User Manual

VOIP FXO Gateway
Model: HT-322

For DC12V Only

Shenzhen HyberTone Technology Co.,Ltd.
Http://www.hybertone.com
Marketing@hybertone.com
Support@hybertone.com

Contents

1 Product Introduction

1.1 General Information

A PSTN FXO (foreign exchange - office) gateway is a PSTN-to-digital adaptor that allows you to use your PSTN line to make VoIP calls. FOX gateway is designed to realize voice communication over a broadband IP network. It offers high voice quality with minimal bandwidth requirement. No matter it could access to the public IP address or not, the device has the advantage of easy installation in simple or complex network Multiple IP phones can also be installed in the same network with only one public IP address. It comes with enriched features for both network and phone call applications, such as broadband router, DHCP, LAN Phone System.

The HT-322 FOX Gateway is an all-in-one VoIP integrated device, which is designed to be a total solution for network's providing VoIP services. Its functions are available via a standard PSTN line. The HT-322 FXO gateway has two PSTN lines port.

1.2 Protocol

TCP/IP V4 (IP V6 auto adapt)
ITU-T H.323 V4 Standard
H.2250 V4 Standard
H.245 V7 Standard
H.235 Standard（MD5，HMAC-SHA1）
ITU-T G.711 Alaw/ULaw, G.729A, G.729AB, and G.723.1 Voice Codec
RFC1889 Real Time Data Transmission
Proprietary Firewall-Pass-Through Technology
SIP V2.0 Standard
Simple Traversal of UDP over NAT (STUN)
Web-base Management
PPP over Ethernet (PPPoE)
PPP Authentication Protocol (PAP)
Internet Control Message Protocol (ICMP)
TFTP Client
Hyper Text Transfer Protocol (HTTP)
Dynamic Host Configuration Protocol (DHCP)
Domain Name System (DNS)
User account authentication using MD5
Out-band DTMF Relay: RFC 2833 and SIP Info

1.3 Hardware Specification

ARM9E Processor
DSP for voice codec and voice processing
Two 100M BaseT Ethernet ports in comply with IEEE 802.3 for both LAN and PC connection.
LEDs for Ethernet port status
Two PSTN Line ports
Build in Ethernet switch

1.4 Software Specification

LINUX OS
Built-in HTTP for programming
PPPoE dial-up
NAT Broadband Router functions
DHCP Client
DHCP Server
Firmware On-line upgrade
Caller ID
Multiple Language Support

1.5 List of the Package

a) One HT-322 main unit
b) One DC12V/500mA power adaptor
c) One Ethernet cable (3M)

1.6 View of the Appearance

1） FXO 1
It is connected to the PSTN with RJ11 connector.

2） FXO 2

It is connected to the PSTN with RJ11 connector.

3） Line1

Internal connected to the Line1; User can connect a spare phone to this socket.

4） Line 2

Internal connected to the Line1; User can connect a spare phone to this socket.

5） LAN

Connect to the switch, DSL modem or other network access equipment.

6） PC

Connect a computer or other terminal.

7） POWER DC12V/500mA

It is the power source, connect to the power adapter.

8） Reset

Press it with a nail to reboot the terminal or long press to reset terminal configuration.

2 Installation

2.1 Installation Steps

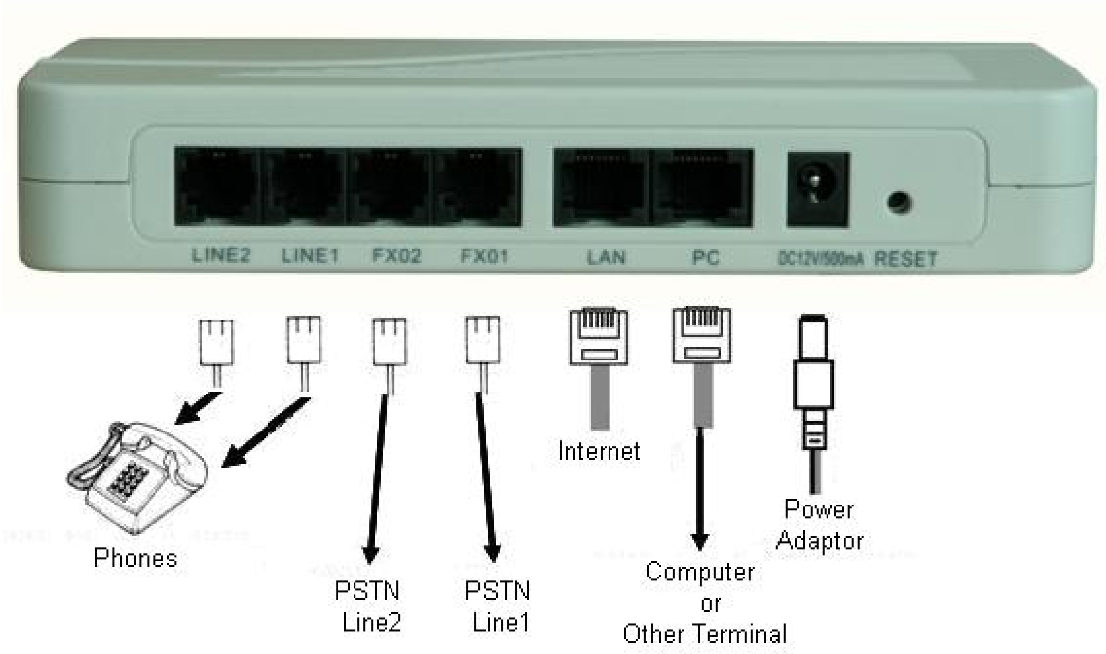
The HT-322 IP ADAPTOR has two PSTN ports（FXO1 and FXO2），one NETWORK port and one PC port. The VoIP channels can login by same SIP (or H.323) account or two different SIP (or H.323) accounts.

Please follow the steps below to install an HT-322 IP ADAPTOR:

a) Connect the PSTN lines to the FXO1 and FXO2.
b) Insert the Ethernet cable into the LAN port of HT-322 IP ADAPTOR and connect the other peer to the internet access equipment.
c) Connect a PC to the PC port of HT-322 IP ADAPTOR.
d) Insert the power adapter into the HT-322 IP ADAPTOR and power up.

2.2 Connection Diagram

The interconnection diagram is as below:



2.3 LED Light Pattern Indication

The following table shows the LED light pattern indication.

LED	DESCRIPTION
RUN	1. When the device is booting up, the LED will flash 100ms ON and 100ms OFF. 2. When the account is logon, the LED will flash 1s ON and 1s OFF. 3. If the ATA does not boot up, the LED will not flash. 4. Normally boot up and login time needs about 30 seconds.
LAN	1. When the LAN port linkup, the LAN LED will lights up 2. when data is transmitting the LAN LED will flashing
PC	1.When the PC port linkup, the LAN LED will lights up 2. When data is transmitting the PC LED will flashing.
FXO1	Lights up when the line is in use

FXO2	Lights up when the line is in use
Line1	No use in this version
Line2	No use in this version

3 Configuration Guide

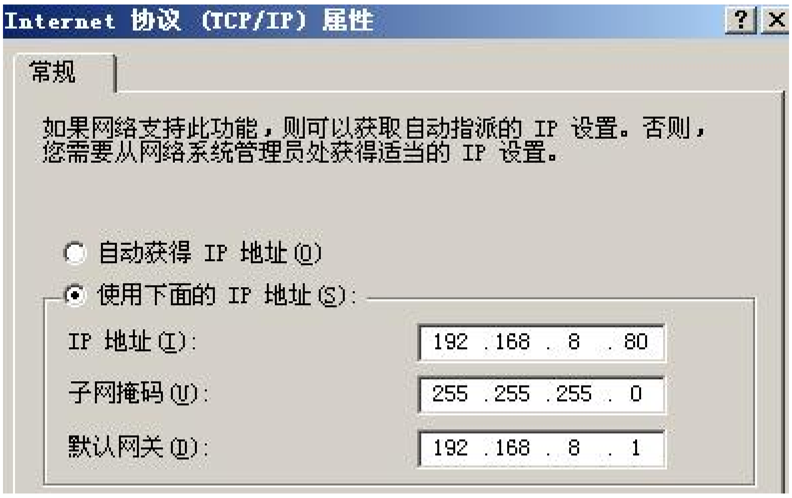
Before configuring the HT-322 IP ADAPTOR, first you must configure you computer's IP in Static Mode and it must use 192.168.8.x. The default network setting of HT-322 is:

