User Manual

Unicorn 610X

Analog IP Gateway

2FXS&2FXO Ports OR 4FXS&4FXO Ports

Hanlong Technology Co., Ltd

http://www.hanlongtek.com

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1 WELCOME

Thank you for purchasing the Hanlong Unicorn 610x Analog FXS IP Gateway. The Unicorn 610x offers an easy to manage, easy to configure IP communications solution for any business with virtual and/or branch locations. The Unicorn 610x supports popular voice codecs 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 Unicorn 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 Unicorn 610x Overview

The new Unicorn 610x 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 Unicorn 610x 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 codecs including G.711 (a/u-law), G.723.1, G.726(16/24/32/48 bit rates), G.729A/B/E and iLBC(Pending).

Caution: Changes or modifications to this product not expressly approved by Hanlong Technology, 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 Unicorn 610x 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 Unicorn 610x package. Using an alternative power adapter may permanently damage the unit.

1.3 Warranty

Hanlong has a reseller agreement with our reseller customer. End users should contact the company from whom you purchased the product for replacement, repair or refund.

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

2 CONFIGURE YOUR UNICORN 610X

Connecting your Unicorn 610x is easy. Before you begin, please verify the contents of the Unicorn 610x package.

2.1 Equipment Packaging

Unpack and check all accessories. The Unicorn 610x package contains:

- One Unicorn 610x VoIP adapter
- One universal power supply
- One Ethernet cable

2.2 Connect The Unicorn 610x

Managing the Unicorn 610x gateway and connecting the unit to the VoIP network is very simple. Follow these four (4) steps to connect your Unicorn 610x gateway to the Internet and access the unit's configuration pages.

1. Connect standard touch-tone analog phones to the FXS1-FXS4 ports.

2. Insert the Ethernet cable into the WAN port of Unicorn 610x 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 Unicorn 610x for initial configuration or if it is being used as a router.

4. Plug the power adapter into the Unicorn 610x and into a power outlet.

2.3 Unicorn 610x Back Panel

Figure 1: Diagram of Unicorn 610x Back Panel

TABLE 1: Definitions Of The Unicorn 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 – FXS4	FXS port to be connected to analog phones / fax machines
FXO1 – FXO4	FXO ports to be connected to physical PSTN lines from a traditional PSTN PBX or PSTN Central Office.

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

2.4 Unicorn 610x Display Panel

Figure 2: Diagram Of Unicorn 610x Display Panel

TABLE 2: Definitions Of The Unicorn 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:

- Flast 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.

3 UNICORN 610x FEATURES

The Unicorn 610x is a next generation IP voice gateway that is interoperable and compatible with leading IP-PBXs, SoftSwitches and SIP platforms. The Unicorn 610x series is auto-configurable, remotely manageable and scalable. There are two models, the Unicorn 610X, each offering superb voice quality, traditional telephony functionality, easy deployment, and 2 or 4 FXS&FXO 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.

3.1 Software Features Overview

- 2 or 4 FXS ports
- 2 or 4 FXO ports
- Two RJ-45 ports (switched or routed)
- Multiple SIP accounts & profiles (2 or 4 accounts / choice of 2 profiles per account)
- Supports Voice Codecs: G711(a/μ, Annex I & II), G723.1A, G726 (ADPCM with 16/24/32/40 bit rates), G729 A/B/E, Ilbc(Pending)
- 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: Unicorn 610x SOFTWARE FEATURES

	Unicorn 610x FXS Analog Gateway Series
Telephone Interfaces	Unicorn 6104: 4 ports, 4 SIP accounts & choice of 2

	profiles Unicorn 6108: 8 ports, 8 SIP accounts & choice of 2 profiles FXS/FXO, 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
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

3.2 Hardware specification

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

TABLE 4: Hardware Specification Of Unicorn 610x

Ports	2 or 4 FXS/FXO Ports
LAN interface	2 x RJ45 10/100Mbps (switched or routed)
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	0.85 kg
Temperature	32~104°F / 0~40°C
Humidity	10% - 90% (non-condensing)
Compliance	FCC, CE

4 CALL FEATURES

The Unicorn 610x 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 CallerID** (for all-config change)
*31	**Send CallerID** (for all-config change)
*67	**Block CallerID** (per call)
*82	**Send CallerID** (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.
***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.

