

# User Manual

## Unicorn 3112

### Analog Telephone Adaptor



**Hanlong Technology Co., Ltd**

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## 1. WELCOME

Unicorn 3112 is an all-in-one VoIP integrated access device that features superb audio quality, rich functionalities, high level of integration, compactness and ultra-affordability. The Unicorn 3112 is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

Special compatibility features include:

- Nortel MCS
- Standard SIP
- Broadsoft
- Howdy

## 2. WHAT IS IN THE PACKAGE

The Unicorn 3112 package contains:

- One Unicorn 3112 VoIP adapter
- One universal power supply
- One Ethernet cable
- One phone cable

## 3. PRODUCT OVERVIEW

### 3.1. Key Features

- Supports SIP 2.0(RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP (both client and server), NTP, PPPoE, STUN, TFTP, etc.
- Built-in router, NAT, Gateway and DMZ port forwarding

- Supports call origination and termination from/to the PSTN network(via FXO Port)
- Powerful digital signal processing (DSP) to ensure superb audio quality; advanced adaptive jitter control and packet loss concealment technology
- Support various vocoders including G.711 (a-law and u-law), G.723.1 (5.3K/6.3K), G.726 (40K/32K/24K/16K), as well as G.728, G.729A/B, and iLBC(Pending).
- Support Caller ID/Name display or block, Hold, Call Waiting/Flash, Call Transfer, Call Forward, in-band and out-of-band DTMF, Dial Plans, etc.
- Support fax pass through T.30 and T.38.
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Support standard encryption and authentication (DIGEST using MD5 and MD5-sess)
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support automated NAT traversal without manual manipulation of firewall/NAT
- Support device configuration via built-in IVR, Web browser or central configuration file through TFTP or HTTP
- Support firmware upgrade via TFTP or HTTP with encrypted configuration files. Ultra compact (wallet size) and lightweight design, great companion for travelers
- Compact, lightweight Universal Power adapter.

## 3.2. Hardware specification

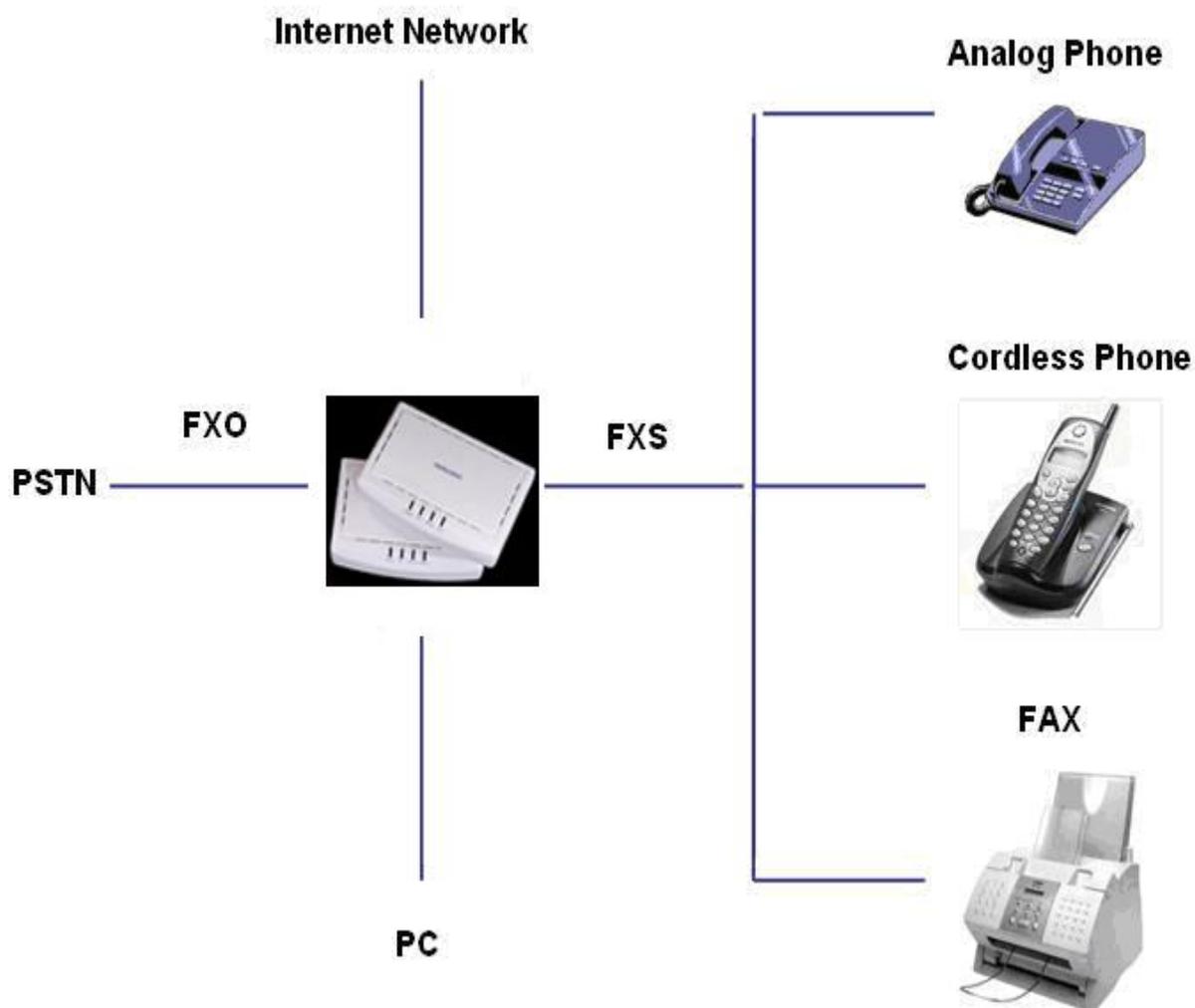
Model	Unicorn 3112
<b>LAN interface</b>	1 x RJ45 100MBase-T
<b>WAN interface</b>	1 x RJ45 100MBase-T
<b>FXS telephone port</b>	1 x FXS
<b>FXO port</b>	1 x FXO
<b>LED light</b>	Green and red color
<b>Universal switching power supply</b>	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA, UL certified
<b>Dimension</b>	70mm (W) × 130mm (D) × 27mm (H)
<b>Weight</b>	0.30kg
<b>Temperature</b>	40 – 130 F 5 – 45 C
<b>Humidity</b>	10 - 90%

## 4. INSTALLATIONS

The Unicorn 3112 is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services. The Unicorn 3112 VoIP features are available when you connect any regular analog telephone to it.

Unicorn 3112 has one FXS port (labeled “Phone”) and one PSTN pass through port (labeled “Line”). After setting up the Unicorn 3112, you can make PSTN calls by pressing \*00. Without pressing \*00, all your calls will be VoIP. You can also receive PSTN calls and VoIP calls.

The following photo illustrates the Interconnection Diagram of the of a Unicorn 3112:



Following are the steps to install a Unicorn 3112:

- Connect a standard touch-tone analog telephone to the “Phone” port.



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- Insert a standard RJ11 telephone cable (included with package) into the “Line” port and connect the other end of the telephone cable to a wall jack.
- Connect a PC to the LAN port of Unicorn 3112 (Ethernet cable is included with package).
- Insert another Ethernet cable into the WAN port of Unicorn 3112 and connect the other end of the Ethernet cable to an uplink port (a router, switch, hub, modem, etc)
- Insert the powers supply (included with package) into the Unicorn 3112 and connect it to a power outlet.

### 4.1. Safety

The Unicorn 3112 is compliant with various safety standards including FCC/CE and C-Tick. Its power adaptor is compliant with UL standard. The Unicorn 3112 should only operate with the universal power adaptor provided in the package.

**Warning:** Please do not use a different power adapter. Using other power adapter may damage the Unicorn 3112 and will void the manufacturer warranty!

**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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## 5. BASIC OPERATIONS

### 5.1. Get Familiar with Voice

Unicorn 3112 has stored a voice prompt menu for quick access to settings and simple configuration. You can enter this voice prompt menu as follows:

- Pick up the receiver (or press the Handsfree button) of the analog telephone and press “\*\*\*”

A voice will say, “Enter the new option.” At this point, you can select from the following menu voice prompt options to begin using the Unicorn 3112:

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 – 07, 12 - 17, 47, 86 or 99 Menu option
<b>01</b>	“DHCP Mode”, “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.
<b>02</b>	“IP Address “ + IP address	The current WAN IP address is announced Enter 12-digit new IP address if in Static IP Mode.
<b>03</b>	“Subnet “ + IP address	Same as Menu option 02
<b>04</b>	“Gateway “ + IP address	Same as Menu option 02
<b>05</b>	“DNS Server “ + IP address	Same as Menu option 02
<b>06</b>	“MAC Address”	Announces the Mac address of the unit.
<b>07</b>	Preferred Vocoder	Enter “9” to go to the next selection in the list: <ul style="list-style-type: none"> <li>➤ PCM U</li> <li>➤ PCM A</li> <li>➤ G-726</li> <li>➤ G-723</li> <li>➤ G-729</li> </ul>
<b>12</b>	WAN Port Web Access	Enter “9” to toggle between enable and disable
<b>13</b>	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
<b>14</b>	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
<b>15</b>	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Enter “9” to toggle between TFTP and HTTP
<b>16</b>	Firmware Version	Firmware version information.
<b>17</b>	Firmware Upgrade	Firmware upgrade mode. Enter “9” to rotate among the following three options: <ol style="list-style-type: none"> <li>1. always check</li> <li>2. check when pre/suffix changes</li> <li>3. never upgrade</li> </ol>
<b>47</b>	“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

Other Menu Prompt Features:

- "\*" shifts down to the next menu option
- "#" returns to the main menu
- "9" functions as the ENTER key in many cases to confirm an option
- All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. Once all of the digits are collected, the input will be processed.
- Incorrect keyed entry cannot be deleted or undone. The Unicorn 3112 will prompt you to start over by telling you that you made an error.

## 5.2. Make Phone call

### 5.2.1. Calling Phone or Extension Numbers

- Dial the number directly and wait for 4 seconds (Default is 4 seconds. To change the default, change the settings via the web configuration page under "No Key Entry Timeout"). Or
- Dial the number directly, and press # (assuming that "Use # as Dial Key" is set to "YES" during web configuration of your Unicorn 3112).

Other functions available during the call are call-waiting/flash, call-transfer, and call-forward. Your SIP gatekeeper/proxy server needs to support these features in order for them to work.

