D-Link
Building Networks for People

DPH-150S
VoIP Phone
User Manual

RECYCLABLE

Ver.1.00
2008/01/02

Table of Contents

Introduction

VoIP (Voice over Internet Protocol; also known as Internet Phone) is a technology that allows anyone to make a telephone call over the Internet environment. This is an operation manual for the DPH-150S IP Phone. It is intended to help you configure the telephone. Please follow the user guide carefully as troubleshooting the telephone can be very difficult and time consuming.

1. Getting Started

1.1. Package Contents

The following materials are included in the package. Please check the package to ensure that all the materials are present, as listed below. Contact your supplier immediately if any item is missing.

Ethernet Cable (1.5 meter)

DPH-150S VoIP Phone

Power Adapter (DC 5V)

CD for User Manual

Quick Installation Guide

1.2. Phone Specifications

Protocol
- IETF SIP (RFC3261)

Network Interface
- RJ45 x 2, 10/100BaseT

LCD Display
- 2 x 16 characters

Key Pad
- 25 keys

Call Features
- Call Hold / Resume
- Call Mute
- Call Transfer (Unattended / Blind & Attended)
- Call Waiting
- Call Forward (Busy / No Answer / Unconditional)
- Caller ID Display
- Anonymous Call
- Anonymous Call Blocking
- In band DTMF / Out-of-band DTMF (RFC 2833) / SIP INFO
- 3-way Conference
- Redial
- Message Waiting Indicator (RFC3842)
- SMS (RFC 3428)
- Auto Answer (Support SIP server required)

Codec
- G.711μ-law
- G.711a-law
- G.729a/b

Phone Functions
- Multi-user (4 SIP accounts)
- Speakerphone communication
- Pre-dial before sending
- Hot Line
- Handset / Speakerphone Volume adjustment
- Pre-dial before sending
- Hot Line
- Handset / Speakerphone Volume adjustment
- Speed dial (10 records)
- Phone book (200 records)
- Call history (Incoming calls / Outgoing calls / Missed calls)
- MP3 Ringer
- Internet Radio

Security
- HTTP 1.1 basic/digest authentication for Web setup
- MD5 for SIP authentication (RFC 2069/ RFC 2617)

Dial Methods
- Direct IP call without SIP registration
- Dial number via SIP server
- Dial URI from phone book / speed dial

Voice Quality
- VAD (Voice Activity Detection)
- CNG (Comfort Noise Generation)
- AEC (Acoustic Echo Cancellation)
- G.168
- Jitter buffer

QoS
- ToS field
- IEEE 802.1Q VLAN

Tone
- DTMF
- Ring Tone, 8 selectable tones
- Ring Back Tone (local and remote)
- Dial Tone
- Busy Tone

NAT Traversal
- UPnP
- STUN
- Static port mapping

TCP/IP
- IP/TCP/UDP/DHCP/RTP/ FTP//HTTP/NTP/TFTP/DNS

Configuration
- Key & LCD configuration
- Web browser configuration
- Auto/Manual provisioning system (Support TFTP/HTTP/FTP)

Firmware Upgrade
- TFTP
- Auto/Manual provisioning system (Support TFTP/HTTP/FTP)

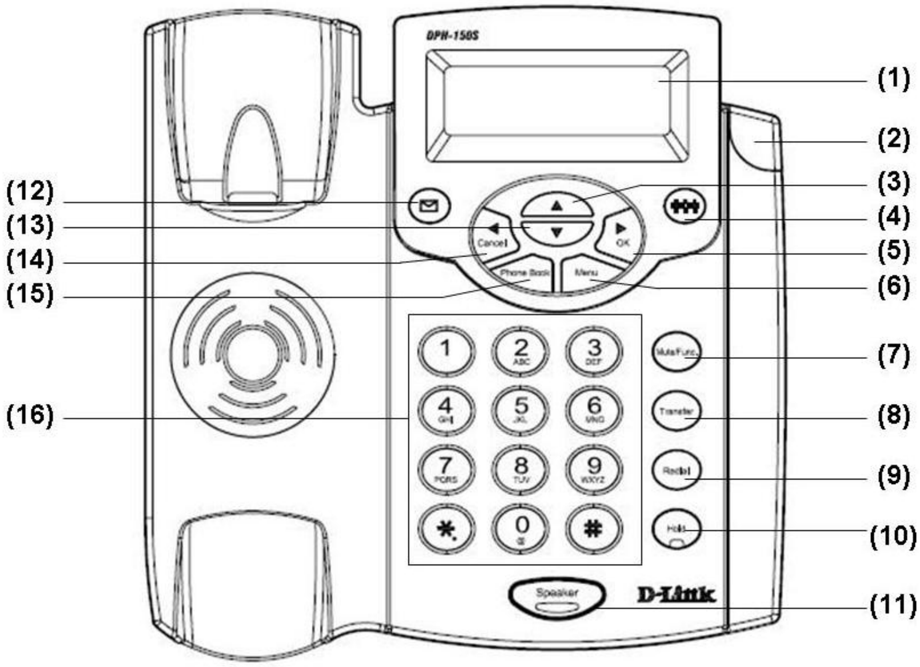
Power
- Input AC 100-120V / 220-240V
- Output DC 5V

Environmental
- Operating temperature: 0～40℃
- Storage temperature: -20～60℃
- Operating humidity: 20%～80%

Physical Dimensions
- Size: 196(L) x 198(W) mm
- Wall Mount
- Weight: 760g
- Color: Dark Gray

Certification Compliance
- FCC Part 15 Class B
- CE Class B
- VCCI Class B
- EN60950

1.3. Phone Diagram



No.	Key	Function
(1)	2 x 16 Characters LCD Display	Displays menu, time, clock, name, phone number, call status
(2)	LED Indicator	Indicates that phone is currently in use or ringing
(3)	Up	Cycle through the phone menu, adjust volume
(4)	3-Way Conference	Enable 3-way conference
(5)	OK / Right	Confirm setting change, exit menu, dial, save changes
(6)	Menu	Access the phone menu
(7)	Mute/Function	Disable user's microphone so that the person on the other line can not hear anything, access the language selection, access the time format
(8)	Transfer	Transfer the person you are currently having a conversation with to another line
(9)	Redial/Call History	Redial last dialed number, access redial menu
(10)	Hold	Place the person on the other line on hold, answer call waiting
(11)	Speaker Phone	Enable user to use the phone without using the handset
(12)	Voice Message	Check for voice messages
(13)	Down	Cycle through the phone menu, adjust volume
(14)	Cancel / Left	Deny changes, cancel phone calls, ignore phone calls, backspace
(15)	Phone Book	Access the phonebook
(16)	Numeric Keypad	Input IP/phone number/alphabet character

1.4. Key Pad Definition and Text Entry

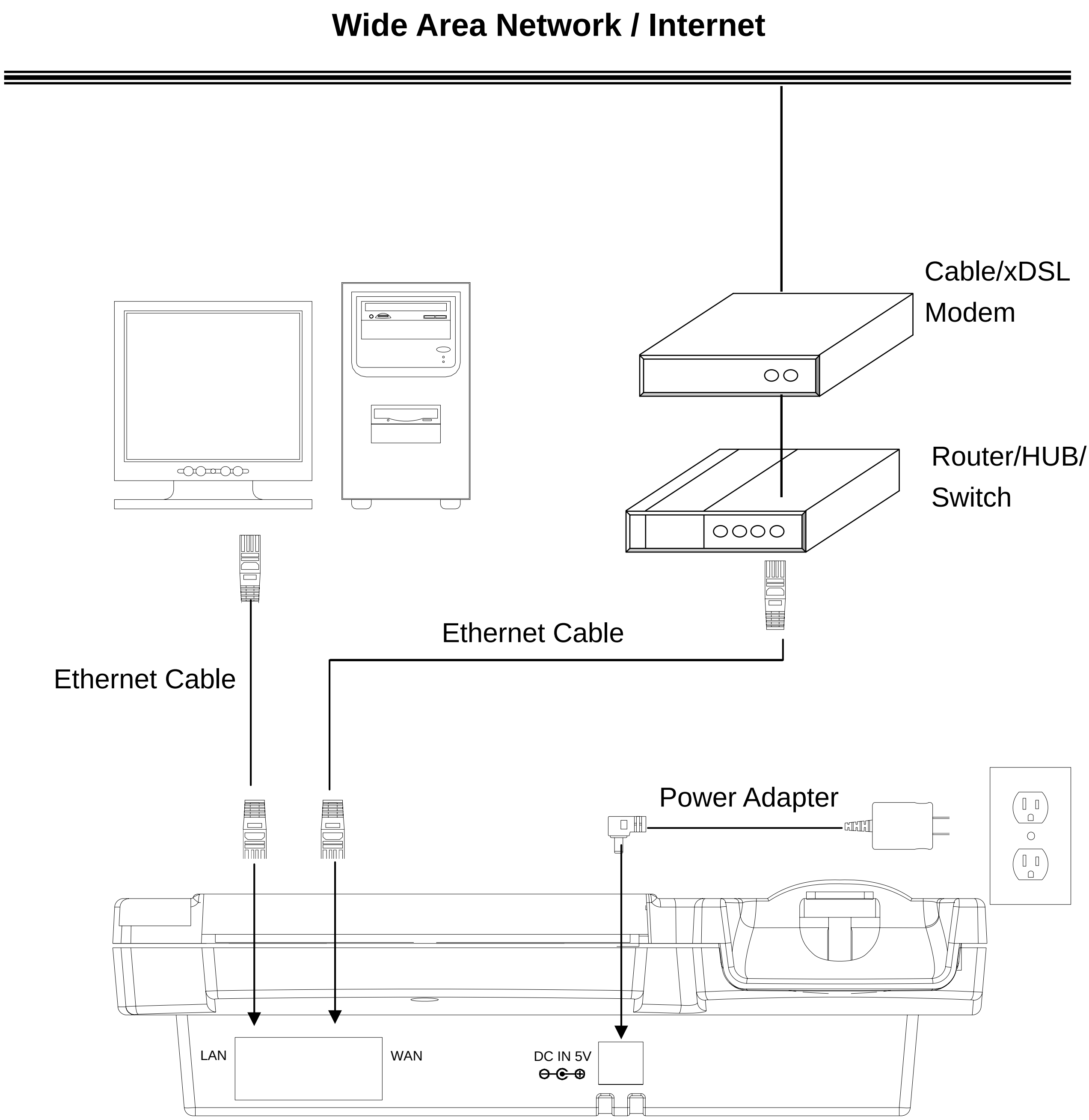
You can use alphanumeric characters to enter details into the phone, including the phone book and other settings. The table below shows the characters that you can enter in the different text modes.

	Text Mode			Text Mode	
Key	**Normal (ABC)**	**Numeric (0-9)**	**Key**	**Normal (ABC)**	**Numeric (0-9)**
1		**1**	7 PQRS	**pqrsPQRS**	**7**
2 ABC	**abcABC**	**2**	8 TUV	**tuvTUV**	**8**
3 DEF	**defDEF**	**3**	9 WXYZ	**wxyzWXYZ**	**9**
4 GHI	**ghiGHI**	**4**	0 @	@ . _ − ∗ # ( ) % & + ／ $,	**0**
5 JKL	**jklJKL**	**5**	∗.	**.**	*****
6 MNO	**mnoMNO**	**6**	#		**#**

In Normal and Numeric modes, each time you quickly the same key, the next character available on that key will be displayed. When you did not press key for more then 1 sec the current character will be selected and the cursor will move right for the next selection. For example, to enter “c” you need to press 2 ABC quickly four times. To enter the displayed character, release the key or press another key.

