PL-300A 1FXS ATA

User's Manual

<Version: V2.0 >

Contents

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 ATA can be directly connected to the IP network, to provide basic accesses for POTS users. This user port gateway can support four telephone lines. It is also applicable for small-size enterprises and IP telephone bars.

2 Packing

The ATA 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 ATA

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
- 1 analog loops starts the FXS interface (RJ-11), used to connect 1 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 remote configuration of Web mode and remote software downloading/upgrading

5 Performance Indices

Description of Product Model	
ATA-100	1FXS port IP voice gateway, SIP protocol
Physical Specifications	
Size	90mm (L) × 65mm (W) × 25mm (H)
Power supply	AC/DC power adapter, 12V DC DC input: 12V DC/0.8A
Power consumption	< 10W
Weight	About 0.1 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 SIP protocol	
Mute processing/four wave processing	
RTP/RTCP voice channel	
Voice compression algorithm G.729, G.711, G.723 and G.726	
Analog voice port (FXS), 1 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

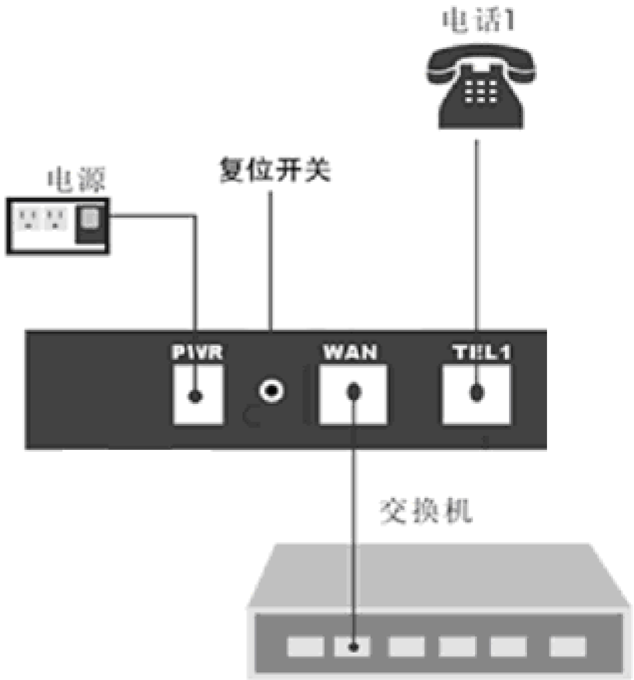
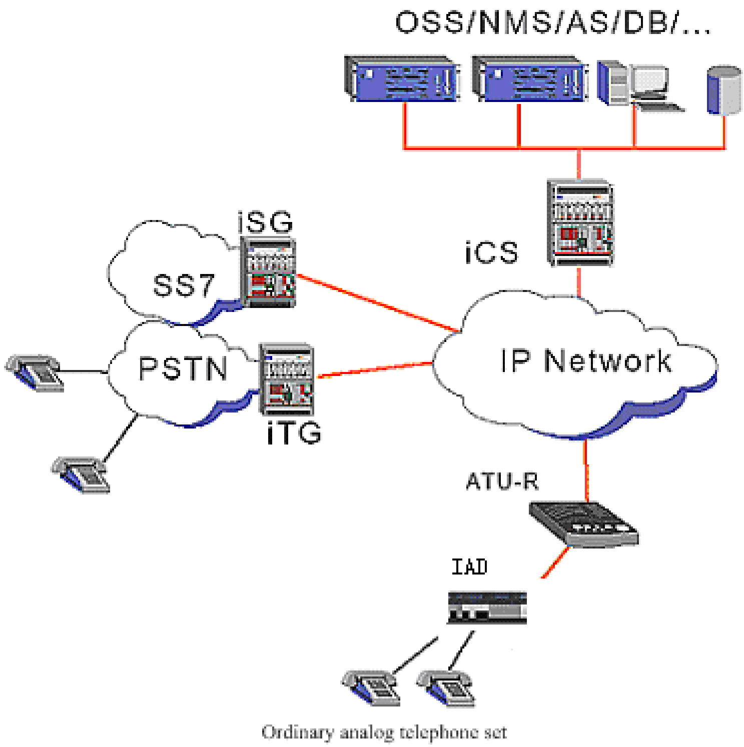


Fig. 1

6.1 Typical Applications

The IAD user gateway integrates the Internet gateway and VoIP gateway into a box. They can use category-5 cables to connect the ISP switch (as shown in Fig. A). Four FXS can be connected to Four ordinary analog telephone sets to provide conversation based on IP network.



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

Fig .A

7 Appearance Description

WAN ----RJ45 PHONE----RJ11 Power-----12V 1A Power Adapter Reset --- button

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 WAN interface (refer to 8.1).

8.1 WAN Configuration Environment

- Configure the "TCP/IP Protocol" of PC according to Fig. 1 with the PC and WAN interface in the same network segment.
- Configure the device according to Fig. 2.
- 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.

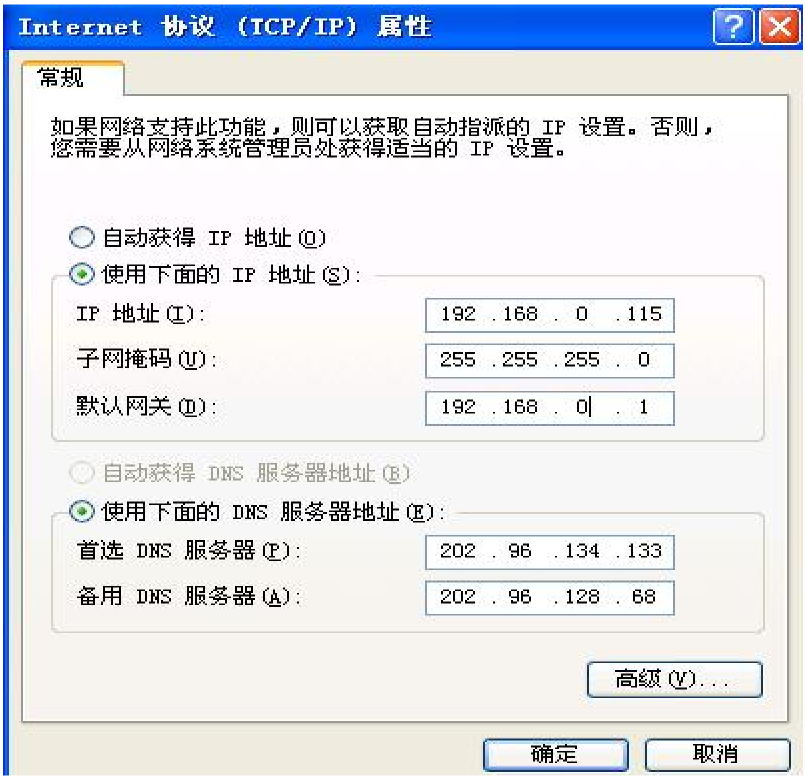


Fig. 1

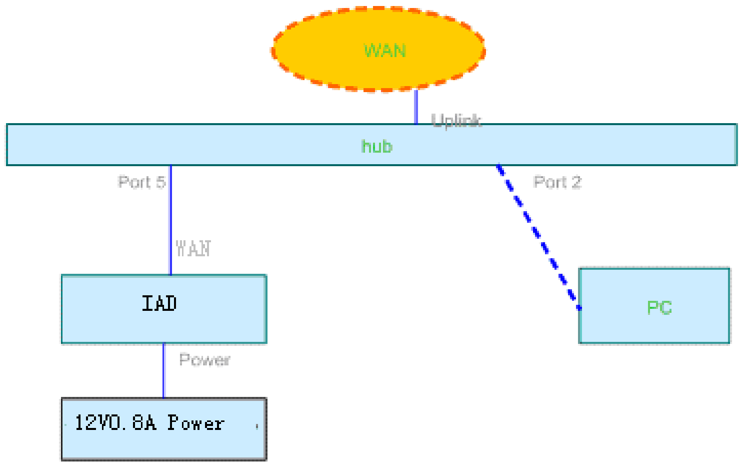


Fig. 2

9 Configuration in WEB Mode

9.1 LOGIN WEB

Through IAD’s Wan 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 3:

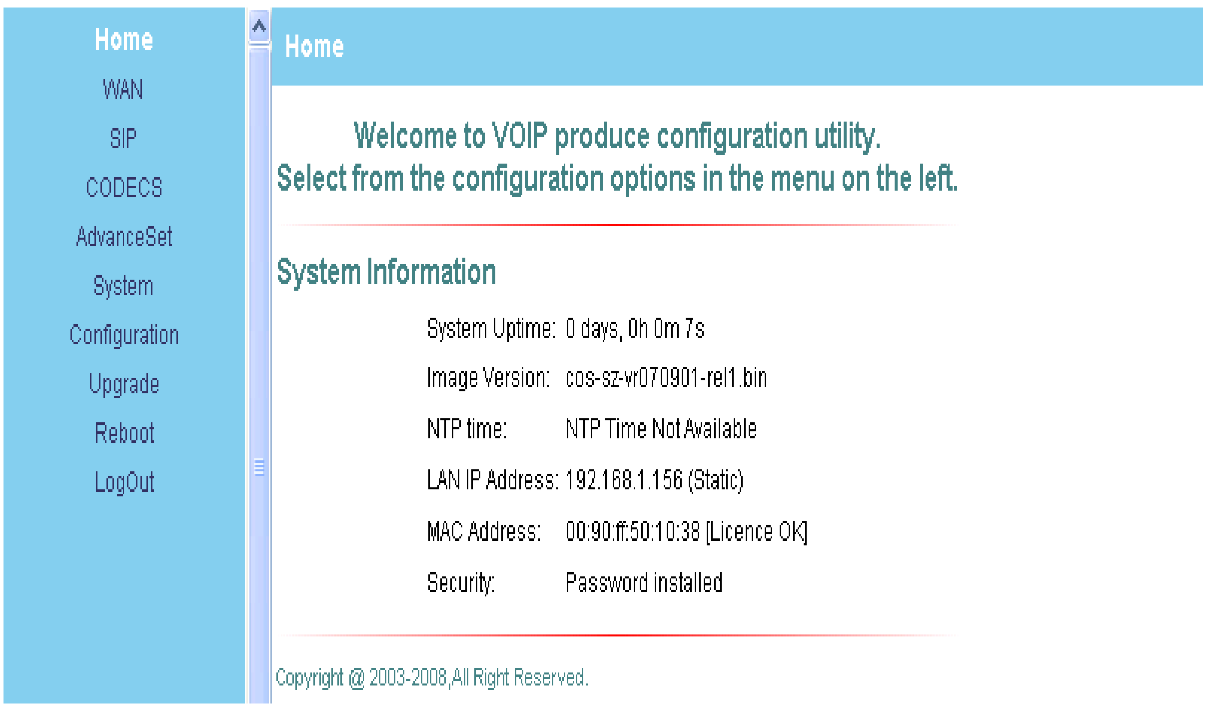


Fig.3

When login as user,will see like figure 4:

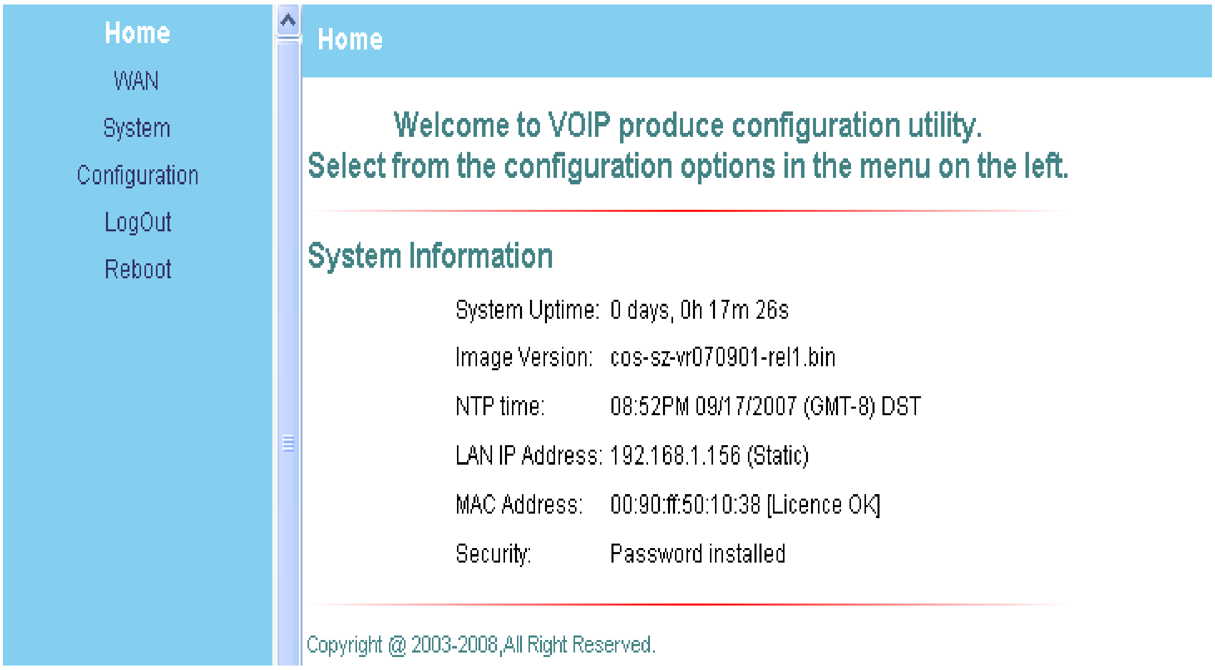
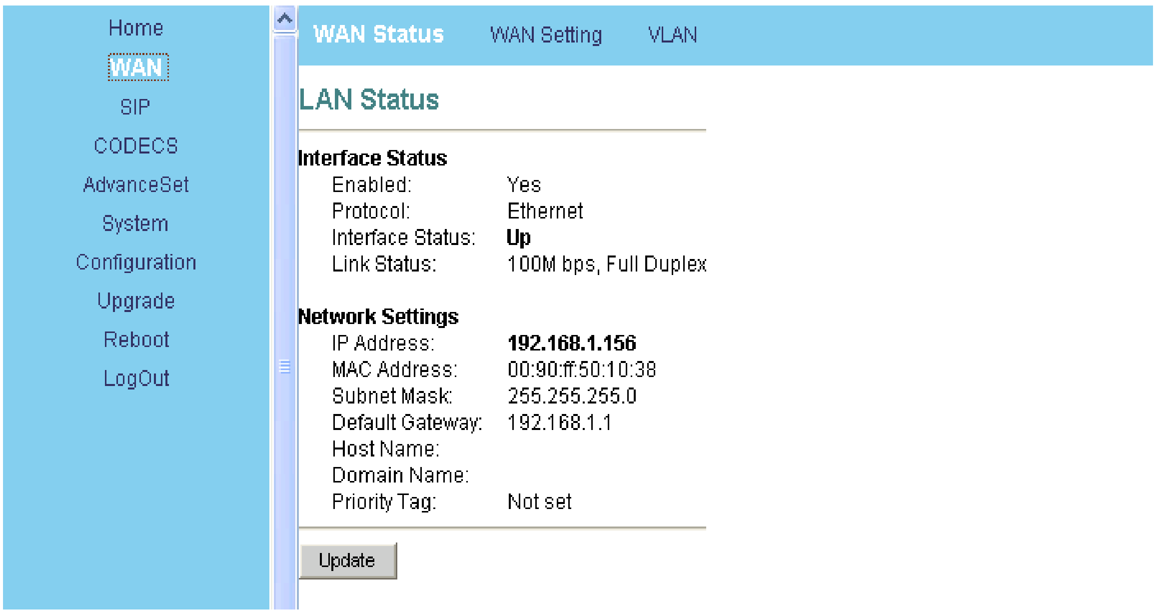
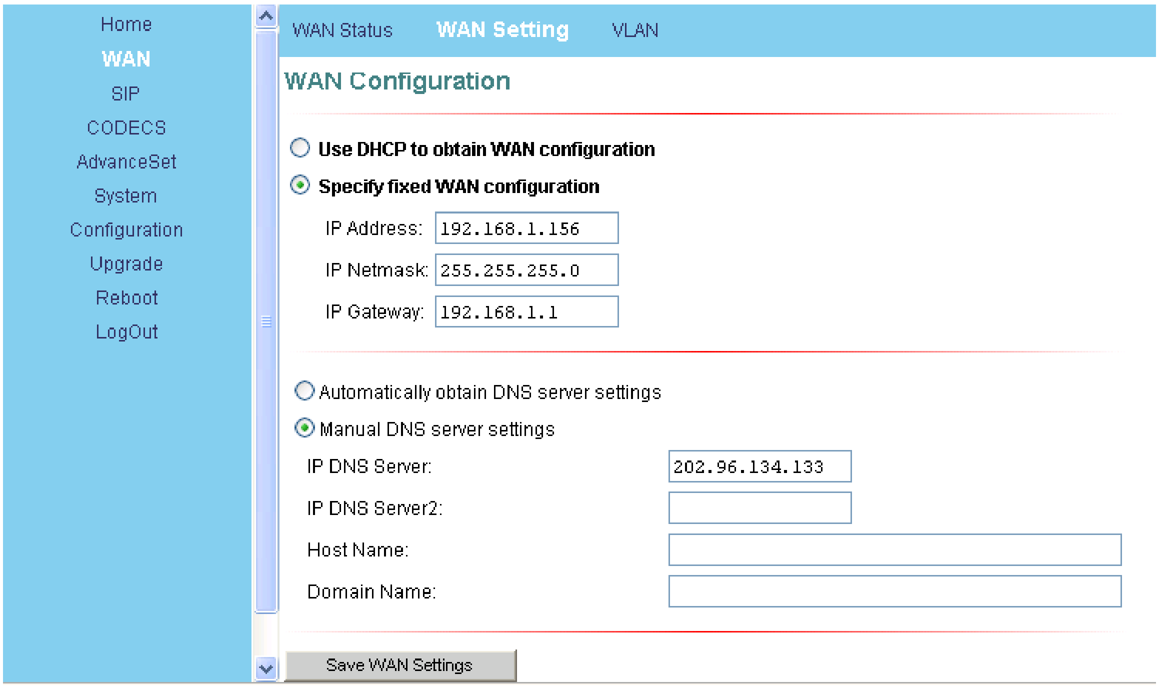