5 BASIC OPERATIONS

5.1 Understanding Unicorn Voice Prompts

Unicorn 610x 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 The Unicorn 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
06	“MAC Address”	Announces the Mac address of the unit.
07	Preferred Vocoder	Enter “9” to go to the next selection in the list: ● PCM U ● PCM A ● G-726 ● G-723 ● G-729
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 can not 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 Unicorn610x and other VoIP Device, have public IP addresses, or

2. Both Unicorn610x and other VoIP Device are on the same LAN using private IP addresses, or

3. Both Unicorn610x and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

Unicorn610x 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 can not 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 Unicorn 610x 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 Unicorn 610x supports Bellcore style 3-way Conference.

Instructions for 3-way conference:

Assuming that call party A and B are in conversation. A (Unicorn 610x) 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 Unicorn 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 4 FXO trunks can be connected to 4 Unicorn 6108 ports configured as a hunt group. The Unicorn 6108 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 Unicorn 6108 will allow 4 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 entered, Hunting group set to "**Active**"
FXS Port #4: SIP UserID and Authenticate ID left blank, Hunting Group set to "**3**"

Please be aware, the choice of 1 for ports 2 , the choice of 3 for ports 4 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 tied to SIP Account configured in Port #1 marked as **"Active"**, and ports 4 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**"

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 Unicorn 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.

Note: The Unicorn 600x support fast mode for inter-port calling by using "*48x". For example PORT 1 call PORT 2, just dail "*482", PORT 2 call PORT 1, just need to dail "*481".

5.9 Sending And Receiving Fax

Unicorn 610x 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 CONFIGURATION GUIDE

6.1 Configuration With Web Browser

The Unicorn 610x has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the gateway through any common web browser.
The GUI interface can be download at bellow:
http://www.hanlongtek.com/books/6108_gui.rar

This gui include bellow pages:
1. SCREENSHOT OF DEVICE STATUS PAGE
2. SCREENSHOT OF BASIC OPTIONS PAGE
3. SCREENSHOT OF SUPPER OPTIONS PAGE
4. SCREENSHOT OF PROFILE 1 (FXS) PAGE
5. SCREENSHOT OF PROFILE 2 (FXS) PAGE
6. SCREENSHOT OF PROFILE 3 (FXO) PAGE
7. SCREENSHOT OF PROFILE 4 (FXO) PAGE
8. SCREENSHOT OF FXS PORTS PAGE
9. SCREENSHOT OF FXO PORTS PAGE

The Unicorn 610x 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 Unicorn 610x using voice prompt menu option 02.

3. Access the Unicorn 610x Web Configuration page by the following URI via WAN port:

http:// Unicorn 610x -IP-Address (the Unicorn 610x IP-Address is the WAN IP address for the Unicorn 610x).

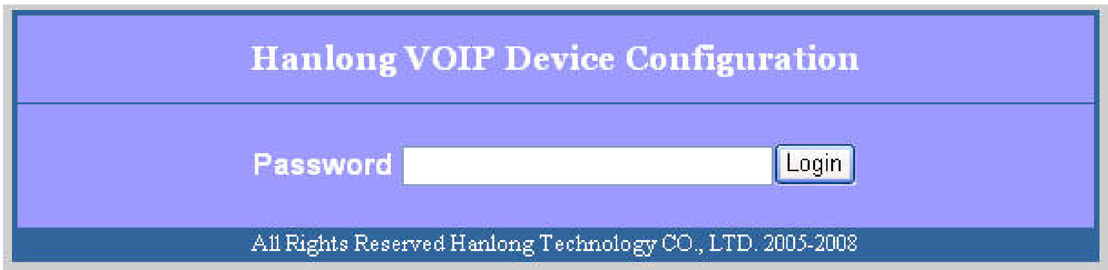
NOTE: To access the configuration page, type the Unicorn IP address into the browser, stripping out the leading “0” because the browser will parse in octet. e.g. if the IP address is: 192.168.001.014, please type in: 192.168.1.14.