LAN Port: DHCP

PC Port: 192.168.8.1

Method as follow:

Windows's **Control Panel-- à Network Connections-- à Local Connectionism's Property-- à TCP/IP Protocol's Property**



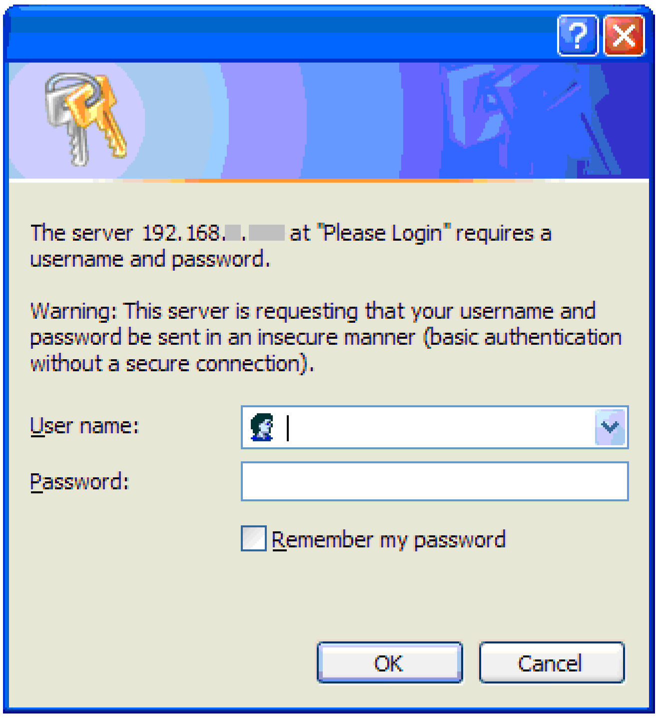
Setup the computer's network setting as the picture above. After this, you can login the HT-322 via the HT-322's PC port IP.

Note: Regarding the HT-322's default setting, please consult the chapter **5.Manufactory Setting.**

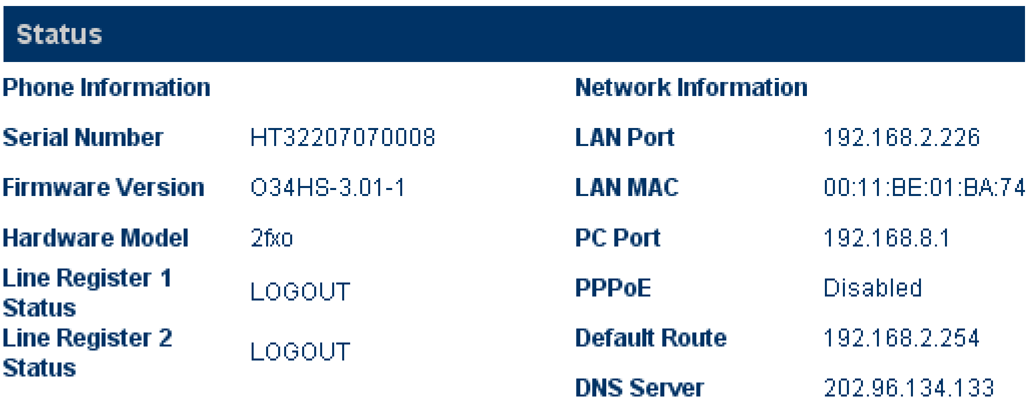
3.1 Access the Web Configuration Menu

After configured your computer, you can login to the HT-322's configuration page by Web browser. Just input the IP: 192.168.8.1 to the Address line.

A login dialog window will pop up:



The default user name is “admin” and the default password is “admin”. Press OK then accessing the configuration page as below.



Administrator (user admin) can setup every thing of the HT-322.

3.2 Status

Status page shows the status of the gateway.

3.2.1 Phone Information

A. Serial Number

Each gateway has a unique serial number assigned by the factory such as HT304O12VTEST01. This number is important for centralized configuration, technical support, and warranty. This number is printed on the bottom of the gateway and is associated with your software license.

B. Firmware Version

Firmware version refers to the firmware version of the gateway such as A34HS-3.07-18, which is used to identify the firmware version.

C. Hardware Mode

This field shows terminal's hardware type.

D. Phone Status

This field shows the status of Line's connection status. If the connection is successful, this field displays LOGIN. Otherwise displays LOGOUT.

3.2.2 Network Information

A. LAN Port Configuration

Display the status of the LAN port.

B. PC Port Configuration

Display the status of the LAN port.

C. PPPoE

When PPPoE is enable the working message will display.

D. Default Route

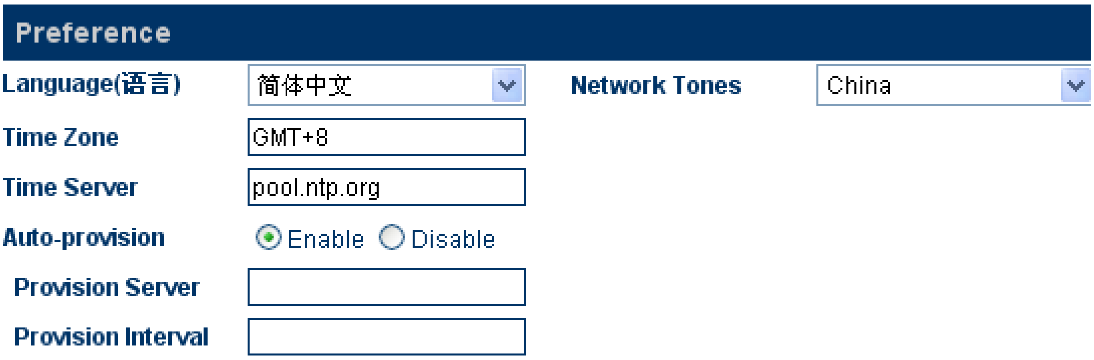
The IP address of the routing gateway.

E. DNS Server

Display the Domain Name Server of the current network.

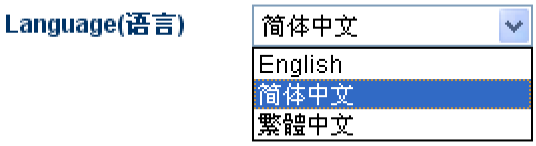
3.3 Configurations Options

Click on the "Configurations", the web page will pop-up a subordinate options. Click on "Preference" in the left menu of the configuration web, and the screen will be displayed as below:



3.3.1 Language

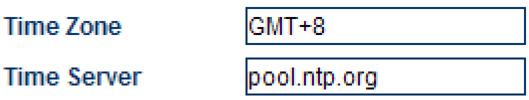
Select web provisioning language as English or Chinese. For example, if you're present language used is English; please click "English" in the "Language" menu, when the terminal was reboot the web page will display all information in English.



You can choice the language use the quick link, the Web page's language will change at soon.

3.3.2 Time Zone and Time Server

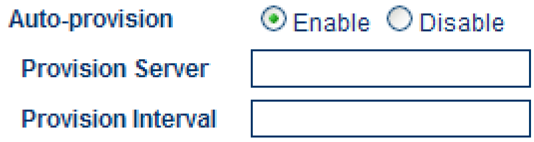
The HT-322 using Network Time Protocol (NTP) to get the date time information from an NTP server (Time Server). The time is in GMT ± offset. For example, Pacific Standard Time is GMT-8, and Pacific Daylight Time is GMT-7.