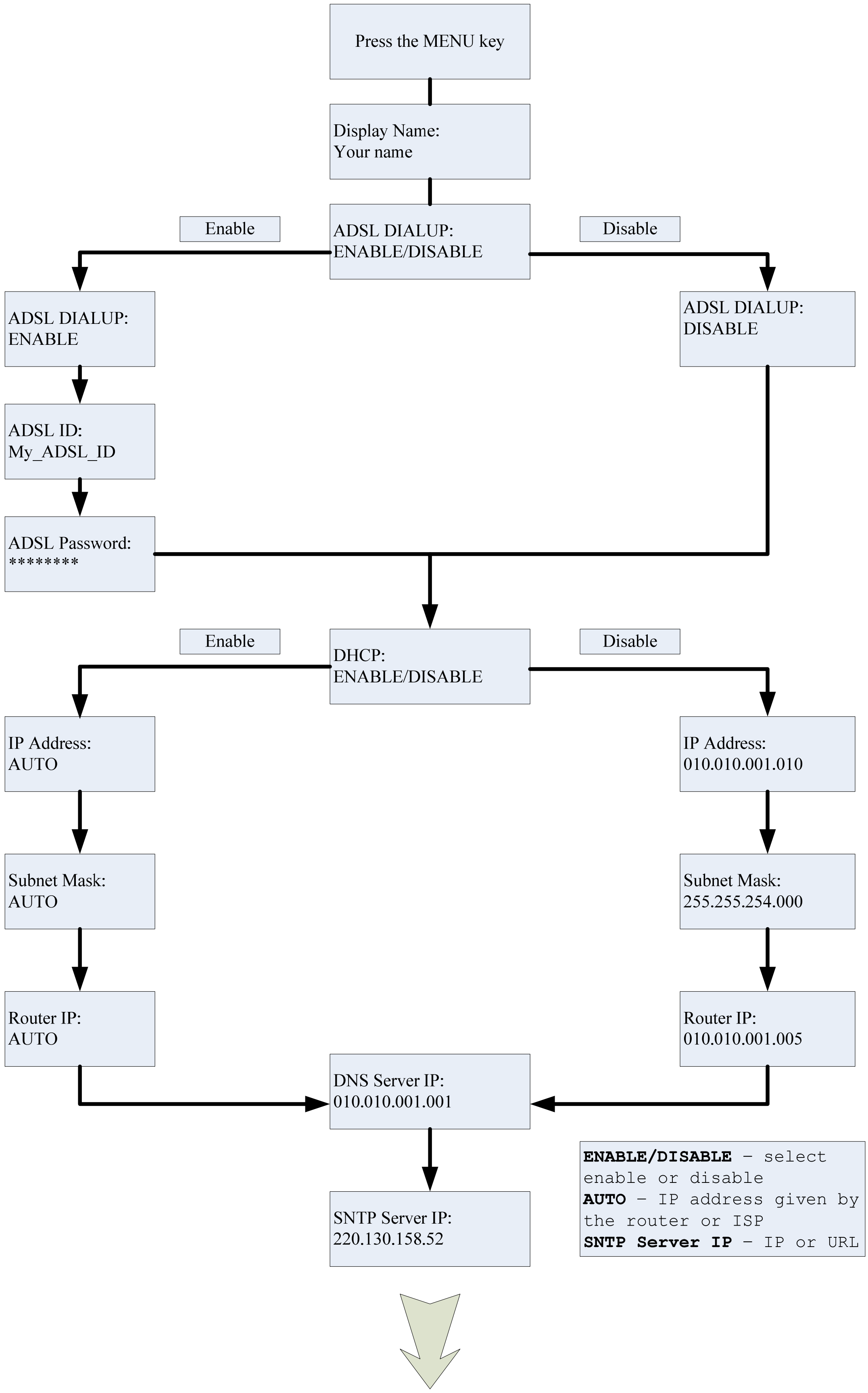
2. Connecting the IP Phone

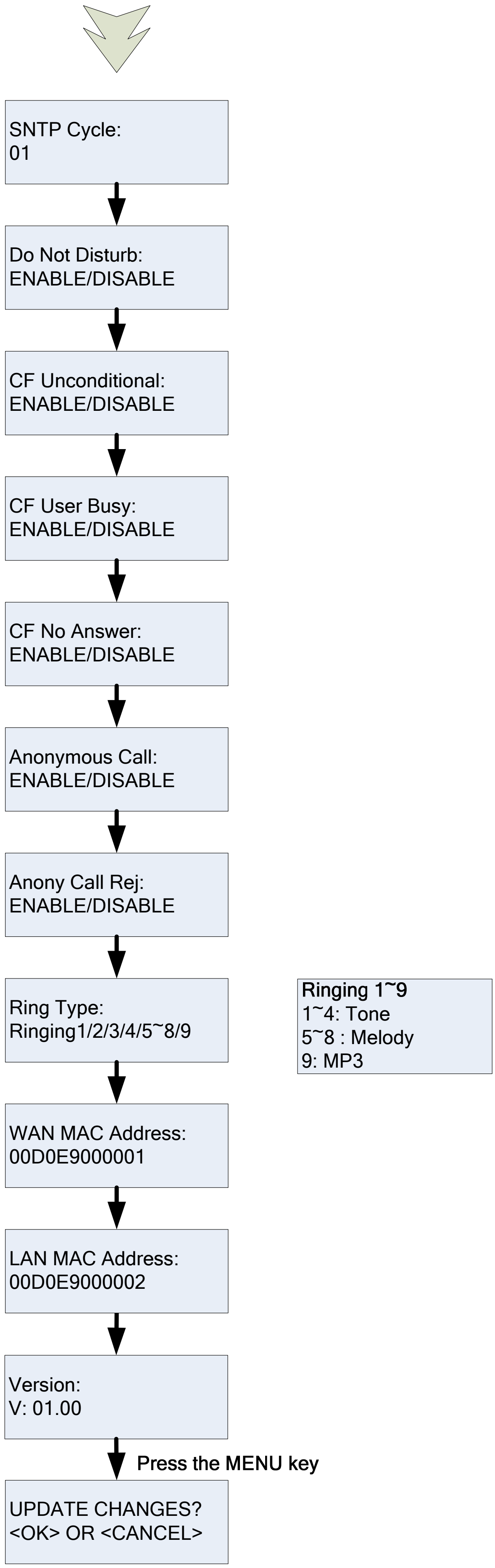
Connect the IP Phone as in the following diagram:



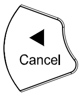
3. Initial Setup

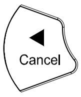
3.1. IP Phone Setup Map



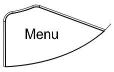

SNTP Cycle:
01
Do Not Disturb:
ENABLE/DISABLE
CF Unconditional:
ENABLE/DISABLE
CF User Busy:
ENABLE/DISABLE
CF No Answer:
ENABLE/DISABLE
Anonymous Call:
ENABLE/DISABLE
Anony Call Rej:
ENABLE/DISABLE
Ring Type:
Ringing1/2/3/4/5~8/9
Ringing 1~9
1~4: Tone
5~8 : Melody
9: MP3
WAN MAC Address:
00D0E9000001
LAN MAC Address:
00D0E9000002
Version:
V: 01.00
Press the MENU key
UPDATE CHANGES?
<OK> OR <CANCEL>

NOTE 1: If you made any modifications, you may quit setup at any time by pressing MENU + OK to save and exit or MENU + CANCEL to quit without saving. The phone will automatically exit from the menu screen if there are no inputs from the user.

NOTE 2: Use  or  to select ENABLE or DISABLE.

NOTE 3: The left arrow key  can be used as the **Backspace** key.

3.2. Display Name

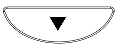
- Press 
- Enter the display name

```
Display Name:
Your name
```

3.3. ADSL Dialup

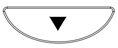
Some Internet Service Providers (mostly ADSL) use PPPoE, which requires that the user enter an ID and a password to access the Internet. In this case, enable ADSL DIALUP and enter the PPPoE ID and PPPoE password.

3.3.1. Enable ADSL Dialup

- Press 
- Use  to select “Enable”

```
ADSL DIALUP:
ENABLE
```

3.3.2. Setup ADSL ID

- Press 
- Use  to select “Enable”

```
ADSL ID:
MY_ADSL_ID
```

3.3.3. Setup ADSL Password

- Press 
- Enter ADSL Password

```
ADSL Password:
*******
```

3.3.4. Disable ADSL Dialup

- Press ▼
- Use ▶ OK to select “Disable”

```
ADSL DIALUP:
DISABLE
```

3.4. DHCP (Dynamic Host Configuration Protocol)

DHCP allows the network administrator to distribute IP addresses when a computer is plugged into a different place in the network. If your ISP provides a static IP address, you must disable DHCP and enter the IP address provided.

3.4.1. Enable DHCP

- Press ▼
- Use ▶ OK or ◀ Cancel to set DHCP “Enable”

```
DHCP:
ENABLE
```

- Press ▼
- IP address automatically acquired

```
IP Address:
  192.168.001.161
```

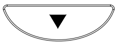
- Press ▼
- Subnet mask automatically acquired

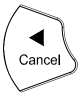
```
Subnet Mask:
  255.255.255.  0
```

- Press ▼
- Router IP automatically acquired

```
Router IP:
  192.168.001.161
```

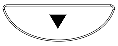
3.4.2. Disable DHCP

- Press 
- Use 

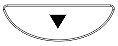
or 

to set DHCP “Disable”

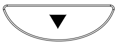
```
DHCP:
DISABLE
```

- Press 
- Enter the IP address

```
IP Address:
192.168.001.161
```

- Press 
- Enter the subnet mask

```
Subnet Mask:
 255.255.255.000
```

- Press 
- Enter the router IP address

```
Router IP:
192.168.001.001
```

3.5. DNS Server IP

The domain name system (DNS) is the way that Internet domain names are located and translated into Internet Protocol addresses. There is probably a DNS server within close geographic proximity to your ISP that maps the domain names in your Internet requests or forwards them to other servers on the Internet.

- Press 

```
DNS Server IP:
 192. 76.144. 66
```

3.6. SNTP Server IP

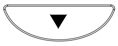
Simple Network Time Protocol (SNTP) is a protocol used to help match your system clock with an accurate time source. If you do not know your SNTP Server IP, please ignore this section. The SNTP Server IP address can be either URL or IP.

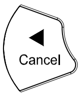
- Press 
- Enter SNTP server IP or URL

```
SNTP Server IP:
220.130.158.52
```

3.7. Do not Disturb

This setting allows the user to reject all incoming phone calls.

- Press 
- Use 

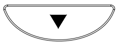
or 

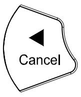
to select “Enable” or “Disable”

```
Do Not Disturb:
ENABLE /DISABLE
```

3.8. CF (Call Forward) Unconditional

Enable CF Unconditional to forward all the incoming calls to another number. Otherwise set to disable. ***You will need to use a web-browser to input the forwarding phone number. Refer to section 6 for more information on using the web configuration.***

- Press 
- Use 

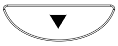
or 

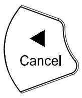
to select “Enable” or “Disable”

```
CF Unconditional:
ENABLE / DISABLE
```

3.9. CF (Call Forward) Busy

Forward all the incoming calls to another number when user is busy on the phone.

- Press 
- Use 

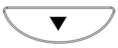
or 

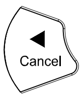
to select “Enable” or “Disable”

```
CF User Busy:
ENABLE / DISABLE
```

3.10. CF (Call Forward) No Answer

Forward all incoming calls to another phone number after a certain number of rings.

- Press 
- Use 

or 

to select “Enable” or “Disable”

```
CF No Answer:
ENABLE / DISABLE
```

3.11. Anonymous Call

Enables the caller (user) to hide the name and phone number from the receiver.

- Press
- Use or o select “Enable” or “Disable”

Anonymous Call:
ENABLE / DISABLE

3.12. Anony Call Rej. (Anonymous Call Rejection)

Reject any anonymous incoming calls.

- Press
- Use or to select “Enable” or “Disable”

Anony Call Rej:
ENABLE / DISABLE

3.13. Ringing Type

Select the ring tone. There are 9 ring tones in total.

- Press
- Use or to select the ring type

Ring Type:
Ringing 1/2/3/4/5/6/7/8/9

NOTE: At this point, you may save the settings and exit. The next two sections explain how to obtain the MAC address and firmware version.

- Press to exit the menu
- When asked to save or cancel, press to save

3.14. MAC Address

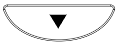
This menu displays the MAC address. You cannot modify the MAC address.

- Press
- The **MAC address** is displayed on the scre

WAN MAC Address:	LAN MAC Address:
000FC9017D4A	000FC9017D4B

3.15. Version

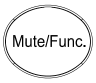
The version menu displays the firmware version. You cannot modify the version number.

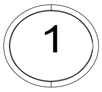
- Press 
- The firmware **version** is displayed on the screen

```
Version:
V: 01.00
```

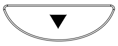
3.16. Language Selection

The VoIP Phone supports 2 languages: English and Russian.

- Press 

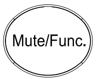
followed by 

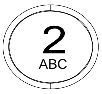

```
Language:
English
```

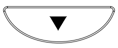
- Use  or  to select the preferred language

3.17. Time Format

You may select the 12hr or 24hr time format.

- Press 

followed by 

- Use  or  to select the time format

```
Time Format:
24Hours
```

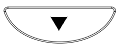
- Press 

when done

3.18. Volume Adjustment

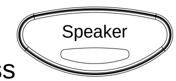
3.18.1. Ringer Volume

While the handset is in place,

- Press  to increase the ringer volume and  to decrease the ringer volume

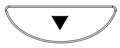
3.18.2. Speaker Volume

While the handset is in place,

- Press 

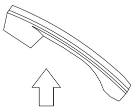
- Press  to increase the speaker volume and  to decrease the speaker volume

3.18.3. Handset Volume

- Pick up the handset and press  to increase the volume or press  to decrease the volume

4. Operating the Phone

4.1. Dialing an IP Address

- Lift the handset  or press the **SPEAKER** button 

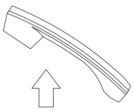
- Dial an IP address

For example: dialing 192.168.0.1

Press OK 

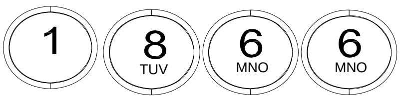
or wait until the timer expires to dial.

