

User Manual

Unicorn 2101

Analog Telephone Adaptor



Hanlong Technology Co., Ltd

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Table of Content

1	WELCOME	3
2	WHAT IS IN THE PACKAGE	3
3	PRODUCT OVERVIEW	3
3.1	Key Features	3
3.2	Hardware specification	4
4	INSTALLATIONS	4
4.1	Safety	6
5	BASIC OPERATIONS	6
5.1	Get Familiar with Voice	6
5.2	Make Phone call	8
5.2.1	Calling Phone or Extension Numbers	8
5.2.2	Call Hold	8
5.2.3	Call Waiting	8
5.2.4	3-way Conferencing.....	8
	Star Code Style 3-way Conference	8
	Bellcore Style 3-way Conference	9
5.2.5	Direct IP-to-IP Calls	9
5.2.6	Blind Transfer	10
5.2.7	Attended Transfer	10
5.2.8	Make and Receive PSTN Calls	11
5.3	Call Features	11
5.3.1	Call Features Tables.....	11
5.3.2	PSTN Pass Through.....	12
5.4	FAX.....	12
5.5	Status Light Indicator	12
6	CONFIGURATION GUIDE	12
6.1	Configuring Unicorn2101 using Web Browser	12
6.1.1	Accessing the Web Configuration.....	13
6.1.2	Programming Unicorn 2101 via the LAN Port.....	13
6.1.3	Programming Unicorn 2101 via the WAN Port	14
6.1.4	User Programming and Configuration	14
6.1.5	Passwords.....	14
6.1.6	Configuration Options and Explanations	15
6.1.6.1	Device Status	15
6.1.6.2	Basic Options Settings	15
6.1.6.3	Super Options Setting	17
6.1.6.4	FXS Port	18
6.1.7	Saving the Configuration Changes	25
6.1.8	Rebooting the Unicorn 2101	25
6.2	Configuring Unicorn 2101 via Voice Prompt	26
6.2.1	DHCP Mode	26

6.2.2	Static IP Mode	26
6.3	Configuration through a Central Server	26
7	SOFTWARE UPGRADE.....	26
8	RESTORE FACTORY DEFAULT SETTINGS.....	27
9	TECHNICAL SUPPORT CONTACT.....	27

1 WELCOME

Unicorn 2101 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 2101 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 2101 package contains:

- One Unicorn 2101 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
- 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, optional), G.726 (32K), as well as 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 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 ,and STUN, UPNP.
- 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.
- Support volume amplification
- Support configurable Call Progress Tones
- Ultra compact (wallet size) and lightweight design, great companion for travelers
- Compact, lightweight Universal Power adapter.

3.2 Hardware specification

Model	Unicorn 2101
LAN interface	1 x RJ45 100MBase-T
WAN interface	1 x RJ45 100MBase-T
FXS telephone port	1 x FXS
PSTN Pass Through	1 x RJ11
LED light	Green and red color
Universal switching power supply	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA, CE certification
Dimension	70mm (W) x130mm (D)x27mm (H)
Weight	0.30kg
Temperature	40 – 130 F 5 – 45 C
Humidity	10 - 90%