### 5.2.2. Call Hold

While in conversation, pressing the "FLASH" button on the attached phone will put the remote end on hold. Pressing the "FLASH" button again will release the previously Hold party and the bi-directional media will resume.

### 5.2.3. Call Waiting

If call waiting feature is enabled, while the user is in a conversation, he will hear a special

stutter tone if there is another incoming call. User can press the flash button to put the current call party on hold and switch to the other call. Pressing flash button toggles between two active calls.

### 5.2.4. 3-way Conferencing

Unicorn3112 supports 3-way conference in two styles: star code style or Bellcore style.

#### Star Code Style 3-way Conference

Assuming that call party A and B are in conversation. A 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 then C's number then # (or wait for 4 seconds).
3. If C answers the call, then A press "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.

#### Bellcore Style 3-way Conference

Bellcore style 3-way conference is also supported. To do this, user needs to enable "Use Bell-style 3-way Conference" in FXS web configuration.

Assuming that call party A and B are in conversation. A 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 C's number then # (or wait for 4 seconds).
3. If C answers the call, then A press "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.

### 5.2.5. Direct IP-to-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 Unicorn3112 and other VoIP Device, have public IP addresses, or
2. Both Unicorn3112 and other VoIP Device are on the same LAN using private IP addresses, or
3. Both Unicorn3112 and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

Unicorn3112 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”.

## 5.2.6. Blind Transfer

Assuming that call party A and party B are talking to each other on the phone. Party A wants to transfer party B to party C:

- Party A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
- Then party A dials \*87 then dials party C’s number, and then # (or wait for 4 seconds) Party A can hang up the phone.
- Note: Call features have to be activated during web configuration by selecting YES to “Enable Call Features”. These features need to be supported by your SIP gatekeeper/proxy server in order to work.
- Party A can hold on to the phone and wait for one of the three following events:
  1. A quick confirmation tone (temporarily using the call waiting indication tone) followed by a dial tone. This indicates the transfer is successful (transferee has received a 200 OK signal from transfer target). At this point, party A can either hang up or make another call.
  2. A quick busy tone followed by a restored call (on supported SIP gatekeeper platforms only). This means the transferee has received a 4xx response signal for the INVITE and will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
  3. Busy tone keeps playing. This means the Unicorn 3112 has failed to receive the

second NOTIFY signal from the transferee and decided to time out.

Note: this does not indicate the transfer has been successful, nor does it indicate the transfer has failed. When transferee uses a device that does not support the second NOTIFY signal, this will be the case. In poor or unstable network scenarios, this could also happen, although the transfer may have been completed successfully.

### 5.2.7. Attended Transfer

Assuming that call party A and party B are in conversation. Party A wants to Attend Transfer party B to party C:

- Party A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
- Party A then dials party C's number then # (or wait for 4 seconds). Party A and party C now are in conversation.
- Party A can hang

Note: When Attended Transfer failed and if party A hangs up, the Unicorn 3112 will ring party A again to remind party A that party B is still on the call, by pressing FLASH or Hook again will restore the conversation between party A and party B.

### 5.2.8. Send and Receive PSTN Calls

Users can send and receive calls from PSTN. To receive PSTN calls, simply take the phone off hook when the analog phone rings. To make a PSTN call, first press \*00 (or your own PSTN Access Code) to get the PSTN line dial tone and dial the PSTN number.

### 5.2.9. VoIP-to-PSTN Calls

To make a VoIP-to-PSTN call, users need to dial the FXO SIP account phone number first. A ring tone is played once followed by a dial tone. At this time, users can dial a PSTN telephone number or a mobile telephone number then # (or wait for 4 seconds). The call will be established afterwards. If no PSTN number is entered after the dial tone, Unicorn 3112 will hang up automatically in 10 seconds.

In the web configuration page, if the Route to PSTN field is configured, the second stage dialing is eliminated. That is, after users dial the FXO SIP account number, the PSTN number will be called automatically.

### 5.2.10. PSTN-to-VoIP Calls

To make a PSTN-to-VoIP call, PSTN callers need to originate a call to the FXO port

telephone number first. If no one answers the FXS phone after 4 (default value, can be configured) ring tones, a dial tone is played. At this time, users can dial a VoIP telephone number then # (or wait for 4 seconds). The call will be established afterwards. If no VoIP number is entered after the dial tone, Unicorn 3112 will hang up automatically in 10 seconds.

In the web configuration page, if the Route to VoIP field is configured, the second stage dialing is eliminated. That is, after users dial the FXO port telephone number, the VoIP number will be called automatically.

## 5.2.11. Route Calls to PSTN

If configured, certain calls will be routed to PSTN line automatically. This call feature is especially useful for emergency calls or local telephone calls. To use this feature, users need to specify a prefix or a telephone number in the Route to PSTN field in the web configuration page. If the dialed digits match one of the specified prefix, outbound calls will be routed to PSTN port.

## 5.3. Call Features

### 5.3.1. Call Features Tables

Following table shows the call features of Unicorn 3112:

Key	Call Features
*23	3-way conference
*87	Blind Transfer
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*67	Block Caller ID (per call)
*82	Send Caller ID (per call)
*50	Disable Call Waiting (for all subsequent calls)
*51	Enable Call Waiting (for all subsequent calls)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward. To use this feature, dial “*72” and get the dial tone. Then dial the forward number and “#” for a dial tone, then hang up.
*73	Cancel Unconditional Call Forward. To cancel “Unconditional Call Forward”, dial “*73” and get the dial tone, then hang up.
*90	Busy Call Forward. To use this feature, dial “*90” and get the dial tone. Then dial the forward

	number and “#” for a dial tone, then hang up.
<b>*91</b>	Cancel Busy Call Forward. To cancel “Busy Call Forward”, dial “*91” and get the dial tone, then hang up.
<b>*92</b>	Delayed Call Forward. To use this feature, dial “*92” and get the dial tone. Dial the forward number and “#” for a dial tone and then hang up.
<b>*93</b>	Cancel Delayed Call Forward. To cancel this feature, dial “*93”, get the dial tone, and then hang up.
<b>Flash/Hook</b>	call waiting indication. When in conversation without an incoming call, this action will switch to a new channel to make a new call.

## 5.3.2. PSTN Pass Through

When Unicorn 3112 is out of power or loses registration or if the network connection is down, the RJ 11 line jack on the side of Unicorn 3112 will function as a pass through connection for PSTN calls. Users will be able to use the same analog phone for PSTN calls.

## 5.4. FAX

Unicorn3112 supports FAX in two modes: T.38 (Fax over IP) and fax pass through. T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting Fax mode to be T.38. If the service provider does not support T.38, pass-through mode may be used. To send or receive faxes in fax pass through mode, users will need to select all the Preferred Codecs to be PCMU/PCMA.

## 5.5. Status Light Indicator

Following tables show the Unicorn 3112 button light pattern indication.

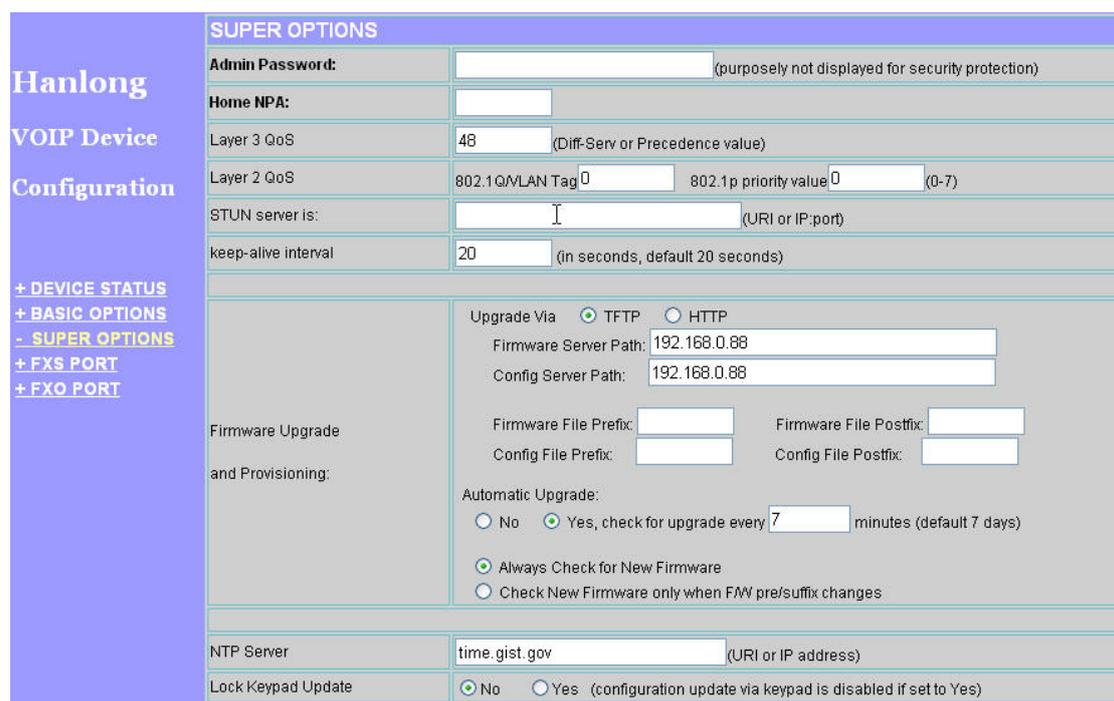
Light Indicator	Signal Pattern	Status Meaning
<b>Red Light</b>	Red light flashes every 2 seconds (if internet connection is configured for DHCP)	DHCP failed or WAN port has no Ethernet connection.
<b>Red Light</b>	Red light flashes every 2	Unicorn 3112 is not able to register

	seconds (if SIP server is configured)	with SIP gatekeeper/proxy server
<b>Green Light</b>	Button flashes every 2 seconds	Message waiting (if feature is available)
<b>Green Light</b>	Button flashes at 1/10 second	Phone is ringing. Incoming call in progress.