4.2. Dialing a SIP Number

- Lift the handset  or press the **SPEAKER** button 

- Dial a SIP Number

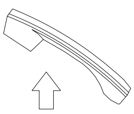
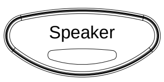
For example: dialing 1866



- Press OK 

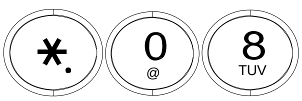
or wait until the timer expires.

4.3. Speed Dialing

- Lift the handset  or press the **SPEAKER** button 

- Dial Speed Dial number with the prefix code "∗".

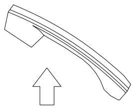
For example: dialing ∗ and speed dial number 08,



4.4. Answer a Phone Call

Note: The CANCEL key may be used to reject a call.

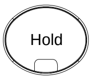
When the phone rings:

- Lift the handset  or press the **SPEAKER** button 

 to begin your conversation.

4.5. Switch to another Line

- While having a conversation:

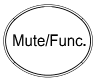
- Press **Hold** 

 to switch to another line.

4.6. Mute

Note: While mute is activated, sound from the caller can be heard from your speaker but your sound can't be heard by the caller.

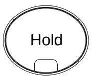
While having a conversation:

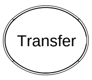
- Press **Mute** 

- Press the Mute key again to resume your conversation.

4.7. Transfer

While having a conversation:

- Press **Hold** 

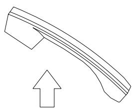
 to put the person on the other line on hold.
- Dial the IP address or the extension number where you would like the call to be transferred.
- Press **Transfer** 

 to transfer the call.

4.8. Redial

Note: To return to idle mode, press the **CANCEL** key

4.8.1. Last Dialed Number

- Lift the handset  or press the **SPEAKER** button 

- Press **Redial** to dial the last dialed number.

4.8.2. Through Call History

- Press **Redial** . Do not lift the handset when you press **Redial**.
- Press **Redial** again to cycle through the dialed, missed, and received calls.
- Press **DOWN** key to scroll down through the dialed, missed, or received lists until the number is displayed on the screen.
- Pick up the handset or press **OK**

4.9. On Hold

Note: To transfer a call while on hold, press the **TRANSFER** key. Dial the extension/phone number and press the **TRANSFER** key again to transfer the call.

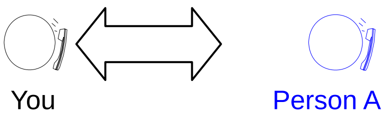
While having a conversation:

- Press **HOLD** (Press **HOLD** again to resume your conversation)

4.10. Call Forward

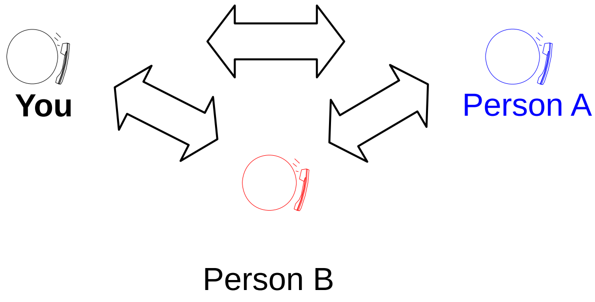
Please refer to Initial Setup (sections 3.8 ~ 3.10) and Web Browser Configuration (section 6) to setup call forwarding.

4.11. Three Way Conference

- Pick up the handset and call Person A.
- 

- After Person A pick up the phone, press **Hold** key to place Person A on hold.
- Dial the extension or phone number of Person B and wait until Person B picks up the phone.

- Press the **Conference** key to begin the 3-way conference.



5. Using the Phone Book

5.1. Dialing from the Phone Book

- Press the **PHONE BOOK** key to access the phone book.
- Press to scroll down through the list until the name is displayed on the screen.
- Press **OK** to dial.

5.2. Storing a Number

- Press and hold the **PHONE BOOK** key until “**Name:**” is displayed on the screen.
- Enter a name then press
- Enter the number that corresponds to the name and press **OK**
- Press **OK** again to save the number into the phone book.
- Repeat Steps 1 to 4 to store another phone number.

5.3. Editing a Number

- Press the **PHONE BOOK** key to access the phone book.
- Press until the name is displayed on the screen.
- Press the **PHONE BOOK** key again.

- Select “**Edit**” and press **OK** to edit.
- Enter a new name and press **OK**
- Enter the new phone number and press **OK**
- Press **OK** to save and override the previous name and phone number.

5.4. Deleting a Number

- Press the **PHONE BOOK** key to access the phone book.
- Press until the name you want to delete is selected.
- Press the **PHONE BOOK** key again.
- Select “Delete” and press **OK** to delete.
- Press **OK** again to save the new list on the phone book.

6. Using the Web Configuration

The web configuration interface can be accessed using a web browser.

6.1. Accessing the Configuration Menu

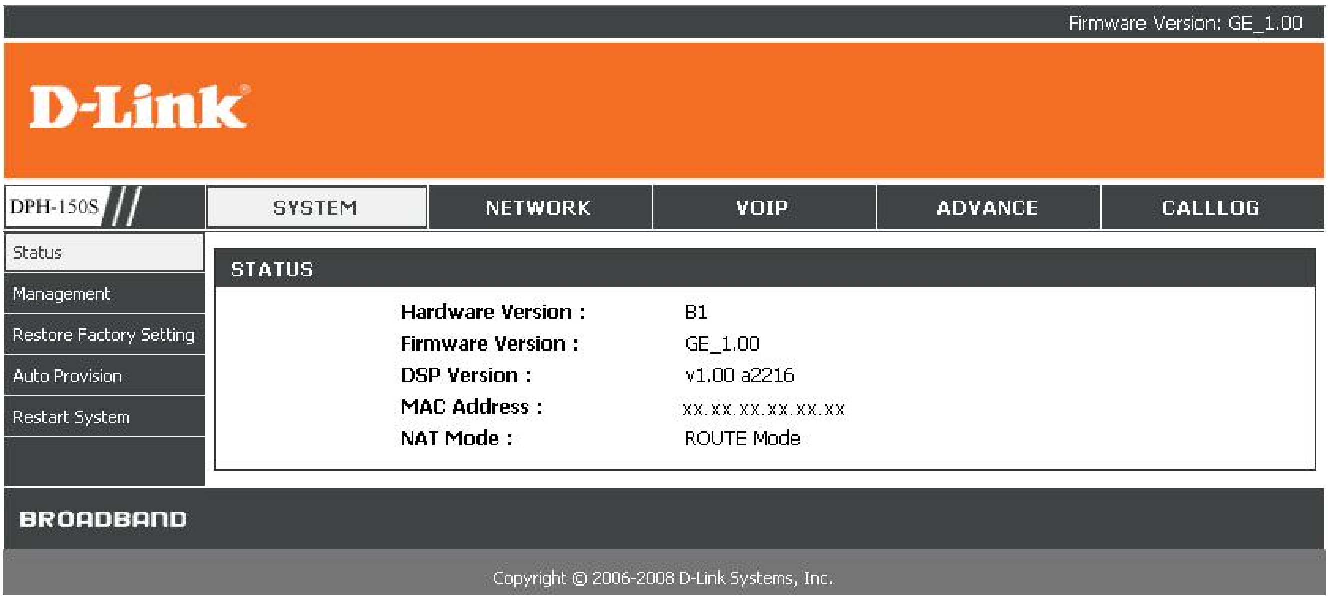
1. Open a web browser (Internet Explorer, Netscape, Opera, Firefox, etc.)
2. Type in the **IP Address** of the phone

The IP address is provided by your Internet Service Provider (ISP). If your ISP supports DHCP, you may obtain the IP address from your phone. Press “ Func.+ 9 ” to get the IP address. It can also login from the LAN port by http://192.168.15.1.

Enter **User Name** and **Password** (enter “admin” as Username and leave Password blank if you are installing the phone for the first time)

Click **OK**

6.2. Web Login



Hardware Version Hardware version of the IP phone

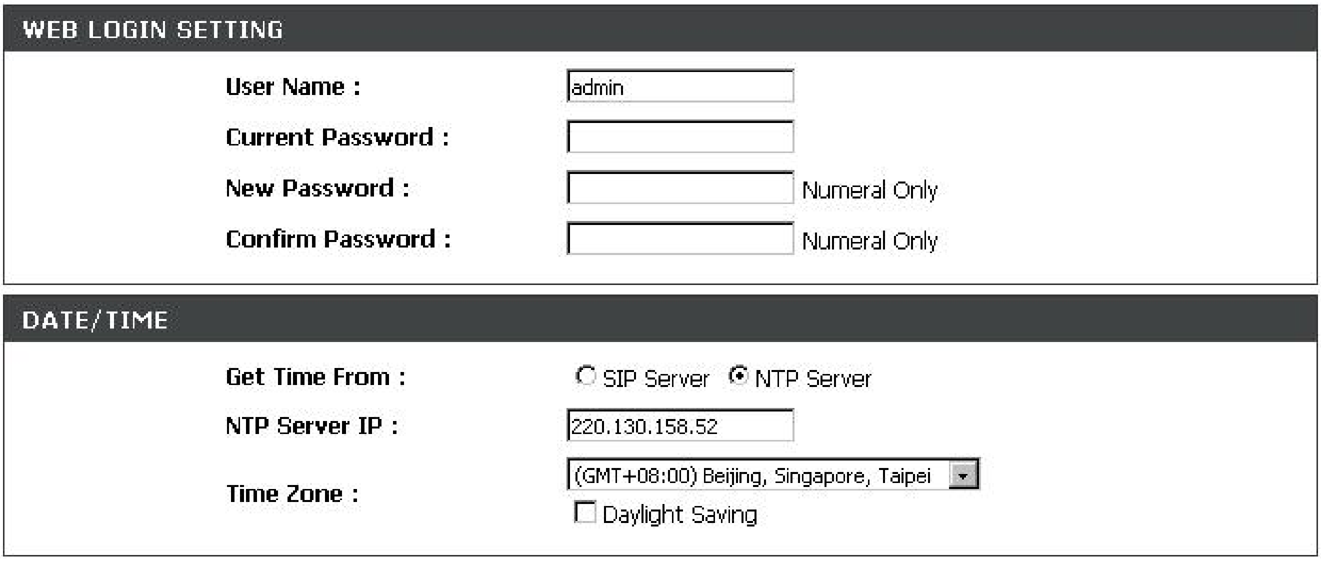
Firmware Version The current firmware version installed on the DPH-150S

DSP Version The current version of the DSP application installed on the DPH-150S

MAC Address MAC address of the IP phone

NAT Mode The NAT mode (Router or Bridge) of the LAN interface

6.3. System – Management



User Name Configuration menu login name

Current Password Configuration menu login password

New Password Enter a new password to replace the current one

Confirm Password Enter the new password again to confirm the change

Get Time From Get time setting from SIP or NTP server

NTP Server IP Network Time Protocol (NTP) is a protocol used to help match your system clock with an accurate time source (ex. atomic clock, time server). It is good practice to have all your networked computers synchronized with one server.

