left menu.

Press "Start Restore Default Factory" button.

After restoring default factory, the web system will redirect to restart device.

10 Dial Model

10. 1 Direct Dial:

Dial through VoIP No: Directly pick up telephone receiver connected with FXS port, and then dial telephone No

11 Troubleshooting

1. Confirm all cables are connected properly.
2. Check whether there is the connection through Ping action of PC
3. Connect the gateway to the LAN interface and check whether the gateway is set properly.
4. If the fault cannot be solved yet, please contact the technicians.

12 Dial Plan And IVR

12.1.1 Dial Plans

The H.323 and SIP code will allow provisioning (via web browser) of the dial plan. A dial plan gives the unit a map to determine when a complete number has been entered and should be passed to the gatekeeper for resolution into an IP address. Dial plans are expressed using the same syntax as used by MGCP NCS specification.

The formal syntax of the dial plan is described by the following notation:

Digit ::= "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" | "8" | "9"

Timer ::= "T" | "t"

Letter ::= Digit | Timer | "#" | "*" | "A" | "a" | "B" | "b" | "C" | "c"

| "D" | "d"

Range ::= "X" | "x" -- matches any digit

| "[" Letters "]" -- matches any of the specified letters

Letters ::= Subrange | Subrange Letters

Subrange ::= Letter -- matches the specified letter

| Digit "-" Digit -- matches any digit between first and last

Position ::= Letter | Range

StringElement ::= Position -- matches any occurrence of the position

| Position "." -- matches an arbitrary number of occurrences including 0

String ::= StringElement | StringElement String

StringList ::= String | String "|" StringList

DialPlan ::= String | "(" StringList ")"

A dial plan, according to this syntax, is defined either by a (case insensitive) string or by a list of strings. Regardless of the above syntax a timer is only allowed if it appears in the last position in a string (12T3 is not valid). Each string is an alternate numbering scheme. The unit will process the dial plan by comparing the current dial string against the dial plan, if the result is under-qualified (partial matches at least one entry) then it will do nothing further. If the result matches or is over-qualified (no further digits could possibly produce a match) then send the string to the gatekeeper and clear the dial string.

The Timer T is activated when it is all that is required to produce a match. The period of timer T is 4 seconds. For example a dial plan of (xxxT|xxxxx) will match immediately if 5 digits are entered, it will also match after a 4 second pause when 3 digits are entered.

D.1 Sample Dial Plans

Simple Dial Plan

This allows dialing of 7 digit numbers (e.g. 5551234) or an operator on 0. Dial plan is (0T|xxxxxxx)

Non-dialed Line Dial Plan

As soon as handset is lifted the unit contacts the gatekeeper (used for systems where DTMF detection is done in-call). Dial plan is (x.) i.e. match against 0 (or more) digits. Note: the dot '.'

Complex Dial Plan

Local operator on 0, long distance operator on 00, four digit local extension number starting with 3,4 or 5, seven digit local numbers are prefixed by an 8, two digit star services (e.g. 69), ten digit long distance prefixed by 91, and international numbers starting with 9011+variable number of digits.

Dial plan for this is:

(0T|00T|[3-5]xxx|8xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)

12.1.2 IVR

To use IVR, one should pick up the phone, and then dial four consecutive asterisks (****.) And hang up the phone will stop IVR.

Code	Status	User input
****	Menu	Enter choice code.
100#	Network status	None.
110#	DHCP setting	1# to enable 2# to disable or # back to menu
120#	IP address setting	Use "*" to instant of ".", and "#" to end. Ex: 172*16*230*227# Or # back to menu.
130#	Gateway setting	Use "*" to instant of ".", and "#" to end. Or # back to menu.
140#	Net mask setting	Use "*" to instant of ".", and "#" to end. Or # back to menu.