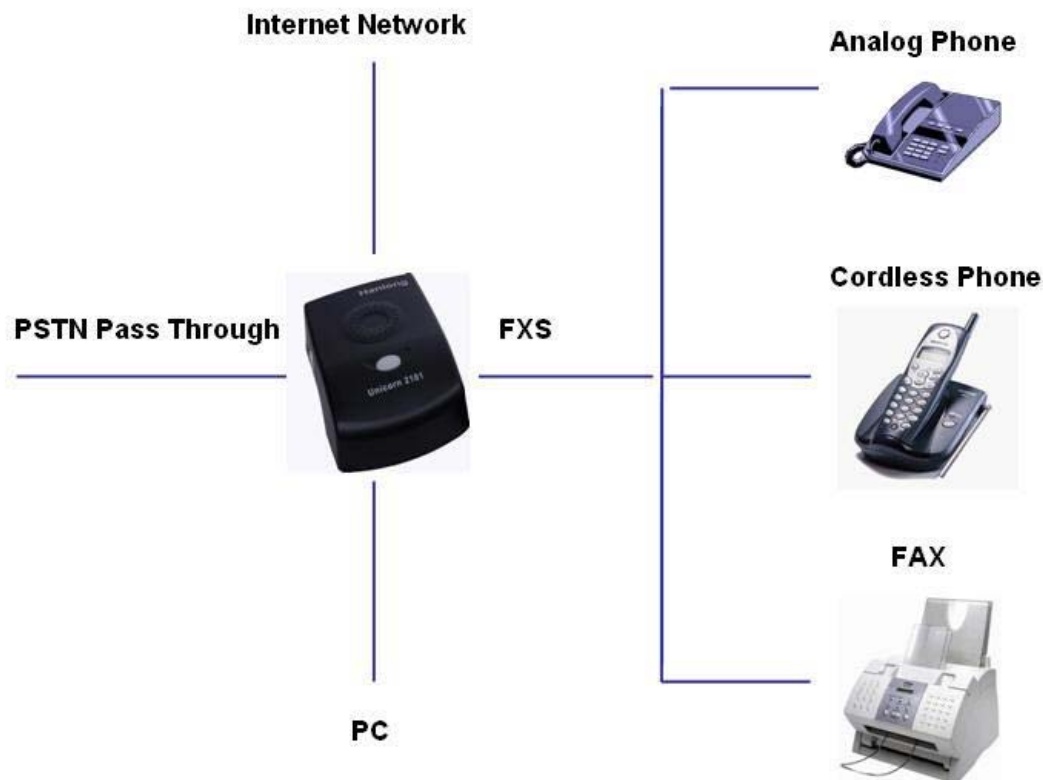
4 INSTALLATIONS

The Unicorn 2101 is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services. The Unicorn 2101 VoIP features are available when

you connect any regular analog telephone to it.

Unicorn 2101 has one FXS port (labeled “Phone”) and one PSTN pass through port (labeled “Line”). After setting up the Unicorn 2101, 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 2101:



Following are the steps to install a Unicorn 2101:

- Connect a standard touch-tone analog telephone to the “Phone” port.
- 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 2101 (Ethernet cable is included with package).
- Insert another Ethernet cable into the WAN port of Unicorn 2101 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 2101 and connect it to a power outlet.

4.1 Safety

The Unicorn 2101 is compliant with various safety standards including FCC/CE and C-Tick. Its power adaptor is compliant with UL standard. The Unicorn 2101 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 2101 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 2101 has stored a voice prompt menu for quick access to settings and simple configuration. You can enter this voice prompt menu one ways

- 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 2101:

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
01	“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.
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: <ul style="list-style-type: none"> ● 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: <ol style="list-style-type: none"> 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

Other Menu Prompt Features:

- Just like pressing "***" on the analog telephone, you will hear "Direct IP Calling", which is just like selecting option 47 from the table above.
- "*" 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 2101 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 2101).

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

Unicorn2101 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 Unicorn2101 and other VoIP Device, have public IP addresses, or
2. Both Unicorn2101 and other VoIP Device are on the same LAN using private IP addresses, or
3. Both Unicorn2101 and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

Unicorn2101 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 2101 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 2101 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 Make and Receive PSTN Calls

Users can send and receive calls from PSTN. To receive PSTN calls, simply pick up the phone receiver (or use handsfree) when the analog phone rings. To make a PSTN call, first press *00 (*00 is default. You can change the “PSTN Access Code” via the web configuration screen) to get the PSTN line dial tone and then dial the PSTN number you want to call.

5.3 Call Features

5.3.1 Call Features Tables

Following table shows the call features of Unicorn 2101:

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.
*91	Cancel Busy Call Forward. To cancel “Busy Call Forward”, dial “*91” and get the dial tone, then hang up.
*92	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.
*93	Cancel Delayed Call Forward. To cancel this feature, dial “*93”, get the dial tone, and then hang up.