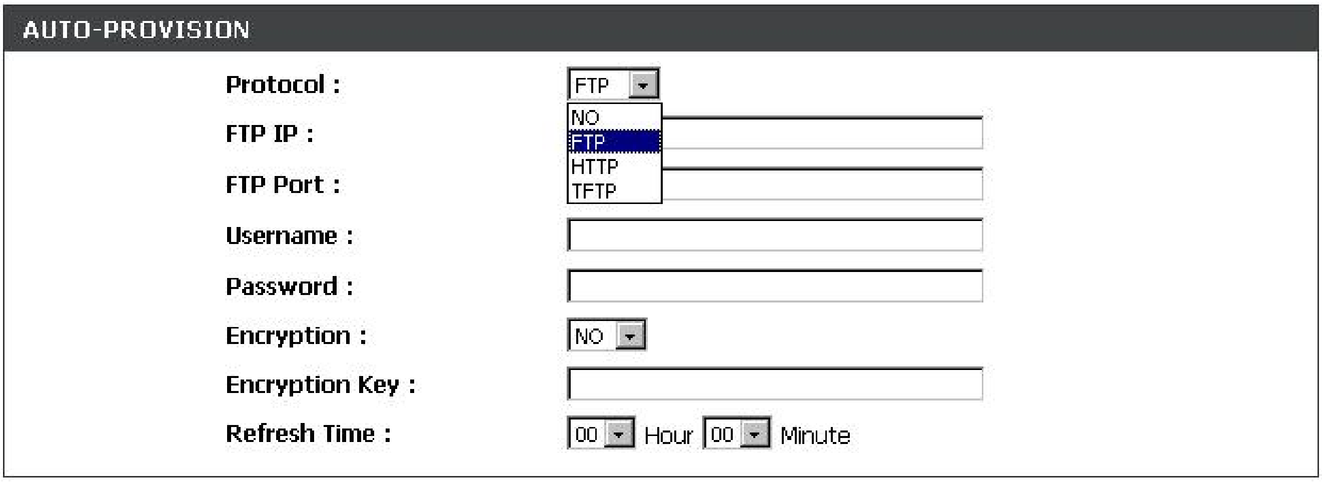
Time Zone Select your time zone. If there is daylight saving in your area, tick the check box

Daylight Saving Check to enable daylight saving

6.4. System – Restore Factory Default

Restore Factory Setting Restores all the settings back to the factory default settings

6.5. System – Auto Provision



Protocol FTP, HTTP and TFTP support for downloading firmware and automatic configuration. The default setting is NO (function disabled)

FTP / HTTP / TFTP IP IP address of the provisioning server

FTP / HTTP / TFTP Port Listening port of the provisioning server

Username The username required by the provisioning server for authorization.

Password The password required by the provisioning server for authorization.

Encryption Choose YES to receive and decrypt the encrypted configuration files

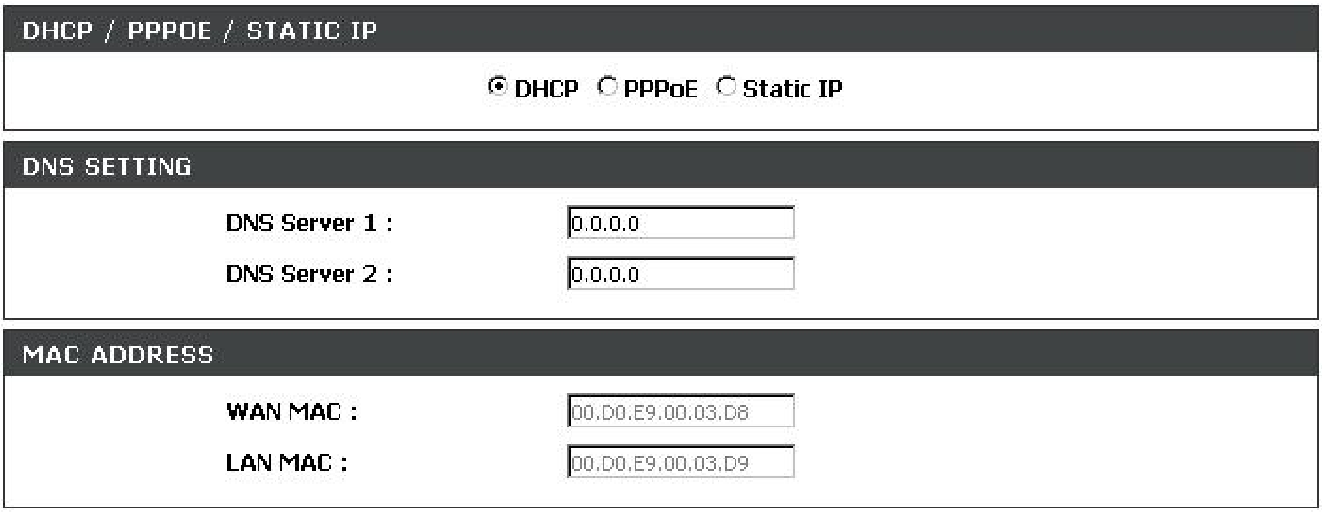
Encryption Key The key which is provided by the administrator for decrypting the encrypted configuration files

Refresh Time The time at which the DPH-150S connects to the auto provision system to check for updates.

6.6. System – Restart System

Restart System Click **Restart** to update all the modifications and reboot the system

6.7. Network – Network Settings / DHCP

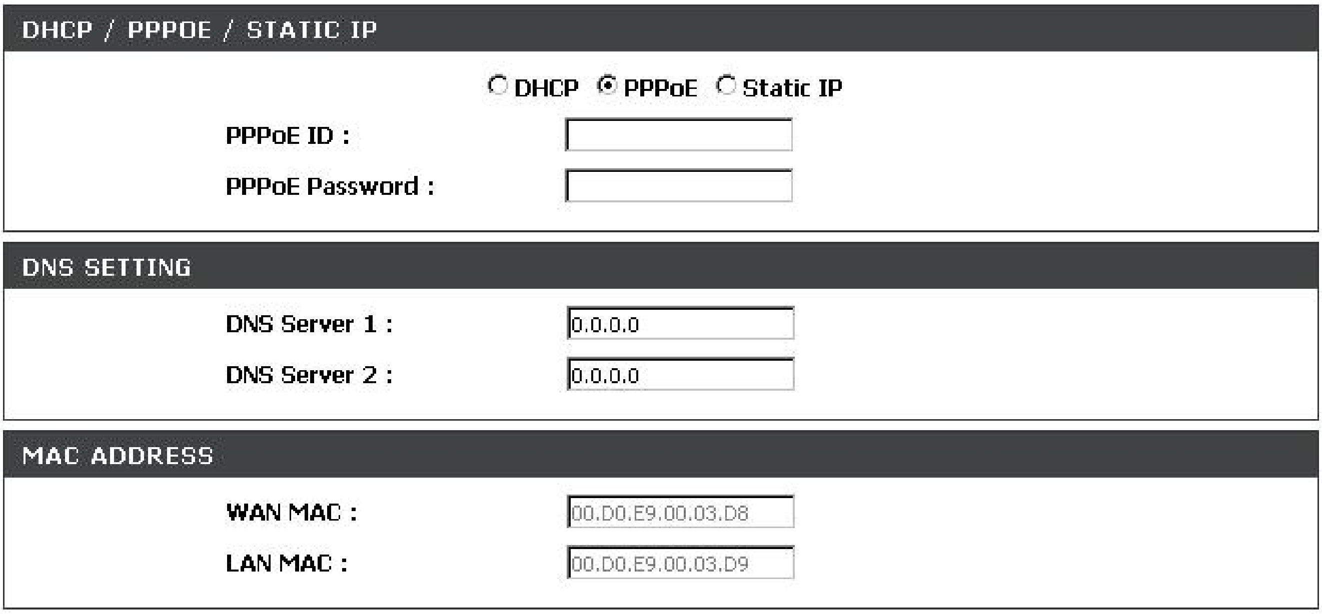


DNS Server 1~2 DNS address provided by your ISP

WAN MAC MAC address of the WAN interface

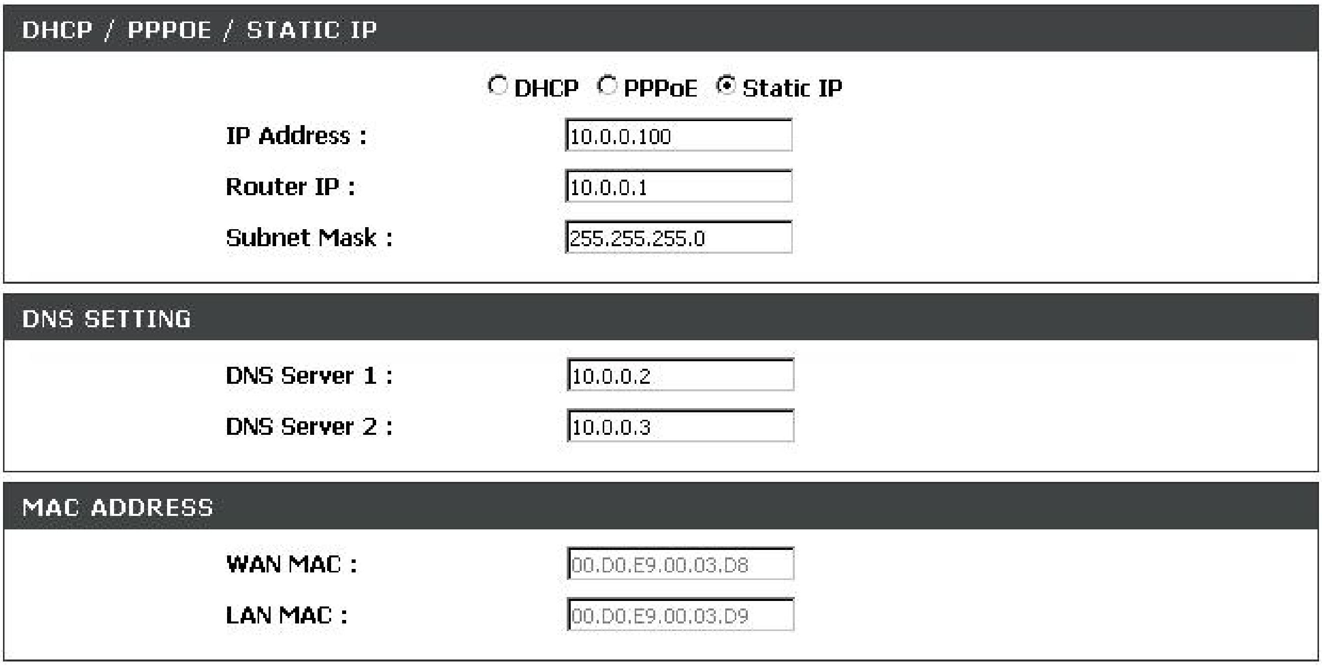
LAN MAC MAC address of the LAN interface

6.8. Network – Network Settings / PPPoE



PPPoE ID	PPPoE ID/username provided by your ISP.
PPPoE Password	PPPoE password.
DNS Server 1~2	DNS address provided by your ISP

6.9. Network – Network Settings / Static IP



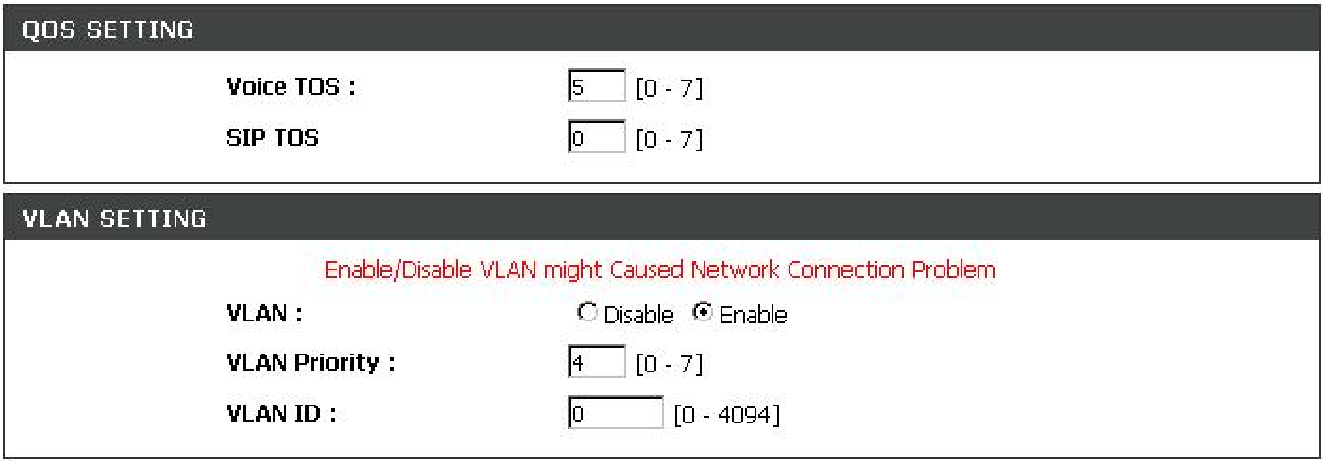
IP Address IP address provided by your ISP.

Router IP Router IP address provided by your ISP

Subnet Mask Subnet mask provided by your ISP

DNS Server 1~2 DNS address provided by your ISP

6.10. Network – QoS Settings



Voice ToS Sets the type of service for this Internet datagram.

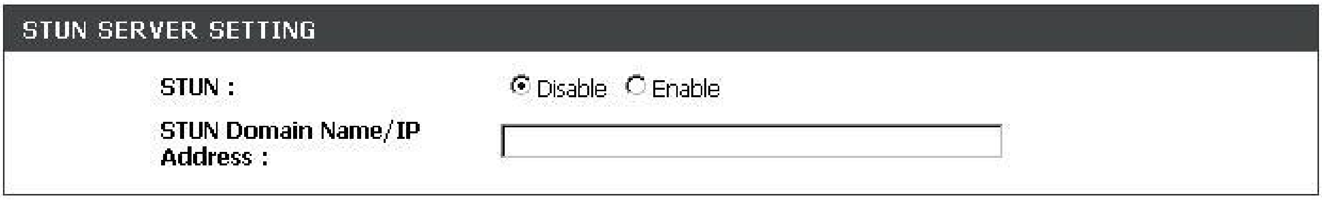
SIP ToS Sets the type of service for this higher priority of signaling packet.

