User Manual

SIP 60X

Analog IP Gateway

4 or 8 FXS Ports

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1 WELCOME

Thank you for purchasing the SIP 60X Analog FXS IP Gateway. The SIP 60X offers an easy to manage, easy to configure IP communications solution for any business with virtual and/or branch locations. The SIP 60X supports popular voice codec's and is designed for full SIP compatibility and interoperability with 3rd party SIP providers, thus enabling you to fully leverage the benefits of VoIP technology, integrate a traditional phone system into a VoIP network, and efficiently manage communication costs.

This manual will help you learn how to operate and manage your SIP 60X FXS Analog IP Gateway and make the best use of its many upgraded features including simple and quick installation, multi-party conferencing, This IP Analog Gateway is very easy to manage and scalable, specifically designed to be an easy to use and affordable VoIP solution for the small – medium business or enterprise.

1.1 Gateway SIP 60X Overview

The SIP 60X series has a compact and quiet design (no fans) and offers superb audio quality, rich feature functionality, strong security protection, and good manageability. It is auto-configurable, remotely manageable and scalable.

The SIP 60X features 4 or 8-port FXS interface for analog telephones, dual 10M/100Mbps network ports with integrated router, PSTN life line in case of power failure,. In addition, it supports the option of 2 SIP Server profiles, caller ID for various countries/regions, T.38 fax, flexible dialing plans, security protection (SIPS/TLS), comprehensive voice codec's including G.711 (a/u-law), G.723.1, G.726(16/24/32/48 bit rates), G.729A/B/E.

Caution: Changes or modifications to this product not expressly approved by the manufacturer, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

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1.2 Safety Compliances

The SIP 60X is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. Warning: use only the power adapter included in the SIP 60X package. Using an alternative power adapter may permanently damage the unit.

1.3 Warranty

Netronix markets its products through reseller partner channels, end users should contact the company from whom you purchased the product for replacement or repair.

If you purchased the product directly from Netronix, contact your Netronix Sales and Service Representative for a RMA (Return Materials Authorization) number. Netronix reserves the right to remedy warranty policy without prior notification.

2 CONFIGURE YOUR SIP 60X

Connecting your SIP 60X is easy. Before you begin, please verify the contents of the SIP 60X package.

2.1 Equipment Packaging

Unpack and check all accessories. The SIP 60X package contains:

- One SIP 60X VoIP adapter
- One universal power supply
- One Ethernet cable

2.2 Connect the SIP 60X

Managing the SIP 60X gateway and connecting the unit to the VoIP network is very simple. Follow these four (4) steps to connect your SIP 60X gateway to the Internet and access the unit’s configuration pages.

1. Connect standard touch-tone analog phones to the FXS1-FXS8 ports.

2. Insert the Ethernet cable into the WAN port of SIP 60X and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)

3. Connect a PC to the LAN port of SIP 60X for initial configuration or if it is being used as a router.

4. Plug the power adapter into the SIP 60X and into a power outlet.

2.3 Figure 1: Diagram of SIP 60X Back Panel

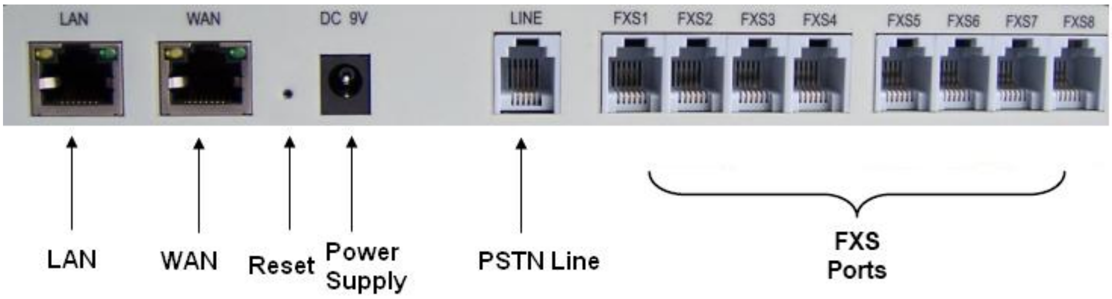


TABLE 1: Definitions of SIP 60X Connectors

LAN	Connect the LAN port with an Ethernet cable to your PC.
WAN	Connect to the internal LAN network or router.
PSTN Line	1 port
RESET	Factory Reset button. Press for 8 seconds to reset factory default settings.
DC 9V 2A	Power adapter connection
FXS1 - FXS8	FXS port to be connected to analog phones / fax machines

Once the SIP 60X is turned on and configured, the front display panel indicates the status of the unit.

2.4 Figure 2: Diagram of SIP 60X Display Panel

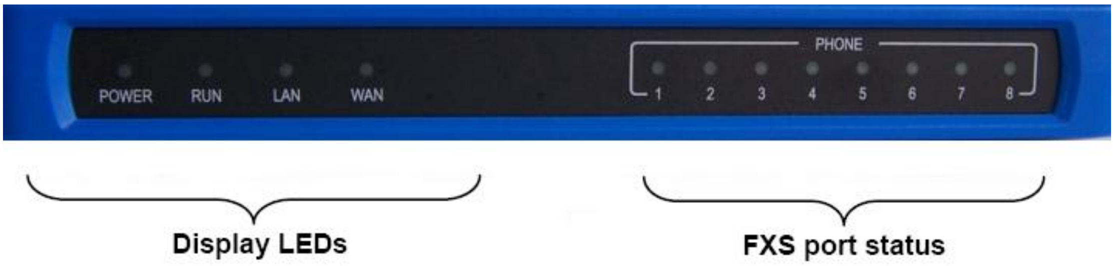


TABLE 2: Definitions of SIP 60X Display Panel

Power LED	Indicates Power. Remains ON when Power is connected

	and turned ON.
RUN LED	Blinking after boot-up.
LAN LED	Indicates LAN port activity
WAN LED	Indicates WAN port activity
LEDs 1 - 8	Indicate status of the respective FXS Ports on the back panel **Busy** - ON (Solid Green) **Available** - OFF Slow blinking FXS LEDs indicates Voice Mail for that port.

NOTE:

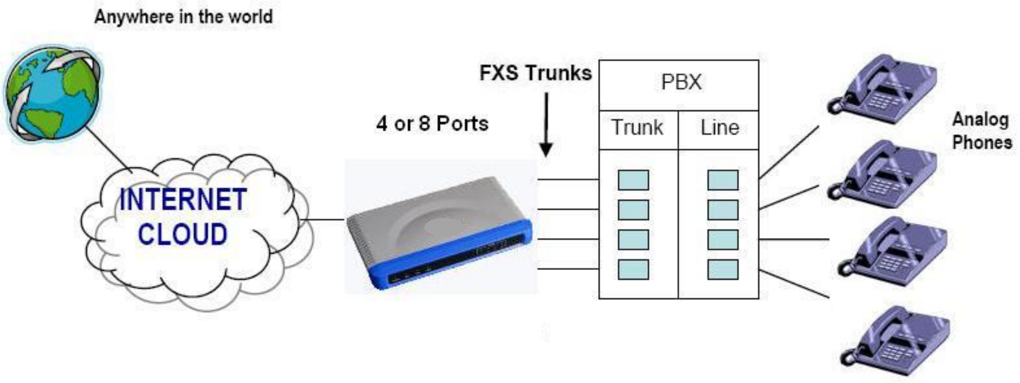
- Flash blinking of RUN, WAN LED together indicates a firmware upgrade or provisioning state.
- LEDs POWER, and WAN are ON and READY blinking when device is up and running and successfully registered to the SIP Server.

3 APPLICATION DESCRIPTION

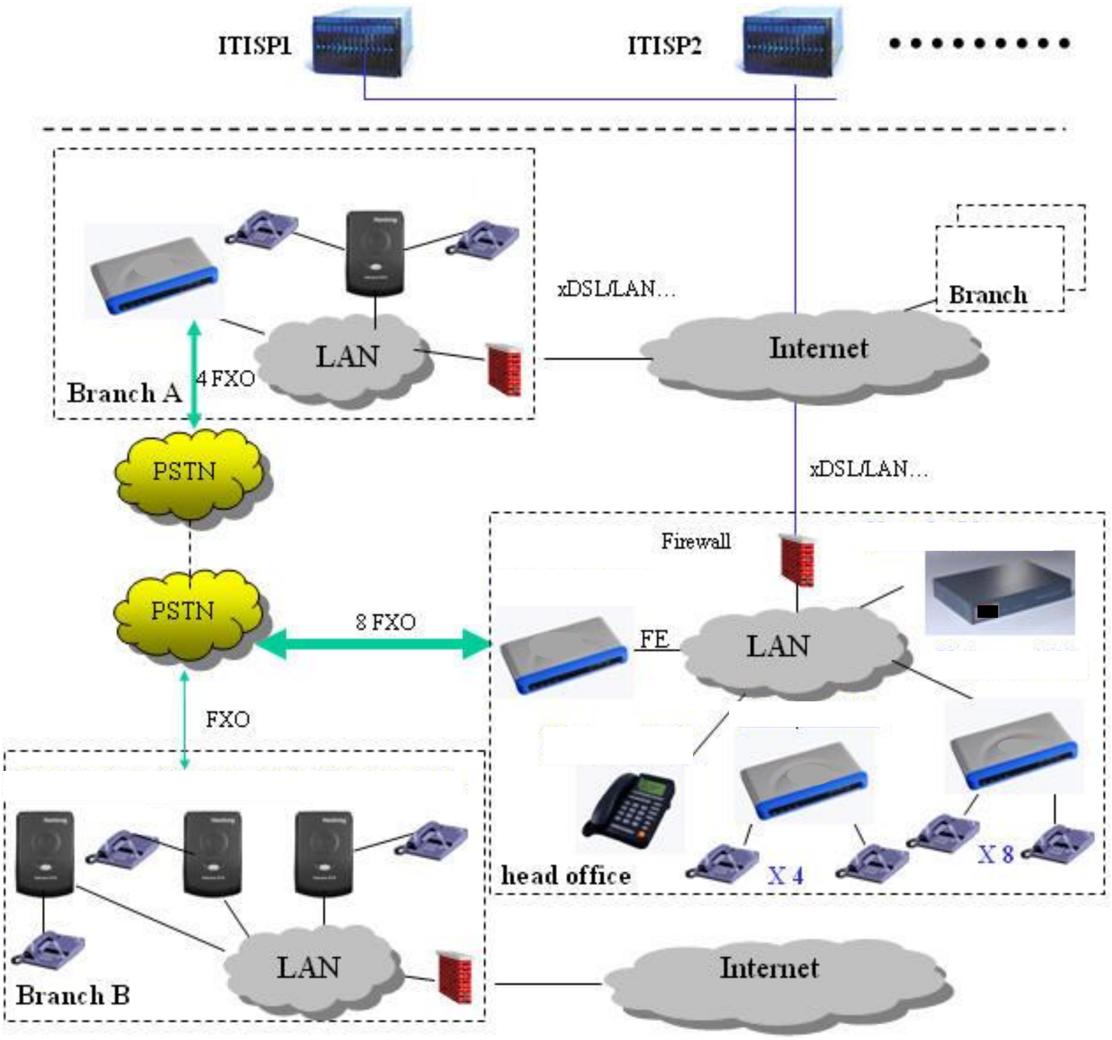
There are two scenarios where the SIP 60X series can be effectively used to enable any business to leverage the benefits of VoIP and the Internet.

3.1 Examples of SIP 60X Configurations

3.1.1 Application : SIP 60X FXS Gateway with PBX Scenario.



3.1.2 Application: SIP 60X & SIP 60X0 Scenario / Toll- Free Calls between Locations



4 SIP 60X FEATURES

The SIP60 x is a next generation IP voice gateway that is interoperable and compatible with leading IP-PBXs, SoftSwitches and SIP platforms. The SIP 60X FXS series is auto-configurable, remotely manageable and scalable. There are two FXS models, the SIP 604 and SIP 608, each offering superb voice quality, traditional telephony functionality, easy deployment, and 4 or 8 FXS ports respectively. Each model features flexible dialing plans, PSTN failover, integrated call routing to support a pure IP network call and an external power supply.