Flash/Hook 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 2101 is out of power or loses registration or if the network connection is down, the RJ 11 line jack on the side of Unicorn 2101 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

Unicorn2101 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 2101 button light pattern indication.

Light Indicator	Signal Pattern	Status Meaning
Red Light	Red light flashes every 2 seconds (if internet connection is configured for DHCP)	DHCP failed or WAN port has no ethernet connection.
Red Light	Red light flashes every 2 seconds (if SIP server is configured)	Unicorn 2101 is not able to register with SIP gatekeeper/proxy server
Green Light	Button flashes every 2 seconds	Message waiting (if feature is available)
Green Light	Button flashes at 1/10 second	Phone is ringing. Incoming call in progress.

6 CONFIGURATION GUIDE

6.1 Configuring Unicorn2101 using Web Browser

Unicorn 2101 has embedded Web server and HTML pages that allow users to configure the Unicorn 2101 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 2101 configuration page:

FXS OPTIONS	
Account Active:	<input type="radio"/> No <input checked="" type="radio"/> Yes
SIP Server:	<input type="text" value="william.xanadu.hanlongtec.cn"/> (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	<input type="text" value="192.168.0.100"/> (e.g., proxy.myprovider.com, or IP address, if any)
NAT Traversal	<input type="radio"/> No <input checked="" type="radio"/> No, but send keep-alive <input type="radio"/> STUN <input type="radio"/> UPNP
SIP User ID:	<input type="text" value="8200"/> (the user part of an SIP address)
Authenticate ID:	<input type="text" value="8200"/> (can be identical to or different from SIP User ID)
Authenticate Password:	<input type="text"/> (purposely not displayed for securityprotection)
Name:	<input type="text" value="nico"/> (optional, e.g., John Doe)
Use DNS SRV	<input checked="" type="radio"/> No <input type="radio"/> Yes
User ID is phone number	<input checked="" type="radio"/> No <input type="radio"/> Yes
SIP Registration	<input type="radio"/> No <input checked="" type="radio"/> Yes
Unregister On Reboot	<input checked="" type="radio"/> No <input type="radio"/> Yes
Register Expiration	<input type="text" value="2"/> (in minutes, default 1 hour, max 45 days)
Outgoing Call without Registration	<input checked="" type="radio"/> No <input type="radio"/> Yes
local SIP port	<input type="text" value="5060"/> (default 5060)

6.1.1 Accessing the Web Configuration

The Unicorn 2101 configuration page can be accessed via the LAN or WAN port.

The GUI interface can be download at bellow:

http://www.hanlongtek.com/books/2101_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 FXS PORT PAGE

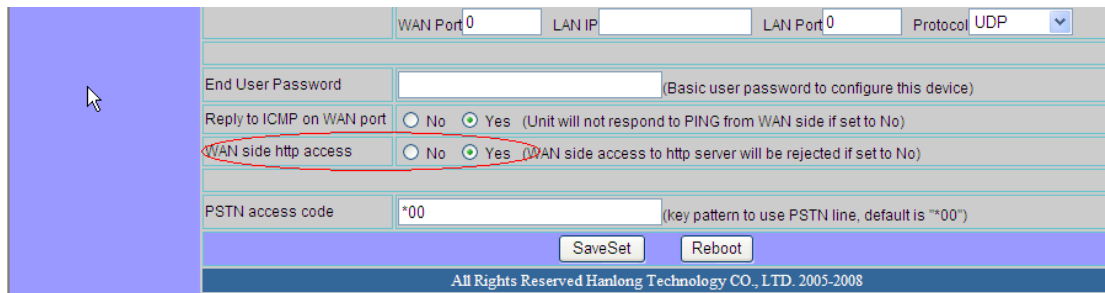
6.1.2 Programming Unicorn 2101 via the LAN Port

To program Unicorn 2101 via the LAN port, directly connect an Ethernet cable from your PC to the LAN port of the Unicorn 2101. 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 2101 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 2101 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 2101 via the LAN port”) and enable the “Enable WAN Web Access” option.