VLAN Enable or disable VLAN

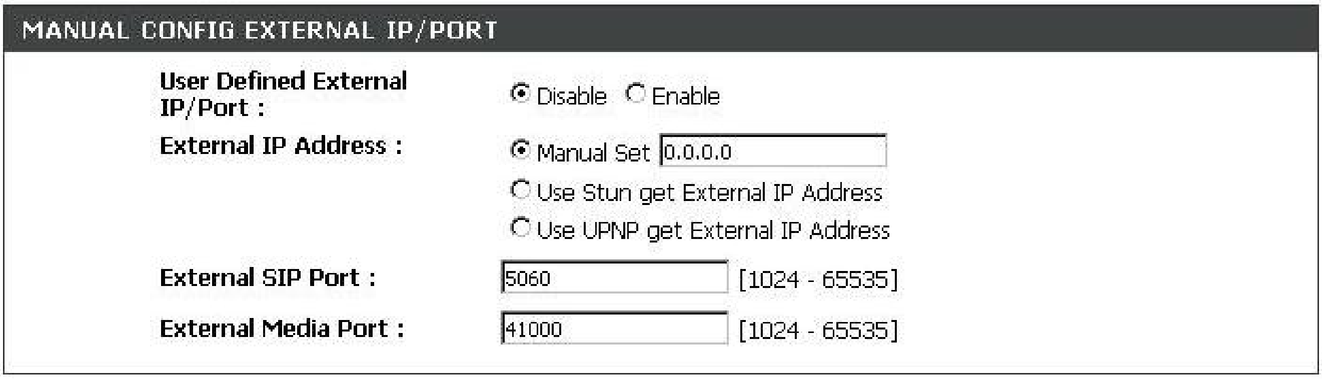
VLAN Priority 8 classes are supported for prioritization on VLAN.

VLAN ID The identification of VLAN.

6.11. Network – NAT Traversal Settings

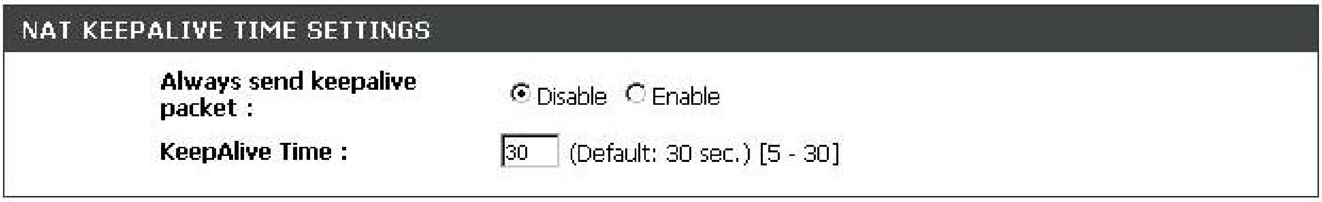


STUN	Simple Traversal of User Datagram Protocol through Network Address Translation (STUN) is a protocol that allows applications to determine the types of NATs and firewalls that are in between them and the Internet. STUN also provides the ability for applications to determine the public IP addresses allocated to them by NAT.
STUN Domain Name / IP Address	Enter the STUN domain name or IP address if STUN is enabled.



User Defined External IP/Port	Enable or disable the settings for configuring the user defined external IP address and port number.
External IP Address	Setup the external IP address manually. Use a STUN server to get external IP address. Use UPnP to get external IP address.
External SIP Port	External SIP port
External Media Port	External media port

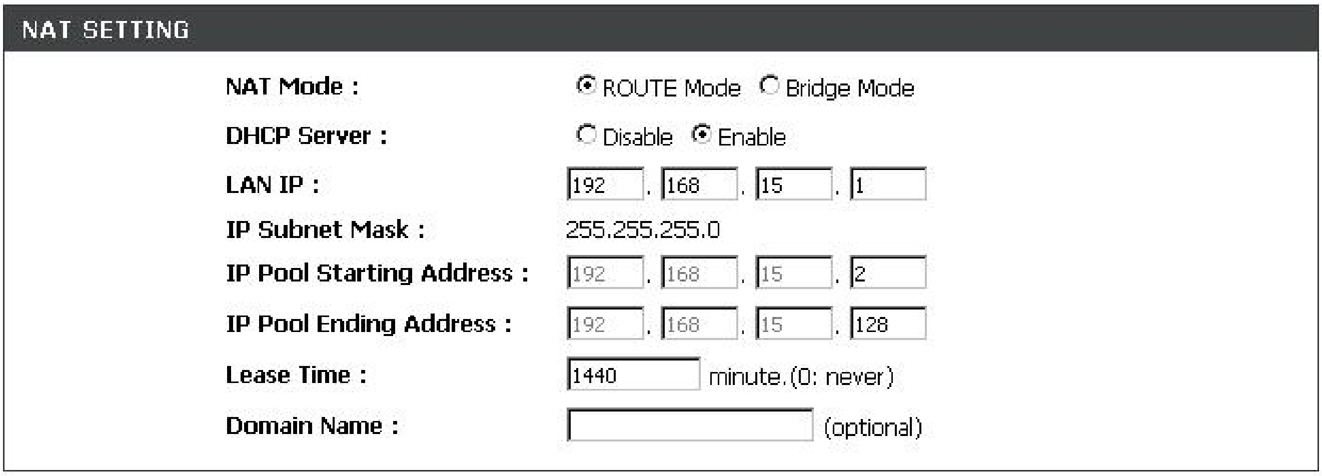
UPnP Enable or disable universal plug and play. Some NAT supports UPnP so STUN is not required and must be disabled



Always send keep-alive packet Enable or disable to keep the SIP signaling channel alive.

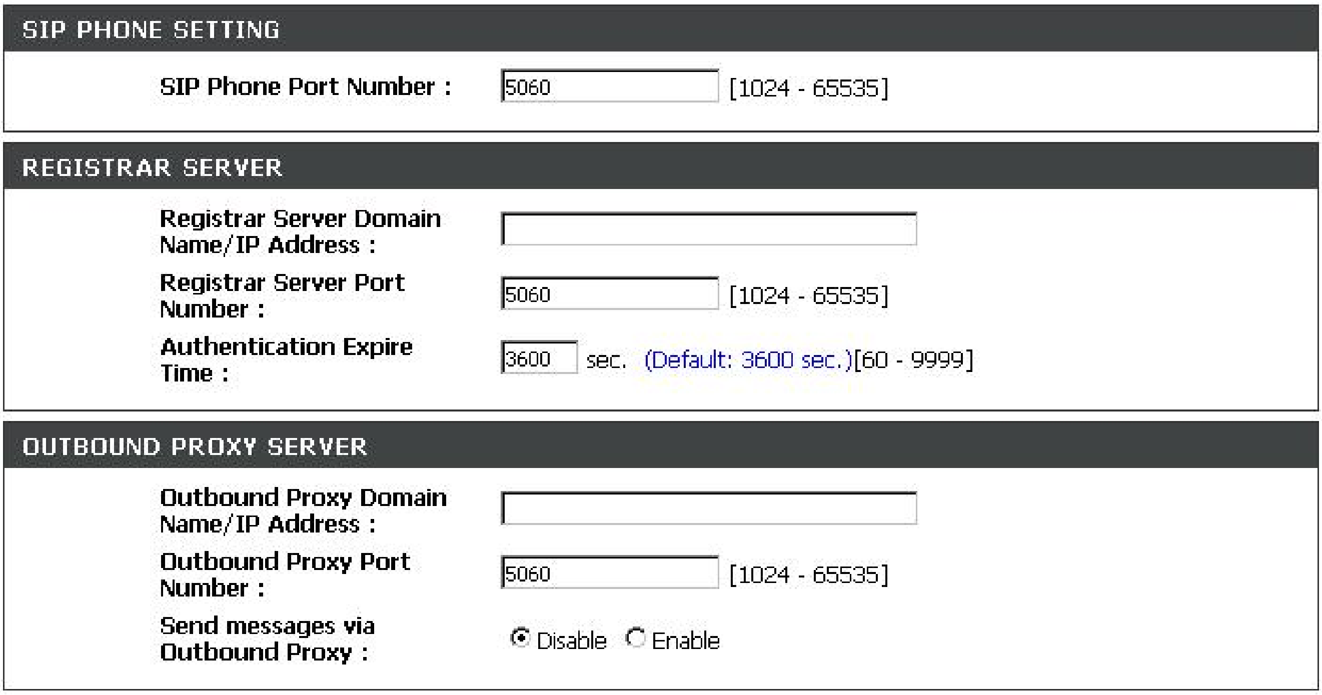
Keep-Alive Time The time interval that the IP phone always sends the keep-alive packet in order to ensure that NAT is working properly.

6.12. Network – NAT



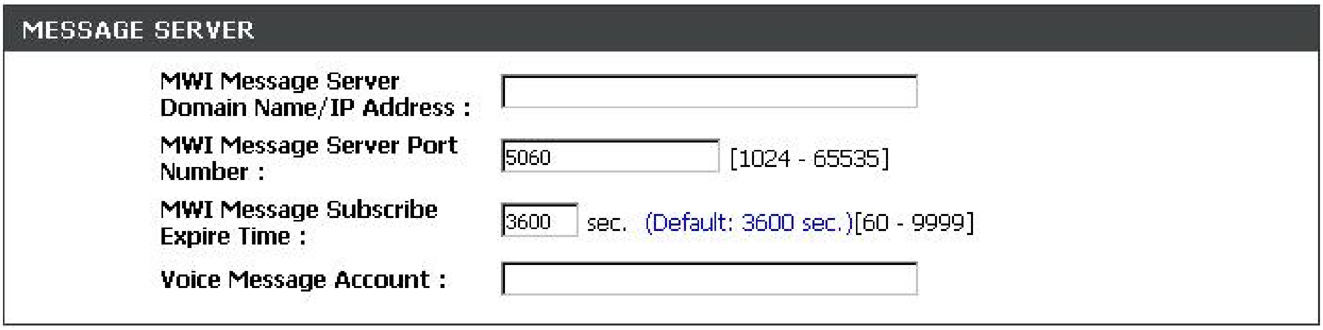
NAT mode can be set to ROUTE Mode or Bridge Mode.

6.13. VoIP – SIP Settings (SIP Phone Setting, Registrar & Outbound Proxy Server)



SIP Phone Port Number	SIP phone port number.
Registrar Server Domain Name/IP Address	Registrar server domain name or IP address.
Registrar Server Port Number	Registrar server listening port.
Authentication Expire Time	The time after which the registration on SIP Registrar expires. The phone must send SIP REGISTER to keep the registration at half of the setting time.
Outbound Proxy Domain Name/IP Address	Outbound proxy domain name or IP address.
Outbound Proxy Port Number	Outbound proxy listening port.
Send message via Outbound Proxy	Select Enable to send all SIP requests through Outbound Proxy.

6.14. VoIP – SIP Settings (Message Server)



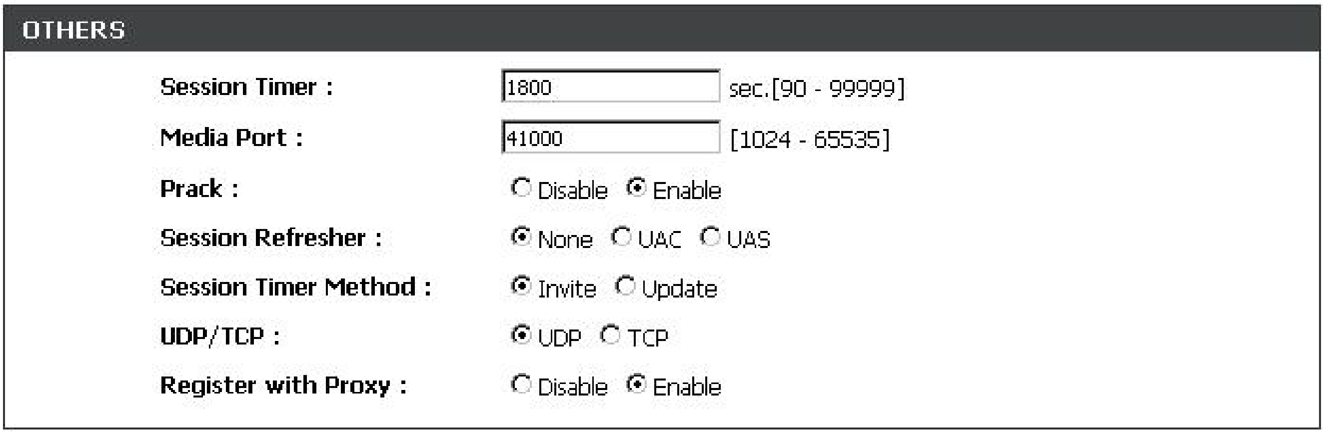
MWI Message Server Domain Name/IP address — Message server domain name or IP address.