6.2 End User Configuration

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

FIGURE 3: Screen-Shot Of Unicorn 610x Log-In Screen



After login, the next configuration page is the Basic Configuration page, explained in detail in Table 5 : Web Log-in Definition.

6.2.1 Basic Settings Page

BASIC OPTIONS SETTING	
Setting options	**Definitions**
Web Port	Default is 80.
IP Address	There are 3 modes under which the Unicorn 600x can operate: - If DHCP mode is enabled, then all the field values for the

	Static IP mode are not used (even though they are still saved in the chipset's memory). The Unicorn 600x will acquire its IP address from the first DHCP server it discovers from the office/home network it is connected to. -To use the PPPoE feature, the PPPoE account settings need to be set. The Unicorn 600x will attempt to establish a PPPoE session if any of the PPPoE fields have been entered with data. - If Static IP mode is enabled, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured by the user. These fields are reset to zero by default.
Time Zone	This parameter controls how the displayed date/time will be adjusted according to the specified time zone.
Allow DHCP Option 2 to override Time Zone setting	If set yes and under DHCP mode,the device will try to get option 2 from DHCP configure and overwrite 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.
Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Need save the setting and reboot the device before the setting start to work
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 120hr (5 Days) The time IP address are assigned to the LAN clients
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 unicorn600x 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.

6.2.2 Status Page Definitions

Device Status

Setting options	Definitions
MAC Address	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
WAN IP Address	This field shows IP address of device
Product Model	This product model is Unicorn600x
Software Version	Information of software
System Up Time	Shows system up time since the last reboot.
PPPoE Link Up	Indicates whether the PPPoE connection is up if the Unicorn600x is connected to DSL modem.
NAT	I Indicate the NAT type behind which the device is when the stun feature is defined.

Port Status

Shows several information regarding the individual FXS ports. Example for Unicorn6108.

Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
1(FXS)	On Hook	Registered	No	613		
2(FXS)	Off Hook	Registered	No		614	
3(FXS)	On Hook	Not Registered	No			
4(FXS)	On Hook	Registered	Yes			615
5(FXO)	On Hook	Registered				
6(FXO)	On Hook	Registered				
7(FXO)	On Hook	Registered				
8(FXO)	On Hook	Registered				

FXS port 4 user has set Do Not Disturb.

FXS port 1 user has set his calls to be forwarded unconditionally to ext 613

FXS port 2 user has set his calls to be forwarded to 614 when his phone is busy.

FXS port 3 user is not registered with his SIP Server.

FXO port1~port4 all registered.

6.3 Super User Settings

The end-user needs to login to the Super user configuration page the same way as for the basic configuration page.

6.3.1 Super Configuration Page Definitions

Super Options	
Setting options	**Definitions**
Admin Password	This contains the password to access the Advanced Web Configuration page. This field is case sensitive.

	Only the administrator can configure the “Advanced Settings” page. Password field is purposely left blank for security reasons after clicking update and saved. The maximum password length is 26 characters,only digit or letter.
Home NPA	Local area code for North American Dial Plan.
Layer3 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.
Layer2 Qos	Value used for layer 2 VLAN tag. Default setting is blank
Data VLAN Tag	When using Bridge Mode, Data VLAN Tag is supported. when your PC connect to LAN Port, data (from your PC to switch) will be tagged with "Data VLAN Tag".
Stun sever is:	IP address or Domain name of the STUN server.
Keep-alive interval	This parameter specifies how often the Unicorn600x sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open. Default is 20 seconds. Minimum value is 20 seconds.
Firmware Upgrade and Provisioning:	Upgrade or provisioning through **TFTP** or **TFTP** server. --Upgrade Via:select HTTP or TFTP mode. --Allow DHCP Option: support 66,128,150. If select yes,device will get server information from DHCP option and ignore the Config Server Path. Option 66--TFTP server name(if you select Upgrade Via->TFTP), HTTP server name(if you select Upgrade Via->HTTP) Option 128--TFPT Server IP address.(if you select Upgrade Via->TFTP), HTTP Server IP address(if you select Upgrade Via->HTTP) Option 150--TFTP server address.(if you selectUpgrade Via->TFTP), HTTP server address(if you select Upgrade Via->HTTP)
Authenticate Conf File	configure file would be authenticated before acceptance if set to Yes
NTP server	This parameter defines the URI or IP address of the NTP server which is used by the Unicorn 600x to set the current date/time.
Allow DHCP Option 42 to override NTP server	If set **Yes**,device can get NTP server from DHCP option 42.
Lock Keypad Update	If set to “Yes”, the configuration update via keypad is disabled.