## 6. CONFIGURATION GUIDE

### 6.1. Configuring Unicorn 3112 using Web Browser (Recommended)

Unicorn 3112 has embedded Web server and HTML pages that allow users to configure the Unicorn 3112 through an easy-to-use Web browser interface such as Microsoft's Internet Explorer or Netscape browser. Below is a screen shot of the Unicorn 3112 configuration page:



The screenshot shows the 'SUPER OPTIONS' configuration page for the Unicorn 3112. On the left is a navigation menu with 'Hanlong VOIP Device Configuration' and several expandable/collapsible options: '+ DEVICE STATUS', '+ BASIC OPTIONS', '- SUPER OPTIONS' (which is currently selected), '+ FXS PORT', and '+ FXO PORT'. The main content area is titled 'SUPER OPTIONS' and contains the following fields and options:

- Admin Password:** A text input field with a note: "(purposely not displayed for security protection)".
- Home NPA:** A text input field.
- Layer 3 QoS:** A text input field containing '48' with a note: "(Diff-Serv or Precedence value)".
- Layer 2 QoS:** Two text input fields: '802.1Q/VLAN Tag' (containing '0') and '802.1p priority value' (containing '0'), with a note: "(0-7)".
- STUN server is:** A text input field with a note: "(URI or IP:port)".
- keep-alive interval:** A text input field containing '20' with a note: "(in seconds, default 20 seconds)".
- Firmware Upgrade and Provisioning:**
  - Upgrade Via:** Radio buttons for 'TFTP' (selected) and 'HTTP'.
  - Firmware Server Path:** Text input field containing '192.168.0.88'.
  - Config Server Path:** Text input field containing '192.168.0.88'.
  - Firmware File Prefix:** Text input field.
  - Firmware File Postfix:** Text input field.
  - Config File Prefix:** Text input field.
  - Config File Postfix:** Text input field.
  - Automatic Upgrade:**
    - Radio buttons for 'No' and 'Yes, check for upgrade every' (selected).
    - Text input field for minutes, containing '7', with a note: "(default 7 days)".
    - Radio buttons for 'Always Check for New Firmware' (selected) and 'Check New Firmware only when FW pre/suffix changes'.
- NTP Server:** Text input field containing 'time.gist.gov' with a note: "(URI or IP address)".
- Lock Keypad Update:** Radio buttons for 'No' (selected) and 'Yes' with a note: "(configuration update via keypad is disabled if set to Yes)".

Disable Voice Prompt	<input checked="" type="radio"/> No <input type="radio"/> Yes (voice prompt is disabled if set to Yes)
Syslog Server	<input type="text"/>
Syslog Level	NONE
Download Device Configuration:	<input type="button" value="Download"/>
<p><b>Call Progress Tones</b></p> <p>Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...]            Note: freq: 0 - 4000Hz; vol: -30 - 0dBm</p> <p>Dial Tone: <input type="text" value="f1=350@-13,f2=440@-13,c=0/0;"/></p> <p>Ringback Tone: <input type="text" value="f1=440@-19,f2=480@-19,c=2000/4000;"/></p> <p>Busy Tone: <input type="text" value="f1=480@-24,f2=620@-24,c=500/500;"/></p> <p>Reorder Tone: <input type="text" value="f1=480@-24,f2=620@-24,c=250/250;"/></p> <p>Confirmation Tone: <input type="text" value="f1=350@-11,f2=440@-11,c=100/100-100/100-100/100;"/></p> <p>Call Waiting Tone: <input type="text" value="f1=440@-13,c=300/10000-300/10000-0/0;"/></p> <p>Default Ring Cadence: <input type="text" value="f1=440@-13,f2=480@-13,c=2000/4000;"/></p> <p><small>Cadence: (Only the cadence is configurable. Syntax: c=on1/off1-on2/off2-on3/off3;[...])</small></p>	
<input type="button" value="SaveSet"/> <input type="button" value="Reboot"/>	
Restore Configuration	<input type="text"/> <input type="button" value="浏览..."/> <input type="button" value="Restore Configuration"/>
Restore License	<input type="text"/> <input type="button" value="浏览..."/> <input type="button" value="Restore License"/>
<small>All Rights Reserved Hanlong Technology CO., LTD. 2005-2008</small>	

## 6.1.1. Accessing the Web Configuration

The Unicorn 3112 configuration page can be accessed via the LAN or WAN port.

## 6.1.2. Programming Unicorn 3112 via the LAN Port

To program Unicorn 3112 via the LAN port, directly connect an Ethernet cable from your PC to the LAN port of the Unicorn 3112. After connecting the cable, confirm that the green light of the LAN port is on. If the green light is not on, this means that your PC is not yet properly connected to the Unicorn 3112 via the LAN port.

For LAN port configuration, use the following default IP address to access the device:  
**http://192.168.22.1**

## 6.1.3. Programming Unicorn 3112 via the WAN Port

The WAN port access for web configuration is disabled by default from the factory. To access the web configuration menu from the WAN port, you must first access the device via the device LAN port (see instructions above “Programming Unicorn 3112 via the LAN port”) and enable the “Enable WAN Web Access” option.

Please see the following screen shot of the Unicorn 3112 basic option page:

Reply to ICMP on WAN port	<input type="radio"/> No <input checked="" type="radio"/> Yes (Unit will not respond to PING from WAN side if set to No)						
WAN side http access	<input type="radio"/> No <input checked="" type="radio"/> Yes (WAN side access to http server will be rejected if set to No)						
Number of Rings	1 (Number of rings for a PSTN incoming call to FXO port before FXO port picks up, default 4)						
PSTN Ring Thru FXS	<input type="radio"/> No <input checked="" type="radio"/> Yes(Default Yes) (If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)						
PSTN Ring Thru Delay(sec)	4 (1-10 seconds. Default 4 seconds)						
PSTN access code	*00 (key pattern to use PSTN line, default is "*00")						
PIN for PSTN Calls	(Enter digits to authorize calling PSTN numbers from VOIP, no default)						
PIN for VOIP Calls	(Enter digits to authorize calling VOIP terminals from PSTN, no default)						
Unconditional Call Forward to PSTN	(VOIP calls will be forwarded to the specified PSTN number)						
Unconditional Call Forward to VOIP	<table border="0"> <tr> <td>User ID</td> <td>Sip Server</td> <td>Sip Destination Port</td> </tr> <tr> <td>fxo001</td> <td>@home.xanadu.hanlongtec.cn</td> <td>5060</td> </tr> </table>	User ID	Sip Server	Sip Destination Port	fxo001	@home.xanadu.hanlongtec.cn	5060
User ID	Sip Server	Sip Destination Port					
fxo001	@home.xanadu.hanlongtec.cn	5060					
<input type="button" value="SaveSet"/> <input type="button" value="Reboot"/>							
All Rights Reserved Hanlong Technology CO., LTD. 2005-2008							

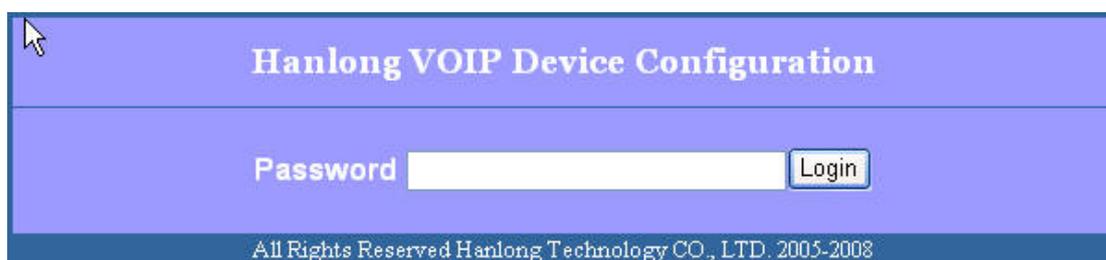
After enabling WAN access, be sure that the WAN port of the Unicorn 3112 is connected to an uplink (i.e. router, hub, switch, etc). Then, get the WAN IP address of the Unicorn 3112 and selecting menu option 02. Then, access the Unicorn 3112 via your web browser by entering the WAN IP address:

## http://Unicorn 3112's IP Address

Be sure that your PC is connected to the router/hub/switch directly or via the LAN port (which also serves as a pass-through connection for internet/network access for your PC) of the Unicorn 3112.

### 6.1.4. User Programming and Configuration

From your web browser, the Unicorn 3112 will show the following login screen:



The login screen features a blue header with the text "Hanlong VOIP Device Configuration". Below the header is a white input field labeled "Password" and a blue "Login" button. At the bottom, there is a footer with the text "All Rights Reserved Hanlong Technology CO., LTD. 2005-2008".

Enter the password and click on the "Login" button

### 6.1.5. Passwords

Passwords are case sensitive and all Unicorn devices come with factory default password as indicated below:

Advanced User Password for access to Super User Options: **admin**

End User Password for access to Basic User Options: **1234**

## 6.1.6. End User Settings:

After a correct password is entered in the login screen, the embedded web server inside the Unicorn 3112 will show the configuration page, which is explained in details below:

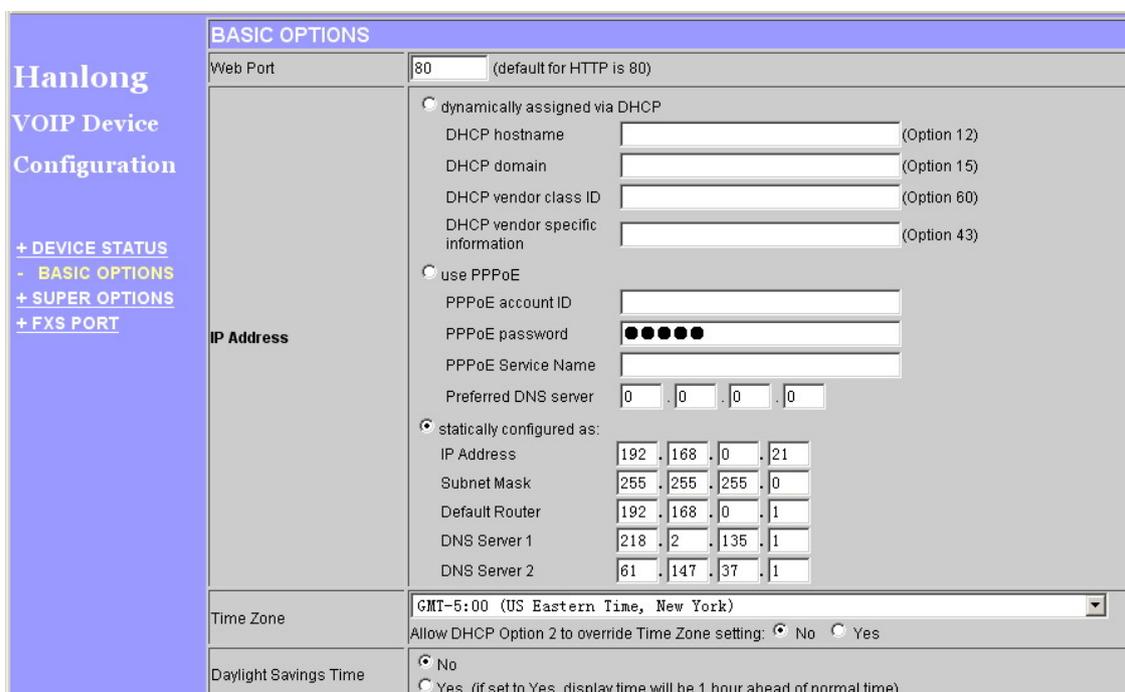
### 6.1.6.1 Device Status:

<b>Hanlong</b> <b>VOIP Device</b> <b>Configuration</b>  - <b>DEVICE STATUS</b> + <b>BASIC OPTIONS</b> + <b>SUPER OPTIONS</b> + <b>FXS PORT</b> + <b>FXO PORT</b>	<b>DEVICE STATUS</b>							
	<b>MAC Address</b>	00:1fc1:08:08:79						
	<b>WAN IP Address</b>	192.168.0.111						
	<b>Product Model</b>	Unicom2112						
	<b>Software Version</b>	BOOT--1.1.0.10(2008-03-20 19:33:00) IMG--1.1.0.10(2008-03-22 09:39:00)						
	<b>System Up Time</b>	0 day(s) 1 hour(s) 47 minute(s) 2 second(s)						
	<b>PPPoE Link Up</b>	Disabled						
	<b>NAT</b>							
	<b>Port Status</b>	Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
		FXS	On Hook	Registered	No			
	FXO	Not Connected	Registered	No				
<input type="button" value="Reboot"/>								
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DEVICE STATUS SETTING	
Setting Options	Meaning
<b>MAC Address</b>	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
<b>WAN IP Address</b>	<p>There are 2 modes under which the Unicorn 3112 can operate:</p> <ul style="list-style-type: none"> <li>- 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 3112 will acquire its IP address from the first DHCP server it discovers from the office/home network it is connected to.</li> <li>- To use the PPPoE feature, the PPPoE account settings need to be set. The Unicorn 3112 will attempt to establish a PPPoE session if any of the PPPoE fields have been entered with data.</li> <li>- 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.</li> </ul>
<b>Product Model</b>	This field contains the product model info, such as Unicorn 3112
<b>Software Version</b>	Program: This is the main software release. Boot and

	Loader are not changed often.
<b>System Up Time</b>	This shows system up time since last reboot.
<b>PPPoE Link Up</b>	This shows whether the PPPoE is up if connected to DSL modem.
<b>NAT</b>	This shows what kind NAT the Unicorn 3112 is connected to. It is based on STUN protocol. If the detected NAT is symmetric NAT, STUN will not work and Outbound Proxy needed to make Unicorn 3112 functioning correctly.
<b>Port Status</b>	Shows several information regarding the FXS and FXO ports.

## 6.1.6.2 Basic Options:



**Hanlong VOIP Device Configuration**

- + DEVICE STATUS
- BASIC OPTIONS
- + SUPER OPTIONS
- + FXS PORT

**BASIC OPTIONS**

Web Port: 80 (default for HTTP is 80)

**IP Address**

dynamically assigned via DHCP

DHCP hostname: \_\_\_\_\_ (Option 12)

DHCP domain: \_\_\_\_\_ (Option 15)

DHCP vendor class ID: \_\_\_\_\_ (Option 60)

DHCP vendor specific information: \_\_\_\_\_ (Option 43)

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 . 21

Subnet Mask: 255 . 255 . 255 . 0

Default Router: 192 . 168 . 0 . 1

DNS Server 1: 218 . 2 . 135 . 1

DNS Server 2: 61 . 147 . 37 . 1

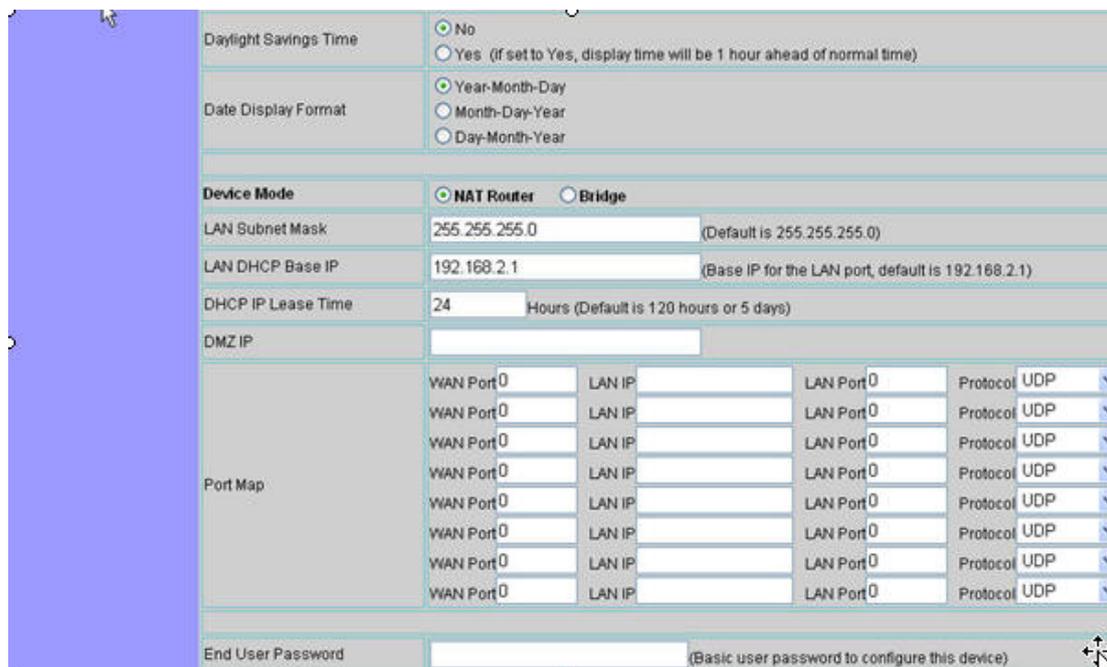
Time Zone: GMT-5:00 (US Eastern Time, New York)

Allow DHCP Option 2 to override Time Zone setting:  No  Yes

Daylight Savings Time:  No  Yes (if set to Yes, display time will be 1 hour ahead of normal time)

BASIC OPTIONS SETTING	
Setting Options	Meaning
<b>Web Port</b>	This is the device's internal HTTP server port. Default is 80.
<b>IP Address</b>	- 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 Flash memory.) The Unicorn 3112 will acquire its IP address from DHCP in the network. PPPoE settings is usually for DSL/ADSL modem users. The Unicorn will attempt to establish a PPPoE session if PPPoE account is set.

	- If Static IP mode is selected, the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (mandatory), DNS Server 2 (optional) fields need to be configured.
<b>Time Zone</b>	This parameter controls how the displayed date/time will be adjusted according to the specified time zone.

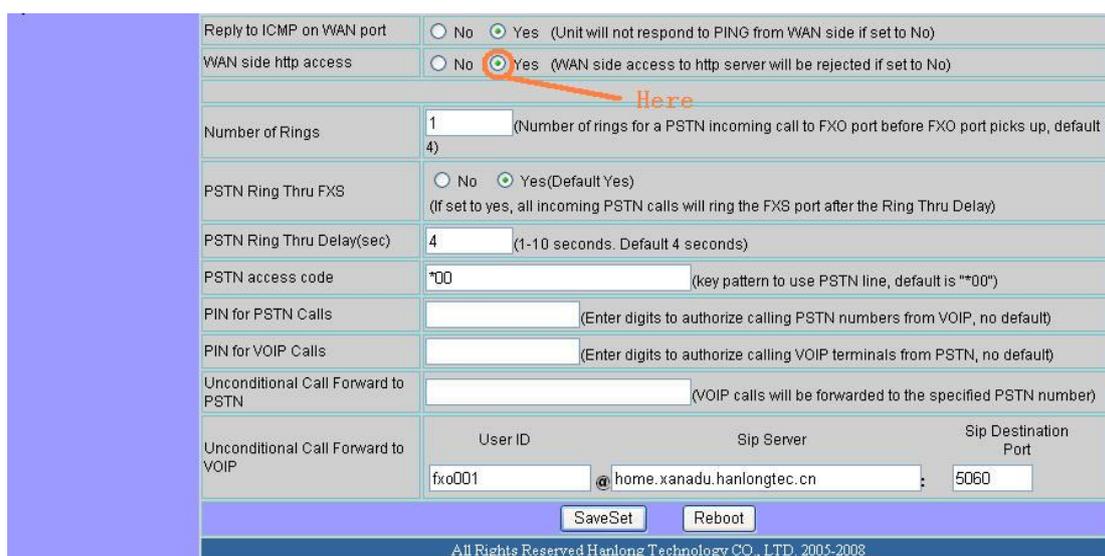


The screenshot shows the configuration page for the Unicorn 3112. It includes the following settings:

- Daylight Savings Time:** Radio buttons for No (selected) and Yes (if set to Yes, display time will be 1 hour ahead of normal time).
- Date Display Format:** Radio buttons for Year-Month-Day (selected), Month-Day-Year, and Day-Month-Year.
- Device Mode:** Radio buttons for NAT Router (selected) and Bridge.
- LAN Subnet Mask:** Text input field with value 255.255.255.0 (Default is 255.255.255.0).
- LAN DHCP Base IP:** Text input field with value 192.168.2.1 (Base IP for the LAN port, default is 192.168.2.1).
- DHCP IP Lease Time:** Text input field with value 24 Hours (Default is 120 hours or 5 days).
- DMZ IP:** Text input field.
- Port Map:** A table with 8 rows, each containing WAN Port 0, LAN IP, LAN Port 0, and Protocol UDP.
- End User Password:** Text input field (Basic user password to configure this device).