For a FXO gateway the real time will appear on the CDR (call detail record), so it is important if user needs rigors CDR.

3.3.3 Auto-provision

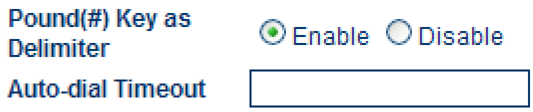
Allow central provisioning of HT-322. Auto-provision is the automatic configuration of devices without manual intervention and without any need for software configuration programs or jumpers. Ideally, auto-provision devices should be "Plug and Play". Auto-provision has been made common because of the low cost of microprocessors and other embedded controller devices. It includes Auto-provision Server and Auto-update.



This is the special service only. It must support by HyberTone's Auto-provision Server.

3.3.4 Pound (#) Key and Auto-dial Timeout

If you want the terminal send out your telephone number shortly, please click the Pound (#) Key enable. If it was disable, the terminal will call out after 5 seconds (or the Auto-dial timeout) after the last digit is input

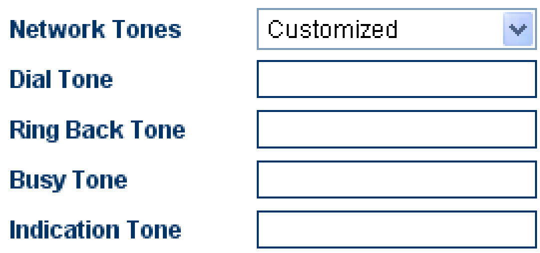


Note: This function only inure when the call from PSTN to VoIP's secondly dial number.

3.3.5 Network Tone

Network Tones is the dial tone of the second dial. Users can choice the tone like the local telephone network.

You can configure the Network Tones as Customized option:



The parameters are fill-in as true digits as below format:

nc, rpt, c1on, c1off, c2on, c2off, c3on, c3off, f1, f2, f3, f4, p1, p2, p3, p4

Chapter Descriptor:

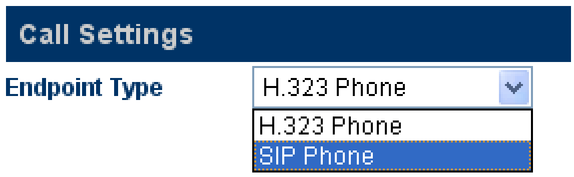
nc: number of cadences
rpt: repeat counter(0 - infinite, 1~n - repeat 1~n times)
c1on: cadence one on (in milliseconds)
c1off: cadence one off (in milliseconds)
c2on: cadence two on (in milliseconds)
c2off: cadence two off (in milliseconds)
c3on: cadence three on (in milliseconds)
c3off: cadence three off (in milliseconds)
f1: tone #1, 300-3000(Hz)
f2: tone #2, 300-3000(Hz)
f3: tone #3, 300-3000(Hz)
f4: tone 34, 300-3000(Hz)
p1: attenuation index #1, 0~31(0=3dB, -1dB increments)
p2: attenuation index #2, 0~31(0=3dB, -1dB increments)
p3: attenuation index #3, 0~31(0=3dB, -1dB increments)
p4: attenuation index #4, 0~31(0=3dB, -1dB increments)

For example, you want add a tone as 450Hz and it is run 700 milliseconds and off 1000 milliseconds. You only add below parameters in corresponding option:

1,0,700,1000,0,0,0,0,450,0,0,0,20,0,0,0

3.4 Call Settings

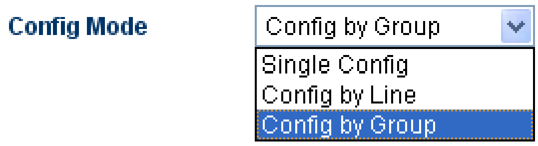
Call setting let users program the parameters between gateway and the server(s).



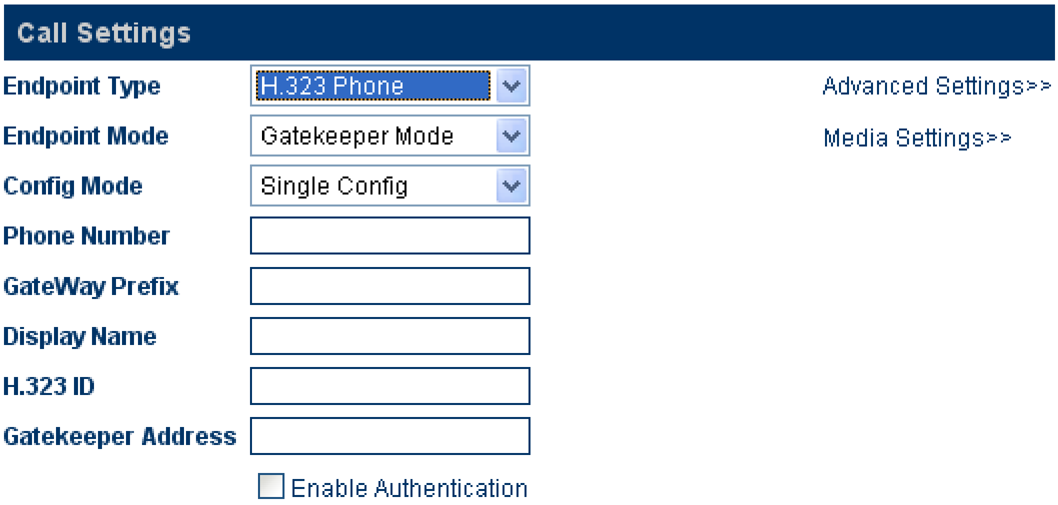
3.4.1 H.323 Phone

Select the end point type to H.323 phone. There are three kinds of login mode can be configured:

1） Single Configuration; 2） Configuration by line ; 3）Configuration by Group .



3.4.1.1 Single Configuration



In this mode, the two lines will login to the server with single number or prefix:

A. H.323 Phone Number

H.323 phone number: fill the login number (E164) here.

B. Gateway Prefix

If login with a Prefix method fill the prefix number (do not fill the Phone number)..

C. Display Name

Display name will display on the other VoIP party fill in the display name of the gateway.

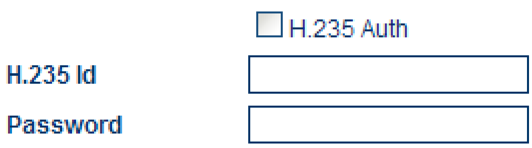
D. H.323 ID

If the system needs H.323ID for the Authenticate fill in the Authentication message.

E. Gatekeeper Address

Fill in the gatekeeper address with IP or domain name. If the registration port is not 1719 then add "**:**" and the port number. Like "192.168.1.70:8080".

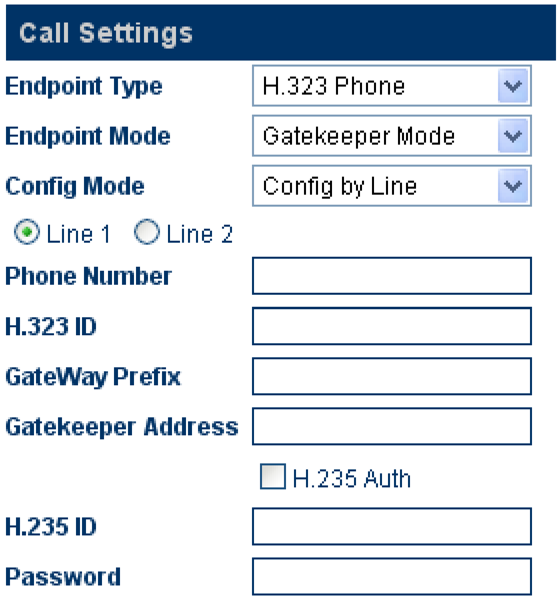
F. Enable Auth



If H.235 Authentication is needed click the “H.235 Auth” and fill the Authenticate message.