Disable Voice Prompt	Default is No
Syslog Sever	The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)
Syslog level	Default is blank, the feature is useful for the Internet Telephone Service Provider.
Download Device Configuration	User can download configuration from the web page and save to configuration file.
Call Progress Tones	Using these settings, users can configure tone frequencies and cadence according to their preference. By default they are set to North American frequencies. Configure these settings 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. Example configuration for N.A. Dial tone: f1=350@-13,f2=440@-13,c=0/0; Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...] (Note: freq: 0 - 4000Hz; vol: -30 - 0dBm)
Restore Configuration	User can restore the before configuration from the configuration file saved at local pc

6.3.2 Configuring The FXS Channels

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 Unicorn 600x to register to (SIP) servers of ITSP.
Name	Name
Profile ID	Select the corresponding Profile ID (1/2)
Hunting Group	This feature enables the gateway to register all existing FXS ports with the same phone number.

	Each incoming call will be routed to first available port in round robin mode. User may configure all ports as members of the same Hunt Group or it may configure different port combinations for more than one Hunt Group. For example: Ports 1, 3 and 5 may be members of the same hunt group, the rest of the ports may have separate numbers and may be reached independently. Select appropriate value for Hunting Group feature. The original SIP account should be set to Active while the group members should be set to the port number of the Active Port. 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.
Off hook Auto-dial	This field default is blank. If set a number, it is mean that according port will auto outgo the setting number when this port off hook.

6.3.3 FXS Profile

PROFILE PAGE DEFINITIONS	
Settings Options	**Definitions**
Account active	When set to Yes this profile is activated

SIP Server	SIP Server’s URI or IP address
Outbound Proxy	SIP Outbound Proxy Server’s URI or IP address
NAT Traversal	This parameter defines whether the Unicorn 600x NAT traversal mechanism will be activated or not. If Choosing **No**,nothing to do. If Choosing **No, but send keep-alive**, the Unicorn 600x 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 choosing **STUN** and a STUN server is also specified, then the Unicorn 600x will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the Unicorn 600x will attempt to detect if and what type of firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the Unicorn 600x will attempt to use its mapped public IP address and port in all its SIP and SDP messages. If choosing **STUN** with no specified STUN server, the Unicorn 600x 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 choosing **UPNP**, the embedded UPNP client inside the Unicorn 600x will attempt to mapping ports with the router by upnp protocol.
Use DNS SRV	Default is No. If set to Yes the client will use DNS SRV for server lookup
User ID is Phone Number	If the Unicorn 600x 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 Unicorn 600x needs to send REGISTER messages to the proxy server. The default setting is “Yes”.
Unregister on Reboot	Default is “No.” If set to “Yes”, then the SIP user will be unregistered on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the Unicorn 600x refreshes 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.even when not registered (if allowed by ITSP) but is unable to receive incoming calls.
Local SIP port	This parameter defines the local SIP port the Unicorn 600x will listen and transmit. The default value for FXS port 1 is 5060. The default value for FXS 2 port is 5062.
Local RTP port	This parameter defines the local RTP-RTCP port pair the Unicorn 600x 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 and port_value+3 for its RTCP. The default value for FXS port 1 is 5004. The default value for FXS 2 port is 5008.
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 Unicorn 600x 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	This parameter sets the payload type for DTMF using RFC2833
DTMF in Audio	This parameter specifies the mechanism to transmit DTMF digit in audio which means DTMF is combined in audio signal(not very reliable with low-bit-rate codec), Default is **YES**.
DTMF via RFC2833	This parameter specifies the mechanism to transmit DTMF digit via RTP (RFC2833). Default **YES**.
DTMF via SIP INFO	This parameter specifies the mechanism to transmit DTMF digit via SIP INFO. Default is **NO**.
Send Flash Event	This parameter allows users to control whether to send an SIP NOTIFY message indicating the Flash event, or just to switch to the voice channel when users press the Flash key.
Enable Call Features	Default is **No**. If set to **Yes**, Call Forwarding & Do-Not-Disturb are supported locally
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 No.
Preferred Vocoder	The Unicorn600x supports up to 5 different Vocoder types including G.711 A-/U-law, G.726 (Supports bit rates 32K), G.723.1, G.729A/B. 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 6”.
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 Unicorn 600x 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	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.