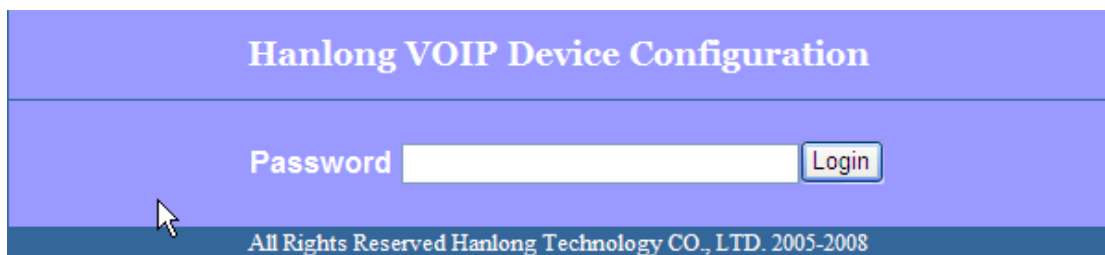
After enabling WAN access, be sure that the WAN port of the Unicorn 2101 is connected to an uplink (i.e. router, hub, switch, etc). Then, get the WAN IP address of the Unicorn 2101 by following section 5.1 of this user manual and selecting menu option 02. Then, access the Unicorn 2101 via your web browser by entering the WAN IP address:

http://Unicorn 2101’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 2101.

6.1.4 User Programming and Configuration

From your web browser, the Unicorn 2101 will show the following login screen:



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 passwords as indicated below:

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

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

Pages available for the end-user are:

- Device Status
- Basic Options

Additional pages available to super user are:

- Super Options.
- FXS port

6.1.6 Configuration Options and Explanations

After a correct password is entered in the login screen, the embedded web server inside the Unicorn 2101 will show the configuration page, which is explained in details below:

6.1.6.1 Device Status

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 Unicorn2101
Software Version	Information of software
System Uptime	Show system uptime since last reboot
PPPoE Link Up	Indicates where the PPPoE connection is up if the Unicorn2101 is connected to the DSL modem.
NAT	Indicate the NAT type behind which the device is when the stun feature is defined.
Port Status	Indicate the current status of the device such as hook (on/off), registration (registered/unregistered) etc.

6.1.6.2 Basic Options Settings

BASIC OPTIONS Setting	
Setting options	Definitions
Web Port	The default is 80.
IP Address	There are 3 modes under which the Unicorn 2101 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 2101 will acquire its IP address from the first DHCP server it

	<p>discovers from the office/home network it is connected to.</p> <ul style="list-style-type: none"> -To use the PPPoE feature, the PPPoE account settings need to be set. The Unicorn 2101 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 120 hrs (5 Days.) The time IP address is 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 Unicorn2101 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 Http access via WAN port is disabled.
PSTN Access Code	This field allows users to customize their own code to access PSTN. Default is *00.

6.1.6.3 Super Options Setting

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.
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48 .
Layer 2 QoS	This contains the value used for layer 2 VLAN tag. Default setting is 0 .
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 server	IP address or domain name of stun server.
keep-alive interval	This parameter specifies how often the Unicorn 2101 sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open.
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 select Upgrade 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 2101 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 this parameter is set to Yes , the configuration update via keypad is disabled.
Disable Voice Prompt	Default is NO .
Syslog Server	The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)
Syslog Level	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. Dialtone: 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

6.1.6.4 FXS Port

FXS PORT	
Setting options	Definitions
Account Active:	Set to the YES , the account can be available
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 2101 NAT traversal mechanism will be activated or not. If Choosing No , nothing to do. If Choosing No, but send keep-alive , the Unicorn 2101 will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to