4.1 Software Features Overview

- 4 or 8 FXS ports
- Two RJ-45 ports (switched or routed)
- Multiple SIP accounts & profiles (4 or 8 accounts / choice of 2 profiles per account)
- Supports Voice Codec's: G711 (a/μ, Annex I & II), G723.1A, G726 (ADPCM with 16/24/32/40 bit rates), G729 A/B/E.
- fax pass through and T.38 Fax
- Comprehensive Dial Plan support for Outgoing calls.
- G.168 Echo Cancellation
- Voice Activation Detection (VAD), Comfort Noise Generation (CNG), and Packet Loss Concealment (PLC)
- Supports PSTN/PBX analog telephone sets or analog trunks

TABLE 3: SIP 60X SOFTWARE FEATURES

	SIP 60X FXS Analog Gateway Series
Telephone Interfaces	SIP 604: 4 ports, 4 SIP accounts & choice of 2 profiles SIP 608: 8 ports, 8 SIP accounts & choice of 2 profiles FXS, RJ-11
Network Interface	Two (2) 10M/100 Mbps, RJ-45
LED Indicators	Power and Line LEDs
Voice over Packet Capabilities	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), AEC with NLP, Packetized Voice Protocol Unit (supports RTP/RTCP and AAL2 protocol), G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711
PSTN Fail-over	PSTN failover port on power failure
Voice Compression	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.723.1A, G.726(ADPCM with 16/24/32/40 bit rates), G.729A/B/E, iLBC G.726 provides proprietary VAD, CNG, and signal power estimation Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet header
DHCP Server/Client	Yes, NAT Router or Switched Mode
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through, Fax Datapump V.17, V.19, V.27ter, V.29 for T.38 fax relay
QoS	Diffserve, TOS, 802.1 P/Q VLAN tagging
IP Transport	RTP/RTCP

DTMF Method	flexible DTMF transmission method, User interface of In-audio, RFC2833, and/or SIP Info
IP Signaling	SIP (RFC 3261)
Provisioning	TFTP, HTTP, HTTPS (pending)
Control	TLS/SIPS
Management	Syslog support, HTTPS (pending), Telnet, remote management using Web browser
Dial Plan	Yes
UPnP Support	Yes
Power	Output: 9VDC / Input: 100–240 VAC/50-60 Hz
Mounting	Rack mount, Wall mount, Desktop
Short and long haul	REN3: Up to150 ft on 24 AWG line
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Polarity Reversal / Wink	Yes
EMC	EN55022/EN55024 and FCC part15 Class B
Safety	UL

4.2 Hardware specification

The hardware specifications of the SIP FXS series are detailed in Table 4.

TABLE 4: Hardware Specification of SIP 60X

Ports	4 or 8 FXS Ports
LAN interface	2 x RJ45 10/100Mbps (switched or routed)
PSTN Port	PSTN fail-over port
LED	4 or 8 LEDs (GREEN)
Universal Switching Power Adaptor	Input: 100-240V AC, 50/60Hz, 0.5A Max Output: 9V DC, 2A UL certified
Dimension	225mm (L) x 135mm (W) x 35mm (H)
Weight	0.29 lbs (3.5 oz)
Temperature	32~104°F / 0~40°C
Humidity	10% - 90% (non-condensing)
Compliance	FCC, CE

5 BASIC OPERATIONS

5.1 Understanding SIP 60X Voice Prompts

SIP 60X has a stored voice prompt menu for quick browsing and simple configuration. To enter the voice prompt menu, press *** on the standard analog phone connected to any FXS port.

TABLE 5: Definitions of SIP 60X Voice Prompts

Menu	Voice Will Say the Following:	
Main Menu	"Enter a Menu Option"	Enter "*" for the next menu option Enter "#" to return to the main menu Enter 01 – 05, 07,10 - 17, 47, 86 or 99 Menu option
01	"DHCP Mode", "PPPoE Mode" or "Static IP Mode"	Enter '9' to toggle the selection If user selects "Static IP Mode", user need configure all the IP address information through menu 02 to 05. If user selects "Dynamic IP Mode", the device will retrieve all IP address information from DHCP server automatically when user reboots the device.
02	"IP Address " + IP address	The current WAN IP address is announced Enter 12-digit new IP address if in Static IP Mode.
03	"Subnet " + IP address	Same as Menu option 02
04	"Gateway " + IP address	Same as Menu option 02
05	"DNS Server " + IP address	Same as Menu option 02
07	Preferred Vocoder	Enter "9" to go to the next selection in the list: ➢ PCM U ➢ PCM A ➢ iLBC ➢ G-726 ➢ G-723 ➢ G-729
10	"MAC Address"	Announces the Mac address of the unit.
12	WAN Port Web Access	Enter "9" to toggle between enable and disable

13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Enter “9” to toggle between TFTP and HTTP
16	Firmware Version	Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Enter “9” to rotate among the following three options: 1. always check 2. check when pre/suffix changes 3. never upgrade
47	“Direct IP Calling”	Enter the target IP address to make a direct IP call, after dial tone. (See “Make a Direct IP Call”.)
99	“RESET”	Enter “9” to reboot the device; or Enter MAC address to restore factory default setting (See Restore Factory Default Setting section)
	“Invalid Entry”	Automatically returns to Main Menu

Five Success Tips when using the Voice Prompt

1. “*” shifts down to the next menu option

2. “#” returns to the main menu

3. “9” functions as the ENTER key in many cases to confirm an option

4. All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. For IP address, add 0 before the digits if the digits are less than 3 (i.e. - 192.168.0.26 should be key in like 192168000026. No decimal is needed).

5. Key entry cannot be deleted but the phone may prompt error once it is detected

5.2 Placing a Phone Call

5.2.1 Phone or Extension Numbers

1. Dial the number directly and wait for 4 seconds (Default “No Key Entry Timeout”); or

2. Dial the number directly and press # (Use # as dial key" must be configured in web configuration).

Examples:

1. Dial an extension directly on the same proxy, and then press the # or wait for 4 seconds.

2. Dial an outside number, first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 4 seconds. Check with your VoIP service provider for further details on prefix numbers.

5.2.2 Direct IP Calls

Direct IP calling allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy.

Elements necessary to completing a Direct IP Call:

1. Both SIP 60X and other VoIP Device, have public IP addresses, or
2. Both SIP 60X and other VoIP Device are on the same LAN using private IP addresses, or
3. Both SIP 60X and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

SIP 60X supports two ways to make Direct IP Calling:

Using IVR

1. Pick up the analog phone then access the voice menu prompt by dial "***"
2. Dial "47" to access the direct IP call menu
3. Enter the IP address using format ex. 192*168*0*160 after the dial tone.

Using Star Code

1. Pick up the analog phone then dial "*47"
2. Enter the target IP address using same format as above.

Note: NO dial tone will be played between step 1 and 2.

Destination ports can be specified by using "*" (encoding for ":") followed by the port number.

Examples:

a) If the target IP address is 192.168.0.160, the dialing convention is

***47 or Voice Prompt with option 47, then 192*168*0*160.**

Followed by pressing the "#" key if it is configured as a send key or wait 4 seconds. In this case, the default destination port 5060 is used if no port is specified.

b) If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be: ***47 or Voice Prompt with option 47, then 192*168*0*160*5062** followed by pressing the "#" key, if it is configured as a send key or wait for 4 seconds.

NOTE: When completing direct IP call, the "**Use Random Port**" should set to "NO". You cannot make direct IP calls between FXS1 to FXS2 since they are using same IP.

5.3 Call Hold

Place a call on hold by pressing the “flash” button on the analog phone (if the phone has that button).Press the “flash” button again to release the previously held Caller and resume conversation. If no “flash” button is available, use “hook flash” (toggle on-off hook quickly). You may drop a call using hook flash.

5.4 Call Waiting

Call waiting tone (3 short beeps) indicates an incoming call, if the call waiting feature is enabled. Toggle between incoming call and current call by pressing the “flash” button. First call is placed on hold. Press the “flash” button to toggle between two active calls.

5.5 Call Transfer

Blind Transfer

Assume that call Caller A and B are in conversation. A wants to Blind Transfer B to C:

3. Caller A presses **FLASH** on the analog phone to hear the dial tone.

4. Caller A dials ***87** then dials caller C’s number, and then # (or wait for 4 seconds)

5. Caller A will hear the confirm tone. Then, A can hang up.

NOTE: “Enable Call Feature” must be set to “Yes” in web configuration page.

Caller A can place a call on hold and wait for one of three situations:

1. A quick confirmation tone (similar to call waiting tone) followed by a dial tone. This indicates the transfer is successful (transferee has received a 200 OK from transfer target). At this point, Caller A can either hang up or make another call.

2. A quick busy tone followed by a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.

3. Continuous busy tone. The phone has timed out. Note: continuous busy tone does not indicate the transfer has been successful, nor does it indicate the transfer has failed. It often means there was a failure to receive second NOTIFY – check firmware for most recent release.

Attended Transfer

Assume that Caller A and B are in conversation. Caller A wants to Attend Transfer B to C:

1. Caller A presses **FLASH** on the analog phone for dial tone.

2. Caller A then dials Caller C's number followed by # (or wait for 4 seconds).

3. If Caller C answers the call, Caller A and Caller C are in conversation. Then A can hang up to complete transfer.

4. If Caller C does not answer the call, Caller A can press "flash" to resume call with Caller B.

NOTE: When Attended Transfer fails and A hangs up, the SIP 60X will ring back user A to remind A that B is still on the call. A can pick up the phone to resume conversation with B.

5.6 3-Way Conferencing

The SIP 60X supports Bellcore style 3-way Conference.

Instructions for 3-way conference:

Assuming that call party A and B are in conversation. A (SIP 60X) wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.

2. A dials *23+C's number then # (or wait for 4 seconds).

3. If C answers the call, then A presses FLASH to bring B, C in the conference.

4. If C does not answer the call, A can press FLASH back to talk to B.

5. Conference end after A hangs up.

5.7 Hunting Group

This feature allows user to setup a single SIP account on the gateway and have the ability to use all FXS ports to make/receive calls. Using this feature, all ports active in same hunt group will have the same phone number and incoming calls will be distributed in a round robin manner among the ports active in that hunt group. The number of hunting groups is limited by the number of ports each SIP 60X gateway model has -i.e. each port can be its own hunt group. The most practical and efficient way to use hunt groups is to assign 2 or 3 ports to separate hunt groups.
One additional and popular way to use the Hunting Group feature is called "*multiplexed analog lines*". In this configuration, a legacy PBX system with 8 FXO trunks can be connected to 8 SIP 608 ports configured as a hunt group. The SIP 608 can be registered to a SIP server provider using only one phone number. If the SIP service provider allows multiple calls to the same number, the SIP 608 will allow 8 concurrent calls to the same SIP number. All office members can be reached remotely using the same phone number

in round robin fashion.

Example Configuration of a typical Hunting Group:

1. Configure the SIP account from your VoIP Service Provider on **FXS port 1** under **FXS Ports** webpage.
2. Select **Active** under the **Hunting Group** drop box for FXS port 1.
3. For the remaining ports (say 2, 3 and 4) select **1** for **Hunting Group**. Ports 2, 3 and 4 are now active members of the hunting group associated with port 1.

This configuration will route all calls directed to FXS port 1 to ports 2, 3 and/or 4 in round robin fashion respectively *if* port 1 is busy. You can configure the ring timeout on the **Profile** page.

Example configuration of a multiple hunt group:

FXS Port #1: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**"
FXS Port #2: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"
FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"
FXS Port #4: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**"
FXS Port #5: SIP UserID and Authenticate ID left blank, Hunting Group set to "**4**"
FXS Port #6: SIP UserID and Authenticate ID left blank, Hunting Group set to "**4**"
FXS Port #7: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**"
FXS Port #8: SIP UserID and Authenticate ID left blank, Hunting Group set to "**7**"

Hunt Group 1 contains ports 1, 2, 3. Hunt Group 4 contains ports 4, 5, 6. Hunt Group 7 contains ports 7, 8.