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	Default is **T.30**(Fax Pass-Through), or T.38 (Auto Detect) FoIP
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.
Distinct Ring Tone	Caller ID must be configured. Select a Distinctive Ring Tone 1 through 3 for a particular Caller ID. The device will ONLY use selected ring tones for particular Caller IDs. For all other calls, the device will use System Ring Tone. When selected and no Caller ID is configured, the selected ring tone will be used for all incoming calls..
Use Bell-style 3-way Conference	Conference mode, default option is No. If set to yes, the feature code for coference *23 would be disabled.
Disable Call -Waiting	Default is **NO**. User can use star codes to enable/disable call waiting.
Disable Call –Waiting Tone	Default is **NO**. This is to disable the stutter call waiting tone when a call waiting call arrived.
Ring Timeout	An incoming call will stop ringing when not picked up given a specific period of time.
No Key Entry Timeout	Default is 4 seconds.
Use # as Send Key	T This parameter allows users to configure the **"#"** key to be used as the **Send** (or **Dial**) key. If set to **Yes**, pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this key is essentially equivalent to the **Re(Dial)** key. If set to **No**, this **"#"** key will then be included as part of the dial string to be sent out.
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*]+ }. More information can be availabled at Dail Plan Notes.
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 in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Check SIP User ID for incoming INVITE	Check the SIP User ID in Request URI. If they don't match, the call will be rejected.
Session Expiration	The session timer extension enables SIP sessions to be periodically "refreshed" via a re-INVITE request. Once the session interval expires, if there is no refresh via a re-INVITE message, the session will be terminated.

	Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is 180 seconds. Default is 180 seconds.
Min-SE	The minimum session expiration (in seconds). Default is **90** seconds.
Caller Request Timer	If selecting "**Yes**" the device will use session timer when it makes outbound calls if remote party supports session timer. Default is **NO.**
Callee Request Timer	If selecting "**Yes**" the phone will use session timer when it receives inbound calls with session timer request. Default is **NO**.
Force Timer	If selecting "**Yes**" the device will use session timer even if the remote party does not support this feature. Selecting "**No**" will allow the device to enable session timer only when the remote party support this feature. To turn off Session Timer, select "**No**" for Caller Request Timer, Callee Request Timer, and Force Timer. Default is **NO**.
UAC Specify Refresher	As a Caller, select UAC to use the device as the refresher, or UAS to use the Callee or proxy server as the refresher. Default is **Omit**
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the device as the refresher. Default is **UAC**.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select "**Yes**" to use INVITE method to refresh the session timer. Default is **NO.**
Special Feature	Choose the selection to meet some special requirements from Soft Switch vendors. Default is **standard.**
FXS Impedance	Select the impedance of anolog telephone connected to phone port
Caller ID Scheme	select caller ID to suit standard of different area.
Onhook Voltage	Select onhook voltage to suit standard of 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	Voice path volume adjustment. Rx is a gain level for signals transmitted by FXS Tx is a gain level for signals received by FXS. Default = 0dB for both parameters. Loudest volume: +6dB Lowest volume: -6dB. User can adjust volume of call on either end using the Rx Gain Level parameter and the Tx Gain Level parameter located on the FXS Port Configuration page. If call volume is too low when using the FXS port (ie. the ATA is at user site), adjust volume using the Rx Gain Level parameter under the FXS Port Configuration page. If voice volume is too low at the other end, user may increase the far end volume using the Tx Gain Level parameter under the FXS Port Configuration page.
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.