Fig. 4

9.2 WAN Configuration

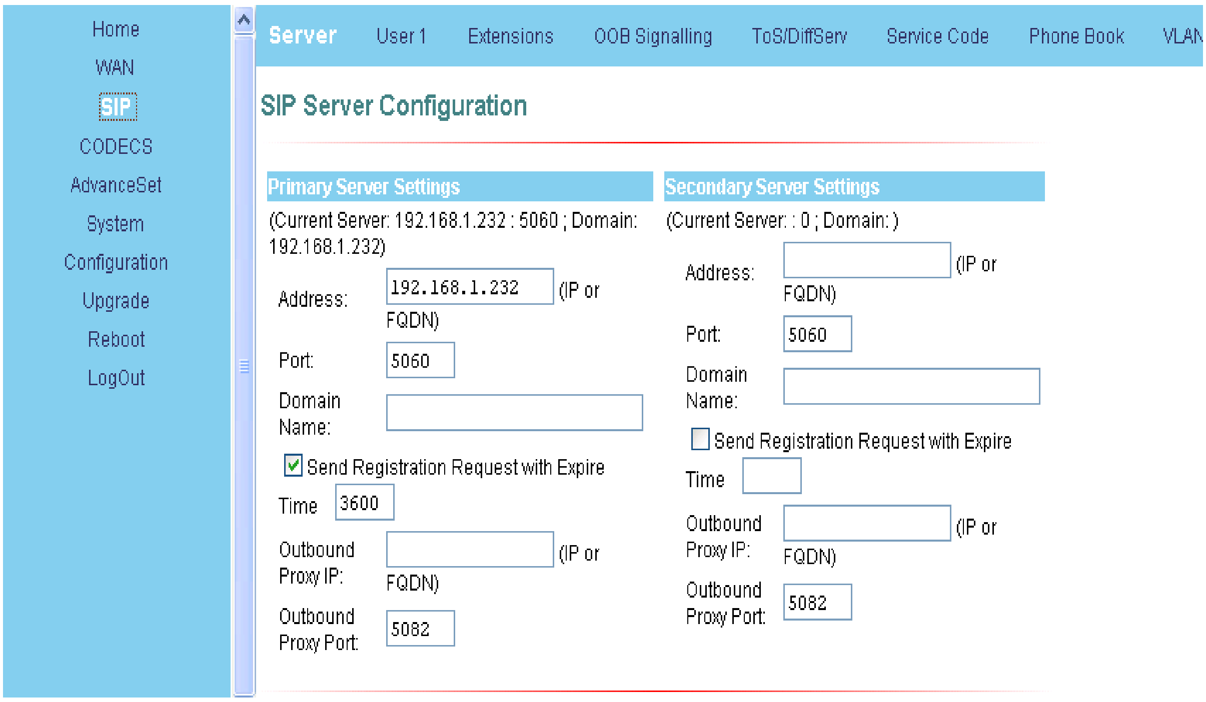


This page is the first page displayed when the device's web pages are accessed. It shows how long the device has been running since its last reboot, the IP address the device is currently using, whether or not the device is password protected, and also displays the main application and downloader application 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 192.168.0.200.Please note that any actions/modifications which alter the topology of the Ethernet Bridge will result in the spanning tree protocol to relearn.

9.3 SIP Configuration



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 self 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 WAN settings page, section 9.2).

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

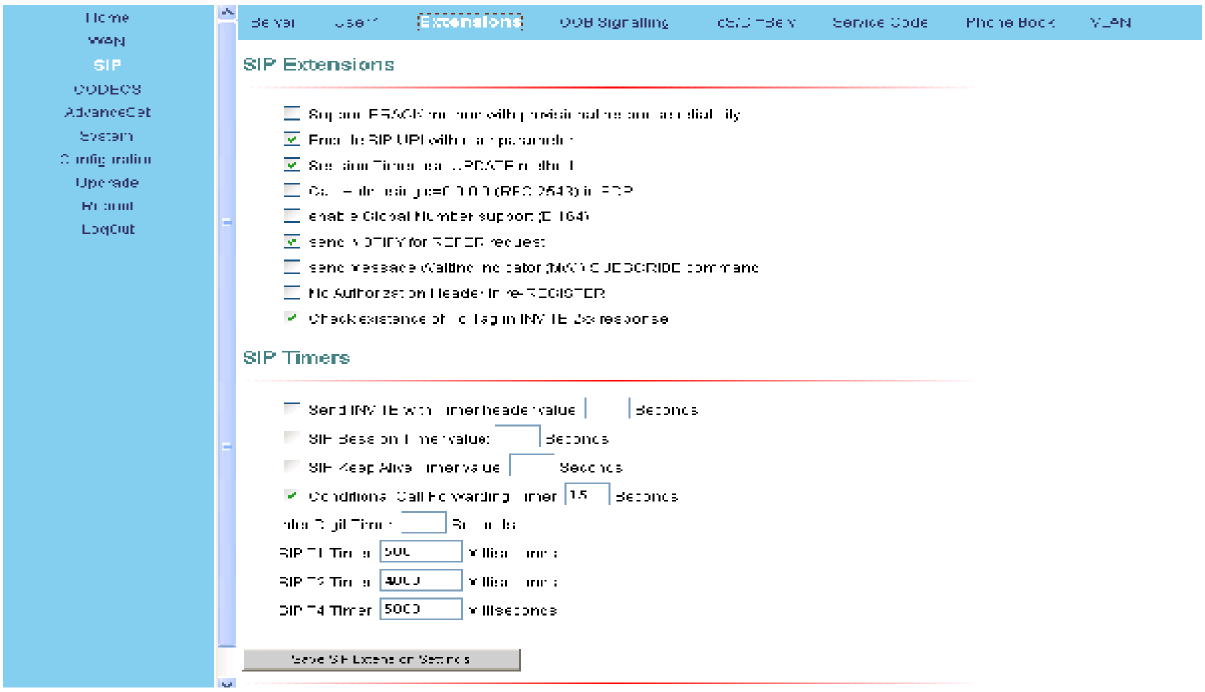
Select whether or not to send a Registration Request to the SIP server by checking the box next to "Send Registration Request".