Please be aware, the choice of 1 for ports 2 and 3, the choice of 4 for ports 5 and 6, the choice 7 for port 8 is required to indicate that the SIP account tied to port market as **"Active"** will be used for all members of the same Hunting group. Needless to say, those members of the same Hunting group may not be sequential ports. In following example ports 3, 5 and 7 tied to SIP Account configured in Port #1 marked as **"Active"**, and ports 4,6,8 tied to SIP Account configured in Port #2 marked as **"Active"** as well.

Example of not sequential configuration of a multiple hunt group:

FXS Port #1: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**"
FXS Port #2: SIP UserID and Authenticate ID entered, Hunting Group set to "**Active**"
FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"
FXS Port #4: SIP UserID and Authenticate ID left blank, Hunting group set to "**2**"
FXS Port #5: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"
FXS Port #6: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**"
FXS Port #7: SIP UserID and Authenticate ID left blank, Hunting group set to "**1**"
FXS Port #8: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**"

Note: A single call directed to the SIP account will NOT result in all ports ringing at the same time. They will ring in the hunting group only. This feature is applicable to incoming calls only.

5.8 Inter-port Calling

In some cases a user may want to make phone calls between SIP 60X gateway ports when the gateway will be used as stand alone unit, without any SIP server. This feature will also be applicable when the gateway is used in mode Hunting Groups and will be registered to SIP server only with one master number. In such cases users still will be able to make inter-port calls by using the IVR feature. For example the user connected to port 1 can reach the user connected to port 3 by dialing *** and 73. Digit 7 indicated using inter-port calling feature, digit 3 indicates port number which should be reached. At the same manner the user connected to port 4 can reach the user connected to port 8 by dialing *** and 78.

5.9 PSTN Pass Through/Life Line

The RJ-11 line jack on the SIP 60X side functions as a pass through jack when the SIP 60X is out of power. The pass through/life line enables the user to use the analog phone for PSTN calls directly without using an access code.

5.10 Sending and Receiving Fax

SIP 60X supports fax in two modes:

1) Fax Pass through. If the service provider does not support T.38, pass-through mode may be used. If you have problems with sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

2) T.38 (Fax over IP)

6 CALL FEATURES

The SIP 60X supports the traditional telephony features available in a PBX as well as additional advanced telephony features.

TABLE 6: Call Features Table (Star Code)

Key	Call Features
***30**	**Block Caller ID** (for all-config change)
***31**	**Send Caller ID** (for all-config change)
***67**	**Block Caller ID** (per call)
***82**	**Send Caller ID** (per call)
***47**	**Direct IP Calling.** Dial "*47" + "IP address". **No** dial tone will be

	played in the middle. Detail see Direct IP Calling section on page 12.
***50**	**Disable Call Waiting** (for all-config change)
***51**	**Enable Call Waiting** (for all-config change)
***69**	**Call Return Service:** Dial *69 and the phone will dial the last incoming phone number received.
***70**	**Disable Call Waiting** (Per Call)
***71**	**Enable Call Waiting** (Per Call)
***72**	**Unconditional Call Forward:** Dial "*72" and then the forwarding number followed by "#". Wait for dial tone and hang up. (dial tone indicates successful forward)
***73**	**Cancel Unconditional Call Forward:** Dial "*73" and wait for dial tone, then hang up.
***74**	**Enable Paging Call:** Dial "*74" and then the destination phone number you want to activate in Paging mode.
***78**	**Enable Do Not Disturb (DND):** When enabled all incoming calls will be rejected.
***79**	**Disable Do Not Disturb (DND):** When disabled, incoming calls will be accepted.
***87**	**Blind Transfer**
***90**	**Busy Call Forward:** Dial "*90" and then the forwarding number followed by "#". Wait for dial tone then hang up.
***91**	**Cancel Busy Call Forward**: Dial "*91". Wait for dial tone. Hang up.
***92**	**Delayed Call Forward:** Dial "*92" and then the forwarding number followed by "#". Wait for dial tone then hang up.
***93**	**Cancel Delayed Call Forward:** Dial "*93" for a dial tone, then hang up.
Flash/Hook	If user hears call waiting beep, flash/hook will switch to the new incoming call. Also used to switch to a new channel for a new call.
#	Pressing pound sign will serve as **Re-Dial** key.

7 CONFIGURATION GUIDE

7.1 Configuring SIP 60X via Voice Prompt

DHCP MODE

Select voice menu option 01 to enable SIP 60X to use DHCP.

STATIC IP MODE

Select voice menu option 01 to enable SIP 60X to use STATIC IP mode, then use option 02, 03, 04, 05 to set up IP address, Subnet Mask, Gateway and DNS server respectively.

FIRMWARE SERVER IP ADDRESS

Select voice menu option 13 to configure the IP address of the firmware server.

CONFIGURATION SERVER IP ADDRESS

Select voice menu option 14 to configure the IP address of the configuration server.

UPGRADE PROTOCOL

Select voice menu option 15 to choose firmware and configuration upgrade protocol. User can choose between TFTP and HTTP.

FIRMWARE UPGRADE MODE

Select voice menu option 17 to choose firmware upgrade mode among the following three options:

1) Always check,

2) Check when pre/suffix changes, and

3) Never upgrade

WAN PORT WEB ACCESS

Select voice menu option 12 to enable WAN Port Wed Access of the device configuration pages.

7.2 Configuring SIP 60X with Web Browser

SIP 60X has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow users to configure the SIP 60X through a Web browser such as Microsoft’s IE and AOL’s Netscape.

7.2.1 Access the Web Configuration Menu

The SIP 60X HTML configuration menu can be accessed via LAN or WAN port:

From the LAN port:

1. Directly connect a computer to the LAN port.

2. Open a command window on the computer

3. Type in “ipconfig /release”, the IP address etc. becomes 0.

4. Type in “ipconfig /renew”, the computer gets an IP address in 192.168.22.x segment by default

5. Open a web browser, type in the default gateway IP address. http://192.168.22.1.

   You will see the login page of the device.

From the WAN port:

The WAN port HTML configuration option is disabled by default from factory. To access the HTML configuration menu from the WAN port:

1. Enable the “WAN Port Web Access” option via IVR option 12.

2. Find the WAN IP address of the SIP 60X using voice prompt menu option 02.

3. Access the SIP 60X Web Configuration page by the following URI via WAN port:

http:// SIP 60X -IP-Address (the SIP 60X IP-Address is the WAN IP address for the SIP 60X).

NOTE: If using a web browser to enter the configuration page, strip the leading “0”s because the browser will parse in octet. (i.e. if the IP address is: 192.168.001.014, please type in: 192.168.1.14).

Once the HTTP request is entered and sent from a Web browser, the user will see a log in screen. There are two default passwords for the login page:

User	Password:	Web pages allowed:
End User Level	1234	Only Status and Basic Settings
Administrator Level	admin	Browse all pages

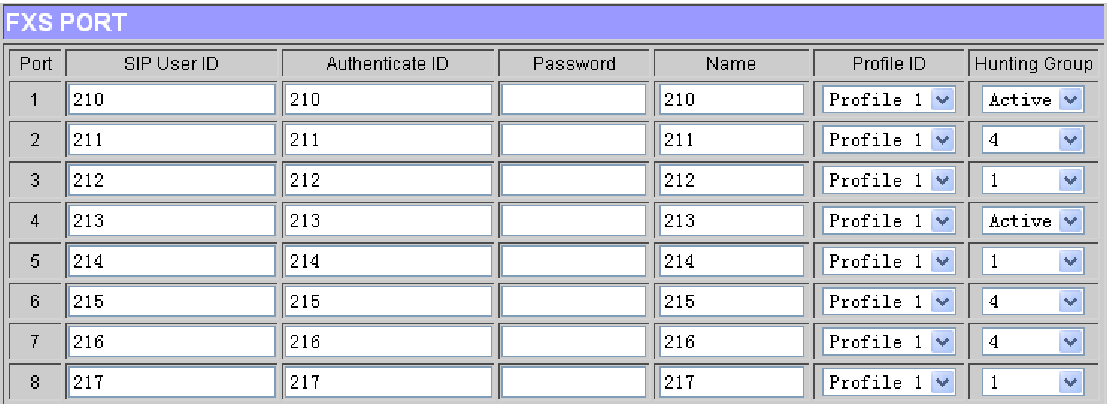
Only an administrator can access the “SUPER SETTINGS” configuration page.

1. There are six different tabs (STATUS, Basic Settings, SUPER Settings, Profile 1, Profile 2 and FXS Ports) on the top of the screen (after login). To open each page, click on the tab.

2. Click on Profile 1 to enter your SIP Server/ SIP Proxy/Registrar information. Enter the IP Address (or FQDN) of the Server under: **SIP Server** and/or **Outbound Proxy**.

3. Click on **FXS ports** to enter the extensions or account information. You will need to fill in the following information for each extension. Once the extensions are configured, you

are finished.



FXS PORT

Port	SIP User ID	Authenticate ID	Password	Name	Profile ID	Hunting Group
1	210	210		210	Profile 1	Active
2	211	211		211	Profile 1	4
3	212	212		212	Profile 1	1
4	213	213		213	Profile 1	Active
5	214	214		214	Profile 1	1
6	215	215		215	Profile 1	4
7	216	216		216	Profile 1	4
8	217	217		217	Profile 1	1

4. Click **save set** after changing any setting and then **Re-boot** to confirm changes.

5. After reboot, check the Status Page to confirm the extensions are successfully registered. You can now use your standard phones connected to ports FXS1 to FXS8 to make calls.

7.3 Important Settings

The end-user must configure the following settings according to the local environment. **NOTE:** Most settings on the web configuration pages are set to the **default values.**

7.3.1 NAT Settings

If you plan to keep the gateway within a private network behind a firewall, we recommend using **STUN Server**. The following three (3) settings are useful in the STUN Server scenario:

1. **STUN Server** (under Super Settings webpage)

Enter a STUN Server IP (or FQDN) that you may have, or look up a free public STUN Server on the internet and enter it on this field. If using Public IP, keep this field blank.

2. **Use Random Ports** (under Super Settings webpage)

It really depends on your network settings, so set this parameter to Yes or No, whichever works. Generally if you have multiple IP devices under the same network, it should be set to Yes. If using a Public IP address, set this parameter to **No**.

3. **NAT Traversal** (under the Profile web pages)

Set this to **Yes** when gateway is behind firewall on a private network.

7.3.2 DTMF Methods

DTMF Settings are **in Profile pages**.

- DTMF in-audio
- DTMF via RTP (RFC2833)
- DTMF via SIP INFO

Enable one or more DTMF methods based on your PBX system.

7.3.3 Preferred VOCODER (Codec)

The SIP 60X supports a broad range of voice codec's. Under Profile web pages, choose your preferred order of different codec's:

- PCMU/A (or G711μ/a)
- G729 A/B/E
- G723
- G726 (16/24/32/40)

7.4 End User Configuration

This section will describe the options in the Web configuration user interface. As mentioned, a user can log in as an administrator or end-user.

Functions available for the end-user are:

- **STATUS:** Displays the network status, account status, software version and MAC-address of the phone
- **BASIC OPTIONS:** Basic preferences such as date and time settings, multi-purpose keys and LCD settings can be set here.