BASIC OPTIONS SETTING	
Setting Options	Meaning
<b>Daylight Savings Time</b>	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.
<b>Date Display Format</b>	Allow user to choose among the following three formats: Year-Month-Day Month-Day-Year Day-Month-Year
<b>Device Mode</b>	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
<b>LAN Subnet Mask</b>	Sets the LAN subnet mask. Default value is 255.255.255.0
<b>LAN DHCP Base IP</b>	Base IP for the LAN port which functions as a Gateway for the subnet. Default value is 192.168.22.1
<b>DHCP IP Lease Time</b>	Value is set in units of hours. Default value is <b>120 hrs</b>

	(5 Days.) The time IP address is assigned to the LAN clients.
<b>DMZ IP</b>	Forward all WAN IP traffic to a specific IP address if no matching port is used by Unicorn 3112 itself or in the defined port forwarding
<b>Port Map</b>	Forwards a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port
<b>End User Password</b>	This contains the password to access the Web Configuration Menu. This field is case sensitive.



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)

Number of Rings  (Number of rings for a PSTN incoming call to FXO port before FXO port picks up, default 4)

PSTN Ring Thru FXS  No  Yes(Default Yes)  
(If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)

PSTN Ring Thru Delay(sec)  (1-10 seconds. Default 4 seconds)

PSTN access code  (key pattern to use PSTN line, default is "\*00")

PIN for PSTN Calls  (Enter digits to authorize calling PSTN numbers from VOIP, no default)

PIN for VOIP Calls  (Enter digits to authorize calling VOIP terminals from PSTN, no default)

Unconditional Call Forward to PSTN  (VOIP calls will be forwarded to the specified PSTN number)

Unconditional Call Forward to VOIP  
 User ID:  Sip Server:  Sip Destination Port:

SaveSet Reboot

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BASIC OPTIONS SETTING	
Setting Options	Meaning
<b>Reply to ICMP on WAN port</b>	If set to "Yes", the Unicorn 3112 will respond to the PING command from other computers, but it also is vulnerable to the DOS attack. Default is <b>No</b> .
<b>WAN side http access</b>	If this parameter is set to "No", the HTML configuration update via WAN port is disabled.
<b>Number of Rings</b>	This parameter specifies the number of FXS phone rings for incoming PSTN calls to FXO port. Default is 4
<b>PSTN Ring Thru FXS</b>	Default is Yes
<b>PSTN Ring Thru Delay(sec)</b>	Default is 4 sec
<b>PSTN access code</b>	This field allows users to customize their own code to access the PSTN line. Default is "*00".
<b>PIN for PSTN Calls</b>	Enter digits to authorize calling PSTN numbers from VOIP, no default
<b>PIN for VOIP Calls</b>	Enter digits to authorize calling VOIP terminals from

		PSTN, no default
<b>Unconditional Forward to PSTN</b>	<b>Call</b>	Calls are unconditionally forwarded to the specified PSTN phone number once users dial the FXO port VoIP number.
<b>Unconditional Forward to VOIP</b>	<b>Call</b>	Calls are unconditionally forwarded to the specified VoIP phone number once users dial the FXO port PSTN number.

### 6.1.6.3 Super Option

**Hanlong**

VOIP Device Configuration

[+ DEVICE STATUS](#)

[+ BASIC OPTIONS](#)

[- SUPER OPTIONS](#)

[+ FXS PORT](#)

[+ FXO PORT](#)

**SUPER OPTIONS**

**Admin Password:**  (purposely not displayed for security protection)

**Home NPA:**

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

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

STUN server is:  (URI or IP:port)

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

---

Firmware Upgrade and Provisioning:

Upgrade Via:  TFTP  HTTP

Firmware Server Path:

Config Server Path:

Firmware File Prefix:  Firmware File Postfix:

Config File Prefix:  Config File Postfix:

Automatic Upgrade:

No  Yes, check for upgrade every  minutes (default 7 days)

Always Check for New Firmware

Check New Firmware only when FW pre/suffix changes

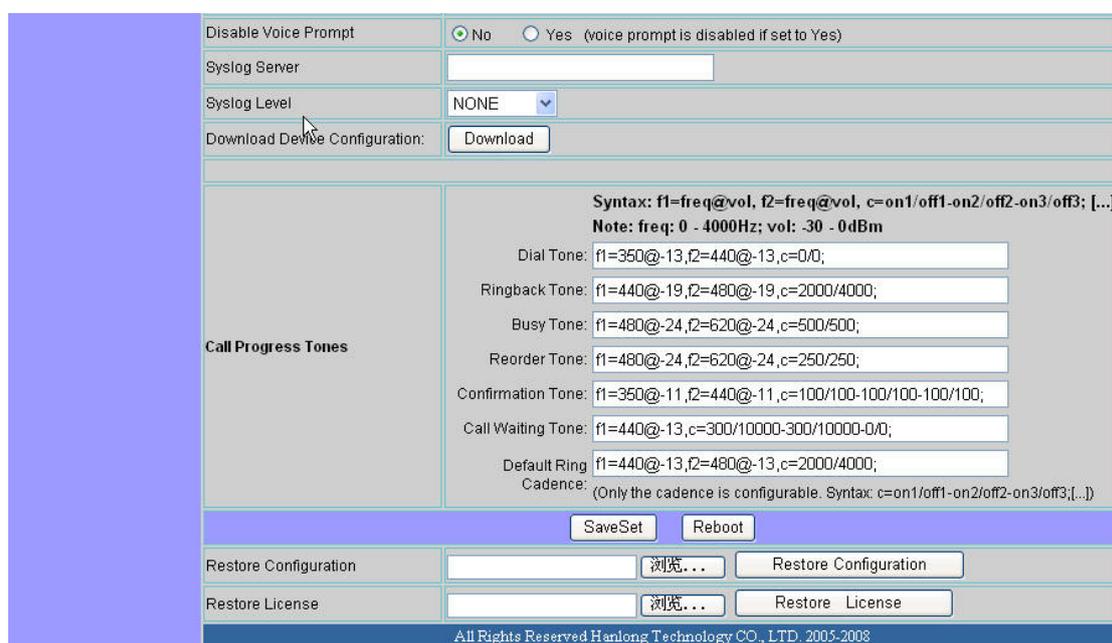
---

NTP Server:  (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
<b>Admin Password:</b>	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.
<b>Home NPA:</b>	Local area code for North American Dial Plan.
<b>Layer 3 QoS</b>	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.
<b>Layer 2 QoS</b>	This contains the value used for layer 2 VLAN tag. Default setting is blank.
<b>STUN server is:</b>	IP address or domain name of stun server

<b>keep-alive interval</b>	This parameter specifies how often the Unicorn 3112 sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open.
<b>Firmware Upgrade and Provisioning:</b>	Default method is HTTP. Firmware upgrade may take up to 10 minutes depending on network environment. Do not interrupt the firmware upgrading process.
<b>NTP Server</b>	This parameter defines the URI or IP address of the NTP server which is used by the Unicorn 3112 to display the current date/time.
<b>Lock Keypad Update</b>	If this parameter is set to “Yes”, the configuration update via keypad is disabled.



The screenshot shows a configuration web page for the Unicorn 3112. It includes the following sections:

- Disable Voice Prompt:** Radio buttons for No (selected) and Yes. Note: (voice prompt is disabled if set to Yes)
- Syslog Server:** A text input field.
- Syslog Level:** A dropdown menu currently set to NONE.
- Download Device Configuration:** A button labeled 'Download'.
- Call Progress Tones:** A section with a syntax guide: `Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...]` and `Note: freq: 0 - 4000Hz; vol: -30 - 0dBm`. Below this are input fields for:
  - 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;
 Note: (Only the cadence is configurable. Syntax: c=on1/off1-on2/off2-on3/off3;[...])
- Buttons:** SaveSet, Reboot, Restore Configuration, and Restore License.
- Footer:** All Rights Reserved Hanlong Technology CO., LTD. 2005-2008

SUPER OPTIONS SETTING	
Setting Options	Meaning
<b>Disable Voice Prompt</b>	Default is NO.
<b>Syslog Server</b>	The IP address or URL of syslog server, especially useful for ITSP (Internet Telephone Service Provider)
<b>Syslog Level</b>	<p>Select the ATA to report the log level. Default is NONE. The level is either one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events:</p> <ul style="list-style-type: none"> <li>product model/version on boot up (INFO level)</li> <li>NAT related info (INFO level)</li> <li>sent or received SIP message (DEBUG level)</li> <li>SIP message summary (INFO level)</li> <li>inbound and outbound calls (INFO level)</li> <li>registration status change (INFO level)</li> </ul>

	<ul style="list-style-type: none"> <li>• negotiated codec (INFO level)</li> <li>• Ethernet link up (INFO level)</li> <li>• SLIC chip exception (WARNING and ERROR levels)</li> <li>• memory exception (ERROR level)</li> </ul> <p>The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components:  GS_LOG: [device MAC address][error code] error message</p> <p>Here is an example: May 19 02:40:38 192.168.1.14  GS_LOG: [00:0b:82:00:a1:be][000]  Ethernet link is up</p>
<b>Download Device Configuration:</b>	User can download configuration from the web page and save to configuration file.
<b>Unregister On Reboot</b>	Default is No. If set to yes, the SIP user will be unregistered on reboot.
<b>Call Progress Tones</b>	<p>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.</p> <p>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.</p>
<b>Restore Configuration</b>	User can restore the before configuration from the configuration file saved at local pc

## 6.1.6.4 FXS Port Settings:

<b>Hanlong</b> <b>VOIP Device</b> <b>Configuration</b>  <a href="#">+ DEVICE STATUS</a> <a href="#">+ BASIC OPTIONS</a> <a href="#">+ SUPER OPTIONS</a> <a href="#">- FXS PORT</a> <a href="#">+ FXO PORT</a>	<b>FXS OPTIONS</b>	
	<b>Account Active:</b>	<input type="radio"/> No <input checked="" type="radio"/> Yes
	<b>SIP Server:</b>	william.xanadu.hanlongtec.cn (e.g., sip.mycompany.com, or IP address)
	<b>Outbound Proxy:</b>	192.168.0.100 (e.g., proxy.myprovider.com, or IP address, if any)
	<b>NAT Traversal</b>	<input type="radio"/> No <input checked="" type="radio"/> No, but send keep-alive <input type="radio"/> STUN <input type="radio"/> UPNP
	<b>SIP User ID:</b>	8203 (the user part of an SIP address)
	<b>Authenticate ID:</b>	8203 (can be identical to or different from SIP User ID)
	<b>Authenticate Password:</b>	(purposely not displayed for securityprotection)
	<b>Name:</b>	(optional, e.g., John Doe)
	<b>Use DNS SRV</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes
	<b>User ID is phone number</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes
	<b>SIP Registration</b>	<input type="radio"/> No <input checked="" type="radio"/> Yes
	<b>Unregister On Reboot</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes
	<b>Register Expiration</b>	6 (in minutes. default 1 hour, max 45 days)
	<b>Outgoing Call without Registration</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>local SIP port</b>	5060 (default 5060)	
<b>local RTP port</b>	5004 (1024-65535, default 5004)	
<b>Use random port</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes	

