ALL7960

User Manual

V1.00

Table Content

Introduction

Voice over IP (also known as Internet Phone) is a technology that allows anyone to make a telephone call over the Internet. This is a quick user guide for the ALL7960 SIP Phone. It is intended to help you configure the telephone and have it ready to run within a few minutes. Please follow the user guide carefully as troubleshooting the telephone can be very difficult and time consuming.

Safety Declaration

1. FCC Part 15 Class B
2. CE Class B
3. VCCI Class B
4. EN60950

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 listed below. Contact your supplier immediately if any item is missing.

IP Phone (Model: All7960)

Power Adaptor (5V DC)

User Manual

Ethernet Cable (1.5 meter)

1.2 Phone Specification

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
- Call Mute
- Call Retrieve
- Call Transfer
- 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
- Message Waiting Indicator
- 3-way Conference
- Redial

Codec

- G.711µ-law
- G.711a-law
- G.729a/b

Phone Functions

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

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 registered 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

Tone

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

IP Assignment

- Static IP
- DHCP
- PPPoE

NAT Traversal

- UPnP
- STUN

TCP/IP

- IP/TCP/UDP/DHCP/RTP/RTCP/ICMP/HTTP/NTP/TFTP/DNS

Configuration

- Key & LCD configuration
- Web browser configuration

Firmware Upgrade

- TFTP

Power

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

Environmental

- Operating temperature: 0□40□
- Storage temperature: -20□60□
- Operating humidity: 20%□80%

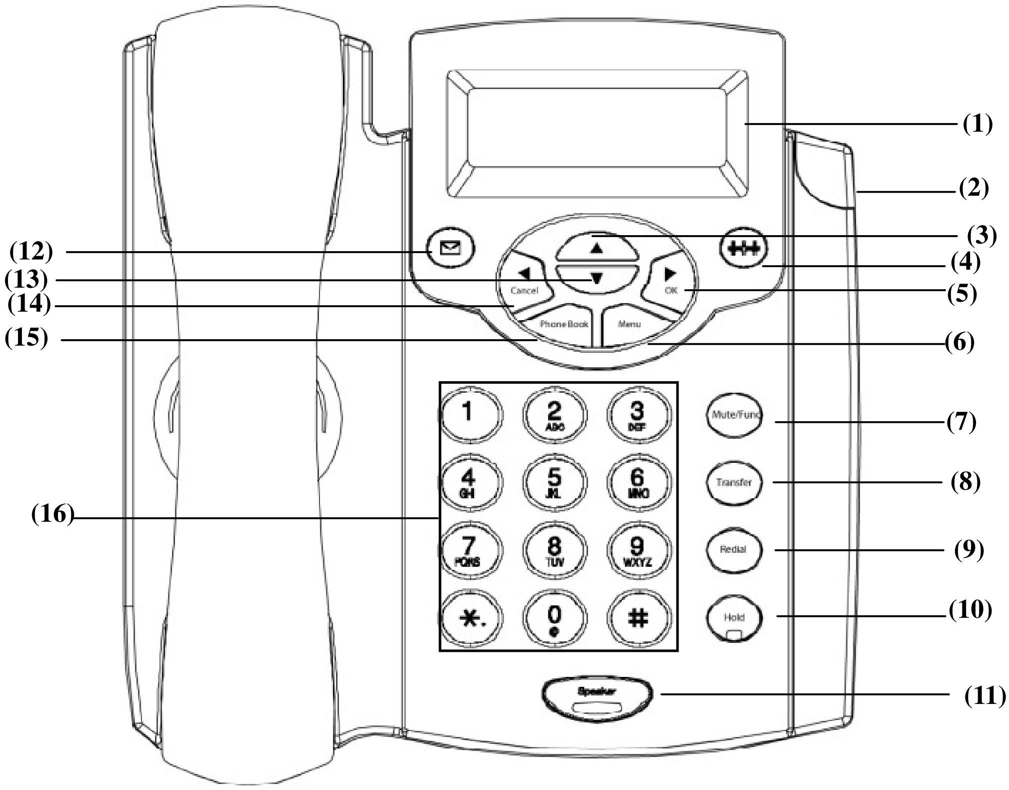
Physical Dimensions

- Size: 196(L) x 198(W) mm
- Weight: 760g
- Color: Black

Certification Compliance

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

1.3 Phone Description



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 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 voice message
(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 define & Text entry