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.

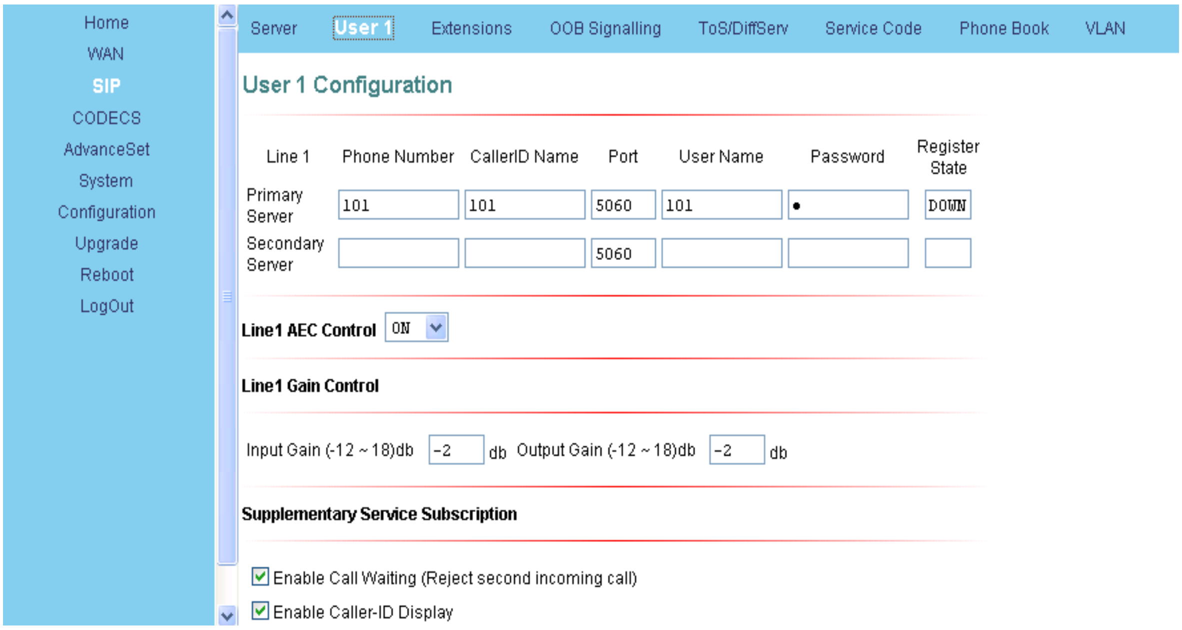
If you wish for the SIP stack to implement reliable transmission of provisional responses according to RFC 3262 (using the PRACK method), check the option "Support PRACK method with provisional response reliability".

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".

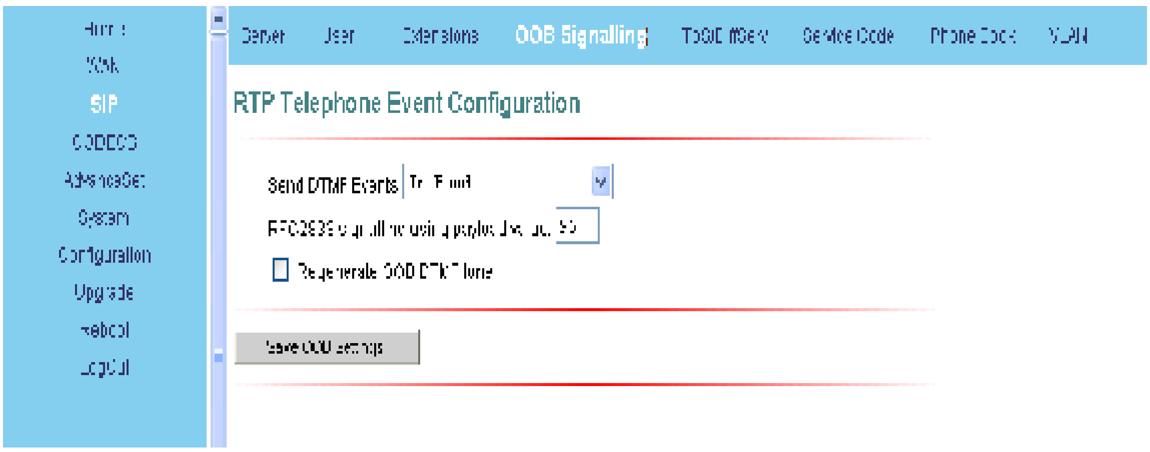
If you wish for the SIP stack to send INVITE messages with the "Timer" header field present, check the option "Send INVITE with Timer header value" and enter the Timer header value.

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.

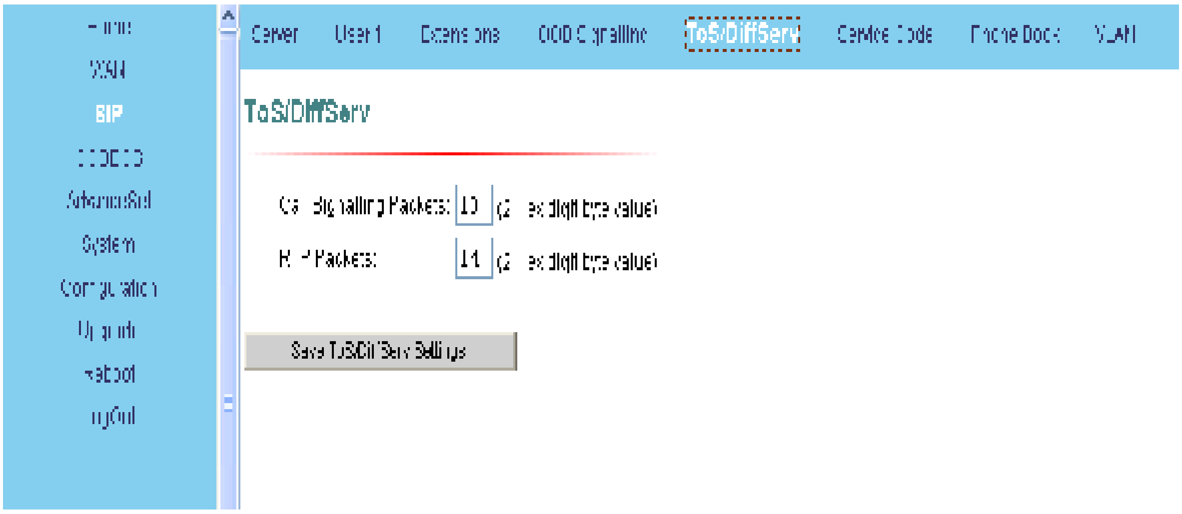
Press "Save SIP Extension Settings" to save the new values.