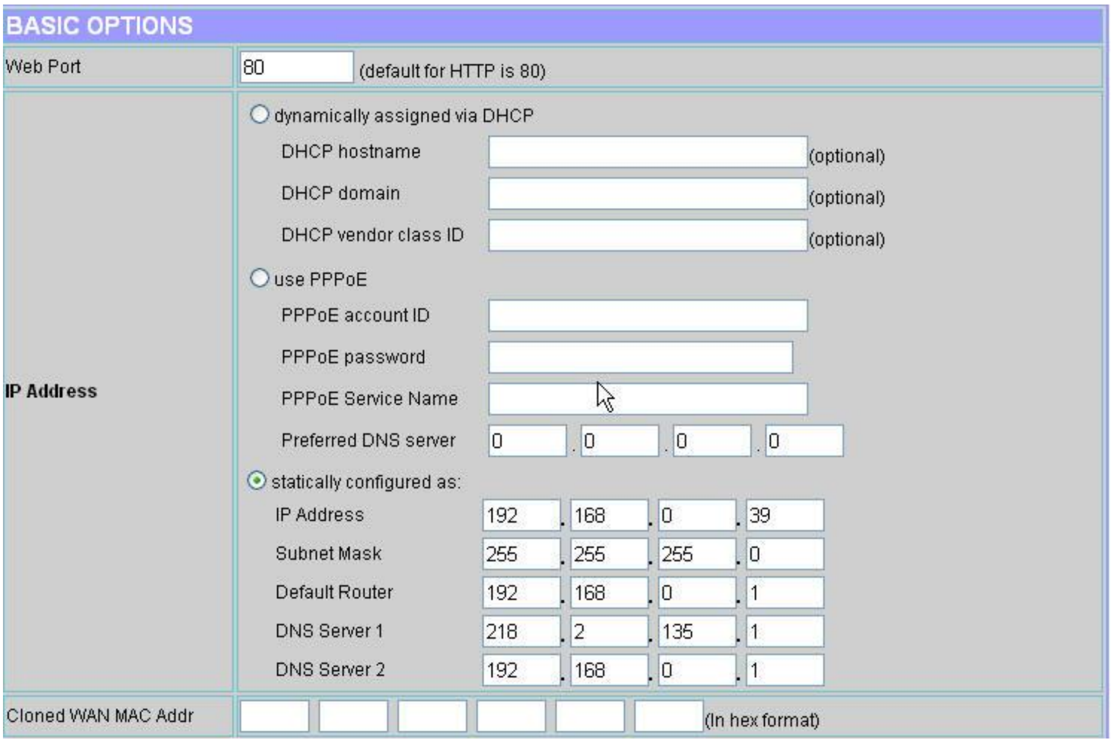
Additional functions available to administrators are:

- **Super OPTIONS:** To set advanced network settings, codec settings and XML

Configuration settings.

- **PROFILE X:** To configure each of the SIP accounts.
- **FXS PORTS:** To configure each of the FXS ports and Hunting Groups etc.

TABLE 7: Basic Settings Page Definitions



BASIC OPTIONS

Web Port: 80 (default for HTTP is 80)

IP Address:

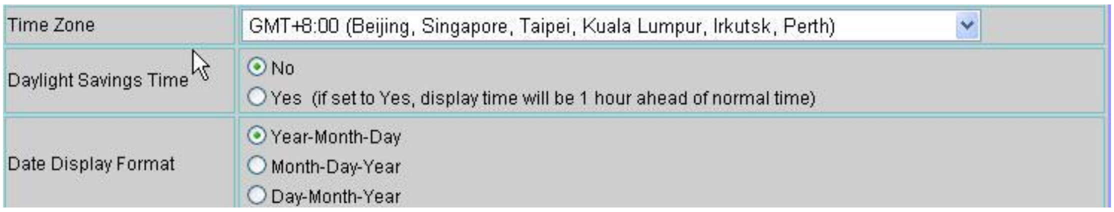
○ dynamically assigned via DHCP
DHCP hostname: (optional)
DHCP domain: (optional)
DHCP vendor class ID: (optional)

○ use PPPoE
PPPoE account ID:
PPPoE password:
PPPoE Service Name:
Preferred DNS server: 0 . 0 . 0 . 0

● statically configured as:
IP Address: 192 . 168 . 0 . 39
Subnet Mask: 255 . 255 . 255 . 0
Default Router: 192 . 168 . 0 . 1
DNS Server 1: 218 . 2 . 135 . 1
DNS Server 2: 192 . 168 . 0 . 1

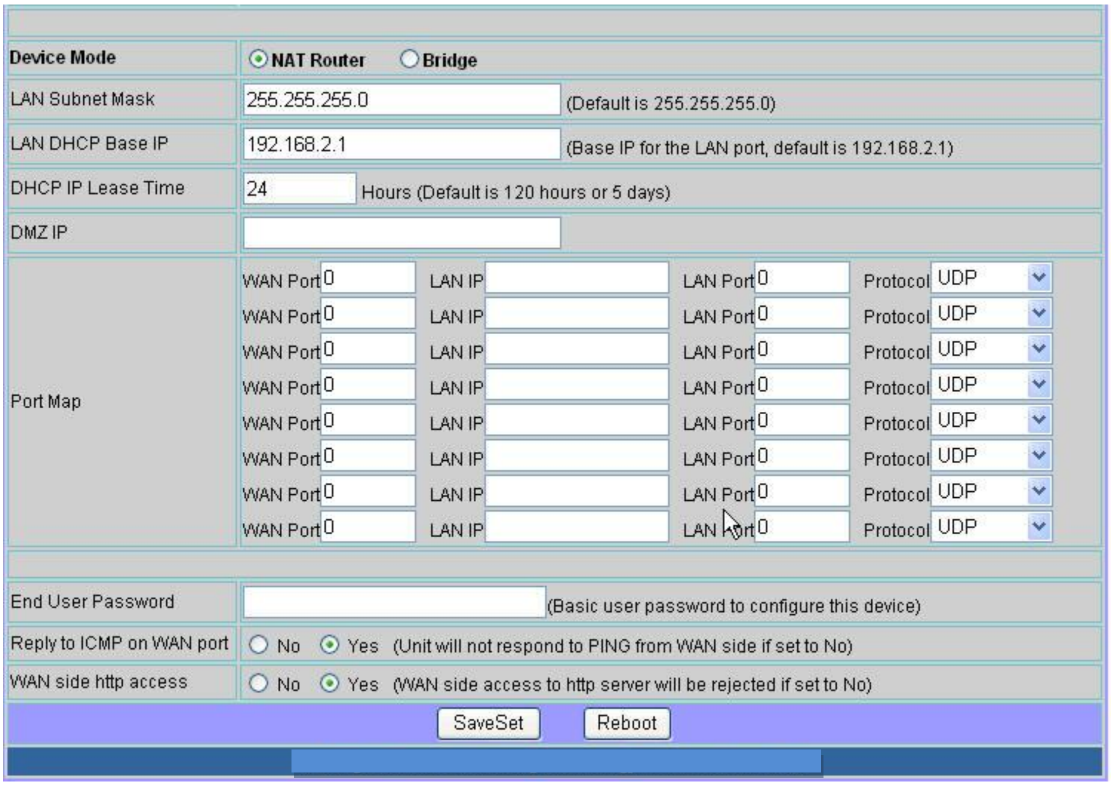
Cloned WAN MAC Addr: (In hex format)

BASIC OPTIONS SETTING	
Setting Options	**Meaning**
Web Port	This is the device's internal HTTP server port. Default is 80.
IP Address	There are two modes to operate the SIP 60X : **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The SIP 60X acquires its IP address from the first DHCP server it discovers from the LAN it is connected. *Using the PPPoE feature*: set the PPPoE account settings. The SIP 60X will establish a PPPoE session if any of the PPPoE fields is set. **Static IP mode:** configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields. These fields are set to zero by default.
Cloned WAN MAC Addr	Allow the user to set a specific MAC address. Set in Hex format



Time Zone	GMT+8:00 (Beijing, Singapore, Taipei, Kuala Lumpur, Irkutsk, Perth)
Daylight Savings Time	● No ○ Yes (if set to Yes, display time will be 1 hour ahead of normal time)
Date Display Format	● Year-Month-Day ○ Month-Day-Year ○ Day-Month-Year

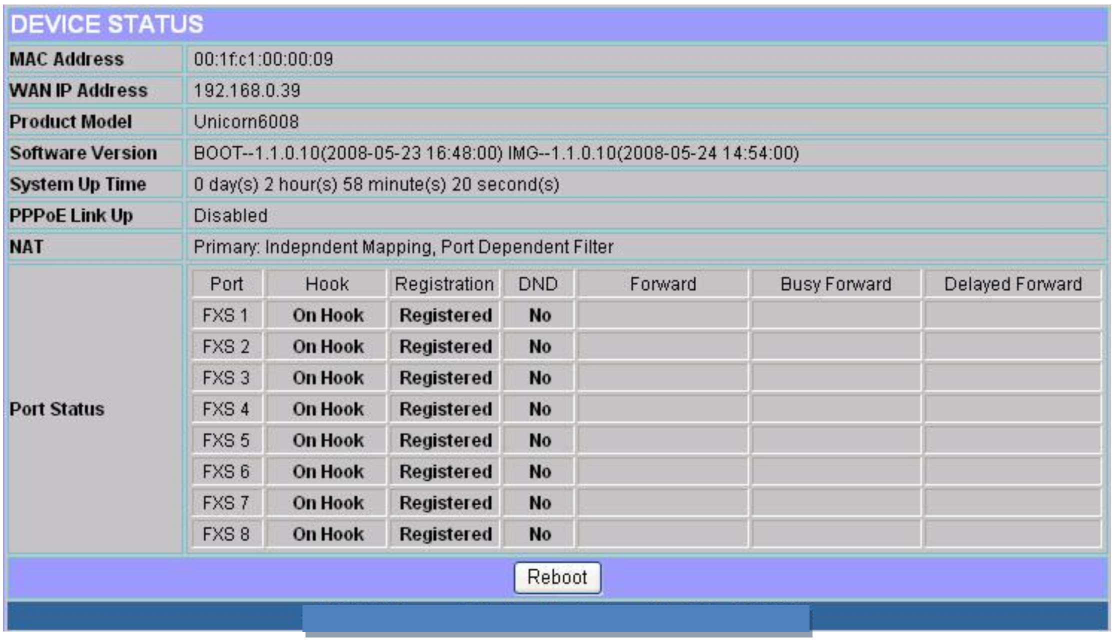
BASIC OPTIONS SETTING	
Setting Options	**Meaning**
Time Zone	Controls how the date/time is displayed according to the specified time zone.
Daylight Savings Time	This parameter controls whether the displayed time will be daylight savings time or not. If set to Yes, then the displayed time will be 1 hour ahead of normal time.
Date Display Format	Allow user to choose among the following three formats: Year-Month-Day Month-Day-Year Day-Month-Year



Device Mode	● **NAT Router** ○ **Bridge**
LAN Subnet Mask	255.255.255.0 (Default is 255.255.255.0)
LAN DHCP Base IP	192.168.2.1 (Base IP for the LAN port, default is 192.168.2.1)
DHCP IP Lease Time	24 Hours (Default is 120 hours or 5 days)
DMZ IP	
Port Map	WAN Port 0 LAN IP LAN Port 0 Protocol UDP WAN Port 0 LAN IP LAN Port 0 Protocol UDP WAN Port 0 LAN IP LAN Port 0 Protocol UDP WAN Port 0 LAN IP LAN Port 0 Protocol UDP WAN Port 0 LAN IP LAN Port 0 Protocol UDP WAN Port 0 LAN IP LAN Port 0 Protocol UDP WAN Port 0 LAN IP LAN Port 0 Protocol UDP WAN Port 0 LAN IP LAN Port 0 Protocol UDP
End User Password	(Basic user password to configure this device)
Reply to ICMP on WAN port	○ No ● Yes (Unit will not respond to PING from WAN side if set to No)
WAN side http access	○ No ● Yes (WAN side access to http server will be rejected if set to No)

BASIC OPTIONS SETTING	
Setting Options	**Meaning**
Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Save the setting and reboot prior to configuring the SIP60X.
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP	Base IP for the LAN port which functions as a Gateway for the subnet. Default value is **192.168.22.1**.
DHCP IP Lease Time	Value is set in units of hours. Default value is **120 hrs** (5 Days.) The time IP address is assigned to the LAN clients.
DMZ IP	Forward all WAN IP traffic to a specific IP address if no matching port is used by SIP 60X or defined in port forwarding.
Port Map	Forwards a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port
End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive.
Reply to ICMP on WAN port	If set to “Yes”, the SIP 60X will respond to the PING command from other computers, but it also is vulnerable to the DOS attack. Default is **No**.
WAN side http access	If this parameter is set to “No”, the HTML configuration update via WAN port is disabled.