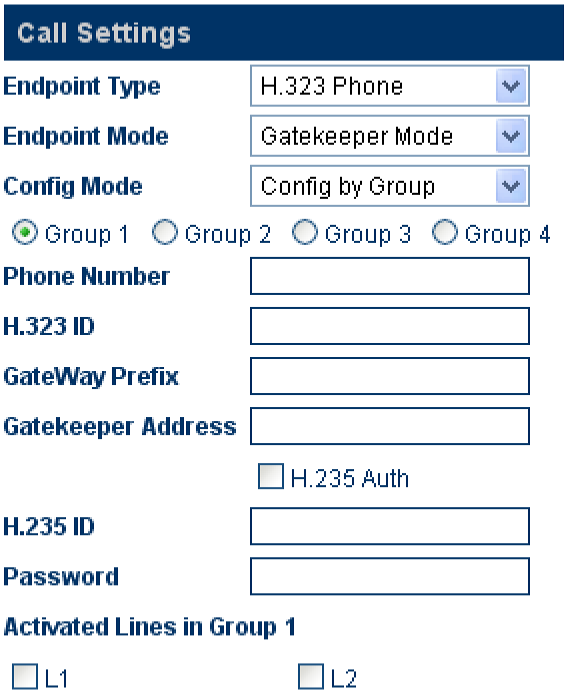
3.4.1.2 Configuration by Line

In the “**Config by line**” mode user can setup each of the two lines with difference numbers or prefixes. For advance using the each line can login difference server.



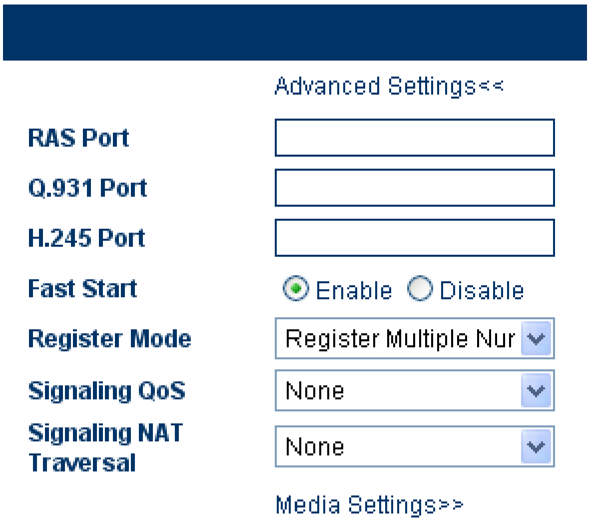
3.4.1.3 Configuration by Group

In the “**Config by Group**” mode user can setup each of the four lines with difference groups. For advance using the each group can login difference server.



3.4.1.4 Advance Settings

Click "**Advance Settings**" in the H.323 menu, the configuration screen of advance settings will be displayed as follows. User can modify all information of H.323 and the preference of the media.



A） RAS Port

RAS Port is connected to an unreliable channel, which is used to convey the registration, admissions, bandwidth change, and status messages between two H.323 entities.

B） Q.931 Port （ Call Signaling Port ）

Call Signaling Port is connected to a reliable channel used to convey the call setup and release messages between two H.323 endpoints.

C） H.245 Port （ Media Control Ports ）

Media control port is the port or port range used by the H.245 media control protocol.

D） Fast Start

Enable or disable the Fast Start in H.225.0. Only few H.323 terminals or gateways do not support Fast start.

E） Register Mode

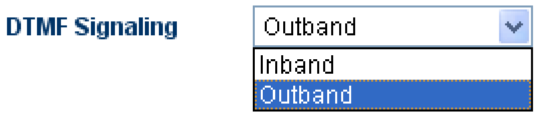
Register Multiple Numbers: The HT-322 send registration request in one signaling packet to the gatekeeper. In the mode, one signaling packet includes two VoIP line's registration information.

Register Multiple Times: In this mode, the HT-322 will register like two terminals.

F） DTMF Signaling

1） DTMF TYPE

DTMF TYPE is used for telephone signaling over the line in the voice frequency band to the call switching center. DTMF means that two groups of tone with different frequencies are united into 16 kinds of dial tone. The telecommunication station, switch or the phone service such as 1860 identify these special tones through DSP analysis to confirm the keys dialed by user. There are two DTMF types: Inband DTMF type and Outband DTMF type.



Inband DTMF type transfers these special dial tones together with the speech tone without doing any special processing to them. So there is not any type as to Inband DTMF.

Outband DTMF type transfers these special dial tones by some special method to confirm its correctness. The special method is the so called protocol such as RFC2833 and SIP information.

2）DTMF Payload Type

DTMF Payload Type is used to carry telephony tones and telephony signals. By using a distinct dynamic RTP payload type in the same RTP stream as the media, it is possible to carry DTMF tones, fax-related tones, standard subscriber line tones, country-specific tones and trunk events.

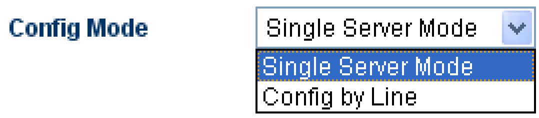
3.4.1.5 H.323 Direct Mode

For the IP to IP call.

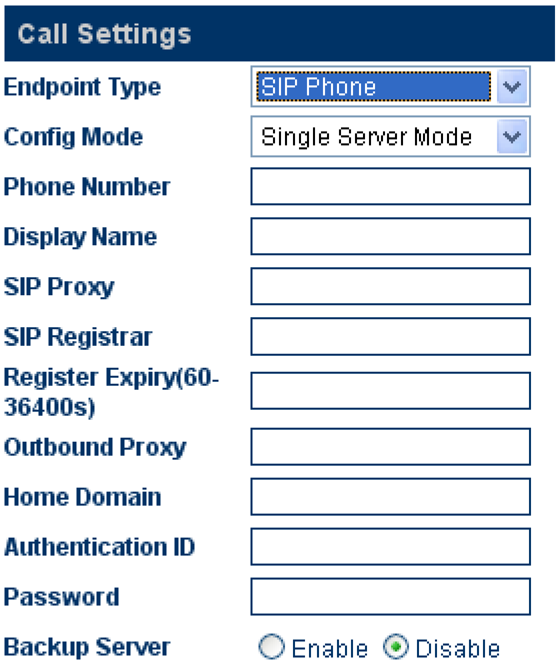
3.4.2 SIP Phone

When user using SIP system, select SIP Phone in the “Endpoint Type”.
There are two configuration modes:

1) Single Server Mode; 2) Configuration by Line



3.4.2.1 Single Server Mode



In this mode, the two lines will login to the server with single number**.**

A）Phone Number

Fill in the login number of SIP.

B）SIP Proxy

Fill in the SIP proxy IP address or domain name. If the registration port isn’t 5060 then add “**:**” and the port number like:”sip.at338.com:8080”

C）SIP Registrar Server

If a registrar server is needed fill in the IP address or domain name.

If the registrar port isn’t 5060 then add “**:**” and the port number like:”sip.at338.com:8080”

D）Home Domain

This field use fill in the IP address of the home domain. Although the IP address is easy to remember, the user still finds it hard to remember because it is long and has no association with its geographical position or organizational relationship. In order to solve the problem, Internet defines the Domain Name System (DNS) by using ASCII string.

E）Authentication ID

Fill in the authentication ID (user name) if it is needed.

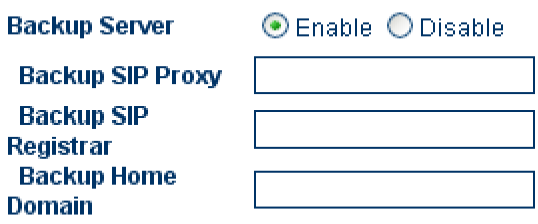
F） Password

Fill in the authentication password if it is needed.

G） Display Name

Fill in the display name and it will display on the other VoIP party.