This sub-page and the next User2,User3,User4 sub-page allow the user to configure the device with phone number, caller ID, username and password specified by the service provider.

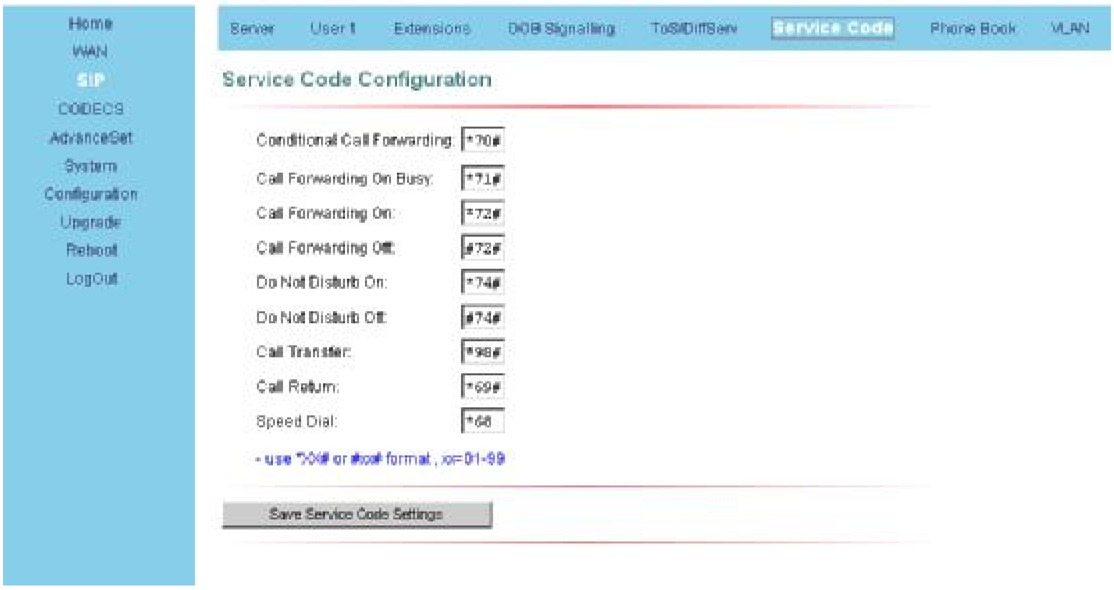


This sub-page allows configuration of the out-of-band signaling options for SIP. Select whether OOB telephone event signaling is to be done using the SIP INFO message, or to be done via RFC2833 RTP signaling



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.



You need to set the service code for using value add service. For example, I set the service code as the above picture。

Condition Call Forwarding: (the call will transfer if no one answer)

a) Set forwarding number: pick up the handset(press *70# (then you will hear the dial tone(press the forwarding number(then you will here three beeps indicating setting finish。
b) Set the timeout: go to the "sip extensions(Conditional call Forwarding timer" and

set the timeout before forwarding， unit: second, and then active this option.

c) Then the call will automatically transfer to the forwarding number if no one answers the call in the timeout.

Call Forwarding on busy

Enable call forwarding on busy: pick up the handset(press *71# (then you will hear the dial tone(press the forwarding number(then you will here three beeps indicating setting finish, then all incoming call will forward to this number automatically when the Phone is busy

Call Forwarding: (forwarding always)

a) Enable call forwarding: pick up the handset **à** press *72# **à** then you will hear the dial tone **à** press the forwarding number **à** then you will here three beeps indicating setting finish, then all incoming call will forward to this number automatically。
b) Disable call forwarding: pick up the handset **à** press #72# **à** then you will here three beeps indicating setting finish

Do not disturb: (DND)

a) Enable DND: pick up the handset **à** press *74# (then you will here three beeps indicating setting finish(then the phone won't ringing when there is an incoming call。
b) Disable DND: pick up the handset (press #74# (then you will here three beeps indicating setting finish

Call transfer:

a) Unattended transfer: A call B(B press *98# and then enter C number(then B will hear three beeps indicating the transfer successfully。

b) Attended transfer： A calls B(B push the hook flash to hold A(B then dial C number to talk with C (then B press *98# to transfer the call(then A can talk with C。

Call Return:

Pick up the handset (and then press *69# to dial the latest received call

3 way conference call:

A calls and talks with B (B push the hook flash to hold A B then dial C number to talk with C (B then push the hook flash again to enable three way conference call(C will leave the call is B push the hook-flash again.

Tone

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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.

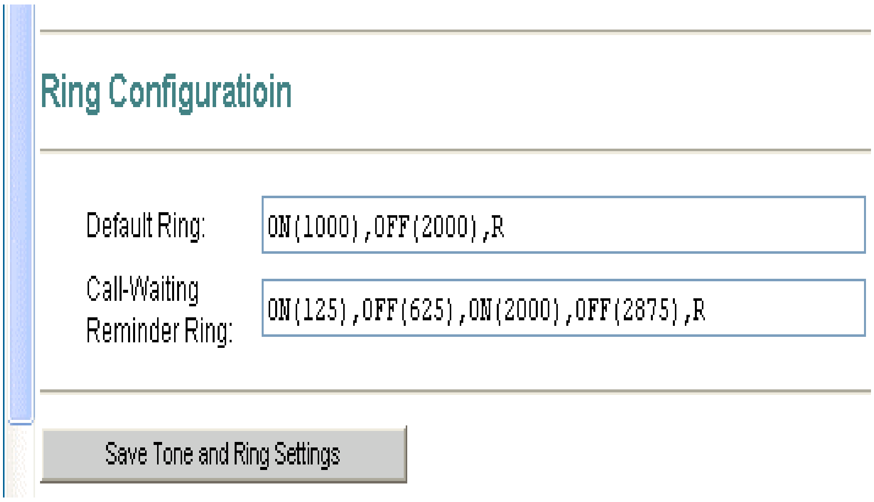
Reorder Tone: The tone you hear if you dial an invalid number or the call is not available.

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



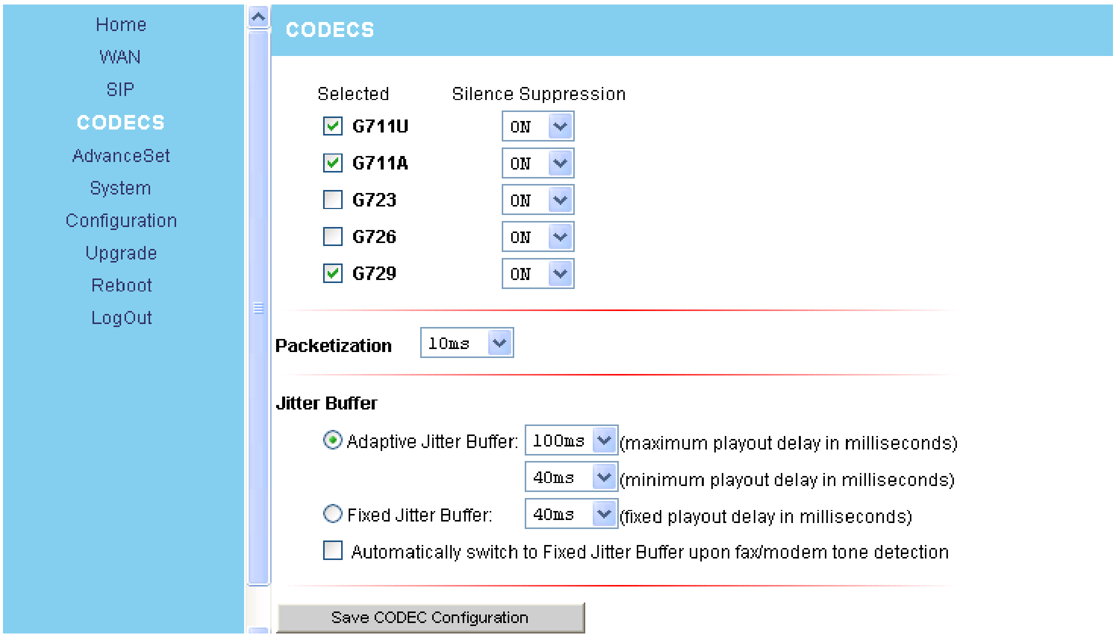
This sub-page is used to configure Ring Cadences required by Rings, Call-Waiting-Indicator, and Distinctive Ring features.