TABLE 8: Status Page Definitions

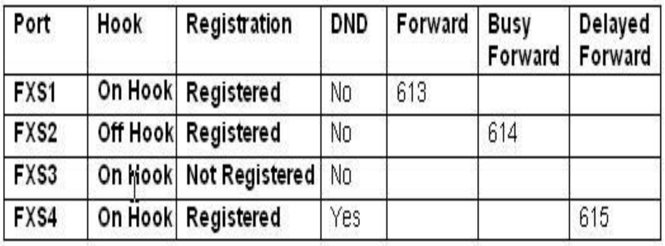


DEVICE STATUS

MAC Address	00:1f:c1:00:00:09
WAN IP Address	192.168.0.39
Product Model	Unicorn6008
Software Version	BOOT--1.1.0.10(2008-05-23 16:48:00) IMG--1.1.0.10(2008-05-24 14:54:00)
System Up Time	0 day(s) 2 hour(s) 58 minute(s) 20 second(s)
PPPoE Link Up	Disabled
NAT	Primary: Indepndent Mapping, Port Dependent Filter

Port Status

Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
FXS 1	On Hook	Registered	No			
FXS 2	On Hook	Registered	No			
FXS 3	On Hook	Registered	No			
FXS 4	On Hook	Registered	No			
FXS 5	On Hook	Registered	No			
FXS 6	On Hook	Registered	No			
FXS 7	On Hook	Registered	No			
FXS 8	On Hook	Registered	No			

<table>
<tr><th colspan="2">STATUS PAGE DEFINITIONS</th></tr>
<tr><th>Setting Options</th><th>Meaning</th></tr>
<tr><td>MAC Address</td><td>The device ID in HEX format.
This is needed for ISP troubleshooting.
Note there are separate MAC addresses for the WAN side and the LAN side.</td></tr>
<tr><td>WAN IP Address</td><td>Shows WAN IP address of SIP 60X</td></tr>
<tr><td>Product Model</td><td>Contains the product model info.</td></tr>
<tr><td>Software Version</td><td>Program: This is the main software release.
Boot and Loader are not changed often.</td></tr>
<tr><td>System Up Time</td><td>Shows system up time since the last reboot.</td></tr>
<tr><td>PPPoE Link Up</td><td>Shows whether the PPPoE connection is running if connected to DSL modem.</td></tr>
<tr><td>NAT</td><td>Shows type of NAT the SIP 60X is connected to via its WAN port. It is based on STUN protocol.</td></tr>
<tr><td>Port Status</td><td>Shows several information regarding the individual FXS ports.
Ex.

<table>
<tr><th>Port</th><th>Hook</th><th>Registration</th><th>DND</th><th>Forward</th><th>Busy Forward</th><th>Delayed Forward</th></tr>
<tr><td>FXS1</td><td>On Hook</td><td>Registered</td><td>No</td><td>613</td><td></td><td></td></tr>
<tr><td>FXS2</td><td>Off Hook</td><td>Registered</td><td>No</td><td></td><td>614</td><td></td></tr>
<tr><td>FXS3</td><td>On Hook</td><td>Not Registered</td><td>No</td><td></td><td></td><td></td></tr>
<tr><td>FXS4</td><td>On Hook</td><td>Registered</td><td>Yes</td><td></td><td></td><td>615</td></tr>
</table>
** FXS port 4 user has set Do Not Disturb.</td></tr>
</table>

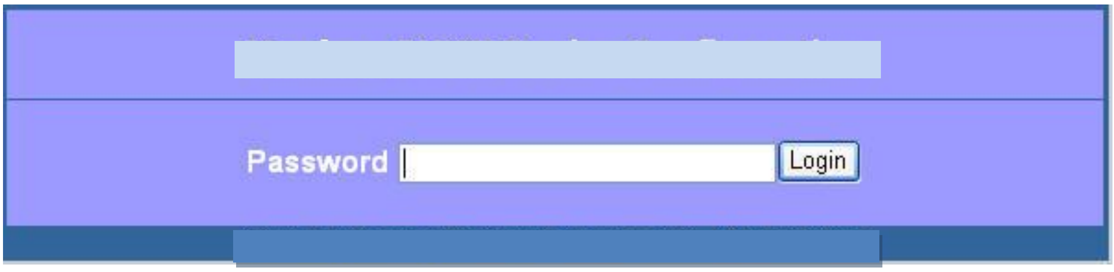
	FXS port 1 user has set his calls to be forwarded unconditionally to ext 613 FXS port 2 users have set his calls to be forwarded to 614 when his phone is busy. FXS port 3 users are not registered with his SIP Server.

Super User configuration includes not only the end user configuration, but also super configurations such as: SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.

7.5 Super User Configuration

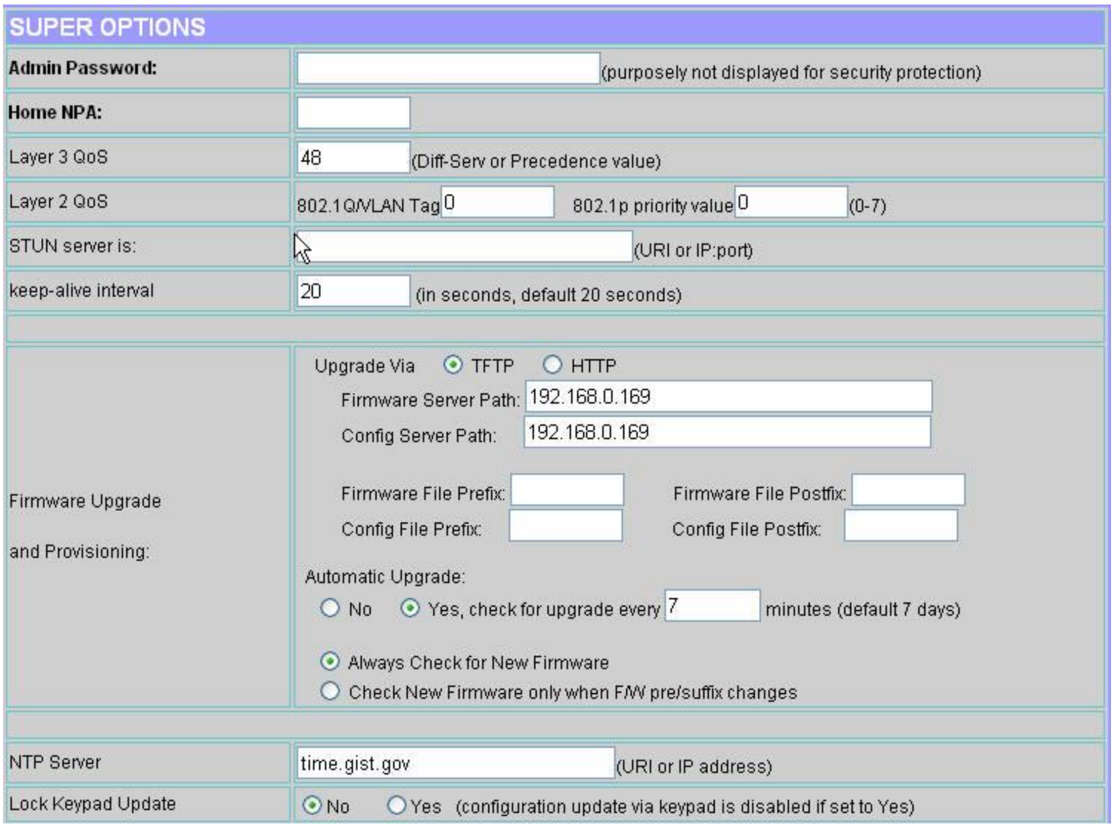
Log-in to the Super User Configuration Page the same way as for the basic configuration page. Log-in using either of the following passwords: “admin” or “123”.

7.6 Figure 3: Screenshot of Super User Configuration Login Screen



Super User configuration includes the end user configuration and Super configurations including: SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.

TABLE 9: Super Configuration Page Definitions



SUPER OPTIONS

Admin Password: (purposely not displayed for security protection)

Home NPA:

Layer 3 QoS: 48 (Diff-Serv or Precedence value)

Layer 2 QoS: 802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)

STUN server is: (URI or IP:port)

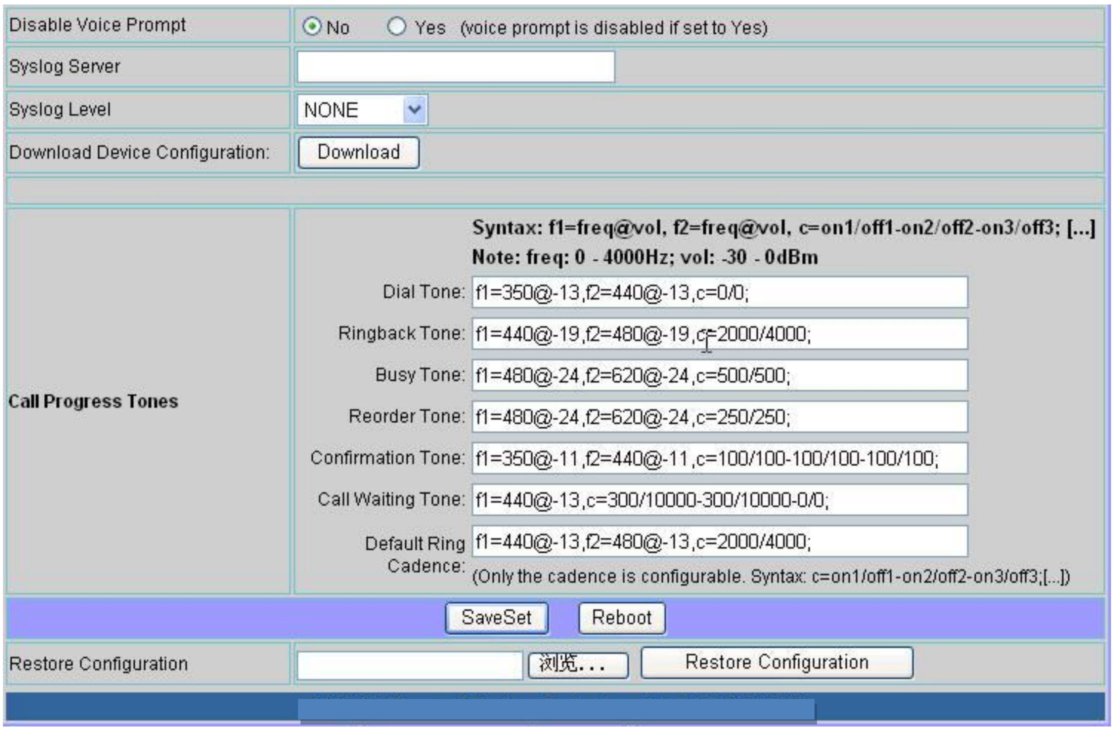
keep-alive interval: 20 (in seconds, default 20 seconds)

Firmware Upgrade and Provisioning:
Upgrade Via ● TFTP ○ HTTP
Firmware Server Path: 192.168.0.169
Config Server Path: 192.168.0.169
Firmware File Prefix: Firmware File Postfix:
Config File Prefix: Config File Postfix:
Automatic Upgrade:
○ No ● Yes, check for upgrade every 7 minutes (default 7 days)
● Always Check for New Firmware
○ Check New Firmware only when F/W pre/suffix changes

NTP Server: time.gist.gov (URI or IP address)

Lock Keypad Update: ● No ○ Yes (configuration update via keypad is disabled if set to Yes)

SUPER OPTIONS SETTING	
Setting Options	**Meaning**
Admin Password:	This contains the password to access the Super Web Configuration page.This field is case sensitive.
Home NPA:	Local area code for North American Dial Plan.
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48.
Layer 2 QoS	This contains the value used for layer 2 VLAN tag. Default setting is blank.
STUN server is:	IP address or domain name of stun server
keep-alive interval	This parameter specifies how often the SIP 60X sends a blank UDP packet to the SIP server to keep the “hole” on the NAT open. Default is **20 seconds**.
Firmware Upgrade and Provisioning:	Default method is HTTP. Firmware upgrade may take up to 10 minutes depending on network environment. Do not interrupt the firmware upgrading process.
NTP Server	This parameter defines the URI or IP address of the NTP server which is used by the SIP 60X to display the current date/time.
Lock Keypad Update	If this parameter is set to “Yes”, the configuration update via keypad is disabled.