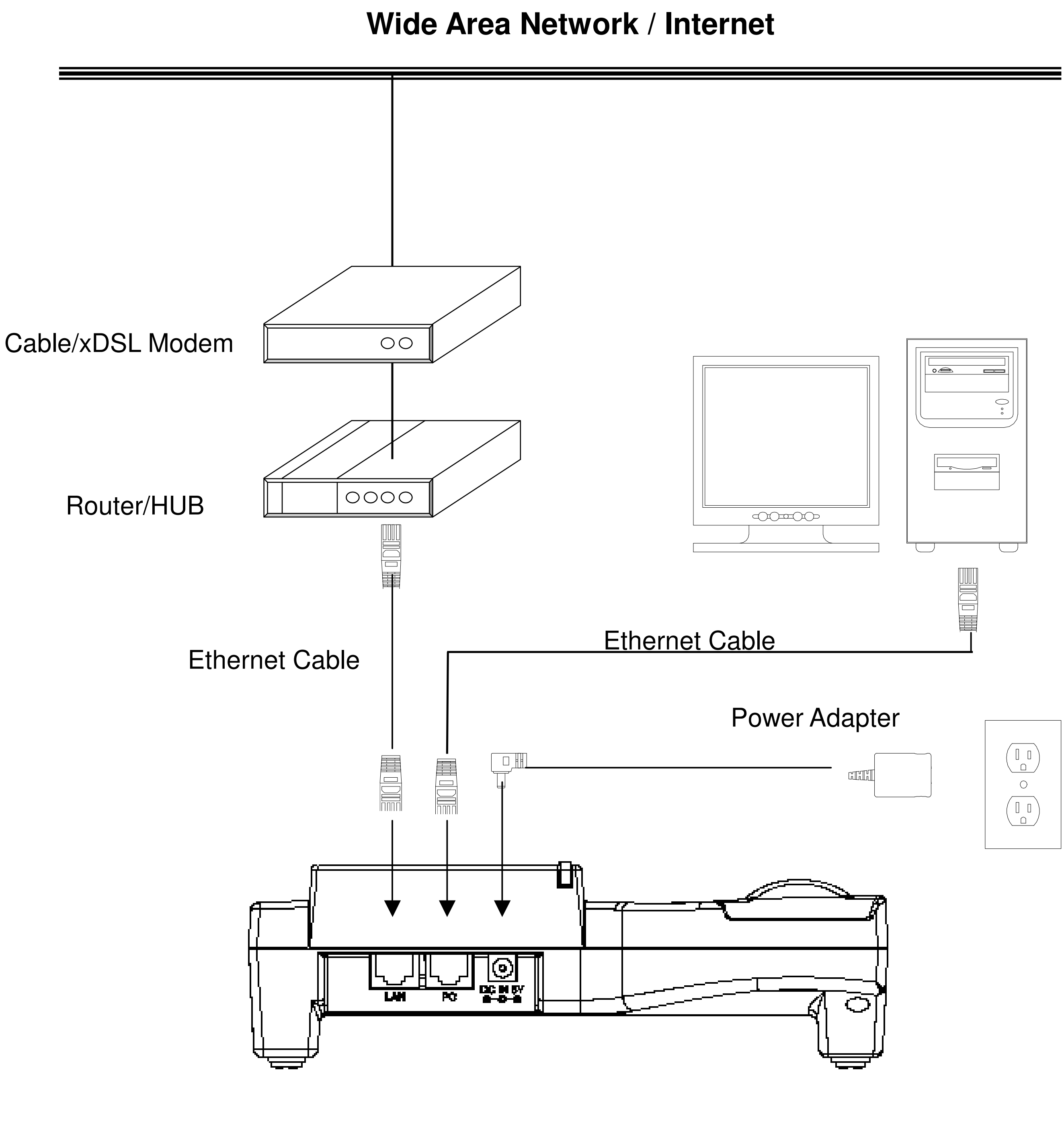
You use alphanumeric characters to enter details into the Phone Book, to create text and e-mail messages. The table below shows the characters that you can enter in the different text modes.