1. Ring Configuration:

Default Ring: Default ring cadence when the phone rings.

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

9.4 CODECS Setting



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. For MGCP, a range of packetizations may be provided for each CODEC (to be advertised in the device's "capabilities" set).

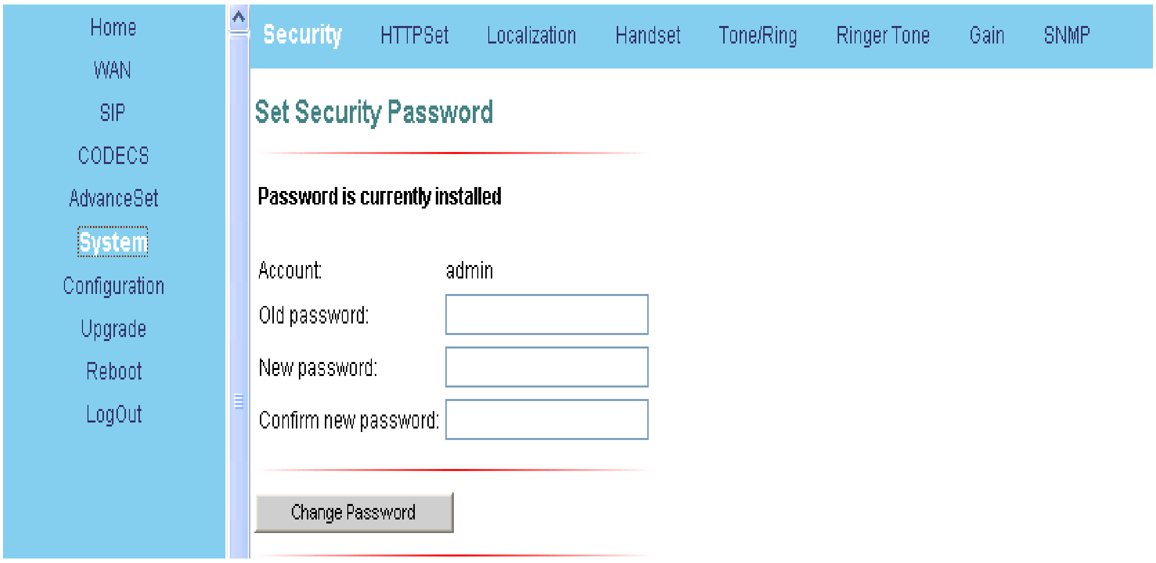
Select whether Silence Suppression is to be supported for each CODEC.

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.

9.5 System Setting



1. Set Security Password:

Click "System" item on the top menu.

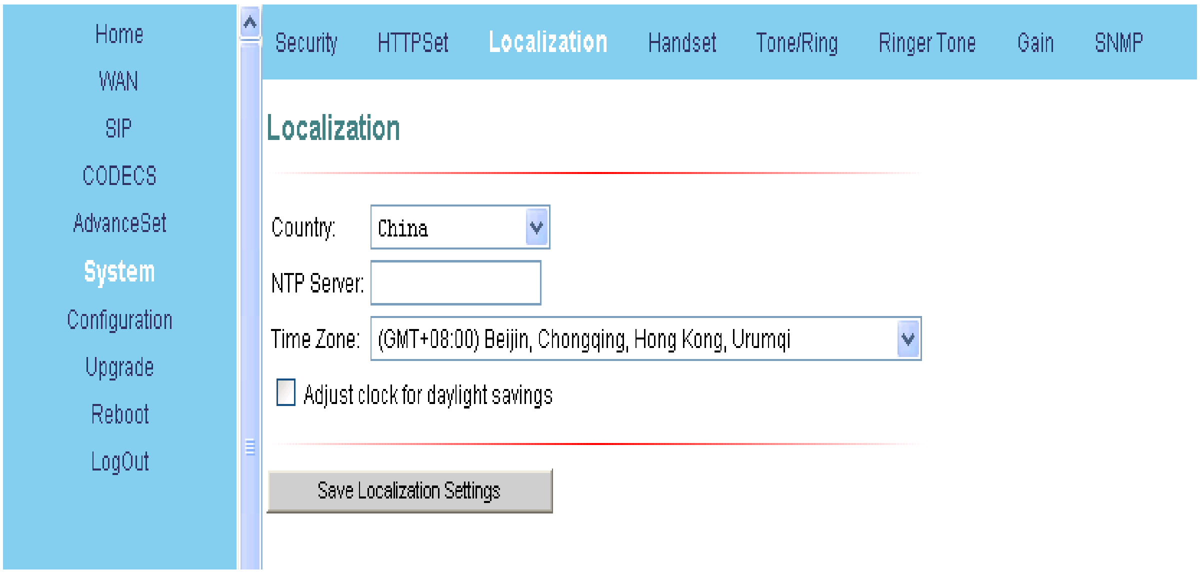
Click "Security" on the left menu.

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.

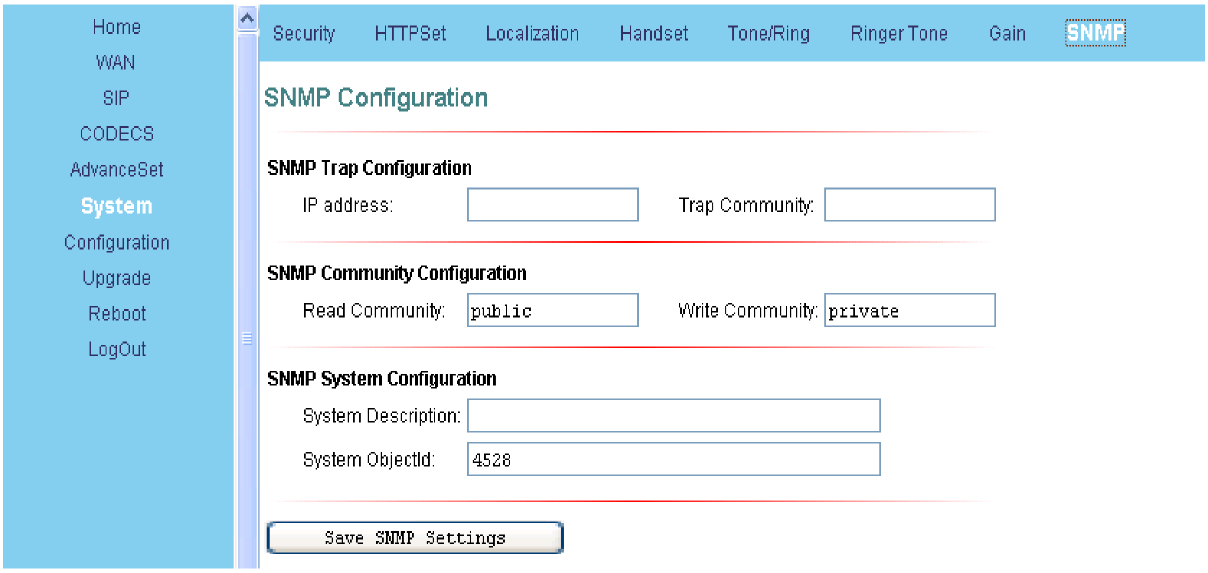
Click "Localization" on the left menu.