Disable Voice Prompt: ● No ○ Yes (voice prompt is disabled if set to Yes)

Syslog Server:

Syslog Level: NONE

Download Device Configuration: Download

Call Progress Tones

Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...]

Note: freq: 0 - 4000Hz; vol: -30 - 0dBm

Dial Tone: f1=350@-13,f2=440@-13,c=0/0;

Ringback Tone: f1=440@-19,f2=480@-19,c=2000/4000;

Busy Tone: f1=480@-24,f2=620@-24,c=500/500;

Reorder Tone: f1=480@-24,f2=620@-24,c=250/250;

Confirmation Tone: f1=350@-11,f2=440@-11,c=100/100-100/100-100/100;

Call Waiting Tone: f1=440@-13,c=300/10000-300/10000-0/0;

Default Ring Cadence: f1=440@-13,f2=480@-13,c=2000/4000;

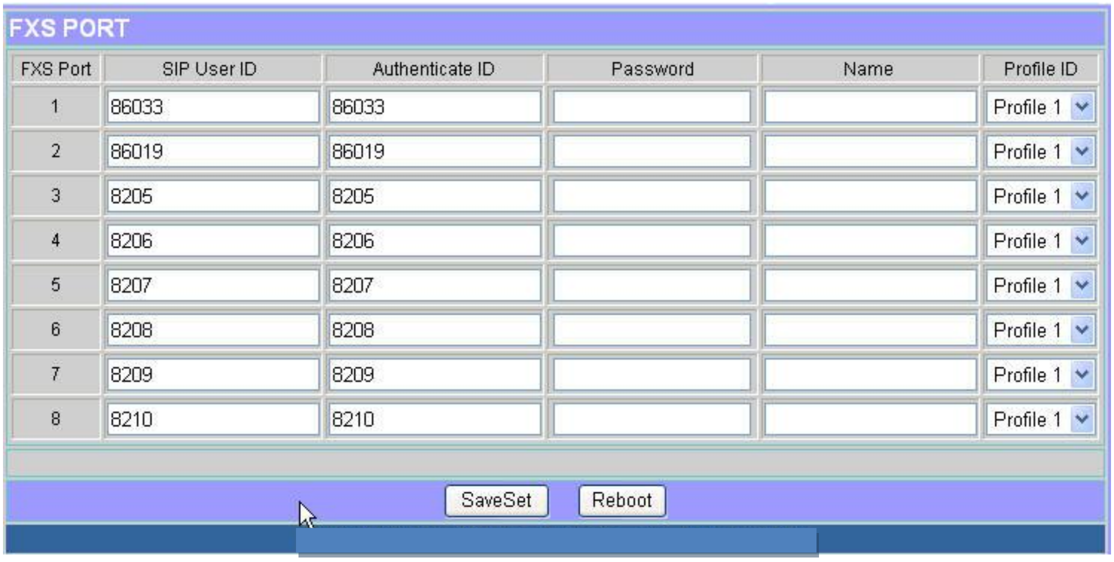
(Only the cadence is configurable. Syntax: c=on1/off1-on2/off2-on3/off3;[...])

Restore Configuration

SUPER CONFIGURATION PAGE DEFINITIONS	
Setting Options	**Meaning**
Disable Voice Prompt	Default is No
Syslog Server	The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)
Syslog Level	Select the SIP 60X to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level)
Download Device Configuration	User can download configuration from the web page and save to configuration file.
Call Progress Tones	Using these settings, user can configure tone frequencies

	according to their preference. By default they are set to North American frequencies .Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous tone, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern.
Restore Configuration	User can restore the before configuration from the configuration file saved at local pc

TABLE 10: FXS Ports Configuration Definitions

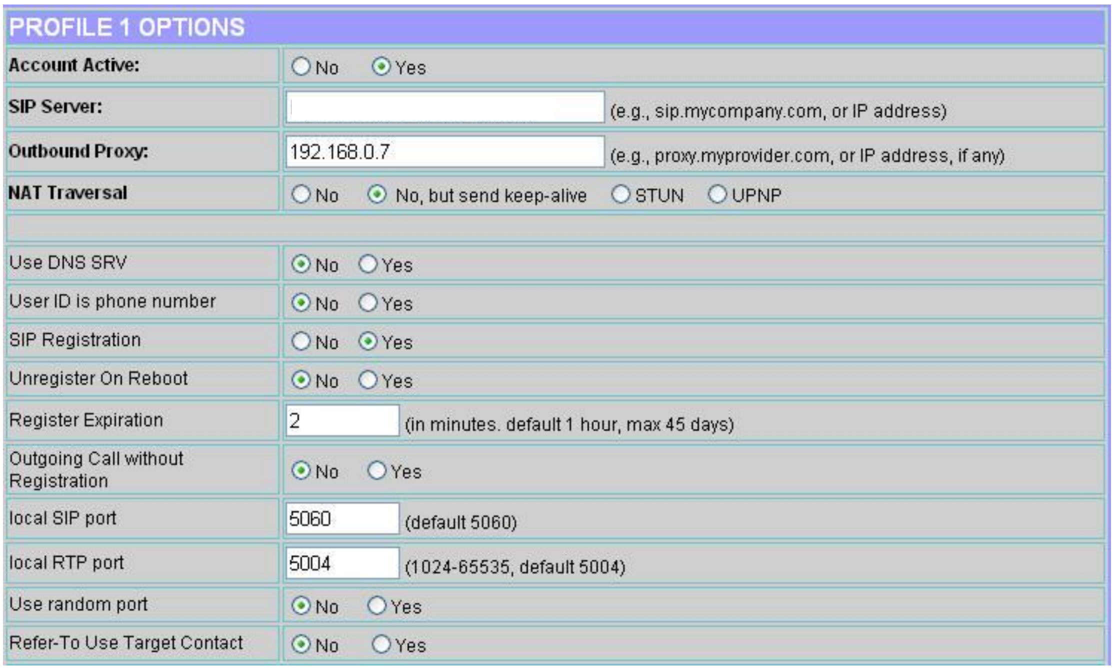


FXS PORT

FXS Port	SIP User ID	Authenticate ID	Password	Name	Profile ID
1	86033	86033			Profile 1
2	86019	86019			Profile 1
3	8205	8205			Profile 1
4	8206	8206			Profile 1
5	8207	8207			Profile 1
6	8208	8208			Profile 1
7	8209	8209			Profile 1
8	8210	8210			Profile 1

FXS PORT SETTING	
Setting Options	**Meaning**
FXS Port	FXS Port Number
SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Password	SIP service subscriber's account password for SIP 60X to register to (SIP) servers of ITSP.
Name	Name
Profile ID	Select the corresponding Profile ID (1/2)

TABLE 11: Profile Page Definitions



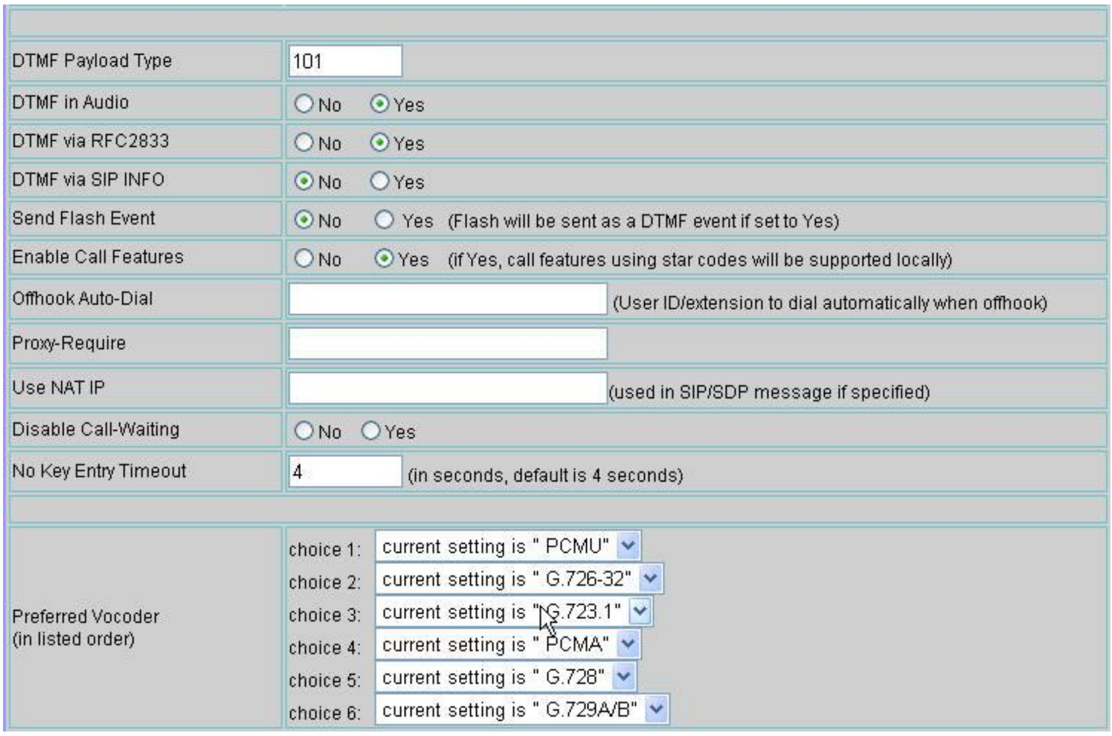
PROFILE 1 OPTIONS

Account Active:	○ No ● Yes
SIP Server:	(e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	192.168.0.7 (e.g., proxy.myprovider.com, or IP address, if any)
NAT Traversal	○ No ● No, but send keep-alive ○ STUN ○ UPNP
Use DNS SRV	● No ○ Yes
User ID is phone number	● No ○ Yes
SIP Registration	○ No ● Yes
Unregister On Reboot	● No ○ Yes
Register Expiration	2 (in minutes. default 1 hour, max 45 days)
Outgoing Call without Registration	● No ○ Yes
local SIP port	5060 (default 5060)
local RTP port	5004 (1024-65535, default 5004)
Use random port	● No ○ Yes
Refer-To Use Target Contact	● No ○ Yes

PROFILE PAGE DEFINITIONS

Setting Options	Meaning
Account Active	When set to Yes the SIP Profile is activated.
SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by SIP 60X for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can correct the problem.
NAT Traversal	This parameter defines whether the SIP 60X NAT traversal mechanism is activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the SIP 60X performs according to the STUN client specification. Under this mode, the embedded STUN client will detect if and what type of firewall/NAT is being used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the SIP 60X will use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" *with no specified STUN server*, the SIP 60X will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.

	If your home or office router can act as a UPNP server, you can select “UPNP” option for NAT traversal.
Use DNS SRV:	Default is **No**. If set to “Yes” the client will use DNS SRV to look up server.
User ID is Phone Number	If the SIP 60X has an assigned PSTN telephone number, this field should be set to “Yes”. Otherwise, set it to “No”. If “Yes” is set, a “user=phone” parameter will be attached to the “From” header in SIP request.
SIP Registration	This parameter controls whether the SIP 60X needs to send REGISTER messages to the proxy server. The default setting is “Yes”.
Unregister on Reboot	Default is **No**. If set to “Yes”, the SIP user’s registration information is cleared on reboot.
Register Expiration	Allows the user to specify the time frequency (in minutes) for the SIP 60X to refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Outgoing Call without Registration	Default is **No**. If set to “Yes,” user can place outgoing calls even when not registered (if allowed by ITSP) but is unable to receive incoming calls.
Local SIP port	Defines the local SIP port the SIP 60X will listen and transmit. The default value for Profile 1 is 5060 and 6060 for Profile 2.
Local RTP Port	Defines the local RTP-RTCP port pair the PROFILE will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP, port_value+3 for its RTCP and so on. The default value for Profile 1 is 5004 and 6004 for Profile 2.
Use random port	This parameter, when set to “YES”, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple SIP 60X are behind the same NAT.
Refer to Use Target Contact	Default is No. If set to Yes, then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact header information.