H） Backup Server



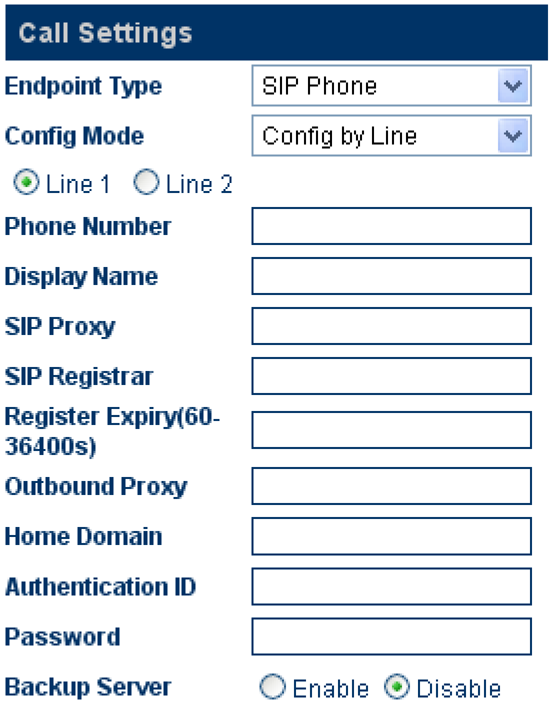
When system support backup server fill in the messages of the backup server. When the first server is failed HT-322 will try the backup server.

I） Outbound Proxy

When Outbound proxy is needed fill in the IP address or domain name..

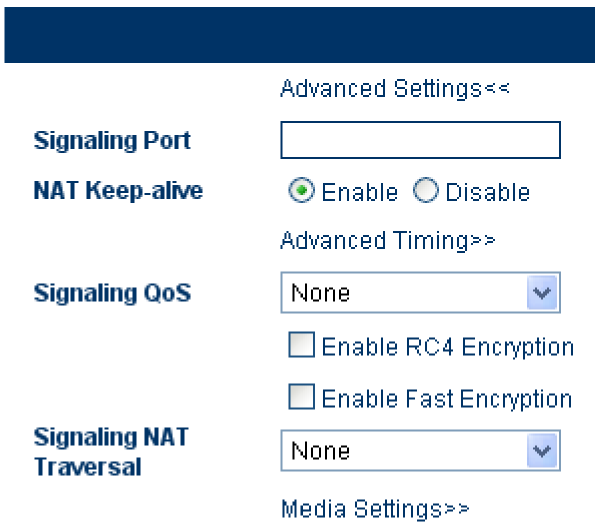
3.4.2.2 Configuration by Line

In the “**Config by line**” mode user can setup each of the two lines with difference numbers. For advance using the each line can login difference server.



3.4.2.3 Advance Settings

Click on "**Advance Settings**" in the SIP menu, the configuration screen of advance settings will be displayed as follows. In this part, you will have more advance control over the SIP signaling and media preference.



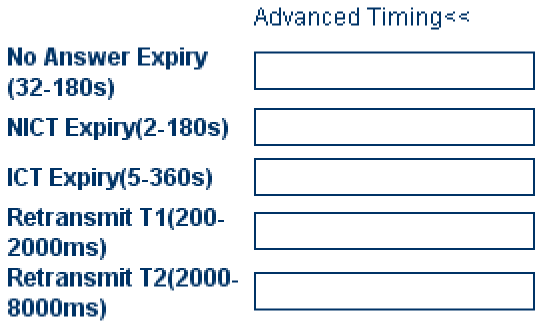
A） Signaling Port (SIP Local port)

Set the local signaling port. If 5060 is not your favor.

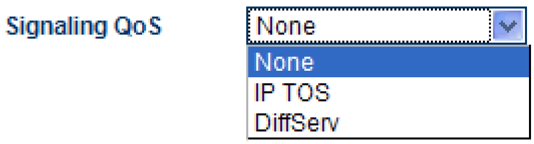
B） NAT Keep-alive

Periodically send out a packet (with no content) to the SIP proxy to keep NAT open for incoming traffic.

C） Advanced Timing



D） Signaling Qos



If local network supporting Qos, Choice one of them.

E） DTMF Signaling

1） DTMF TYPE

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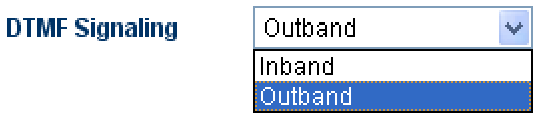

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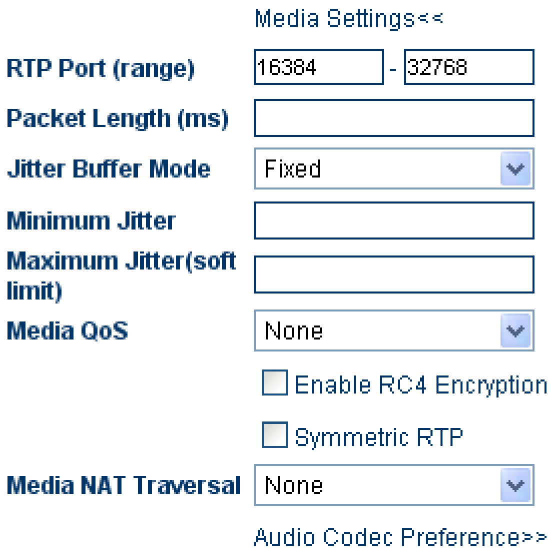
3.4.3 Media Setting

Click on “**Media Settings**” in the “Call Setting” menu, Media setting allow user change the preference of the media and RTP etc.



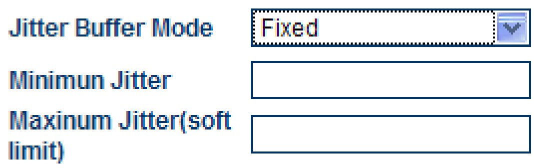
A） RTP Ports

RTP (Real Time Protocol) Port id the Flexed port of the RTP.

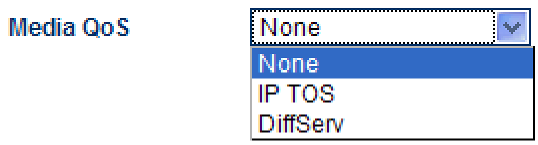
B） Packet Length

Default length is 20mS; user can change the packet size from 5mS to 40mS with 5mS per step.

C） Jitter Buffer Mode



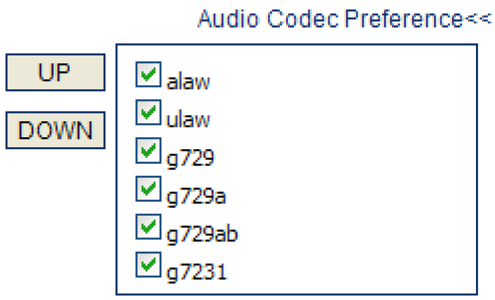
D） Media Qos



If local network supporting Qos, Choice one of them.

3.4.4 Codec Preference

Codec Preference let user to choice the audio codec sequence and disable some of the codec.

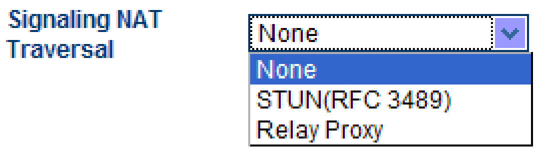


To upper or lower one of the codec of the sequence; highlight one of them then kick UP or DOWN button. To disable or enable one of the codec click the "□" front of the codec names.

3.4.5 NAT Traversal

3.4.5.1 Signaling NAT Traversal

Signaling traversal is setting the method of the signaling traversal between NAT.



A） None

None means there is no traversal mechanism supported.

B） Port-forwarding Support

Port-forwarding is the action of forwarding network ports on the outside network interface (public IP) to inside network (private IP). Check your NAT (router)setting before user choice this function.