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.

9.6 Download Setting

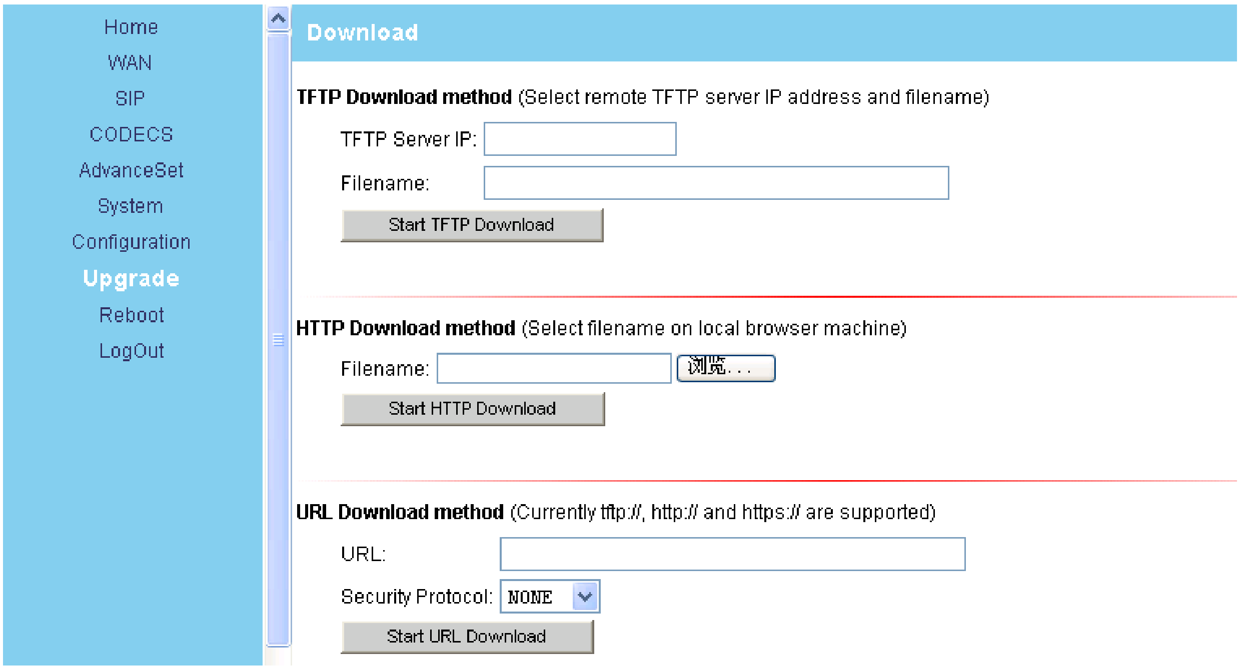
This page provides two options for downloading a new firmware application image to the device. If you wish to download the new firmware image using TFTP, enter the filename of the ROM image and enter the IP address of the TFTP server on which this file resides.

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:

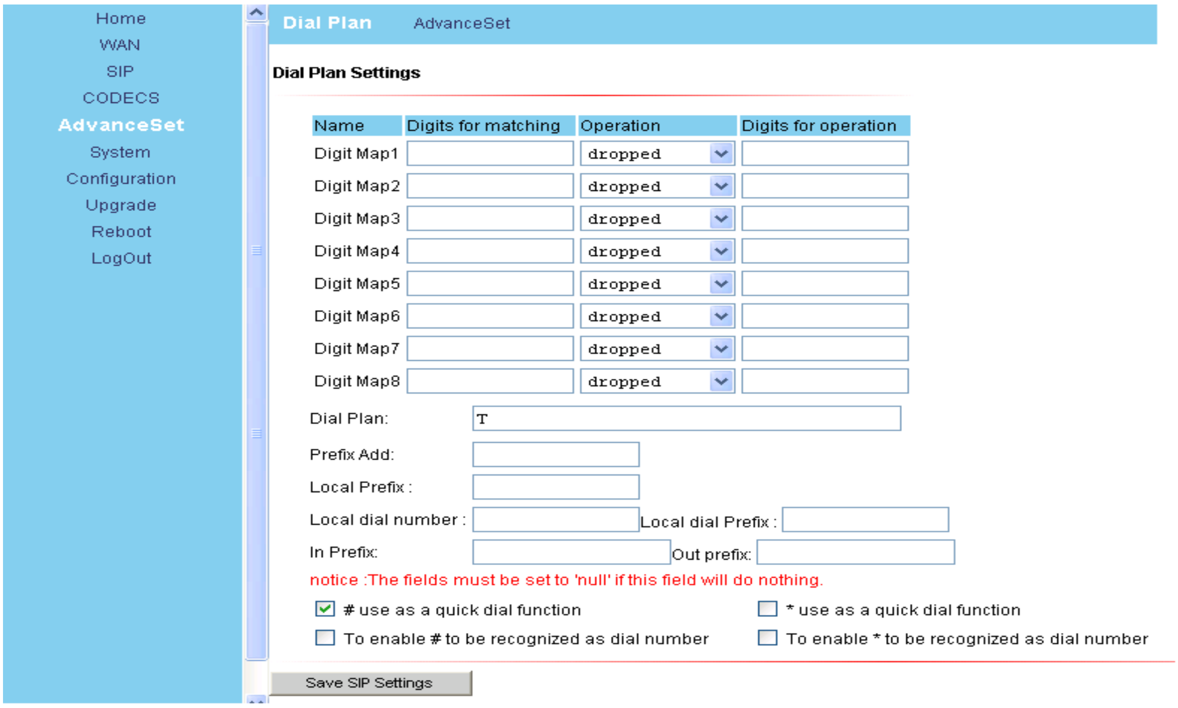
When using http to upgrade firmware, it will check firmware version before starting download process.

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.

9.7 Advance Setting



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

  eg., when call 1368611111, the send no will be 01368611111
- Digit Map 2: For the No prefixed with 8, 07558 will substitute

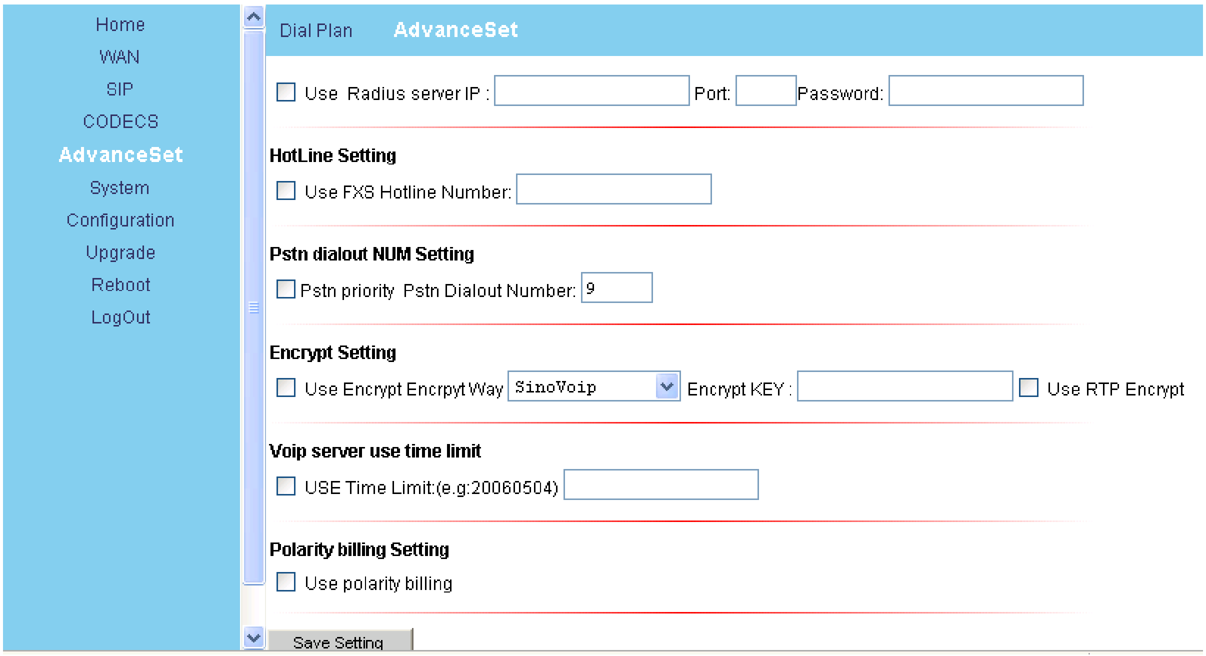
  But due to the setting of Digit Map 1 & Digit Map 2， this rule

  exclude the No prefixed with 86 & 88.

  eg., when call 81971111, the send no will be 075581971111
- Digit Map 3: For the No prefixed with 86, 86 will be dropped.