<b>FXS PORT SETTING</b>	
Setting Options	Meaning
<b>Account Active:</b>	Set to the YES, the account can be available
<b>SIP Server</b>	SIP Server's URI or IP address
<b>Outbound Proxy</b>	SIP Outbound Proxy Server's URI or IP address
<b>NAT Traversal</b>	This parameter defines whether the Unicorn 3112 NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the Unicorn 3112 will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the Unicorn 3112 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 3112 will attempt to use its mapped public IP address and port in all its SIP and SDP messages. If the NAT Traversal field is set to "Yes" with no specified STUN server, the Unicorn 3112 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.
<b>SIP User ID</b>	SIP service subscriber's User ID
<b>Authenticate ID</b>	SIP service subscriber's Authenticate ID. Can be identical

	to or different from SIP User ID
<b>Authenticate Password</b>	SIP service subscriber's account password
<b>Name</b>	SIP service subscriber's name which will be used for Caller ID display
<b>Use DNS SRV</b>	Default is No. If set to Yes the client will use DNS SRV for server lookup
<b>User ID is phone number</b>	If the Unicorn 3112 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
<b>SIP Registration</b>	This parameter controls whether the Unicorn 3112 needs to send REGISTER messages to the proxy server. The default setting is "Yes"
<b>Unregister On Reboot</b>	Default is No. If set to yes, the SIP user will be unregistered on reboot.
<b>Register Expiration</b>	This parameter allows the user to specify the time frequency (in minutes) the Unicorn 3112 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).
<b>Outgoing Call without Registration</b>	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.
<b>Local SIP port</b>	This parameter defines the local SIP port the Unicorn 3112 will listen and transmit. The default value for FXS port is 5060. The default value for FXO port is 5062.
<b>Local RTP port</b>	This parameter defines the local RTP-RTCP port pair the Unicorn 3112 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 is 5004. The default value for FXO port is 5008.
<b>Use random port</b>	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 3112 are behind the same NAT.

Refer-To Use Target Contact	<input checked="" type="radio"/> No <input type="radio"/> Yes
SIP T1 Timeout	0.5 sec
SIP T2 Interval	2 sec
DTMF Payload Type	101
DTMF in Audio	<input type="radio"/> No <input checked="" type="radio"/> Yes
DTMF via RFC2833	<input type="radio"/> No <input checked="" type="radio"/> Yes
DTMF via SIP INFO	<input checked="" type="radio"/> No <input type="radio"/> Yes
Send Flash Event	<input checked="" type="radio"/> No <input type="radio"/> Yes (Flash will be sent as a DTMF event if set to Yes)
Enable Call Features	<input type="radio"/> No <input checked="" type="radio"/> Yes (if Yes, call features using star codes will be supported locally)
Offhook Auto-Dial	<input type="text"/> (User ID/extension to dial automatically when offhook)
Proxy-Require	<input type="text"/>
Use NAT IP	<input type="text"/> (used in SIP/SDP message if specified)
Disable Call-Waiting	<input type="radio"/> No <input type="radio"/> Yes
No Key Entry Timeout	4 (in seconds, default is 4 seconds)

FXS PORT SETTING	
Setting Options	Meaning
<b>Refer-To Use Target Contact</b>	Used for Attended transfer Feature. Default is NO. If set to YES, the "Refer-To" header uses the transferred target's "Contact" header information.
<b>SIP T1 Timeout</b>	Default is 0.5 sec
<b>SIP T2 Interval</b>	Default is 2 sec
<b>DTMF Payload Type</b>	This parameter sets the payload type for DTMF using RFC2833
<b>DTMF in Audio</b>	Default is YES
<b>DTMF via RFC2833</b>	Default is YES
<b>DTMF via SIP INFO</b>	Default is NO
<b>Send Flash Event</b>	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.
<b>Enable Call Features</b>	Default is No. If set to Yes, Call Forwarding & Do-Not-Disturb are supported locally
<b>Offhook Auto-Dial</b>	This parameter allows users to configure a User ID or extension number to be automatically dialed upon offhook. Please note that only the user part of a SIP address needs to be entered here. The Unicorn 3112 will automatically append the "@" and the host portion of the corresponding SIP address.
<b>Proxy-Require</b>	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
<b>Use NAT IP</b>	NAT IP address used in SIP/SDP message. Default is blank.
<b>Disable Call-Waiting</b>	Default is No.
<b>No Key Entry Timeout</b>	Default is 4 seconds.

Preferred Vocoder (in listed order)	choice 1: <input pcmu"="" type="text" value="current setting is "/> choice 2: <input g.726-32"="" type="text" value="current setting is "/> choice 3: <input g.723.1"="" type="text" value="current setting is "/> choice 4: <input pcma"="" type="text" value="current setting is "/> choice 5: <input g.728"="" type="text" value="current setting is "/> choice 6: <input b"="" g.729a="" type="text" value="current setting is "/>
Voice Frames per TX	<input type="text" value="2"/> (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
G723 rate	<input checked="" type="radio"/> 6.3kbps encoding rate <input type="radio"/> 5.3kbps encoding rate
iLBC frame size	<input checked="" type="radio"/> 20ms <input type="radio"/> 30ms
iLBC payload type	<input type="text" value="97"/> (between 96 and 127, default is 97)
G726-16 Payload Type	<input type="text" value="100"/> (between 96 and 127, default is 100)
G726-24 Payload Type	<input type="text" value="99"/> (between 96 and 127, default is 99)
G726-40 Payload Type	<input type="text" value="103"/> (between 96 and 127, default is 103)
G729E Payload Type:	<input type="text" value="102"/> (between 96 and 127, default is 102)
VAD	<input checked="" type="radio"/> No <input type="radio"/> Yes
Symmetric RTP	<input type="radio"/> No <input type="radio"/> Yes
Fax Mode	<input checked="" type="radio"/> T.38 (Auto Detect) <input type="radio"/> Pass-Through
Fax Tone Detection Mode	<input type="radio"/> Caller <input checked="" type="radio"/> Callee

FXS PORT SETTING	
Setting Options	Meaning
<b>Preferred Vocoder (in listed order)</b>	<p>The Unicorn 3112 supports up to 7 different Vocoder types including G.711 A-/U-law, G.723.1, G.726, G.728, G.729A/B, iLBC(Pending). Depending on the product model, some of these Vocoders may not be provided in standard release.</p> <p>Users can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder in this list can be entered by choosing the appropriate option in "Choice 1". Similarly, the last Vocoder in this list can be entered by choosing the appropriate option in "Choice 7".</p>
<b>Voice Frames per TX</b>	<p>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</p>

	maximum allowed value, the Unicorn 3112 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.
<b>G723 Rate</b>	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.
<b>iLBC frame size</b>	This sets the iLBC size in 20ms or 30ms
<b>iLBC payload type</b>	This defines payload time for iLBC. Default value is 98. The valid range is between 96 and 127.
<b>G726-16 Payload Type</b>	default is 100
<b>G726-24 Payload Type</b>	Default is 99
<b>G726-40 Payload Type</b>	Default value is103, range is from 96 to 123
<b>G729E Payload Type:</b>	Default value is102, range is from 96 to 127
<b>VAD</b>	Default is No. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network
<b>Symmetric RTP</b>	Default is <b>No</b> . 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.
<b>Fax Mode</b>	T.38 (Auto Detect) FoIP by default, or fax Pass-Through.
<b>Fax Tone Detection Mode</b>	Default is Callee. This decides whether Caller or Callee sends out the re INVITE for T.38 or Fax Pass Through.

Jitter Buffer Type	<input checked="" type="radio"/> Fixed <input type="radio"/> Adaptive
Jitter Buffer Length	<input checked="" type="radio"/> Low <input type="radio"/> Medium <input type="radio"/> High
Distinctive Ring Tone	Ring Tone 1 <input type="text" value=""/> used if incoming caller ID is <input type="text" value=""/>
	Ring Tone 1 <input type="text" value=""/> used if incoming caller ID is <input type="text" value=""/>
	Ring Tone 1 <input type="text" value=""/> used if incoming caller ID is <input type="text" value=""/>
Disable Call-Waiting	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Call-Waiting Tone	<input checked="" type="radio"/> No <input type="radio"/> Yes
Ring Timeout	60 (10-300 seconds, default is 60 seconds)
No Key Entry Timeout	4 (in seconds, default is 4 seconds)
Early Dial	<input checked="" type="radio"/> No <input type="radio"/> Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix	<input type="text" value=""/> (this prefix string is added to each dialed number)
Use # as Dial Key	<input type="radio"/> No <input checked="" type="radio"/> Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
Dial Plan	<input type="text" value=""/>
SUBSCRIBE for MWI	<input checked="" type="radio"/> No, do not send SUBSCRIBE for Message Waiting Indication <input type="radio"/> Yes, send periodical SUBSCRIBE for Message Waiting Indication
Send Anonymous	<input checked="" type="radio"/> No <input type="radio"/> Yes (caller ID will be blocked if set to Yes)
Anonymous Call Rejection	<input checked="" type="radio"/> No <input type="radio"/> Yes
Session Expiration	180 (in seconds, default 180 seconds)
Min-SE	90 (in seconds, default and minimum 90 seconds)
Caller Request Timer	<input checked="" type="radio"/> No <input type="radio"/> Yes (Request for timer when making outbound calls)

FXS PORT SETTING	
Setting Options	Meaning
<b>Jitter Buffer Type</b>	Select either Fixed or Adaptive based on network conditions.
<b>Jitter Buffer Length</b>	Select Low, Medium or High based on network conditions.
<b>Distinctive Ring Tone</b>	Default is NO.
<b>Disable Call-Waiting</b>	Default is NO.
<b>Disable Call-Waiting Tone</b>	Default is NO. This is to disable the stutter call waiting tone when a call waiting call arrived
<b>Ring Timeout</b>	Incoming call will stop ringing
<b>No Key Entry Timeout</b>	Default is 4 seconds.
<b>Early Dial</b>	Default is No. Use only if proxy supports 484 response
<b>Dial Plan Prefix</b>	Sets the prefix added to each dialed number
<b>Use # as Dial Key</b>	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.
<b>Dial Plan</b>	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;