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

2. Connecting IP Phone

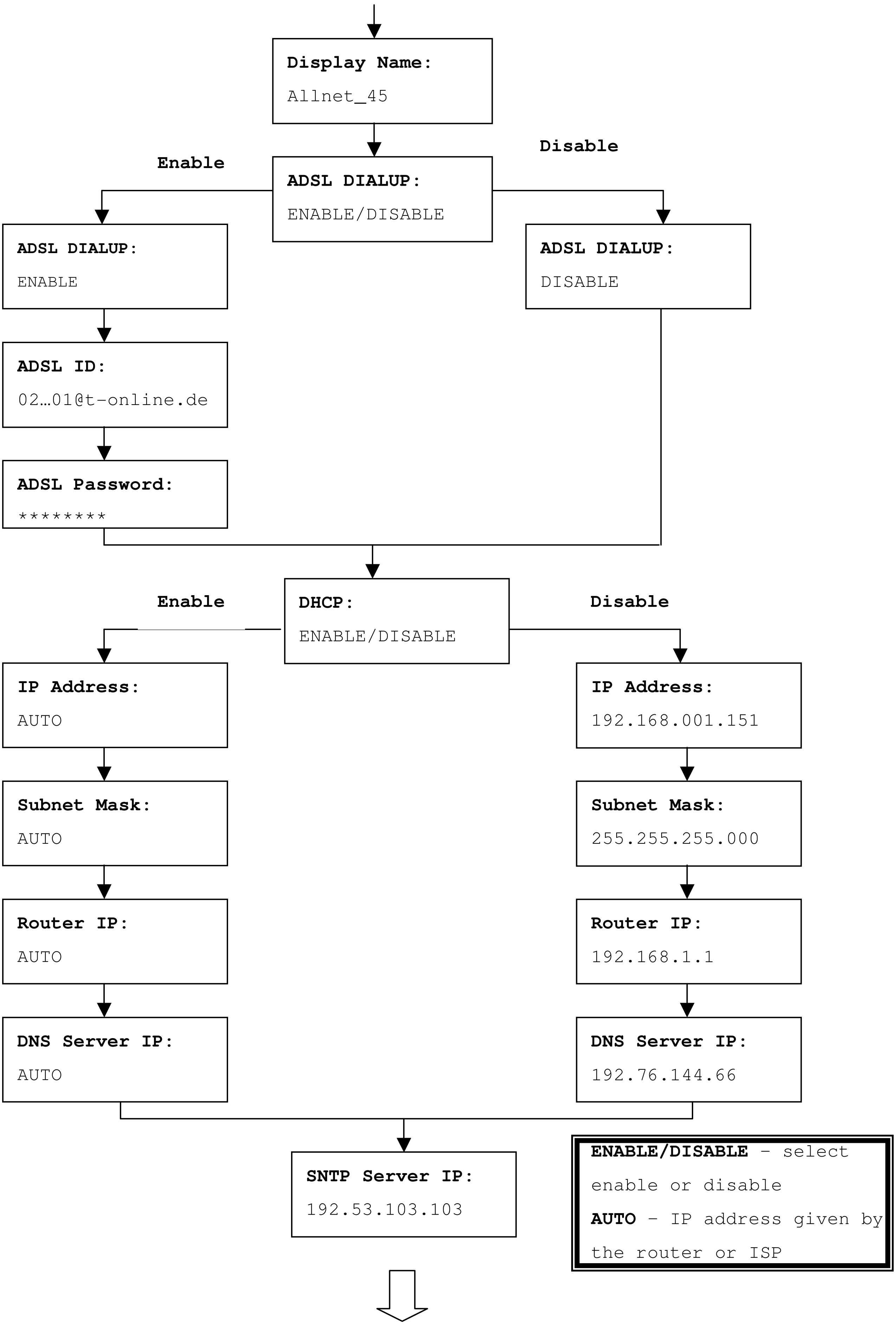
Connect the IP Phone as the following diagram:



3. Setting up

3.1 IP Phone Setup Map

Press the MENU key



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

↓

Ringing Type:
Ringing1/2/3…8

Ringing1/2/3…8 – select ringing1, 2, 3… 8

↓

MAC Address:
000FC9022B04

↓

Version:
V: 01.03.03

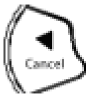
↓

Press the MENU key

↓

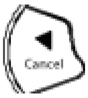
UPDATE CHANGES?
<OK> OR <CANCEL>

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: Left arrow key 

can be used as **Backspace** key.

3.2 Display Name

- Press 

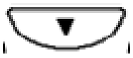
- Enter the display name

```
Display Name:
Your name
```

3.3 ADSL Dialup

Some Internet Service Provider (mostly ADSL) uses 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.

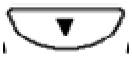
3.3.1 Enable ADSL Dialup

- Press 
- Use 

to select "Enable"

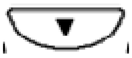
```
ADSL DIALUP:
ENABLE
```

3.3.2 Setup ADSL ID

- Press 
- Enter ADSL ID

```
ADSL ID:
provider_ID
```

3.3.3 Setup ADSL Password

- Press 
- Enter ADSL Password

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

3.3.4 Disable ADSL Dialup

- Press
- Use 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 static IP address, you must disable DHCP and enter the IP address provided.

3.4.1 Enable DHCP

- Press
- Use or to set DHCP "Enable"

```
DHCP:
ENABLE
```

- Press
- IP address automatically acquired

```
IP Address:
  192.168.1.161
```

- Press
- Subnet mask automatically acquired

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

- Press
- Router IP automatically acquired

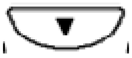
```
Router IP:
  192.168.1.1
```

3.4.2 Disable DHCP

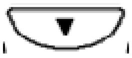
- Press
- Use or to set DHCP "Disable"

```
DHCP:
DISABLE
```

- Press

- Enter the IP address
- Press 

```
IP Address:
192.168.1.161
```

- Enter the subnet mask
- Press 

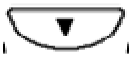
```
Subnet Mask:
 255.255.255.0
```

- Enter the router IP address

```
Router IP:
192.168.1.1
```

3.5 DNS Server IP

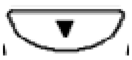
The domain name system (DNS) is the way that Internet domain names are located and translated into Internet Protocol addresses. There is probably a DNS server within close geographic proximity to your ISP that maps the domain names in your Internet requests or forwards them to other servers in the Internet.

- Press 

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

3.6 SNTP Server IP

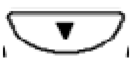
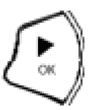
Simple Network Time Protocol (SNTP) is a protocol used to help match your system clock with an accurate time source. If you do not know your SNTP Server IP, please ignore this section. SNTP Server IP address can be either URL or IP.

- Press 
- Enter SNTP server IP or URL

```
SNTP Server IP:
216.133.140.78
```

3.7 Do Not Disturb

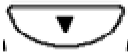
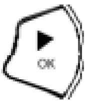
This setting allows the user to reject all incoming phone calls.