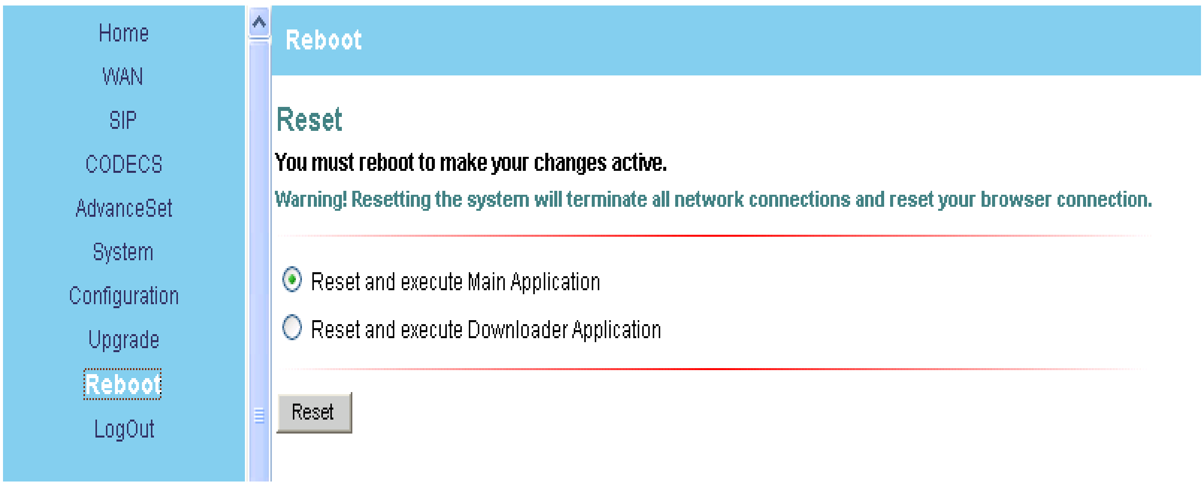
  eg., when call 8607552647xxxx, the send no will be 07552647xxxx
- Digit Map 4: For the No prefixed with 88, no change

  eg., when call 88011000, the send no will be 88011000



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: Lifelive setting
4. Encrytp 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

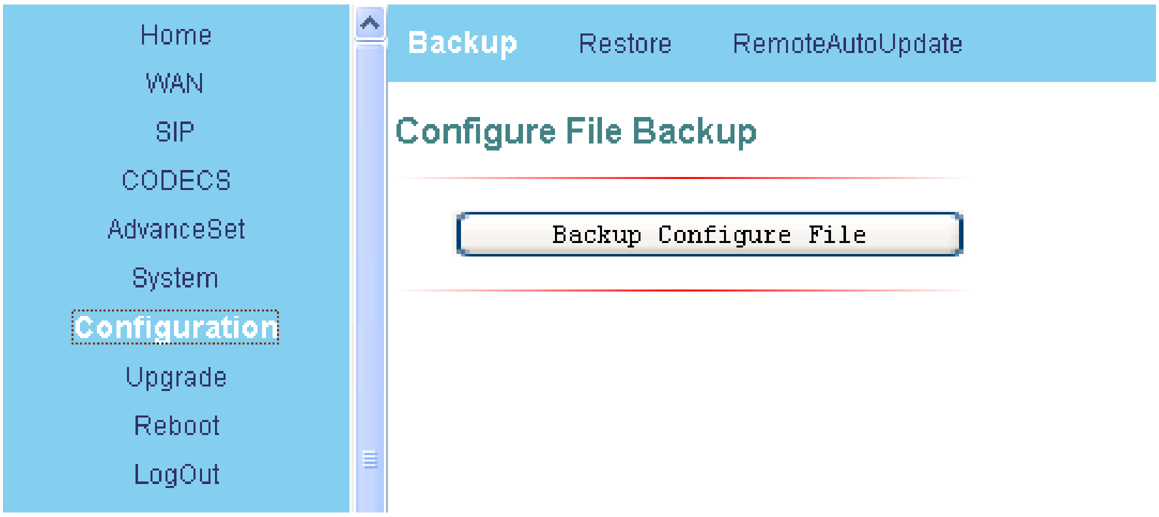
9.8 Reboot Setting



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

9.9 Config Backup



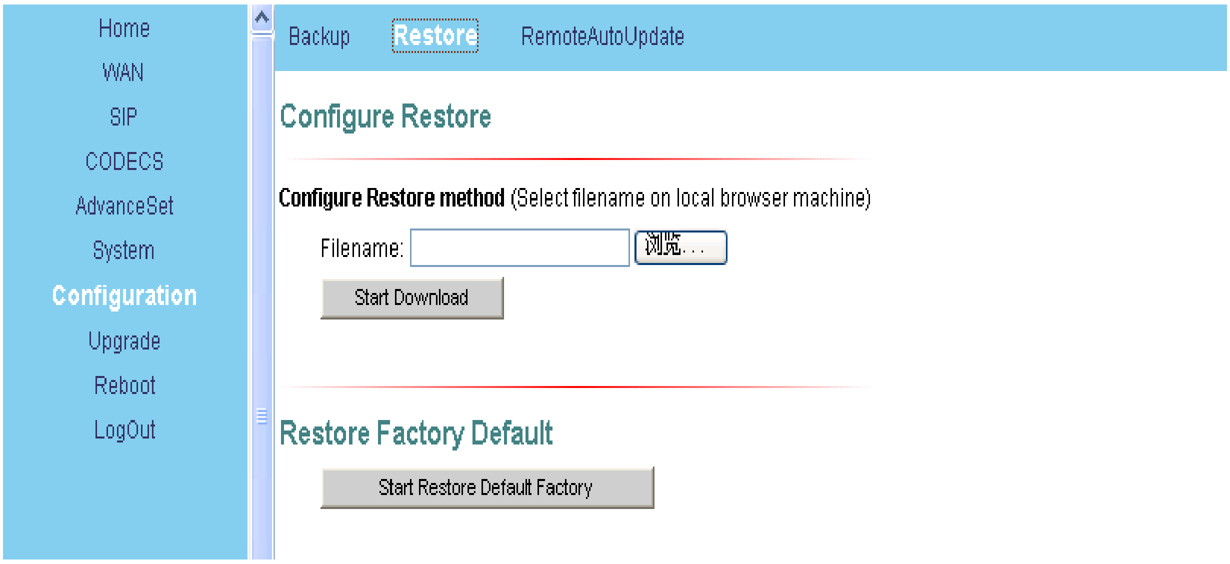
Backup configuration values of system settings to a file from the device.

Click “Configuration” item on the top menu.

Click “Backup” item on the left menu.

Press Backup Configure File button to save configuration file.

9.10 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. If the fault cannot be solved yet, please contact the technicians.

12 Dial Plan And IVR

12.1.1 Dial Plans

The 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.