	<p>c. ^ - exclude;</p> <p>d. [3-5] - any digit of 3, 4, or 5;</p> <p>e. [147] - any digit 1, 4, or 7;</p> <p>f. &lt;2=011&gt; - replace digit 2 with 011 when dialing</p> <p>Example 1: {[369]11   1617xxxxxxx} Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617</p> <p>Example 2: {^1900x+   &lt;=1617&gt;xxxxxxx} Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers</p> <p>Example 3: {1xxx[2-9]xxxxxx   &lt;2=011&gt;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.</p> <p>3. Default: Outgoing - {x+}</p> <p>Example of a simple dial plan used in a Home/Office in the US:</p> <p>{ ^1900x.   &lt;=1617&gt;[2-9]xxxxxx   1[2-9]xx[2-9]xxxxxx   011[2-9]x.   [3469]11 }</p> <p>Explanation of example rule (reading from left to right):</p> <p>^1900x. - prevents dialing any number started with 1900</p> <p>&lt;=1617&gt;[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically</p> <p>1[2-9]xx[2-9]xxxxxx  - allows dialing to any US/Canada Number with 11 digits length</p> <p>011[2-9]x. - allows international calls starting with 011</p> <p>[3469]11 - allow dialing special and emergency numbers 311, 411, 611 and 911</p> <p>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*]+ }.</p>
<p><b>SUBSCRIBE for MWI</b></p>	<p>Default is No. When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.</p>
<p><b>Send Anonymous</b></p>	<p>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.</p>
<p><b>Anonymous Call</b></p>	<p>Default is NO, if set to YES, the anonymous call will be</p>

<b>Rejection</b>	rejected with busy message.
<b>Session Expiration</b>	Default is 180 seconds.
<b>Min-SE</b>	Default is 90 seconds
<b>Caller Request Timer</b>	Default is NO

Callee Request Timer	<input checked="" type="radio"/> No <input type="radio"/> Yes (When caller supports timer but did not request one)
Force Timer	<input checked="" type="radio"/> No <input type="radio"/> Yes (Use timer even when remote party does not support)
UAC Specify Refresher	<input type="radio"/> UAC <input type="radio"/> UAS <input checked="" type="radio"/> Omit (Recommended)
UAS Specify Refresher	<input checked="" type="radio"/> UAC <input type="radio"/> UAS (When UAC did not specify refresher tag)
Force INVITE	<input checked="" type="radio"/> No <input type="radio"/> 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	36V
Polarity Reversal	<input checked="" type="radio"/> No <input type="radio"/> Yes (reverse polarity upon call establishment and termination)
Hook Flash Timing	minimum: 200 maximum: 600 (Note: In 50-1200 milliseconds range)
Volume Amplification	TX: DdB default RX: DdB default

<b>Ring Tones</b>	<b>Syntax: c=on1/off1-on2/off2-on3/off3; [...]</b>	
	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;
<input type="button" value="SaveSet"/> <input type="button" value="Reboot"/>		
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FXS PORT SETTING	
Setting Options	Meaning
<b>Callee Request Timer</b>	Default is NO
<b>Force Timer</b>	Default is NO
<b>UAC Specify Refresher</b>	Default is Omit
<b>UAS Specify Refresher</b>	Default is UAC
<b>Force INVITE</b>	Default is NO
<b>Special Feature</b>	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel,

	Broadsoft, etc.
<b>FXS Impedance</b>	Selects the impedance of the analog telephone connected to the Phone port.
<b>Caller ID Scheme</b>	Select the Caller ID Scheme to suit the standard of different area. <ul style="list-style-type: none"> <li>• Bellcore (North America)</li> <li>• ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA)</li> <li>• ETSI-DTMF (Finland, Sweden)</li> <li>• DTMF (Denmark)</li> </ul>
<b>Onhook Voltage</b>	Select the onhook voltage to suit different area or PBX
<b>Polarity Reversal</b>	Select Polarity Reversal to adapt some call charge/billing system. Default is No.
<b>Hook Flash Timing</b>	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.
<b>Volume Amplification</b>	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.
<b>Ring Tones</b>	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.

### 6.1.6.5 FXO Port Settings:

<b>Hanlong</b> <b>VOIP Device</b> <b>Configuration</b>  <a href="#">+ DEVICE STATUS</a> <a href="#">+ BASIC OPTIONS</a> <a href="#">+ SUPER OPTIONS</a> <a href="#">+ FXS PORT</a> <a href="#">- FXO PORT</a>	<b>FXO OPTIONS</b>	
	<b>Account Active</b>	<input type="radio"/> No <input checked="" type="radio"/> Yes
	<b>SIP Server</b>	william.xanadu.hanlongtec.cn (e.g., sip.mycompany.com, or IP address)
	<b>Outbound Proxy</b>	192.168.0.100 (e.g., proxy.myprovider.com, or IP address, if any)
	<b>NAT Traversal</b>	<input type="radio"/> No <input type="radio"/> No, but send keep-alive <input checked="" type="radio"/> STUN <input type="radio"/> UPNP
	<b>SIP User ID</b>	nj001 (the user part of an SIP address)
	<b>Authenticate ID</b>	nj001 (can be identical to or different from SIP User ID)
	<b>Authenticate Password</b>	(purposely not displayed for securityprotection)
	<b>Name</b>	(optional, e.g., John Doe)
	<b>Use DNS SRV</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes
	<b>User ID is phone number</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes
	<b>SIP Registration</b>	<input type="radio"/> No <input checked="" type="radio"/> Yes
	<b>Unregister On Reboot</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes
	<b>Register Expiration</b>	6 (in minutes, default 1 hour, max 45 days)
	<b>Outgoing Call without Registration</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes
	<b>local SIP port</b>	5062 (default 5062)
	<b>local RTP port</b>	5008 (1024-65535, default 5008)
	<b>Use random port</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes

FXO PORT SETTING	
Setting Options	Meaning
<b>Account Active:</b>	Set to the YES, the account can be available
<b>SIP Server</b>	SIP Server's URI or IP address
<b>Outbound Proxy</b>	SIP Outbound Proxy Server's URI or IP address
<b>NAT Traversal</b>	This parameter defines whether the Unicorn 3112 NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the Unicorn 3112 will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the Unicorn 3112 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 3112 will attempt to use its mapped public IP address and port in all its SIP and SDP messages. If the NAT Traversal field is set to "Yes" with no specified STUN server, the Unicorn 3112 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.
<b>SIP User ID</b>	SIP service subscriber's User ID
<b>Authenticate ID</b>	SIP service subscriber's Authenticate ID.

	Can be identical to or different from SIP User ID
<b>Authenticate Password</b>	SIP service subscriber's account password
<b>Name</b>	SIP service subscriber's name which will be used for Caller ID display
<b>Use DNS SRV</b>	Default is No. If set to Yes the client will use DNS SRV for server lookup
<b>User ID is phone number</b>	If the Unicorn 3112 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
<b>SIP Registration</b>	This parameter controls whether the Unicorn 3112 needs to send REGISTER messages to the proxy server. The default setting is "Yes".
<b>Unregister On Reboot</b>	Default is No. If set to yes, the SIP user will be unregistered on reboot.
<b>Register Expiration</b>	This parameter allows the user to specify the time frequency (in minutes) the Unicorn 3112 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).
<b>Outgoing Call without Registration</b>	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.
<b>Local SIP port</b>	This parameter defines the local SIP port the Unicorn 3112 will listen and transmit. The default value for FXS port is 5060. The default value for FXO port is 5062.
<b>Local RTP port</b>	This parameter defines the local RTP-RTCP port pair the Unicorn 3112 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 is 5004. The default value for FXO port is 5008.
<b>Use random port</b>	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 3112 are behind the same NAT.

Refer-To Use Target Contact	<input checked="" type="radio"/> No <input type="radio"/> Yes
SIP T1 Timeout	0.5 sec
SIP T2 Interval	4 sec
DTMF Payload Type	101
DTMF in Audio	<input type="radio"/> No <input checked="" type="radio"/> Yes
DTMF via RFC2833	<input type="radio"/> No <input checked="" type="radio"/> Yes
DTMF via SIP INFO	<input checked="" type="radio"/> No <input type="radio"/> Yes
Send Flash Event	<input checked="" type="radio"/> No <input type="radio"/> Yes (Flash will be sent as a DTMF event if set to Yes)
Proxy-Require	
Use NAT IP	(used in SIP/SDP message if specified)
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"
Voice Frames per TX	2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
G723 rate	<input checked="" type="radio"/> 6.3kbps encoding rate <input type="radio"/> 5.3kbps encoding rate

FXO PORT SETTING	
Setting Options	Meaning
<b>Refer-To Use Target Contact</b>	Default is NO. If set to YES, then for Attended Transfer, the "Refer-To" header uses the transferred target's Contact header information.
<b>SIP T1 Timeout</b>	Default is 0.5 sec
<b>SIP T2 Interval</b>	Default is 4 sec
<b>DTMF Payload Type</b>	This parameter sets the payload type for DTMF using RFC2833
<b>DTMF in Audio</b>	Default is YES
<b>DTMF via RFC2833</b>	Default is YES
<b>DTMF via SIP INFO</b>	Default is NO
<b>Send Flash Event</b>	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.
<b>Proxy-Require</b>	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
<b>Use NAT IP</b>	NAT IP address used in SIP/SDP message. Default is blank.
<b>Preferred Vocoder (in listed order)</b>	The Unicorn 3112 supports up to 7 different Vocoder types including G.711 A-/U-law , G.723.1, G.726, G.728, G.729A/B, iLBC(Pending). Depending on the product model, some of these Vocoders may not be provided in standard release. Users can configure Vocoders in a preference list that will be included with the same preference order in SDP

	<p>message. The first Vocoder in this list can be entered by choosing the appropriate option in “Choice 1”. Similarly, the last Vocoder in this list can be entered by choosing the appropriate option in “Choice 7”.</p> <p>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 3112 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.</p> <p>This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.</p>
<b>Voice Frames per TX</b>	This sets the iLBC size in 20ms or 30ms
<b>G723 Rate</b>	This defines payload time for iLBC. Default value is 98. The valid range is between 96 and 127.