MWI Message Server Port Number — Message server listening port.

MWI message Subscribe Expire Time — The time after which the subscription expires. It is included in SIP SUBSCRIBE and is used to negotiate with Message server.

Voice Message Account — Voice message account

6.15. VoIP – SIP Settings (Others)



Session Timer — The time interval in which the phone periodically refreshes SIP sessions by sending repeated INVITE requests. These INVITE requests allow the user agent or proxies to determine the status of the SIP session.

Media Port — Real-time Transport Protocol port number. Provides end-to-end transfer of data with real-time audio.

Prack A SIP method which is applied to the condition of acknowledging provisional responses like 180 Ringing. Select Enable for a more reliable connection.

Session Refresher Select None to disable SIP session timer support.

Select UAC to initiate SIP request.

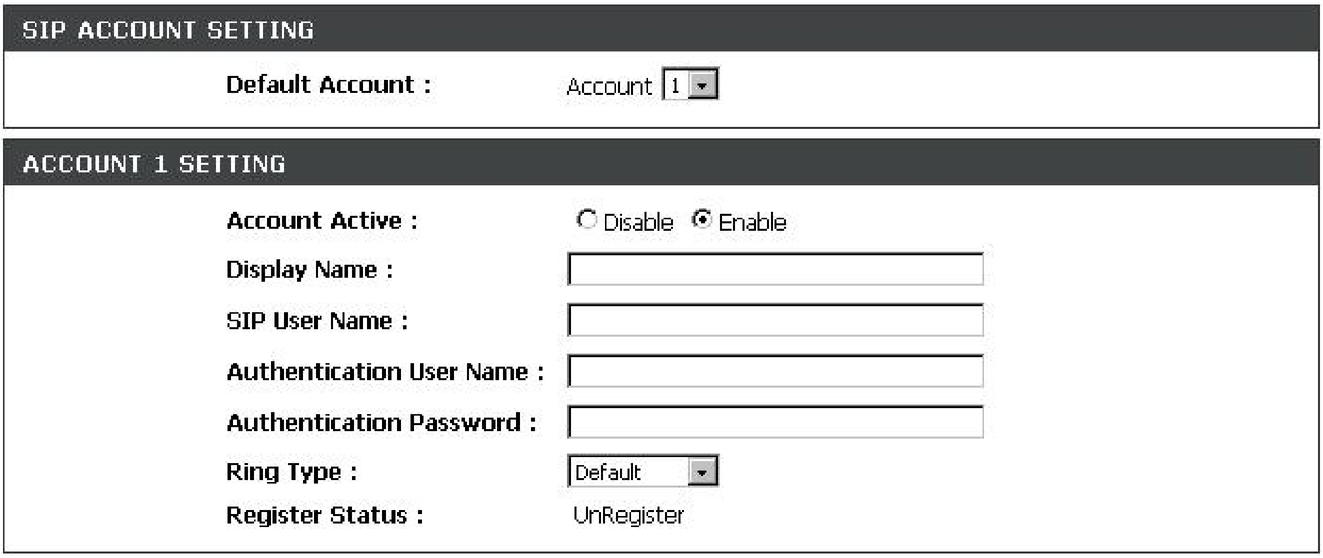
Select UAS to receive SIP request and then return a response.

Session Timer Method Select SIP request method. Default method is Invite.

UDP/TCP Select SIP signal transmission method. Default method is UDP.

Register with Proxy When “Send messages via Outbound Proxy” is enabled, all the SIP requests including Register will be sent through Outbound Proxy. Enabling “Register with Proxy” will be against this rule and send SIP Register directly to the Registrar as described in section 6.13.

6.16. VoIP – SIP Account Settings



Default Account When you dial a number, the default account is used to dial. The User Name of default account is displayed on the receiver’s IP phone.

Account Active Enable or disable this account.

Display Name Name displayed on the LCD screen of the called party.

SIP User Name The number in the URI displayed on the LCD screen for the caller.

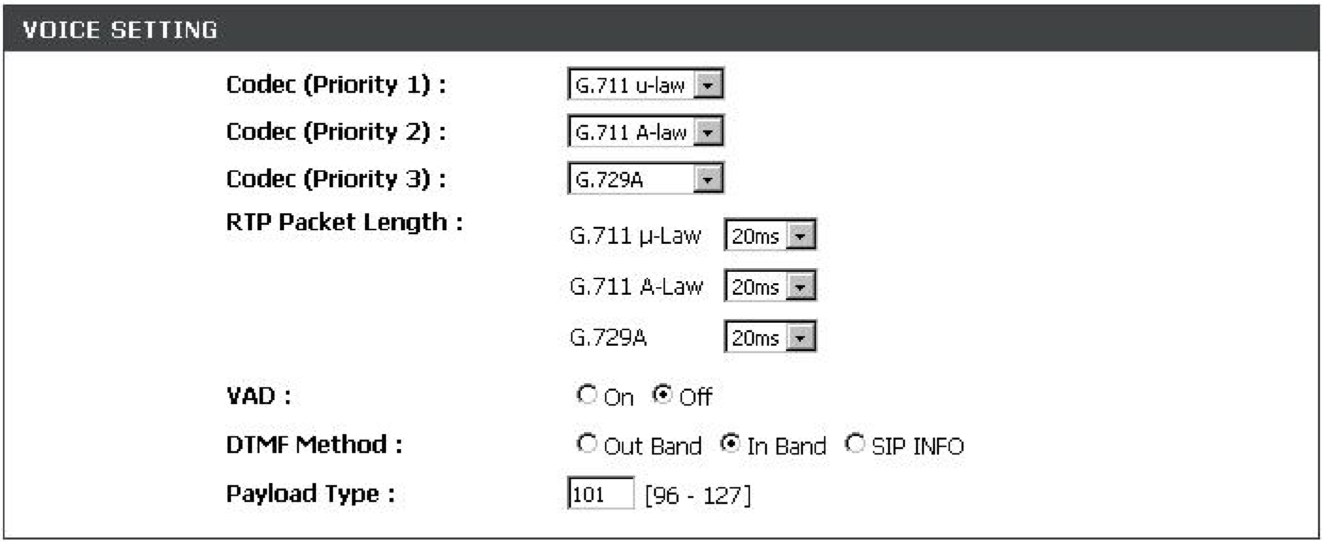
Authentication User Name User name to log into the SIP server.

Authentication Password Password to log into the SIP server.

Ring Type Nine types of tones, melodies, and MP3s can be chosen for the specified account

Register Status Displays if the current phone is registered or unregistered with SIP server.

6.17. Advance – Voice Settings



Codec (Priority 1 ~ 3) Voice Compression Algorithm priority settings. Select from the most used codec to the least used codec.

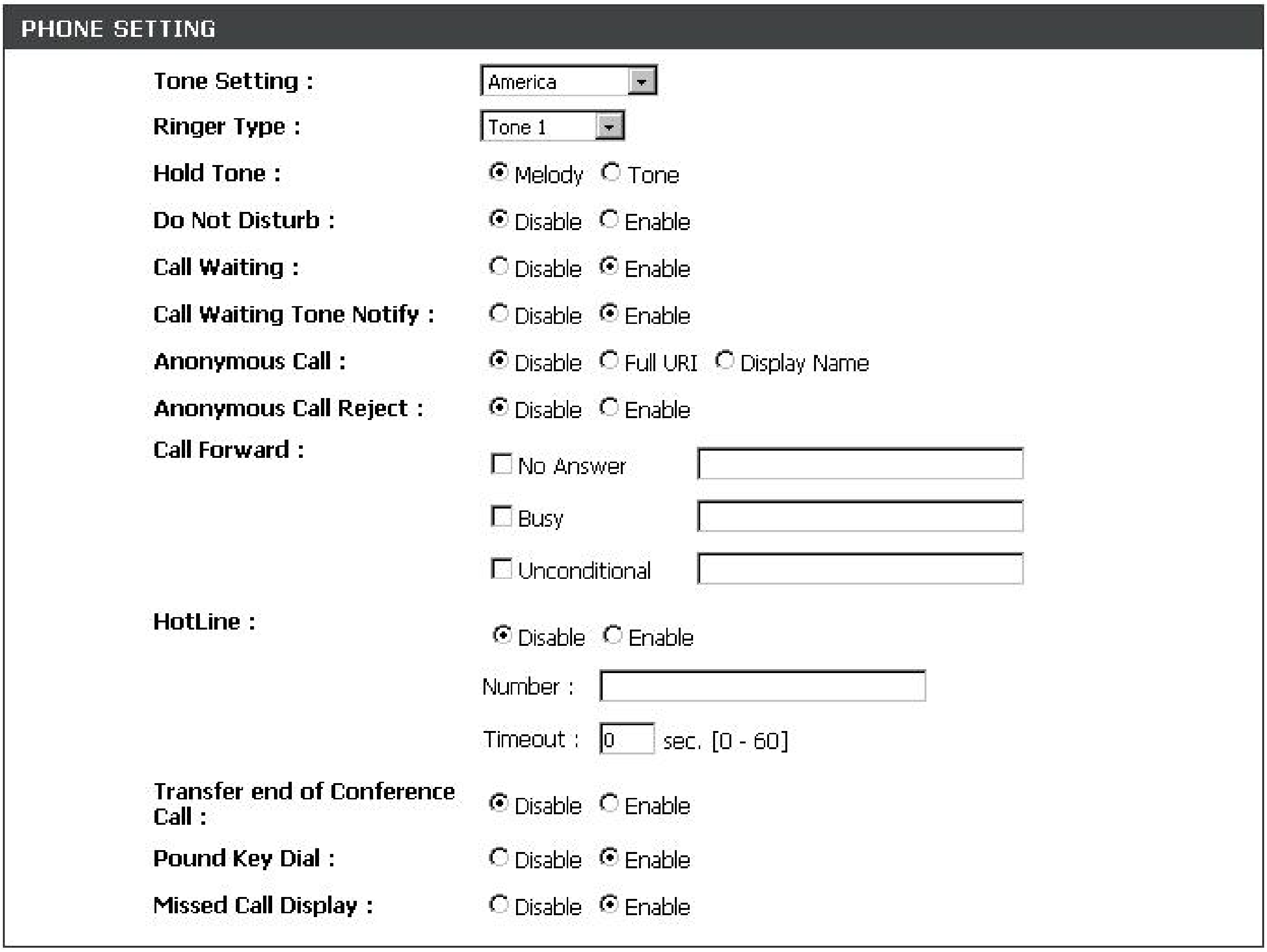
RTP Packet Length The payload size for each RTP packet.

VAD VAD is supported for silence suppression. When Enable is selected, it also supports SID frame for CNG.

DTMF Method Select the method to generate DTMF. Out Band DTMF is based on RFC2833.

Payload Type Set the payload type for the Out Band DTMF (Default is 101).