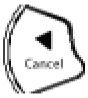
- Press 
- Use  or  to select “Enable” or “Disable”

```
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 forwarded phone number. Refer to section 7.0 for more information on call forwarding.***

- Press 
- Use 

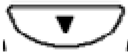
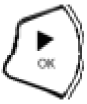
or 

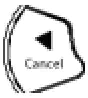
to select "Enable" or "Disable"

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

3.9 CF (call forward) User Busy

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

- Press 
- Use 

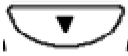
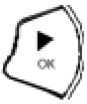
or 

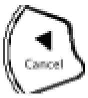
to select "Enable" or "Disable"

```
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 

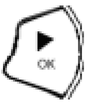
or 

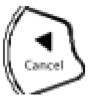
to select "Enable" or "Disable"

```
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 

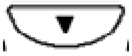
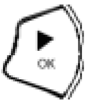
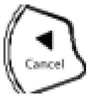
or 

to select "Enable" or "Disable"

```
Anonymous Call:
ENABLE / DISABLE
```

3.12 Anoy Call Rej. (Anonymous Call Rejection)

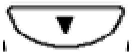
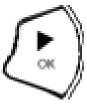
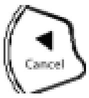
Reject any anonymous incoming calls.

- Press 
- Use  or  to select "Enable" or "Disable"

Anony Call Rej:
ENABLE / DISABLE

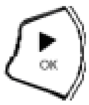
3.13 Ringing Type

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

- Press 
- Use  or  to select the ring type

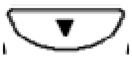
Ring Type:
Ringing 1/2/3/4/5/6/7/8

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 menu
- When asked to save or cancel, press  to save

3.14 MAC Address

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

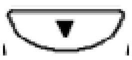
- Press 
- **MAC address** is displayed on the screen

MAC Address:
000FC9017D4A

LAN MAC Address:
000FC9017D4B

3.15 Version

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

- Press 
- Firmware **version** is displayed on screen

Version:
V: 01.03.03

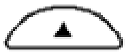
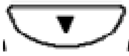
3.16 Language Selection

The VoIP Phone (model no. Allnet 7960) supports 4 languages: English, German, Italian and Spanish.

- Press 

followed by 


```
Language:
English
```

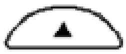
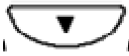
- Use  or  to select the preferred language

3.17 Time Format

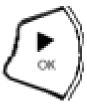
You may select the 12hr or 24hr time format.

- Press 

followed by 

- Use  or  to select the time format