6.3.4 Configuring The FXO Channels

FXS PORT SETTING	
Setting Options	**Meaning**
FXO 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 Unicorn 600x to register to (SIP) servers of ITSP.
Name	Name
Profile ID	Select the corresponding Profile ID (1/2)
Unconditional Call Forward to VOIP	Calls are unconditionally forwarded to the specified VoIP phone number once users dial the FXO port PSTN number
Enable PSTN Disconnect Tone	If set to **Yes**, arrived Busy Tone is used as the disconnect signal.

Detection	
PSTN Disconnect Tone	This configuration should be configured by the VoIP service provider. Some country use single frequency tone to signal PSTN disconnection, some country use double frequency tone. This setting can be configured to suit the telephone company's standard in different country.
Enable Polarity Reversal Disconnect	If set to **Yes**, the Polarity Reversal is used as the disconnect signal.
Enable Terminate Call After PSTN Silence Timeout	If set **Yes**, the device terminate the call when Silence timer expire.
PSTN Silence Timeout	Silence timer value, default is 60 minutes.
Number of Rings	Number of rings for a PSTN incoming call to FXO port before FXO port picks up, default 2
Min Delay Before Dial PSTN(ms)	Default is 500ms. This needs to be equal to or greater than the Current Disconnect threshold setting. Once the threshold is reached the gateway can dial out. This parameter should only be used if there are PSTN line detection issues.
DTMF Digit Volume(dB)	Default value is 11dB.
DTMF Digit Length(x10ms)	Digit length and Dial Pause are port digit dialing configurations; FXO needs to dial out digits for VOIP to PSTN 1 stage calls, and unconditional call forward to PSTN, and route to PSTN. Digit Length is the play time for each digit. Note: In order to receive the caller ID information, the delay should be set to a value larger than the delay required to complete the PSTN caller ID delivery. Please note that the value will be multiplied by 10ms.
DTMF Dial Pause(x10ms)	Dial pause is the time between 2 digits for the same scenario as explained above. Please note that the value will be multiplied by 10ms.
Stage Method(1/2)	This configuration is applicable for VoIP to PSTN calls and indicates one or two stage dialing methods.
Unconditional Call Forward to PSTN	Calls are unconditionally forwarded to the specified PSTN phone number once users dial the FXO port VoIP number. Each port can be setted independence.

6.3.5 FXO Profiles

PROFILE PAGE DEFINITIONS	
Settings Options	**Definitions**
Account active	When set to Yes this profile is activated
SIP Server	SIP Server's URI or IP address
Outbound Proxy	SIP Outbound Proxy Server's URI or IP address
NAT Traversal	This parameter defines whether the Unicorn 600x NAT traversal mechanism will be activated or not. If Choosing **No**,nothing to do. If Choosing **No, but send keep-alive**, the Unicorn 600x 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 choosing **STUN** and a STUN server is also specified, then the Unicorn 600x will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the Unicorn 600x will attempt to detect if and what type of firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the Unicorn 600x will attempt to use its mapped public IP address and port in all its SIP and SDP messages. If choosing **STUN** with no specified STUN server, the Unicorn 600x 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 choosing **UPNP**, the embedded UPNP client inside the Unicorn 600x will attempt to mapping ports with the router by upnp protocol.
Ports Using The Profile Share With One Common Account	If set to "**Yes**", Unicorn60x0 will use the first account among the FXO PORTS of using the same profile. If set to "**No**", you need configure one port one account in FXO PORTS page.
Auto Select Idle Port (For Outgoing Call)	If set to "**Yes**", Unicorn60x0 will auto-select an idle Line to make outbound call to PSTN. If set to "**No**", you need configure one port one account in FXO PORTS page, then it is the business of SIP SERVER that it decide which idle FXO Port for outbound call.
Use DNS SRV	Default is No. If set to Yes the client will use DNS SRV for server lookup
User ID is Phone	If the Unicorn 600x has an assigned PSTN telephone