Port-forwarding support includes gateway address and echo server address. Gateway address id the IP of the network gateway. Echo server is a standard service implementing the ECHO protocol.

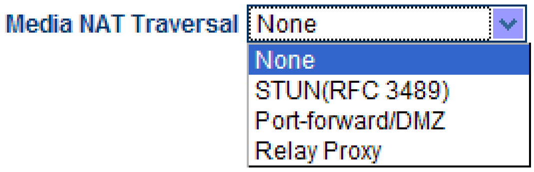
C） Relay Proxy

Relay proxy is HyberTone's proprietary NAT traversal technology which allow HyberTone's VoIP terminals to achieve anywhere of the complicated network environments. It includes address, ports, user name, and password.

Note: This RELAY proxy is only working with the RELAY Server software developed by HyberTone independently. It was free software; user can get it from our technical support.

3.4.5.2 Media NAT Traversal

Click on the "Media NAT Traversal" in the "Call Settings" configuration page's "Media Settings NAT Traversal" screen will be displayed as below.



Media NAT traversal is setting the method of the RTP traversal between NAT:

A） None

None means there is no traversal mechanism supported.

B） Port-forwarding Support

Port-forwarding is the action of forwarding network ports on the outside network interface (public IP) to inside network (private IP). Check your NAT (router)setting before user choice this function.

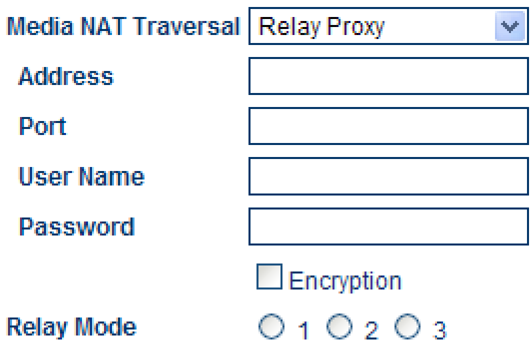
Port-forwarding support includes gateway address and echo server address. Gateway address id the IP of the network gateway. Echo server is a standard service implementing the ECHO protocol.

C） STUN （RFC 3489 ）

When system support STUN then choice it and fill in the STUN server IP (or domain name) address.

D） Relay Proxy

Relay proxy is HyberTone's proprietary NAT traversal technology which allow HyberTone's VoIP terminals to achieve anywhere of the complicated network environments. It includes address, ports, user name, and password..



The RELAY can run in 3 kind of packaging mechanism:

Mode 1: The media use UDP packets and (or) encrypt with multiple UDP port;

Mode 2: The media use UDP packets and (or) encrypt with single UDP port;

Mode 3: The media use TCP packets and (or) encrypt (UDP over TCP).

3.5 Call divert

The two VoIP and PSTN channels both can be program to divert mode.

Forward Number (VoIP to PSTN)

When forward number (VoIP to PSTN) is filled, the VoIP incoming call will forward to the number of the PSTN immediately. For the PSTN number "**,**" is supported, it means 500mS delay between two digits.

Forward Password (VoIP to PSTN)

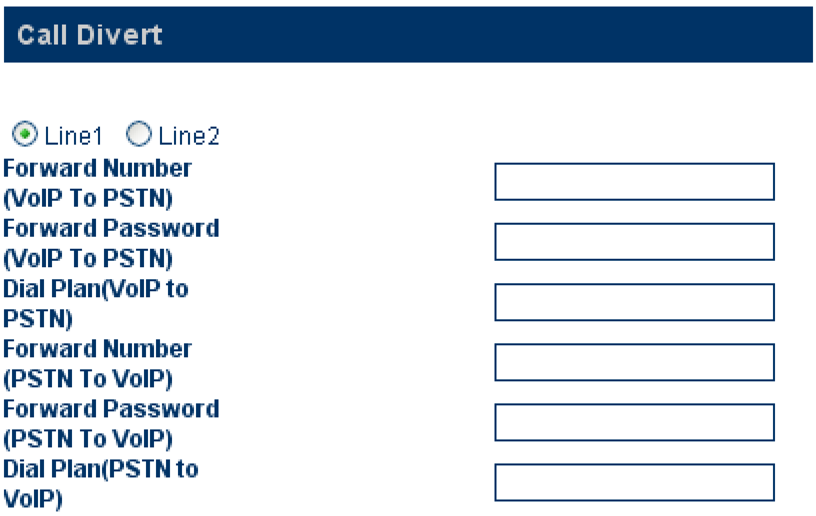
When the forward Password is filled, the call to PSTN needs input the "Password" before the call is making. The call flow is: call the HT-322 -> hears the indication tone -> input the password by press the keypad (by DTMF) -> get the PSTN dialing tone -> make call.

Forward Number (PSTN to VoIP)

When forward number (PSTN to VoIP) is filled, the PSTN incoming call will forward to the number of the VoIP immediately. And when the VoIP side pickup the call will be establish.

Forward Password (PSTN to VoIP)

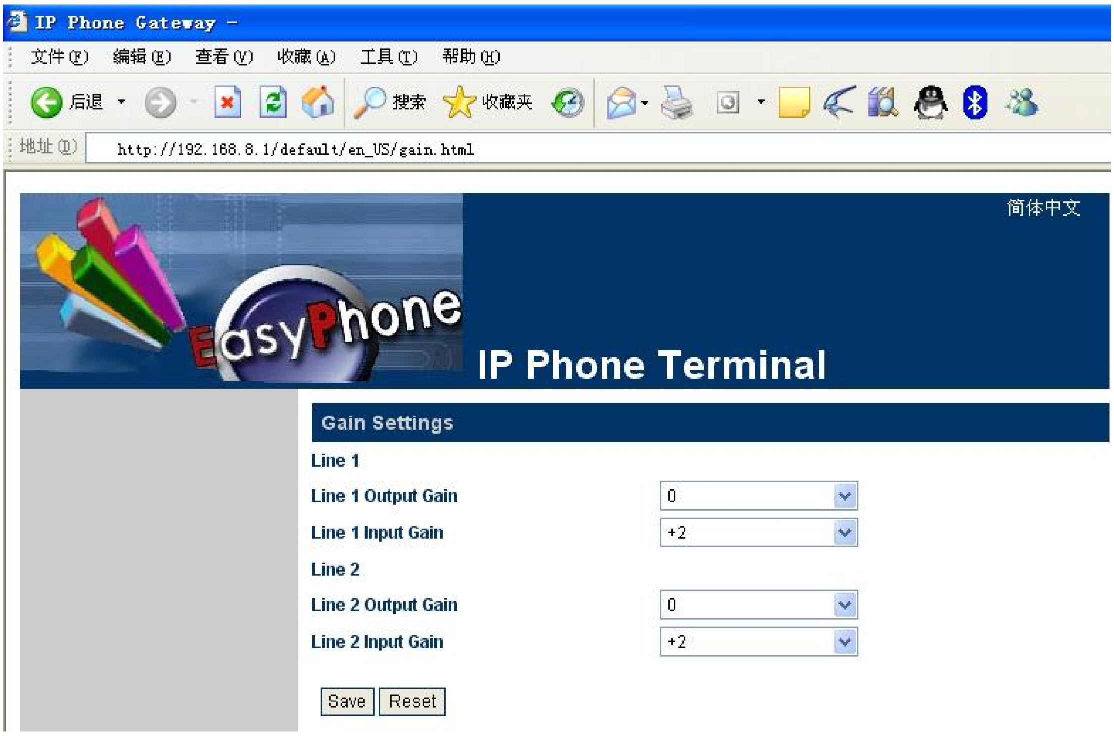
When the forward Password is filled, the call to VoIP needs input the "Password" before the call is making. The call flow is: call the HT-322 -> hears the indication tone -> input the password by press the keypad (by DTMF) -> get the VoIP dialing tone -> make call.