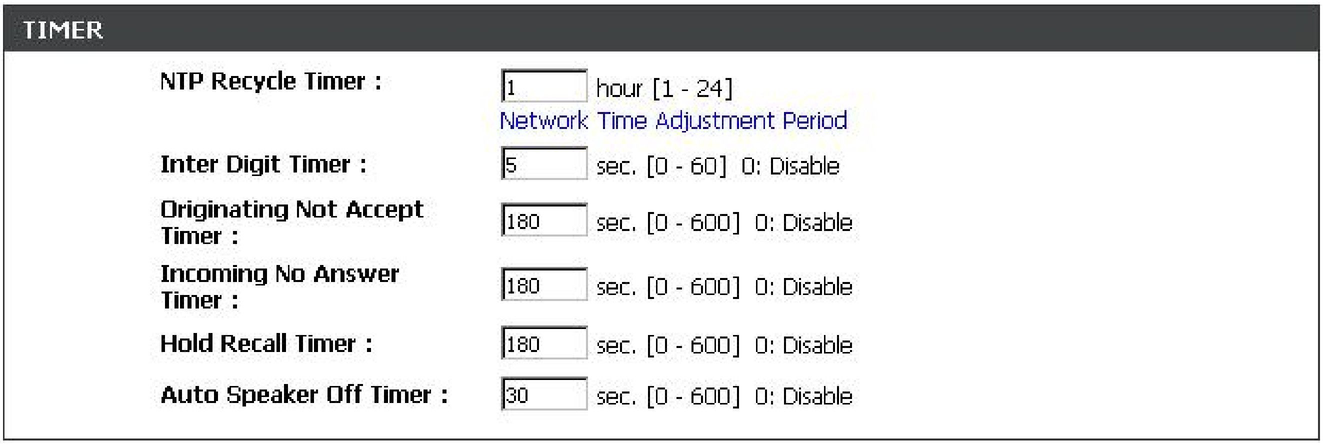
6.18. Advance – Phone Settings (Phone Setting)



Tone Setting	Select the tone for your particular country
Ringer Type	Select the ring type (Tone 1 ~ 4, Melody 5 ~ 8, and MP3 9).
Hold Tone	Select melody or tone when the phone is on hold.
Do Not Disturb	Reject all incoming calls.
Call Waiting	Enable or disable call waiting.
Call Waiting Notify	Enable or disable the reminding tone for Call Waiting

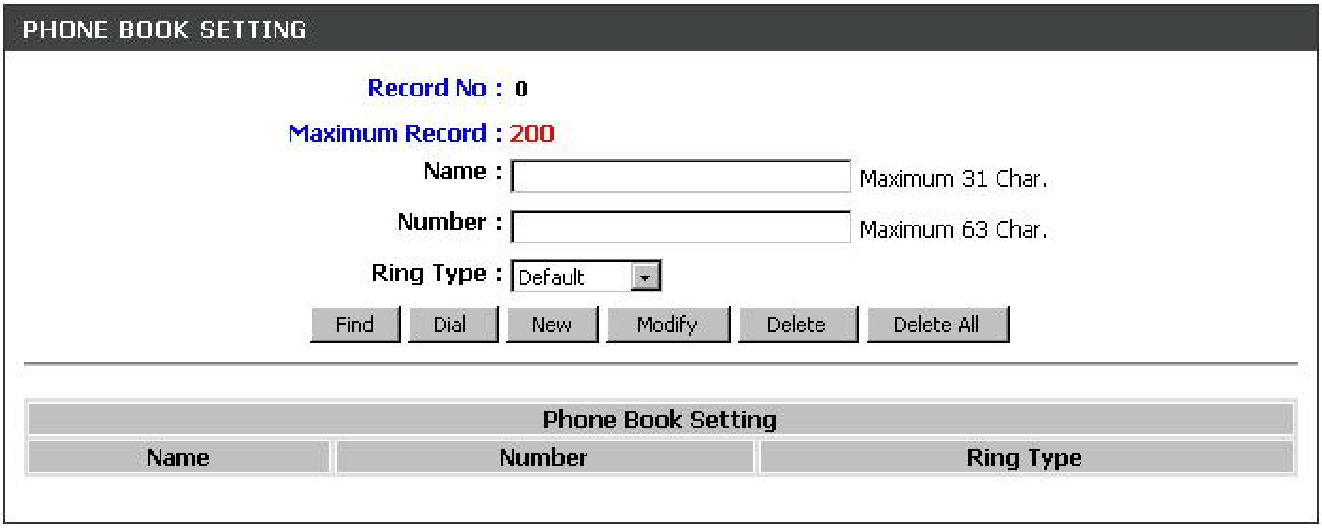
Anonymous Call	1. If DISABLE is selected, full URI and name are sent to the receiver's phone when the user makes a phone call. The URI and name of the caller are displayed on the receiver's phone. 2. When Full URI is selected, it uses "Anonymous" as its display name and URI when the user makes a phone call. It may display "Anonymous" or nothing on the receiver's phone. 3. When Display Name is selected, only the display name is replaced by "Anonymous" when the user makes a phone call. It may display "Anonymous" or nothing on the receiver's phone.
Anonymous Call Reject	Select Enable to reject anonymous calls.
Call Forward	1. Click No Answer to enable call forwarding to another number when no one answers the phone after 180s (default). The timer can be changed from 0-600s. Refer to section 6.18 to change the timer. 2. Click Busy to enable call forwarding to another number when you are busy on the phone. 3. Click Unconditional to transfer all incoming calls to another number. Enter the call forwarding number in the text box.
Hot Line	1. Enable or disable Hot Line 2. **Number**: a phone number which is the destination of the Hot Line 3. **Timeout**: the time after which the phone will dial the pre-configured phone number automatically
Transfer end of Conference Call	Enable or disable the feature of transferring calls after the three-way conference call is ended.
Pound Key Dial	Enable or disable Pound key Dial. Pound Key (#) can be defined as a <send> key.
Missed Call Display	Enable or disable to display missed calls on the LCD screen.

6.19. Advance – Phone Settings (Timer)



NTP Recycle Timer	The time interval that the IP phone synchronizes with the NTP server.
Inter Digit Timer	The time interval that the IP phone waits to detect the end of DTMF digits. No more digits are accepted after this period and the phone begins to dial.
Originating Not Accept Timer	The time interval that the caller’s phone waits to establish a call. If the receiver fails to answer the phone during this time interval, the caller’s phone will automatically disconnect.
Incoming No Answer Timer	The time interval that the receiver’s phone will ring. If the receiver fails to answer the phone during this time interval, the phone will automatically disconnect.
Hold Recall Timer	The recall time interval for the call party which is put on hold.
Auto Speaker Off Timer	The time interval that the speaker phone is on before turning off automatically (due to inactivity).

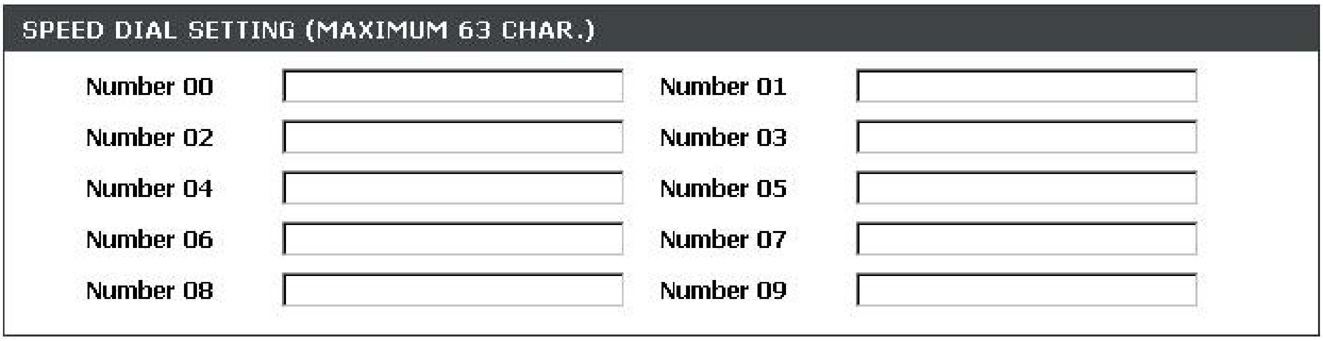
6.20. Advance – Phone Book



Phonebook menu allows the user to add, modify, and delete phone numbers. To add, type in the name and number then click NEW to add. To modify/delete, select the name from the list and click modify/delete.

Name	Name that you would like to add.
Number	Phone number that corresponds to the name.
Ring Type	Ring type of the number

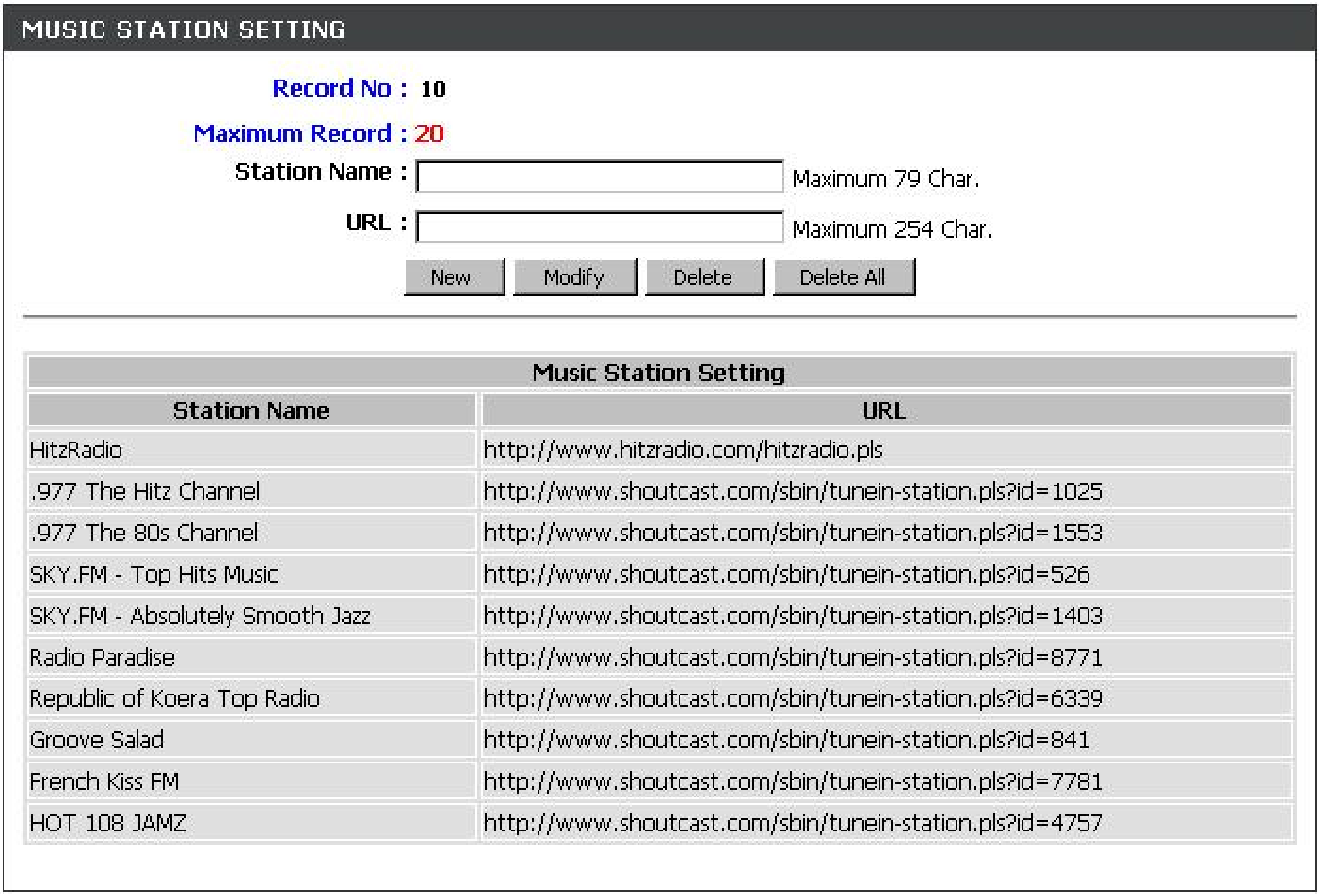
6.21. Advance – Speed Dial



Speed dial numbers can be accessed from the IP phone.

Number 0x	Speed dials phone number. 0x is the speed dial number.

6.22. Advance – Music Station



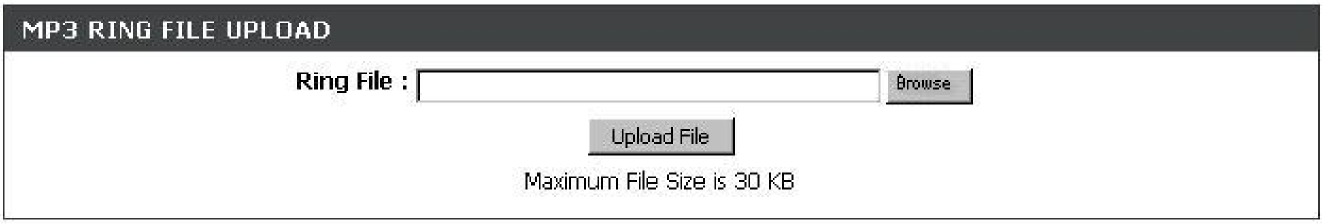
Station Name	URL
HitzRadio	http://www.hitzradio.com/hitzradio.pls
.977 The Hitz Channel	http://www.shoutcast.com/sbin/tunein-station.pls?id=1025
.977 The 80s Channel	http://www.shoutcast.com/sbin/tunein-station.pls?id=1553
SKY.FM - Top Hits Music	http://www.shoutcast.com/sbin/tunein-station.pls?id=526
SKY.FM - Absolutely Smooth Jazz	http://www.shoutcast.com/sbin/tunein-station.pls?id=1403
Radio Paradise	http://www.shoutcast.com/sbin/tunein-station.pls?id=8771
Republic of Koera Top Radio	http://www.shoutcast.com/sbin/tunein-station.pls?id=6339
Groove Salad	http://www.shoutcast.com/sbin/tunein-station.pls?id=841
French Kiss FM	http://www.shoutcast.com/sbin/tunein-station.pls?id=7781
HOT 108 JAMZ	http://www.shoutcast.com/sbin/tunein-station.pls?id=4757

Station Name An easy-to-memorize name for the station, ex: Station1.