Number	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 Unicorn 600x needs to send REGISTER messages to the proxy server. The default setting is “Yes”.
Unregister on Reboot	Default is “No.” If set to “Yes”, then the SIP user will be unregistered on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the Unicorn 600x refreshes 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.even when not registered (if allowed by ITSP) but is unable to receive incoming calls.
Local SIP port	This parameter defines the local SIP port the Unicorn 600x will listen and transmit. The default value for FXS port 1 is 5060. The default value for FXS 2 port is 5062.
Local RTP port	This parameter defines the local RTP-RTCP port pair the Unicorn 600x 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 and port_value+3 for its RTCP. The default value for FXS port 1 is 5004. The default value for FXS 2 port is 5008.
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 Unicorn 600x 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	This parameter sets the payload type for DTMF using RFC2833
DTMF in Audio	This parameter specifies the mechanism to transmit DTMF digit in audio which means DTMF is combined in audio signal(not very reliable with low-bit-rate codec), Default is **YES**.
DTMF via RFC2833	This parameter specifies the mechanism to transmit DTMF digit via RTP (RFC2833). Default **YES**.

DTMF via SIP INFO	This parameter specifies the mechanism to transmit DTMF digit via SIP INFO. Default is **NO**.
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.
Preferred Vocoder	The Unicorn600x supports up to 5 different Vocoder types including G.711 A-/U-law, G.726 (Supports bit rates 32K), G.723.1, G.729A/B. 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 6”.
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 Unicorn 600x 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	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.
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	Default is **T.30**(Fax Pass-Through), or T.38 (Auto Detect) FoIP
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.
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*]+ }. More information can be availabled at Dail Plan Notes.
Send Anonymous	If this parameter is set to "Yes", the "From" header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Session Expiration	The session timer extension enables SIP sessions to be periodically "refreshed" via a re-INVITE request. Once the session interval expires, if there is no refresh via a re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is 180 seconds. Default is 180 seconds.
Min-SE	The minimum session expiration (in seconds). Default is **90** seconds.
Caller Request Timer	If selecting "**Yes**" the device will use session timer when it makes outbound calls if remote party supports session timer. Default is **NO.**
Callee Request Timer	If selecting "**Yes**" the phone will use session timer when it receives inbound calls with session timer request. Default is **NO**.
Force Timer	If selecting "**Yes**" the device will use session timer even if the remote party does not support this feature. Selecting "**No**" will allow the device to enable session timer only when the remote party support this feature. To turn off Session Timer, select "**No**" for Caller Request