3.6 Gain Settings…

The gain setting must be cautious to using. It was a hidden web page. If you want adjust the VoIP Lines's volume. Please rewrite the URL address to *http://xxx.xxx.xxx.xxx/default/en_US/**gain.html*** and enter. The web browser will show up a GAIN SETTINGS screen.

You can adjust the volume of the two PSTN Lines to different values. The range you can adjust is from 5 to -5.



Note: A too low or too high input gain will make DTMF detector insensitive.

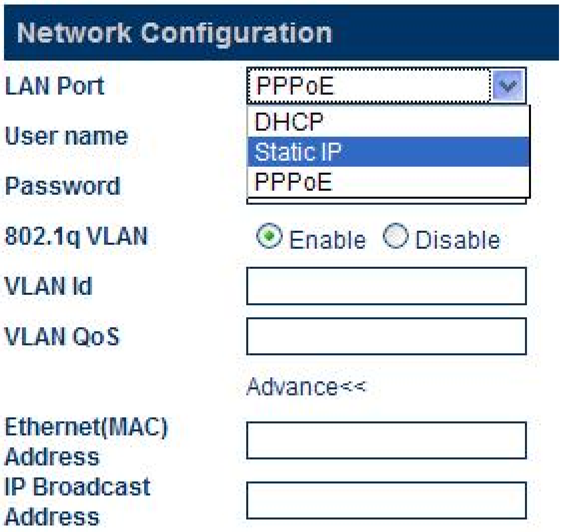
3.7 Network Configurations

Click on Network Configuration in the left menu of the Configuration page and the network configurations screen will be displayed as below.

The Network Configurations screen allows user setup the IP addresses of the LAN and PC port. User also can setup PC port to a DHCP server or working in Bridge mode.

3.7.1 LAN Port configurations

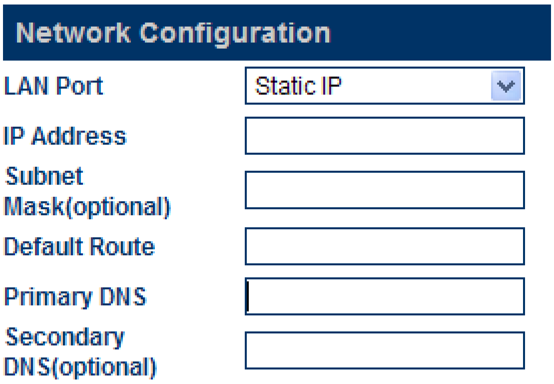
Choice one of the LAN configuration that meet your network environment, they are:
1） DHCP, 2） Static IP, 3） PPPoE.



1） DHCP

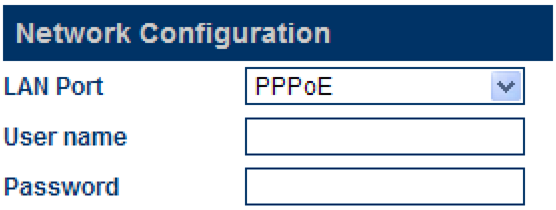
Choice DHCP when the network offering DHCP service.

2） Static IP



Choice Static IP when network needs a static IP and fill in the messages of then.

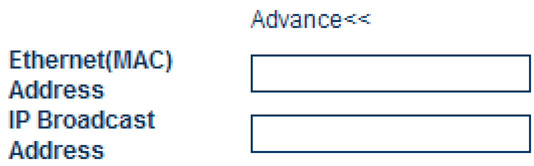
3） PPPoE



PPPoE, point-to-point protocol over Ethernet, is a network protocol for encapsulating PPP frames in Ethernet frames. It is used mainly with cable modem and DSL (Digital Subscriber Line) services. Click PPPOE, and fill in the user name and password accurately.

4）. Advance…

Click Advance and you will get the two items: Hardware address and Broadcast address.

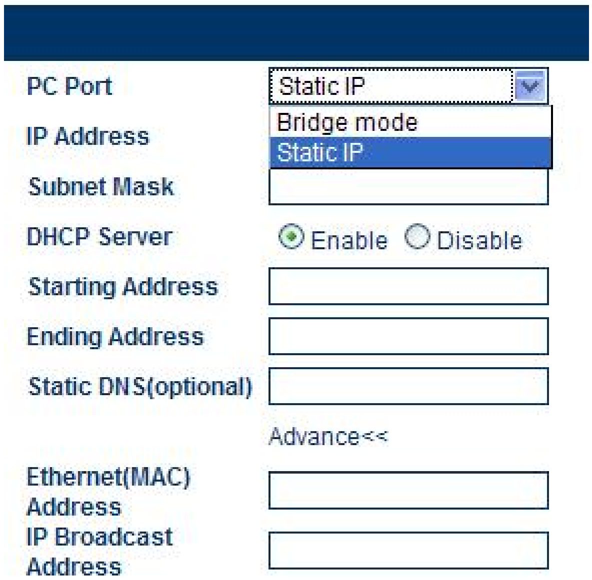


Fill in if the MAC address and broadcast need to change to specify.

3.7.2 PC port configurations

PC port can setup into Bridge Mode or Static IP. Strongly recommend user using static IP mode because of in bridge mode PC port don't has a IP for login the configuration page.

The default setting is Static IP in 192.168.8.1.

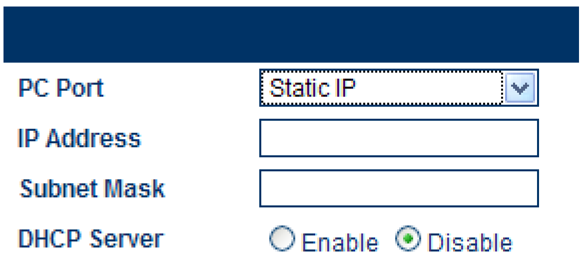


1） **Bridge Mode**

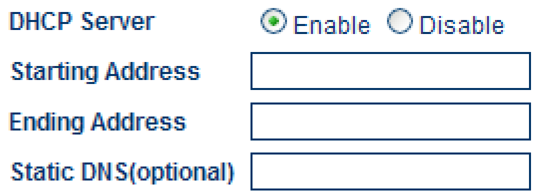
In the Bridge Mode, PC port is link to LAN port with a internal switch, device which connect to PC port same as connect to the LAN cable.

2） **Specify an IP Address Manually**

Click Static IP, and enter IP Address and Subnet Mask.



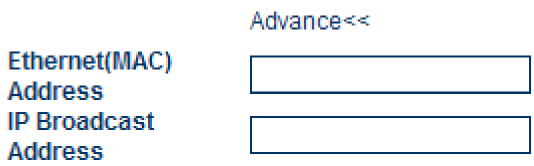
3） Enable DHCP Service



When users enable the DHCP service, PC port become a DHCP server and offer DHCP service. Please also fill in the starting address and ending address of the DHCP service. If the LAN port setting to static IP and user want to offer DNS field in DHCP please also fill in the DNS server IP address.

4） Advance…

Click Advance and you will get the two items: Hardware address and Broadcast address.



Fill in if the MAC address and broadcast address need to change to specify.

3.7.3 Primary DNS

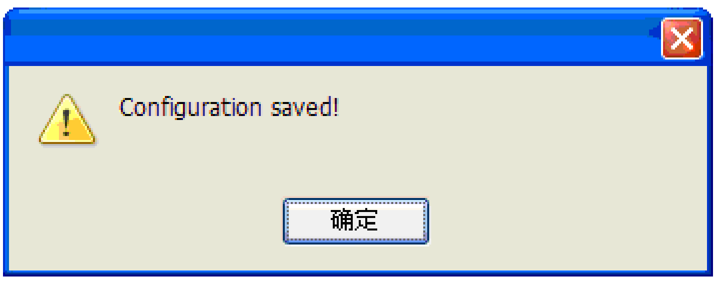
When LAN port is setting to static IP DNS must be set. If there is not DNS IP filled HT-322 and the device(S) behind the HT-322 can not access any address with **Domain Name**.