iLBC frame size	<input checked="" type="radio"/> 20ms <input type="radio"/> 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	<input checked="" type="radio"/> No <input type="radio"/> Yes
Symmetric RTP	<input type="radio"/> No <input type="radio"/> Yes
Fax Mode	<input checked="" type="radio"/> T.38 (Auto Detect) <input type="radio"/> Pass-Through
Fax Tone Detection Mode	<input type="radio"/> Caller <input checked="" type="radio"/> Callee
Jitter Buffer Type	<input type="radio"/> Fixed <input type="radio"/> Adaptive
Jitter Buffer Length	<input checked="" type="radio"/> Low <input type="radio"/> Medium <input type="radio"/> High
Early Dial	<input checked="" type="radio"/> No <input type="radio"/> Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix	<input type="text"/> (this prefix string is added to each dialed number)
Use # as Dial Key	<input type="radio"/> No <input checked="" type="radio"/> Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
SUBSCRIBE for MWI	<input checked="" type="radio"/> No, do not send SUBSCRIBE for Message Waiting Indication <input type="radio"/> Yes, send periodical SUBSCRIBE for Message Waiting Indication

FXO PORT SETTING	
Setting Options	Meaning
<b>iLBC frame size</b>	This sets the iLBC size in 20ms or 30ms
<b>iLBC payload type</b>	This defines payload time for iLBC. Default value is 98. The valid range is between 96 and 127.
<b>G726-16 Payload Type</b>	default is 100
<b>G726-24 Payload Type</b>	Default is 99
<b>G726-40 Payload Type</b>	Default value is 103, range is from 96 to 123
<b>G729E Payload Type:</b>	Default value is 102, range is from 96 to 127
<b>VAD</b>	Default is No. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network
<b>Symmetric RTP</b>	Default is <b>No</b> . 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.
<b>Fax Mode</b>	T.38(Auto Detect) FoIP by default, or fax Pass-Through.
<b>Fax Tone Detection Mode</b>	Default is Callee. This decides whether Caller or Callee sends out the re INVITE for T.38 or Fax Pass Through.
<b>Jitter Buffer Type</b>	Select either Fixed or Adaptive based on network conditions.
<b>Jitter Buffer Length</b>	Select Low, Medium or High based on network conditions.
<b>Early Dial</b>	Default is No. Use only if proxy supports 484 response
<b>Dial Plan Prefix</b>	Sets the prefix added to each dialed number
<b>Use # as Dial Key</b>	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.
<b>SUBSCRIBE for MWI</b>	Default is No. When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.

Send Anonymous	<input type="radio"/> No <input type="radio"/> Yes (caller ID will be blocked if set to Yes)
Session Expiration	180 (in seconds, default 180 seconds)
Min-SE	90 (in seconds, default and minimum 90 seconds)
Caller Request Timer	<input checked="" type="radio"/> No <input type="radio"/> Yes (Request for timer when making outbound calls)
Callee Request Timer	<input checked="" type="radio"/> No <input type="radio"/> Yes (When caller supports timer but did not request one)
Force Timer	<input checked="" type="radio"/> No <input type="radio"/> Yes (Use timer even when remote party does not support)
UAC Specify Refresher	<input type="radio"/> UAC <input type="radio"/> UAS <input checked="" type="radio"/> Omit (Recommended)
UAS Specify Refresher	<input checked="" type="radio"/> UAC <input type="radio"/> UAS (When UAC did not specify refresher tag)
Force INVITE	<input checked="" type="radio"/> No <input type="radio"/> Yes (Always refresh with INVITE instead of UPDATE)
Special Feature	Standard
Caller ID Minimum RX Level (dB)	(-50 - 0dB, Default -30dB)
Caller ID Transport Type	Relay via SIP P-Asserted-Identity
Volume Amplification	TX: 0dB default RX: +6dB
PSTN AC Termination	600 Ohm (North America) impedance
Enable PSTN Disconnect Tone Detection	<input type="radio"/> No <input checked="" type="radio"/> Yes (Default No) (If set to yes, the following tone is used as the disconnect signal)

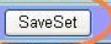
PSTN Disconnect Tone	f1=480@-32,f2=620@-32,c=500/500; (Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...]) (Allowed Range: freq = 0 to 4000Hz; vol = -40 to -24dBm) (Default: Busy Tone: f1=480@-32,f2=620@-32,c=500/500;)
PSTN Silence Timeout	60 (sec, terminate call after long silence detected, default is 60 sec, max 65536)
<input type="button" value="SaveSet"/> <input type="button" value="Reboot"/>	
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FXO PORT SETTING	
Setting Options	Meaning
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.
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	Default is Omit

Refresher	
UAS Specify Refresher	Default is UAC
Force INVITE	Default is NO
Special Feature	Default is standard
Caller ID Minimum RX Level (dB)	Default is -30dB
Caller ID Transport Type	Default is Relay via SIP P-Asserted-Identity
Volume Amplification	<p>Voice path volume adjustment.</p> <p>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.</p> <p>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.</p> <p>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.</p>
PSTN AC Termination	Selects the impedance of the analog PSTN line connected to the Line port.
Enable PSTN Disconnect Detection	If set to Yes, arrived Busy Tone is used as the disconnect signal.
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.
PSTN Silence Timeout	Terminate call after long silence detected. Default setting is 60 sec, max 65536

## 6.1.7. Saving the Configuration Changes

Once a change is made, users should click on the “SaveSet” button in the Configuration page, as follow:

End User Password	<input type="text"/>	(Basic user password to configure this device)						
Reply to ICMP on WAN port	<input type="radio"/> No <input checked="" type="radio"/> Yes	(Unit will not respond to PING from WAN side if set to No)						
WAN side http access	<input type="radio"/> No <input checked="" type="radio"/> Yes	(WAN side access to http server will be rejected if set to No)						
Number of Rings	<input type="text" value="1"/>	(Number of rings for a PSTN incoming call to FXO port before FXO port picks up, default 4)						
PSTN Ring Thru FXS	<input type="radio"/> No <input checked="" type="radio"/> Yes(Default Yes)	(If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)						
PSTN Ring Thru Delay(sec)	<input type="text" value="4"/>	(1-10 seconds. Default 4 seconds)						
PSTN access code	<input type="text" value="*00"/>	(key pattern to use PSTN line, default is "**00")						
PIN for PSTN Calls	<input type="text"/>	(Enter digits to authorize calling PSTN numbers from VOIP, no default)						
PIN for VOIP Calls	<input type="text"/>	(Enter digits to authorize calling VOIP terminals from PSTN, no default)						
Unconditional Call Forward to PSTN	<input type="text"/>	(VOIP calls will be forwarded to the specified PSTN number)						
Unconditional Call Forward to VOIP	<table border="0"> <tr> <td>User ID</td> <td>Sip Server</td> <td>Sip Destination Port</td> </tr> <tr> <td><input type="text" value="fxo001"/></td> <td><input type="text" value="@home.xanadu.hanlongtec.cn"/></td> <td><input type="text" value="5060"/></td> </tr> </table>	User ID	Sip Server	Sip Destination Port	<input type="text" value="fxo001"/>	<input type="text" value="@home.xanadu.hanlongtec.cn"/>	<input type="text" value="5060"/>	
User ID	Sip Server	Sip Destination Port						
<input type="text" value="fxo001"/>	<input type="text" value="@home.xanadu.hanlongtec.cn"/>	<input type="text" value="5060"/>						
<span style="color: red; font-weight: bold;">Here</span> 								
<input type="button" value="SaveSet"/> <input type="button" value="Reboot"/>								
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The Unicorn 3112 will then display the following screen to confirm that the changes have been saved. Please allow 5 to 10 seconds before rebooting the device.

<b>Hanlong</b> VOIP Device Configuration  <a href="#">+ DEVICE STATUS</a> <a href="#">+ BASIC OPTIONS</a> <a href="#">+ SUPER OPTIONS</a> <a href="#">+ FXS PORT</a> <a href="#">+ FXO PORT</a>	SAVING...
	<b>Your changes have been saved.</b>  <span style="color: red; font-weight: bold;">Please wait 5 second and then reboot the device.</span>
	<input type="button" value="Reboot"/>
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## 6.1.8. Rebooting the Unicorn 3112

You can reboot the Unicorn 3112 by clicking on the “Reboot” button after each update to the configuration page. Alternatively, you can reboot by unplugging the power supply of the Unicorn 3112 and then powering it on again. If your Unicorn 3112 ever becomes “stuck” or un-responsive, you can unplug the power supply to reboot it. Frequent rebooting by unplugging the power supply is not recommended and should not be necessary.

## 6.2. Configuring Unicorn 3112 via Voice Prompt

### 6.2.1. DHCP Mode

Follow section 5.1 with voice menu option 01 to enable Unicorn 3112 to use DHCP

### 6.2.2. Static IP Mode

Follow section 5.1 with voice menu option 01 to enable Unicorn 3112 to use STATIC IP mode, then use option 02, 03, 04 to set up Unicorn 3112's IP, Subnet Mask, Gateway respectively.

### 6.2.3. Configuration through a Central Server

Unicorn 3112 devices can be automatically configured from a central provisioning system.

When Unicorn 3112 boots up, it will send TFTP or HTTP request to download configuration files. There are two configuration files, one is "cfg.txt" and the other is "cfg001fc1xxxxxx", where "001fc1xxxxxx" is the MAC address of the Unicorn 3112.

For more information regarding configuration file format, please refer to the related technical documentation.

The configuration file can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of Unicorn 3112s can easily manage the configuration and service provisioning of individual devices remotely and automatically from a central server. The central provisioning system uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual Unicorn 3112 for firmware upgrade, etc.

## 7. SOFTWARE UPGRADE

To upgrade software, Unicorn 3112 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 3112.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the Unicorn 3112'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 3112. 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 3112.

TFTP process may take as long as 1 to 2 minutes over the Internet, or just 20+ 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.txt" and the other is "cfg001fc1xxxxx", where "001fc1xxxxx" is the MAC address of the Unicorn 3112. 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 to the factory default settings will delete all configuration information of the device.

Steps to follow in restoring to factory default settings:

- a) Press "\*" or for voice prompt.
- b) Enter "99" and then you will hear the voice prompt "Reset".
- c) Enter the number "862584658050". A "click" sound will be heard.
- d) Wait for 15 seconds.

The device is now restored to the factory default setting.

## 9. TECHNICAL SUPPORT CONTACT

Email: [Support@mail.hanlongtek.com](mailto:Support@mail.hanlongtek.com)