DTMF Payload Type: 101

DTMF in Audio: ○ No ● Yes

DTMF via RFC2833: ○ No ● Yes

DTMF via SIP INFO: ● No ○ Yes

Send Flash Event: ● No ○ Yes (Flash will be sent as a DTMF event if set to Yes)

Enable Call Features: ○ No ● Yes (if Yes, call features using star codes will be supported locally)

Offhook Auto-Dial: (User ID/extension to dial automatically when offhook)

Proxy-Require:

Use NAT IP: (used in SIP/SDP message if specified)

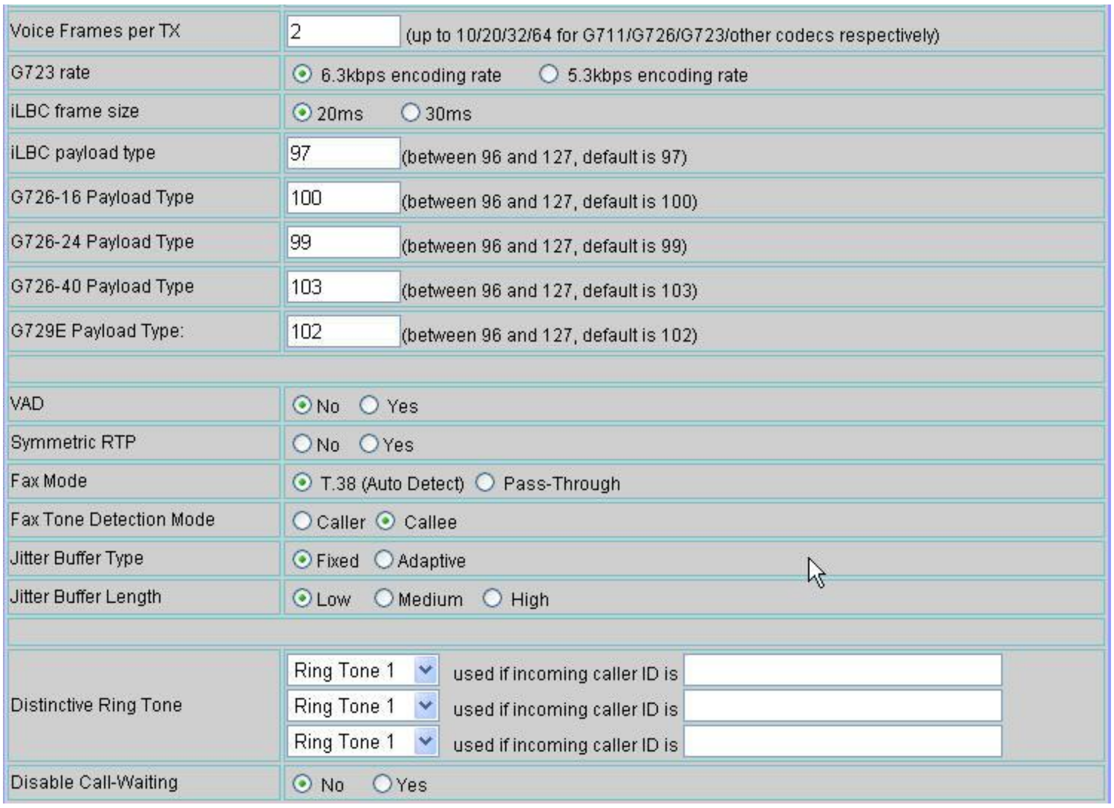
Disable Call-Waiting: ○ No ○ Yes

No Key Entry Timeout: 4 (in seconds, default is 4 seconds)

Preferred Vocoder (in listed order)
- choice 1: current setting is " PCMU"
- choice 2: current setting is " G.726-32"
- choice 3: current setting is " G.723.1"
- choice 4: current setting is " PCMA"
- choice 5: current setting is " G.728"
- choice 6: current setting is " G.729A/B"

PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
DTMF Payload Type	Sets the payload type for DTMF using RFC2833.
DTMF in-audio	Send DTMF as inband (in-audio).
DTMF via RFC2833	Default "**YES**".
DTMF via SIP INFO	Send DTMF via SIP INFO message.
Send Flash Event	Default is **No**. If set to yes, flash will be sent as DTMF event.
Enable Call Features	Default is **Yes**. (If Yes, call features using star codes will be supported locally)
Off-Hook Auto Dial	Allows the user to configure a User ID or extension number to be automatically dialed upon off-hook. Only the user part of a SIP address needs to be entered here. The phone will automatically append the "@" and the host portion of the corresponding SIP address.
Proxy Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
Disable Call Waiting	Default is **No**.
No Key Entry Timeout	Default is **4 seconds**.
Preferred Vocoder	The SIP 60X supports up to 5 different Vocoder types including G.711 A-/U-law, G.726 (Supports bit rates 16, 24,

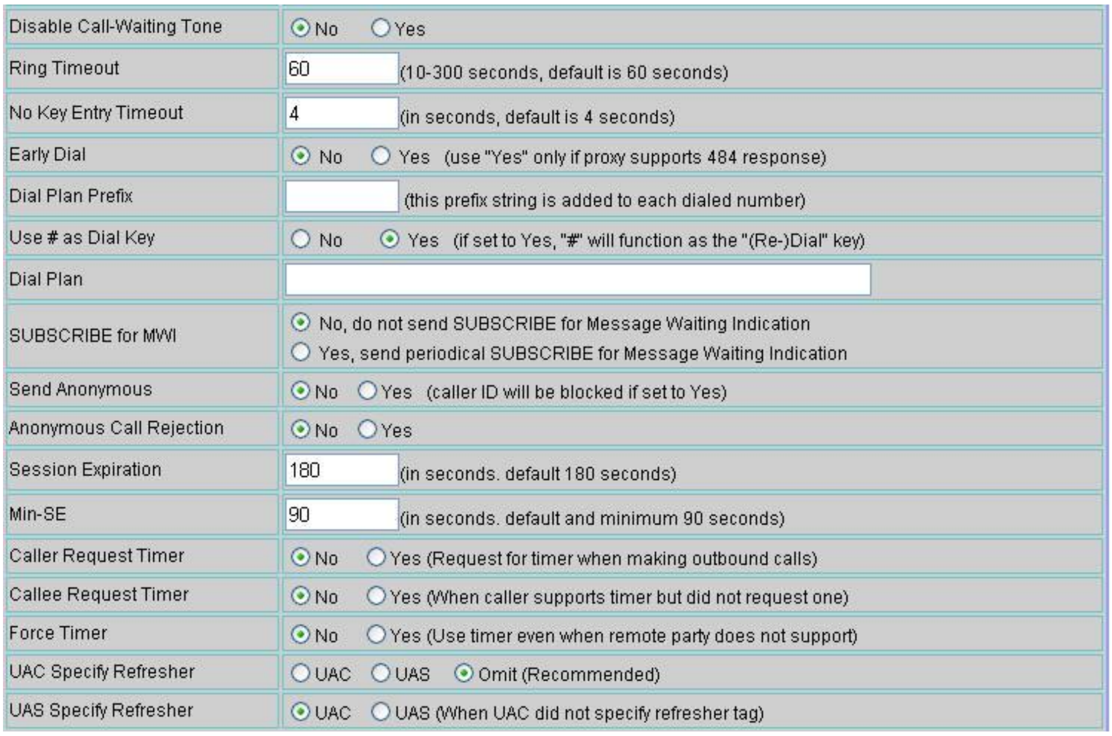
	32 and 40), G.723.1, G.729A/B/E and iLBC. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder is entered by choosing the appropriate option in “Choice 1”. The last Vocoder is entered by choosing the appropriate option in “Choice 8”.



Voice Frames per TX	2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
G723 rate	● 6.3kbps encoding rate ○ 5.3kbps encoding rate
iLBC frame size	● 20ms ○ 30ms
iLBC payload type	97 (between 96 and 127, default is 97)
G726-16 Payload Type	100 (between 96 and 127, default is 100)
G726-24 Payload Type	99 (between 96 and 127, default is 99)
G726-40 Payload Type	103 (between 96 and 127, default is 103)
G729E Payload Type:	102 (between 96 and 127, default is 102)
VAD	● No ○ Yes
Symmetric RTP	○ No ○ Yes
Fax Mode	● T.38 (Auto Detect) ○ Pass-Through
Fax Tone Detection Mode	○ Caller ● Callee
Jitter Buffer Type	● Fixed ○ Adaptive
Jitter Buffer Length	● Low ○ Medium ○ High
Distinctive Ring Tone	Ring Tone 1 used if incoming caller ID is Ring Tone 1 used if incoming caller ID is Ring Tone 1 used if incoming caller ID is
Disable Call-Waiting	● No ○ Yes

PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first vocoder is configured as G723 and the "Voice Frames per TX" is set to be 2, then the "ptime" value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the "ptime" value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the SIP 60X will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.
G723 Rate	Defines the encoding rate for G.723 vocoder. By default, 6.3kbps rate is chosen.
iLBC Frame Size	Sets the iLBC frame size in 20ms or 30ms
iLBC Payload type	Defines payload type for iLBC. Default value is 97. The valid range is between 96 and 127.
G726-16 Payload type	Default value is 98. Range is from 96 to 127.
G726 – 24 Payload type	Default value is 99. Range is from 96 to 127.
G726 – 40 Payload type	Default value is 103. Range is from 96 to 127.
G729E payload type	Default value is 102. Range is from 96 to 127.
VAD	Default is **No**. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.
Symmetric RTP	Default is **No**. When set to Yes the device will change the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device.
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)

Fax Tone Detection Mode	Default is Callee. This decides whether Caller or Callee sends out the re-INVITE for T.38 or Fax Pass Through.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions.
Distinctive Ring tone	Custom Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY uses this ring tone when the incoming call is from the Caller ID. System Ring Tone is used for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for **all** incoming calls.
Disable Call Waiting	Default is **No**.