```
Time Format:
24Hours
```

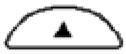
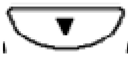
- Press 

when done

3.18 Volume Adjustment

3.18.1 Ringer Volume

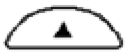
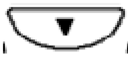
While the handset is in place,

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

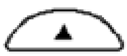
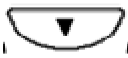
3.18.2 Speaker Volume

While the handset is in place,

- Press 

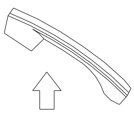
- Press  to increase the speaker volume and  to decrease the speaker volume

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 IP Address

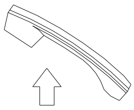
- Lift handset  or press SPEAKER button 

- Dial IP address.

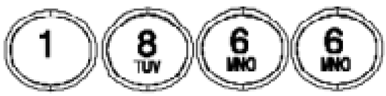
  For example: dialing 192.168.0.1

- Press OK  or wait until the timer expires to dial.

4.2 Dialing SIP Number

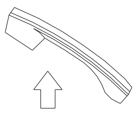
- Lift handset  or press SPEAKER button 
- Dial SIP Number

  For example: dialing 1866

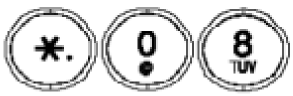


- Press OK  or wait until the timer expires.

4.3 Speed Dialing

- Lift handset  or press SPEAKER button 
- Dial Speed Dial number.

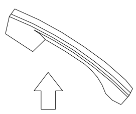
  For example: dialing speed dial number 08,



4.4 Answering a Phone Call

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

When phone rings:

- Lift handset  or press SPEAKER button  to begin conversation.

4.5 Switching to another Line

While having a conversation:

- Press Hold to switch to another line.

4.6 Mute

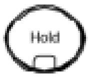
Note: While mute is activated, sound from the caller can be heard from your speaker but your sound can't be heard by the caller.

While having a conversation:

- Press Mute  You may press Mute key again to resume conversation.

4.7 Call Transfer

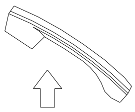
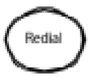
While having a conversation:

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

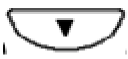
4.8 Redial

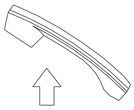
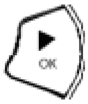
Note: To return to idle mode, press **CANCEL** key

4.8.1 Last Dialed Number

- Lift handset  or press **SPEAKER** button 
- Press **Redial**  to dial the last dialed number.

4.8.2 Through Call History

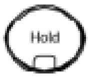
- Press **Redial** . Does not lift the handset when you press **Redial**.
- Press **Redial**  again to cycle through the dialed, missed and received calls.