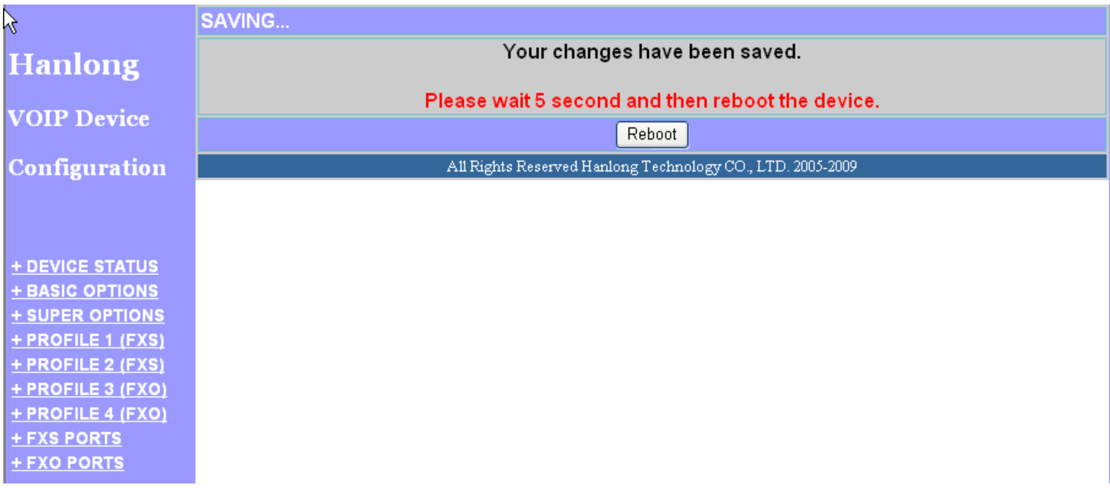
	Timer, Callee Request Timer, and Force Timer. Default is **NO**.
UAC Specify Refresher	As a Caller, select UAC to use the device as the refresher, or UAS to use the Callee or proxy server as the refresher. Default is **Omit**
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the device as the refresher. Default is **UAC**.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select "**Yes**" to use INVITE method to refresh the session timer. Default is **NO.**
Send 200 OK Until FXO Has Detected Polarity Reversal	Default No. Check with your PSTN carrier before set to Yes
FXO Pick Up Incoming Call After Receive 200 OK Form Server	Default No.
Special Feature	Choose the selection to meet some special requirements from Soft Switch vendors. Default is **standard.**
Volume Amplification	Voice path volume adjustment. Rx is a gain level for signals transmitted by FXS Tx is a gain level for signals received by FXS. Default = 0dB for both parameters. Loudest volume: +6dB Lowest volume: -6dB. User can adjust volume of call on either end using the Rx Gain Level parameter and the Tx Gain Level parameter located on the FXS Port Configuration page. If call volume is too low when using the FXS port (ie. the ATA is at user site), adjust volume using the Rx Gain Level parameter under the FXS Port Configuration page. If voice volume is too low at the other end, user may increase the far end volume using the Tx Gain Level parameter under the FXS Port Configuration page.
PSTN AC Termination	You can select the AC termination by Country or by Impedance.
Caller ID Scheme	select caller ID to suit standard of different area.
Caller ID Minimum RX Level (dB)	An adjustable value for the Caller ID signal to help this device to recognize Caller ID from different networks. (-50 -0dB. Default -30dB)
Caller ID Transport	According to customer's choice CID information will be

Type	transferred from PSTN network to VoIP network using following rules: 1. via SIP from - PSTN CID is in the SIP From field 2. via P-Asserted-Identity - SIP From field uses the pre-configured account user Id. PSTN CID is in the P-Asserted-Identity field 3. Send anonymous - SIP From field uses "anonymous". PSTN CID is put in the P-Asserted-Identity field 4. Disable - PSTN CID will not be sent. SIP From field uses the pre-configured account user ID
PIN for PSTN Calls	Enter digits to authorize calling PSTN numbers from VOIP, default is no.
PIN for VOIP Calls	Enter digits to authorize calling VOIP terminals from PSTN, default is no.

6.4 Saving The Configuration Changes

Once a change is made, press the “Update” button in the Configuration Menu. The Unicorn 610x will display the following screen to confirm that the changes have been saved. To activate changes, reboot or power cycle the Unicorn 610x after all changes are made.

FIGURE 5: Screen-Shot Of Save Configuration



6.5 Rebooting From Remote

The administrator can remotely reboot the unit by pressing the “Reboot” button at the bottom of the configuration menu. The following screen will indicate that rebooting is underway.

FIGURE 6: Screen-Shot Of Rebooting

The user can re-login to the unit after waiting for about 30 seconds.

7 SOFTWARE UPGRADE

To upgrade software, Unicorn 610x 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 Unicorn 610x.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the Unicorn 610x'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 Unicorn 610x. 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 Unicorn 610x.

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. For those who do not have a local TFTP server, Hanlong technology provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Service section of Hanlong's Web site to obtain this TFTP server's IP address.

NOTES:

When Hanlong ATA boot 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 Unicorn 610x. 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.

8 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. Hanlong 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 Unicorn 610x WAN side HTTP access is disabled. After a factory reset, the device's web configuration page can be accessed only from its LAN port.

9 TECHNICAL SUPPORT CONTACT

Email: Support@mail.hanlongtek.com