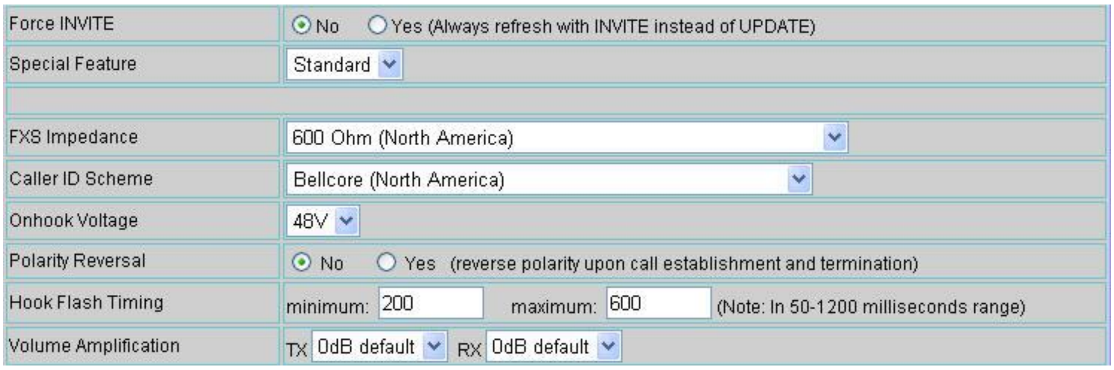


Disable Call-Waiting Tone	● No ○ Yes
Ring Timeout	60 (10-300 seconds, default is 60 seconds)
No Key Entry Timeout	4 (in seconds, default is 4 seconds)
Early Dial	● No ○ Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix	(this prefix string is added to each dialed number)
Use # as Dial Key	○ No ● Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
Dial Plan	
SUBSCRIBE for MWI	● No, do not send SUBSCRIBE for Message Waiting Indication ○ Yes, send periodical SUBSCRIBE for Message Waiting Indication
Send Anonymous	● No ○ Yes (caller ID will be blocked if set to Yes)
Anonymous Call Rejection	● No ○ Yes
Session Expiration	180 (in seconds. default 180 seconds)
Min-SE	90 (in seconds. default and minimum 90 seconds)
Caller Request Timer	● No ○ Yes (Request for timer when making outbound calls)
Callee Request Timer	● No ○ Yes (When caller supports timer but did not request one)
Force Timer	● No ○ Yes (Use timer even when remote party does not support)
UAC Specify Refresher	○ UAC ○ UAS ● Omit (Recommended)
UAS Specify Refresher	● UAC ○ UAS (When UAC did not specify refresher tag)

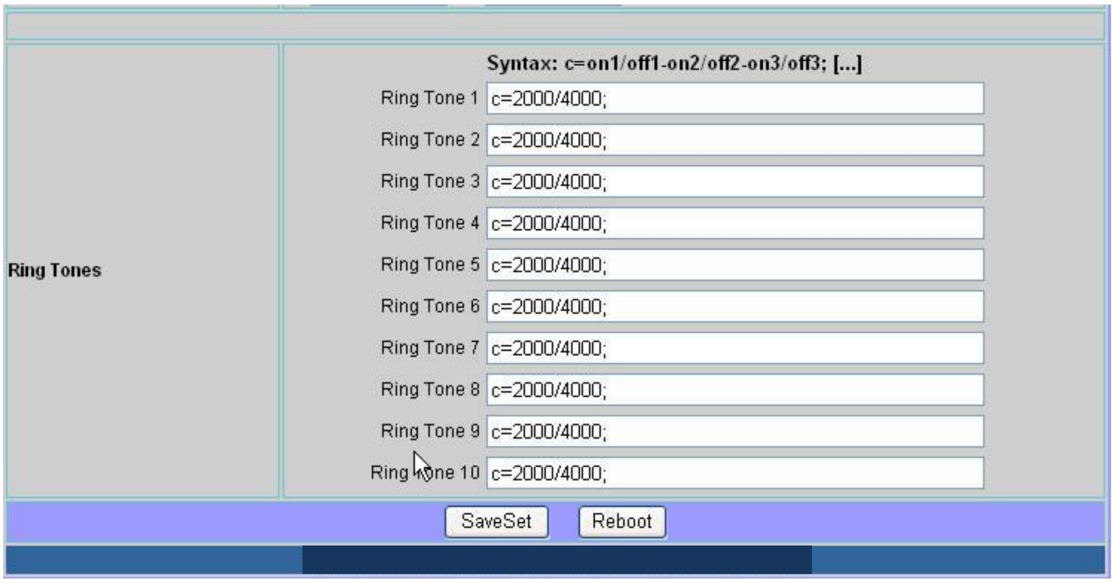
PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
Disable Call Waiting Tone	Default is **No**. This is to disable the stutter Call Waiting Tone when a Call Waiting call arrives. The CWCID will still be displayed.
Ring Timeout	Incoming call will stop ringing when not picked up given a specific period of time.
No Key Entry Timeout	Default is **4 seconds**.
Early Dial	Default is **No**. Use only if proxy supports 484 response.

	This parameter controls whether the phone will send an early INVITE each time a key is pressed when a user dials a number. If set to “Yes”, an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the “(Re-)Dial” button is pressed or after about 5 seconds have elapsed if the user forgets to press the “Re-Dial” button. The “Yes” option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address responses. Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). *This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling*.
Dial Plan Prefix	Sets the prefix added to each dialed number.
Use # as Dial Key	Allows users to configure the “#” key as the “Send” (or “Dial”) key. If set to “Yes”, “#” will send the number. In this case, this key is essentially equivalent to the “Dial” key. If set to “No”, this “#” key can be included as part of number.
Dial Plan	Dial Plan Rules: 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d 2. Grammar: x - any digit from 0-9; a. xx+ - at least 2 digits number; b. xx. ?at least 2 digits number; c. ^ - exclude; d. [3-5] - any digit of 3, 4, or 5; e. [147] - any digit 1, 4, or 7; f. <2=011> - replace digit 2 with 011 when dialing Example 1: {[369]11 | 1617xxxxxxx} Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617 Example 2: {^1900x+ | <=1617>xxxxxxx} Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers Example 3: {1xxx[2-9]xxxxxx | <2=011>x+} Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; if leading digit is 2, replace leading digit 2 with 011 before dialing. 3. Default: Outgoing - {x+} Example of a simple dial plan used in a Home/Office in the US: { ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 }

	Explanation of example rule (reading from left to right): ^1900x. - prevents dialing any number started with 1900 <=1617>[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically 1[2-9]xx[2-9]xxxxxx |- allows dialing to any US/Canada Number with 11 digits length 011[2-9]x. - allows international calls starting with 011 [3469]11 - allow dialing special and emergency numbers 311, 411, 611 and 911 Note: In some cases user wishes to dial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature and the Dial Plan should be: { [x*]+ }.
Subscribe for MWI	Default is **No**. When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Send Anonymous	If this parameter is set to “Yes”, the “From” header along with Privacy and P_Asserted_Identity headers in outgoing INVITE message will be set to anonymous, blocking Caller ID.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 600X Busy message.
Session Expiration	Default is 180 seconds.
Min-SE	Default is 90 seconds
Caller Request Timer	Default is NO
Callee Request Timer	Default is NO
Force Timer	Default is NO
UAC Specify Refresher	Default is Omit
UAS Specify Refresher	Default is UAC



Force INVITE	● No ○ Yes (Always refresh with INVITE instead of UPDATE)
Special Feature	Standard
FXS Impedance	600 Ohm (North America)
Caller ID Scheme	Bellcore (North America)
Onhook Voltage	48V
Polarity Reversal	● No ○ Yes (reverse polarity upon call establishment and termination)
Hook Flash Timing	minimum: 200 maximum: 600 (Note: In 50-1200 milliseconds range)
Volume Amplification	TX 0dB default RX 0dB default



Ring Tones

Syntax: c=on1/off1-on2/off2-on3/off3; [...]

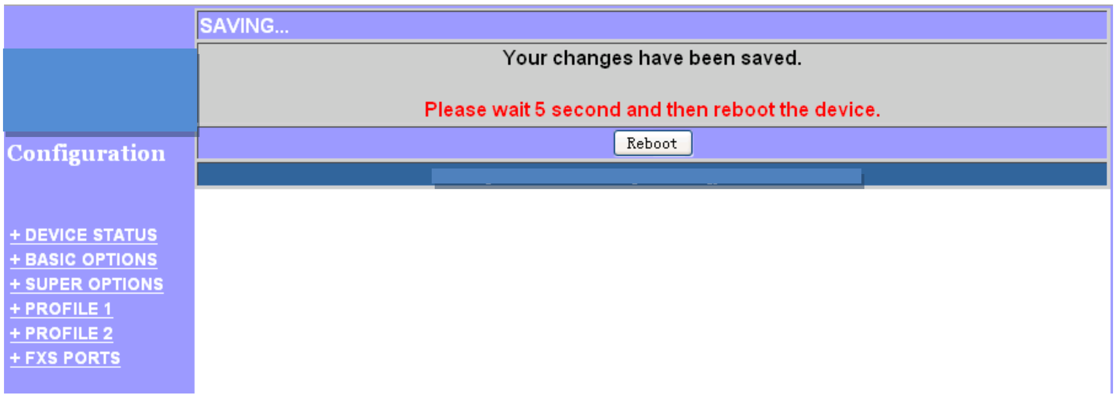
Ring Tone 1	c=2000/4000;
Ring Tone 2	c=2000/4000;
Ring Tone 3	c=2000/4000;
Ring Tone 4	c=2000/4000;
Ring Tone 5	c=2000/4000;
Ring Tone 6	c=2000/4000;
Ring Tone 7	c=2000/4000;
Ring Tone 8	c=2000/4000;
Ring Tone 9	c=2000/4000;
Ring Tone 10	c=2000/4000;

PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
Force INVITE	Default is NO
Special Feature	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port.
Caller ID Scheme	Select the Caller ID Scheme to suit the standard of different area. • Bellcore (North America) • ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA) • ETSI-DTMF (Finland, Sweden) • DTMF (Denmark)
On hook Voltage	Select the on hook voltage to suit different area or PBX
Polarity Reversal	Select Polarity Reversal to adapt some call charge/billing system. Default is No.
Hook Flash Timing	Time period when the cradle is pressed (Hook Flash) to simulate FLASH. To prevent unwanted activation of the Flash/Hold and automatic phone ring-back, adjust this time value.
Volume Amplification	Handset volume adjustment. RX is for receiving volume, TX is for transmission volume. Default values are 0dB for both parameters. +6dB generates the highest volume and -6dB generates the lowest volume.
Ring Tones	This function lets you configure ring tone cadence preferences. User has 10 choices. The configuration, completed in Distinctive Ring Tones block in the same page, applies to ring tones cadences configured here.

7.7 Saving the Configuration Changes

Once a change is made, press the “Update” button in the Configuration Menu. The following screen will confirm that the changes have been saved. To activate changes, reboot or power cycle the SIP 60X after changes are made.

7.8 Figure 4: Screen-Shot Of Save Configuration Page



Your changes have been saved.

Please wait 5 second and then reboot the device.

7.9 Rebooting From Remote

The administrator can remotely reboot the unit by pressing the “Reboot” button at the bottom of the configuration menu. The user can re-login to the unit after waiting for about 30 seconds.

7.10 Figure 5: Screen-Shot of Rebooting Page

Reboot in progress...
You may login after 30 seconds by clicking the link below.

8 SOFTWARE UPGRADE

To upgrade software, SIP 60X can be configured with a TFTP server where the new code image is located. The TFTP upgrade can work in either static IP or DHCP mode using private or public IP address. It is recommended to set the TFTP server address in either a public IP address or on the same LAN with the SIP 60X.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the SIP 60X’s Web configuration interface. To configure the TFTP server via voice prompt, follow section 5.1 with option 06, once set up the TFTP IP address, power cycle the ATA, the firmware will be fetched once the ATA boots up.

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the SIP 60X. Input the admin password to enter the configuration screen. From there, enter the TFTP server address in the designated field towards the bottom of the configuration screen.

Once the TFTP server is configured, please power cycle the SIP 60X.

TFTP process may take as long as 1 to 2 minutes over the Internet or just 30+ seconds if it is performed on a LAN. Users are recommended to conduct TFTP upgrade in a controlled LAN environment if possible.

NOTES:

When SIP 60X boots up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is “cfg.bin” and the other is “cfg001fc1xxxxxx”, where “001fc1xxxxxx” is the MAC address of the SIP 60X. These two files are for initial automatically provisioning purpose only, for normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.

9 RESTORE FACTORY DEFAULT SETTINGS

WARNING! Restoring the Factory Default Setting will DELETE all configuration information of the phone.

Please BACKUP or PRINT out all the settings before you approach to following steps. Netronix will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.

FACTORY RESET

There are two (2) methods for resetting your unit:

Reset Button

Reset default factory settings following these four (4) steps:

1. Unplug the Ethernet cable.

2. Locate a needle-sized hole on the back panel of the gateway unit next to the power connection.

3. Insert a pin in this hole, and press for about 8 seconds.

4. Take out the pin. All unit settings are restored to factory settings.

IVR Command

Reset default factory settings using the IVR Prompt (Table 5):

1. Dial “***” for voice prompt.
2. Enter “99” and wait for “reset” voice prompt.
3. Enter 862584658050

NOTE:

1. Factory Reset will be disabled if the **“Lock keypad update”** is set to “Yes”.

2. Please be aware by default the SIP 60X WAN side HTTP access is disabled. After a factory reset, the device’s web configuration page can be accessed only from its LAN port.