	<p>keep the “hole” on the NAT open.</p> <p>If choosing STUN and a STUN server is also specified, then the Unicorn 2101 will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the Unicorn 2101 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 2101 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 2101 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.</p> <p>If choosing UPNP, the embedded UPNP client inside the Unicorn 2101 will attempt to mapping ports with the router by upnp protocol.</p>
SIP User ID	SIP service subscriber’s User ID
Authenticate ID	SIP service subscriber’s Authenticate ID. Can be identical to or different from SIP User ID
Authenticate Password	SIP service subscriber’s account password
Name	SIP service subscriber’s name which will be used for Caller ID display
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 2101 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 2101 needs to send REGISTER messages to the proxy server. The default setting is Yes .
Unregister On Reboot	Default is No . If set to Yes , the SIP user will be unregistered on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the Unicorn 2101 refreshes its registration with the specified registrar. The default interval is 6 minutes, The maximum interval is 65535 minutes (about 45 days).
Outgoing Call without Registration	Default is No. If set to “Yes,” user can place outgoing calls even when not registered (if allowed by ITSP) but is unable to receive incoming calls.

Local SIP port	This parameter defines the local SIP port the Unicorn 2101 will listen and transmit. The default value for FXS port is 5060 . The default value for FXO port is 5062 .
Local RTP port	This parameter defines the local RTP-RTCP port pair the Unicorn 2101 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 .
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 2101 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	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
Offhook Auto-Dial	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 2101 will automatically append the “@” and the host portion of the corresponding SIP address.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is

	blank.
Preferred Vocoder	The Unicorn2101 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 Vcoders 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 2002 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	Default is Callee . This decides whether Caller or Callee

Mode	sends out the re INVITE for T.38 or Fax Pass Through.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions.
Distinctive Ring Tone	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 Dial Key	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	<p>Dial Plan Rules:</p> <ol style="list-style-type: none"> 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; <ol style="list-style-type: none"> 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 <p>Example 1: {[369]11 1617xxxxxxx} Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617</p> <p>Example 2: {^1900x+ <=1617>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 <2=011>x+} Allow any</p>

	<p>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> <pre>{ ^1900x. <=1617>[2-9]xxxxxx 1[2-9]xx[2-9]xxxxxx 011[2-9]x. [3469]11 }</pre> <p>Explanation of example rule (reading from left to right):</p> <p>^1900x. - prevents dialing any number started with 1900</p> <p><=1617>[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>More information can be available at Dail Plan Notes.</p>
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 , the anonymous call will be rejected with 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	<p>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.</p> <p>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.</p>

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 analog 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.

<p>Volume Amplification</p>	<p>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.</p>
<p>Ring Tones</p>	<p>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.</p>

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:



The Unicorn 2101 will then display the following screen to confirm that the changes have been saved. Please allow 5 to 10 seconds before rebooting the device.



6.1.8 Rebooting the Unicorn 2101

You can reboot the Unicorn 2101 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 2101 and then powering it on again. If your Unicorn 2101 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 2101 via Voice Prompt

6.2.1 DHCP Mode

Follow section 5.1 with voice menu option 01 to enable Unicorn 2101 to use DHCP

6.2.2 Static IP Mode

Follow section 5.1 with voice menu option 01 to enable Unicorn 2101 to use STATIC IP mode, then use option 02, 03, 04 to set up Unicorn 2101's IP, Subnet Mask, Gateway respectively.

6.3 Configuration through a Central Server

Unicorn 2101 devices can be automatically configured from a central provisioning system.

When Unicorn 2101 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 "cfg001fc1xxxxx", where "001fc1xxxxx" is the MAC address of the Unicorn 2101.

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 2101s 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 2101 for firmware upgrade, etc.

7 SOFTWARE UPGRADE

To upgrade software, Unicorn 2101 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 2101.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the Unicorn 2101'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 2101. 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 2101.

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 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 2101. 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 "*" 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.

9 TECHNICAL SUPPORT CONTACT

Email: Support@mail.hanlongtek.com