URL A complete URL used to access the station

It accepts **20 stations** maximum. (*10 default stations are provided*). Please see “Appendix B” for more details.

6.23. Advance – MP3 Ring



Ring File Click “Browse” to choose one MP3 file and click “Upload File”. The maximum size of the MP3 file is 30KB.
The MP3 file is used for the Ringer type “MP3 Ring 9” (in sections 6.16, 6.18 and 6.20)

6.24. Call Log – Call Tracing Log

CALL TRACING LOG

No.	Trace Log
000	==============================
001	V: 01.00 2007.11.09
002	==============================
003	DSP V:v1.00 a2216
004	==============================
005	!0 Language(0), len(3262), size(4303)
006	!6 Basic number for random: (1)
007	total if=3
008	!6 WriteSetupInfo: 0. len(000014DC)
009	!6 IPConfig Fin!
010	!6 PB_ClearAll.
011	!0 phone_task: 0.
012	!0 alloc xcall(100D4664)
013	!0 Call state: x(100D4664), (dial)
014	!6 RtpPlayToneBase: tone(1)
015	!6 RtpPlayToneBase: sdSetGain
016	!6 RtpPlayToneBase: tone(20)
017	!6 RtpPlayToneBase: sdSetGain
018	DspChanClose
019	!0 free xcall(100D4664): 1
020	!6 Force delay
021	!6 Force delay

Call Tracing Log keeps a record of all the phone activities. This log is used by engineers to troubleshoot hardware problems.

7. Troubleshooting

The following troubleshooting information can be used to help solve most common problems.

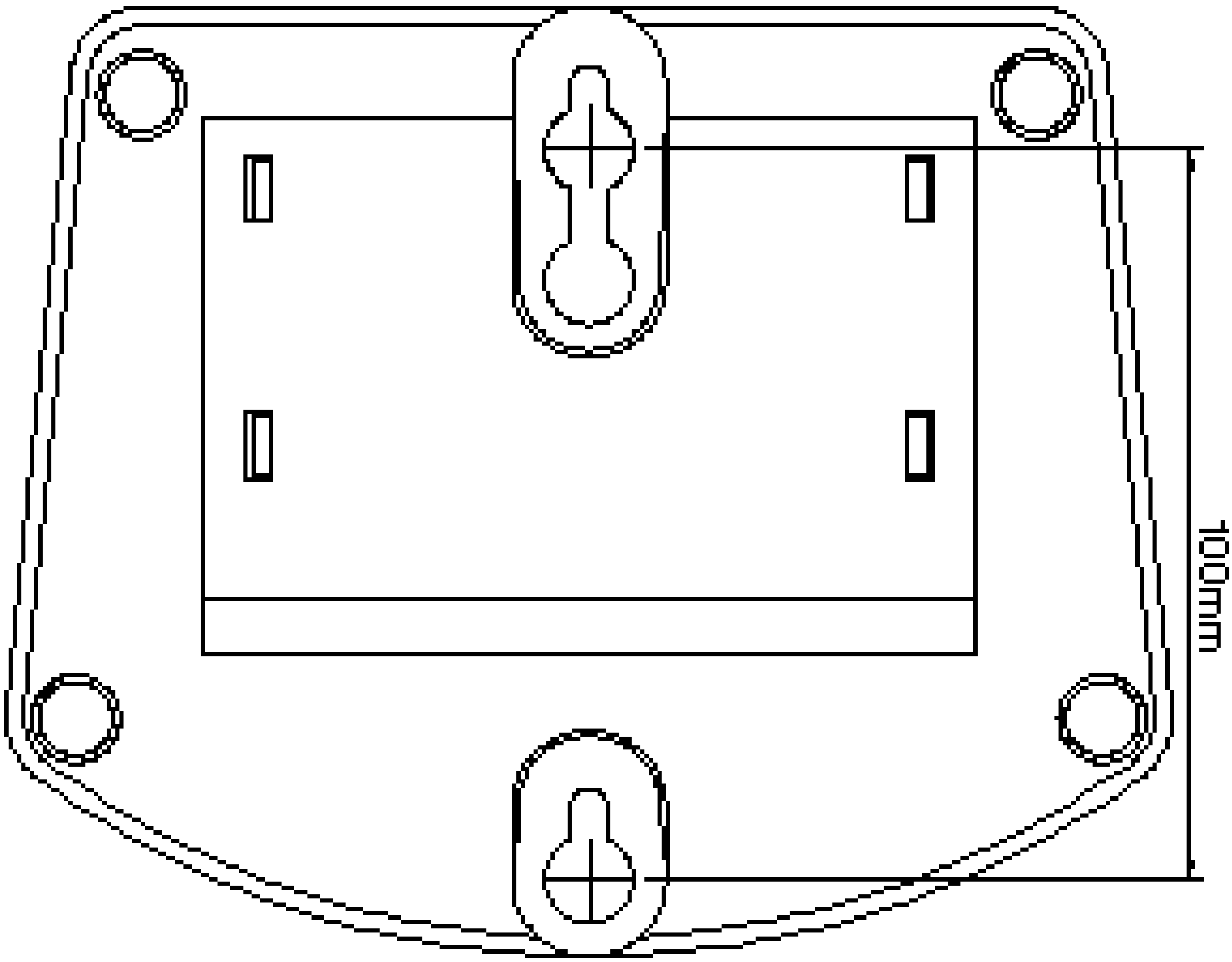
QUESTION	RECOMMENDED ACTION
There is no DIAL tone	1. Check if there are any loose connections.
Nothing is displayed on the LCD screen	1. Check if the power cord is connected properly. 2. Check if there is proper AC power coming from the power outlet.
Why can't I dial my friend's SIP number?	1. Check Registrar Server Domain Name/IP address and Outbound Proxy Domain Name/IP Address (under SIP Settings in Configuration Menu). Make sure you have the right Name or IP Address. 2. Check the LCD display on your phone to see if there is a name or number displayed on the screen. If the name or number is not displayed, use a web browser and access the configuration menu. Make sure that the Registrar Server Domain Name/IP Address is correct. 3. Check the register status under SIP Account Settings in the configuration menu (from a web browser). If your status is unregistered, it means you do not have a SIP account. Contact your SIP service provider to get an account.
I accidentally set DSL to enable and now the phone does not boot up	Unplug the power cord from the IP phone. Wait 2 seconds and plug the power cord back in the IP phone. Press and hold the MENU key. The system should bypass boot up and go straight into the phone setup menu. Modify the phone setting and make sure you save it before you exit.
Why do I get "Can't Upgrade Now" screen when I click [Submit] in the configuration menu?	Make sure you exit setting mode (phonebook, menu, speed dial…) before you click [Submit] in the configuration menu.
The WAN port of my DPH-150S (PhoneB) is connected with the LAN port of another DPH-150S (PhoneA). Then, my DPH-150S	To solve this problem, please change the IP segment of the PhoneA LAN port to something other than "192.168.15.xxx" (for example, "192.168.10.xxx").

(PhoneB) became disabled on the network so that I can not get VoIP services. What can I do to fix it?	Then, the PhoneB will automatically start to get the VoIP connection and the associated VoIP services. It is because that in the factory default settings the DPH-150S has an integrated DHCP server to assign the IP address of the LAN port with the IP segment of "192.168.15.xxx". That is, for this kind of connection of PhoneA & PhoneB (the WAN port of PhoneB is connected with the LAN port of PhoneA), the WAN port & the LAN port of PhoneB will be in the same IP segment (192.168.15.xxx), which will get the system of PhoneB confused so as to become disabled on the network. For this reason, we should change the IP segment of the PhoneA LAN port.

Appendix A: Wall Mount Installation

This appendix herein illustrates the installation step by step if you would like to mount the DPH-150S on the wall. Please print out this page (Figure A1) before the installation

1. Put the template (Figure A1), which you have printed before the installation on the wall. The template shows the two keyholes with plus sign indicating the center where the screw must be located.



> ***Attention***
> ***Do not scale the size of this page when you are printing. Be sure that the distance between the two keyholes is 100 mm.***

2. Use a screwdriver to fasten the screw on the wall. Please use the screw with the suitable size and reserve sufficient distance between the wall and the underside of the screw head as shown in Figure A2.

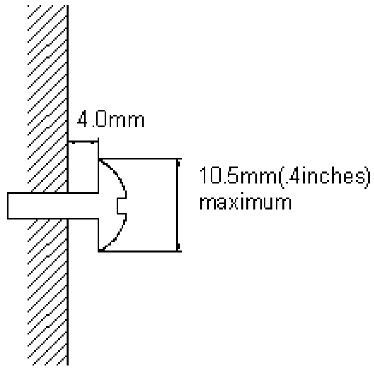


Figure A2

3. Place the mount on the wall as in Figure A3 and the keyholes of the mount are above the mounting screws.
4. Slide down the mount until it stops against the top of the keyhole
5. Place the DPH-150S on the wall mount as in Figure A4.

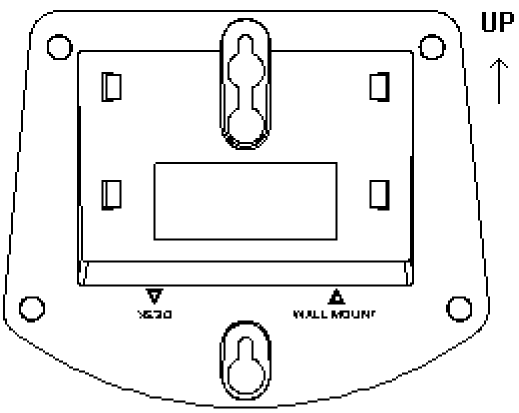


Figure A3

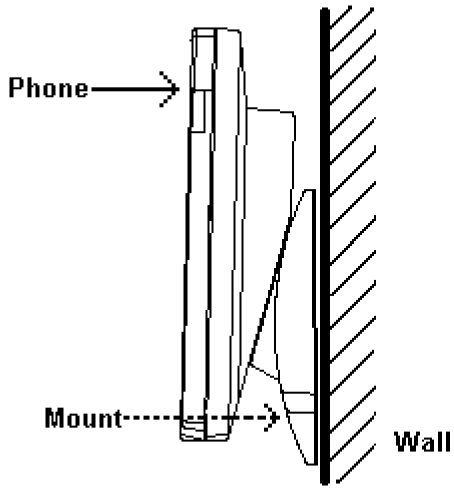
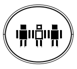
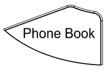
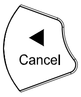


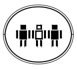
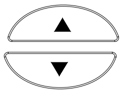
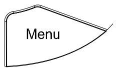
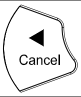
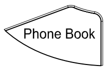
Figure A4

Appendix B: Internet Radio

1. How do I use the Internet Radio?

- Press  to turn on the Internet Radio
- Use  to select the preferred station
- Press  to turn off the Internet Radio.

2. Key Definition

Key	Definition	Key	Definition
	Turn on the Internet Radio		Increase / decrease the volume
	Pause / Play		Display the name of the current station
	Turn off the Internet Radio		Tune the Internet Radio to the preferred station
Numeral keys	The ten numeral keys 0, 1~9 are the quick access keys to the first ten preferred stations on web configuration "**Music Station**".		

3. Information about Internet Radio

- All the keys related to the Internet Radio are described in "Key Definition" (please refer to the above columns). Those key functions will be only available when the phone is on hook. If the phone is off hook, those key functions will back to the original designed which has stated in the User Manual.
- When the phone is receiving an incoming call, the Internet Radio function will turn off automatically.
- When the user picks up the handset or presses "SPEAKER" to make a phone call, the Internet Radio will also turn off automatically.
- Please turn off the Internet Radio before you are going to do any of the following:
  i. Use pre-dialing to make a phone call.
  ii. enter MENU to configure
  iii. access the Phone Book
  iv. adjust the Ringer Volume
- When the user is listening to the Internet Radio, the phone will display the current song and singer's name on the LCD screen.

FCC Statement:

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communication. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by tuning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

FCC Caution:

Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment.

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:
(1) This device may not cause harmful interference, and (2) this device must accept any interference received, include interference that may cause undesired operation.

IMPORTANT NOTICE:

FCC Radiation Exposure Statement:

This equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment. This equipment should be installed and operated with minimum distance 20cm between the radiator & your body. This transmitter must not be co-located or operating in conjunction with any other antenna or transmitter.
The availability of some specific channels and/or operational frequency bands are country dependent and are firmware programmed at the factory to match the intended destination. The firmware setting is not accessible by the end user.