3.7.4 Secondary DNS

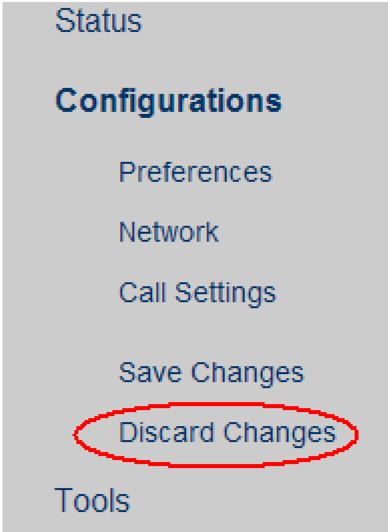
Fill in a backup DNS server IP address here.

3.8 Save Configuration

Once a change is confirm, users should click on the “Save Configuration” button in the Configuration page. Otherwise, your configuration will not take effect. After user click the “Save configuration” the screen will be:

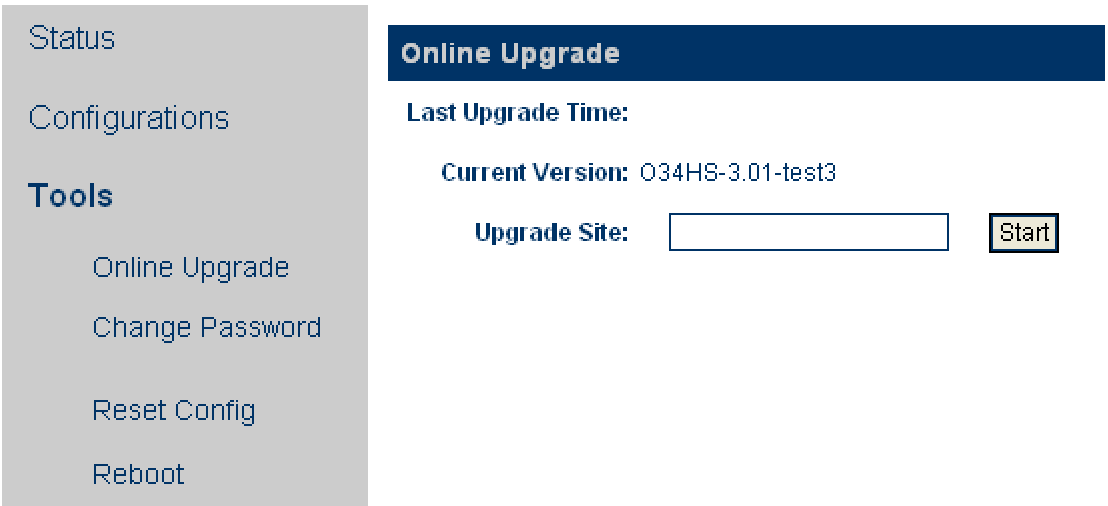


3.9 Discard Changes



3.10 Tools Menu

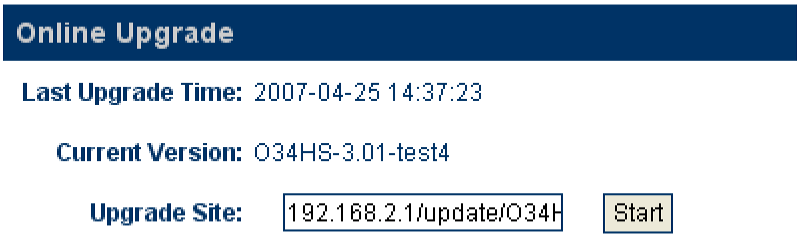
Click on the “Tools”, the web page will pop-up a subordinate options and the screen will be displayed as below:



3.10.1 Online Upgrade

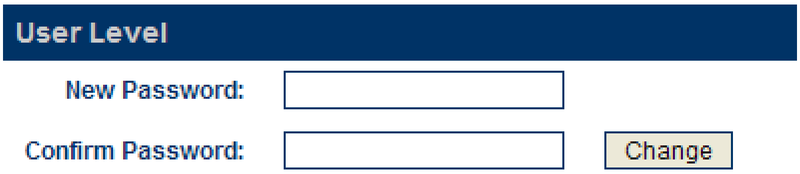
WARNING! Performing an online upgrade is for experienced users or administrators only!

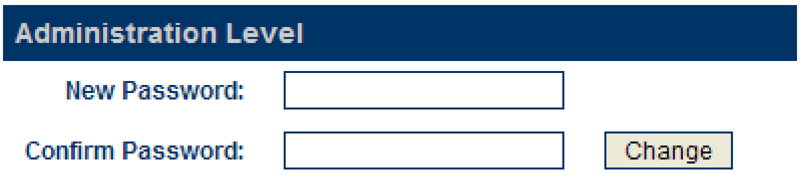
Click Online Upgrade. Enter the update link which provide by HyberTone like: *http://202.96.136.145/update/A34HS-3.07-18.pkg*， And click the Start button.



WARNING! When the terminal is updating, never power down the HT-322. It will destroy the equipment.

3.10.2 Change Password





A） **User Password (default:1234)**

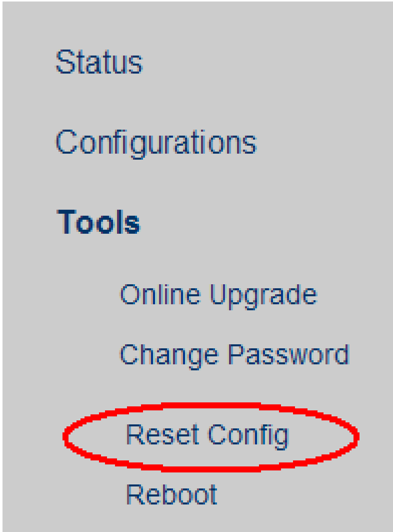
User password is for the user of “user”. “user” is the lower level user of the configuration, when “user” login, “Call setting” will not show. It means “user” only can setup something like network etc.

B） **Administrator Password (default: admin)**

Administrator Password is the password of “admin”. ”admin” can change anything of HT-322 so please reminded change the password of “admin” after you finish configure the device.

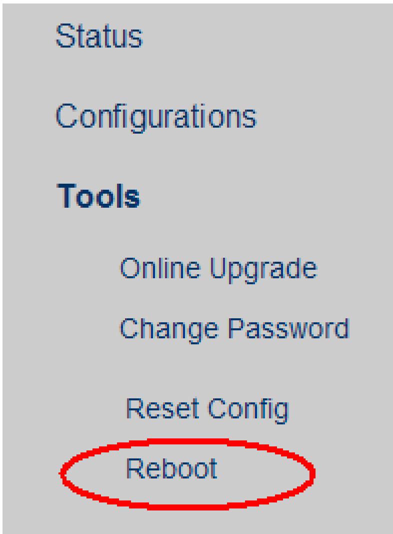
3.10.3 Reset Configuration

When reset configuration is click HT-322 will reset to factory default.



After reset all the configured will be lost.

3.10.4 Reboot the Device



Click on this field to reboot the device.

4 Products Parameter

Characteristics of the hardware	Parameter	Remarks
Processor	ARM9E 133MHz	
DSP	VDS924PM4 200MHz	
RAM	16M	
FLASH	4M	
Power	DC12V/500mA +-10%	
Consumption	The maximum 3 W	
Impedance	600 ohm	
Ring Detect Range	30-150Vrms	18Hz-55Hz
PSTN Port Voltage Upper Limit	160V	
LED	Operate, network , circuit	
Network card	2	100/10BASE-T
Weight	230 grams	WHITOUT DC ADAPTER
Working temperature	0－42℃	
Working humidity	40％－90％ not congealed	
Colour	White	
PSTN Ports	2	
VoIP Chunnel	2	

5 Manufactory Parameter

Parameter		Default
Network	LAN	DHCP
	PC	Static IP:192.168.8.1
Password	admin	admin
	user	1234
Time Zone		GMT＋8

This default parameter are unsuitability the customization’s products.