- Press **DOWN** key  to scroll down the dialed, missed or received lists until the number is displayed on the screen.

- Pickup the handset  or press **OK** 

4.9 On Hold

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

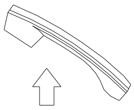
While having a conversation:

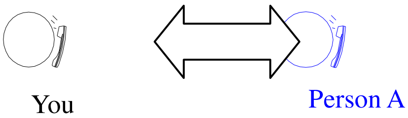
- Press **HOLD**  (Press **HOLD** again to resume conversation)

4.10 Call Forward

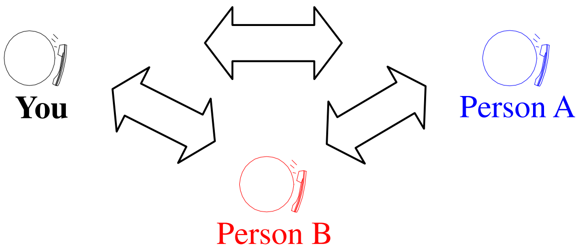
Please refer to IP Phone Setup and Web Browser Configuration section to setup call forwarding.

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



5. Using the phone book

5.1 Dialing from the Phonebook

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

- Press **OK** to dial.

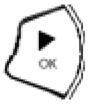
5.2 Storing a Number

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

5.3 Editing a Number

- Press the **PHONE BOOK** key to access the phonebook.
- 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

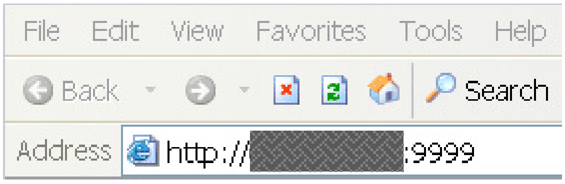
- Press the **PHONE BOOK** key  to access the phonebook.
- Press  until the name you want to delete is selected.
- Press the **PHONE BOOK** key  again.
- Select “Delete” and press **OK**  to delete.
- Press **OK**  again to save the new list on the phonebook.

6 Using the web configuration

The configuration web can be accessed using a web browser.

6.1 Accessing Configuration Menu

1. Open the web browser (ie. Internet Explorer, Netscape…)
2. Type in the **IP Address** of the phone followed by :9999

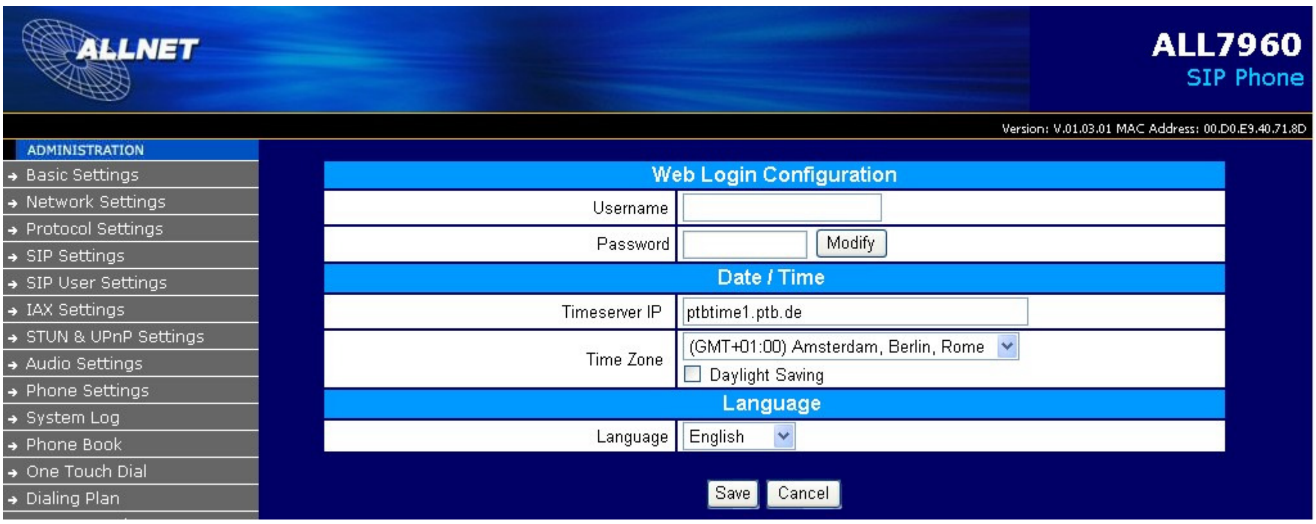


IP address is provided by your Internet Service Provider (ISP). If your ISP supports DHCP, you may obtain the IP address from you phone. Press " Func.+ 9 " to get IP address. Also can login from LAN port by http://192.168.15.1:9999 .

Enter **User Name** and **Password** (leave User Name and Password blank if you are installing the phone for the first time)

Click **OK**

6.2 Web Login Setting



User Name	Configuration menu login name.
Password	Configuration menu login password.
NTP Server IP	Network Time Protocol (NTP) is a protocol used to help match your system clock with an accurate time source (eg atomic clock, time server). It is good practice to have all your networked computers synchronized with one server.

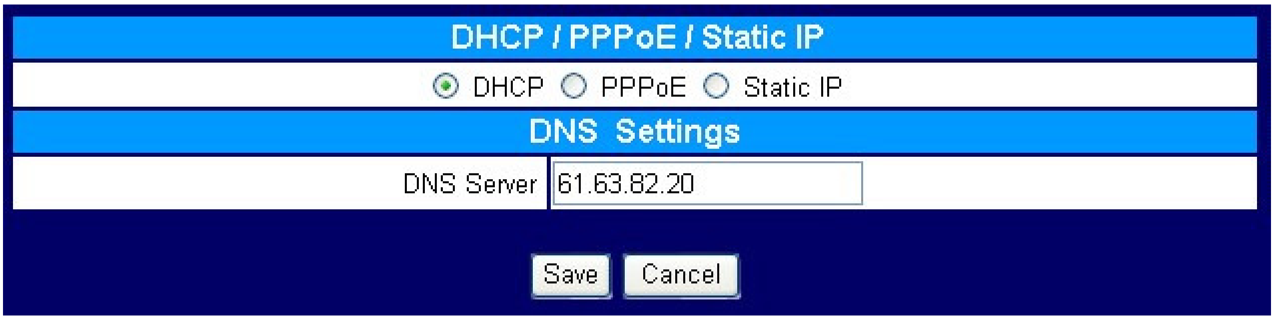
Time Zone Select your time zone. If there is daylight saving in your area, click the check box.

6.3 Management Setting –Factory Setting

Click on “**Management**”, Select “**Factory Setting**” and the above screen will display on the screen.

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

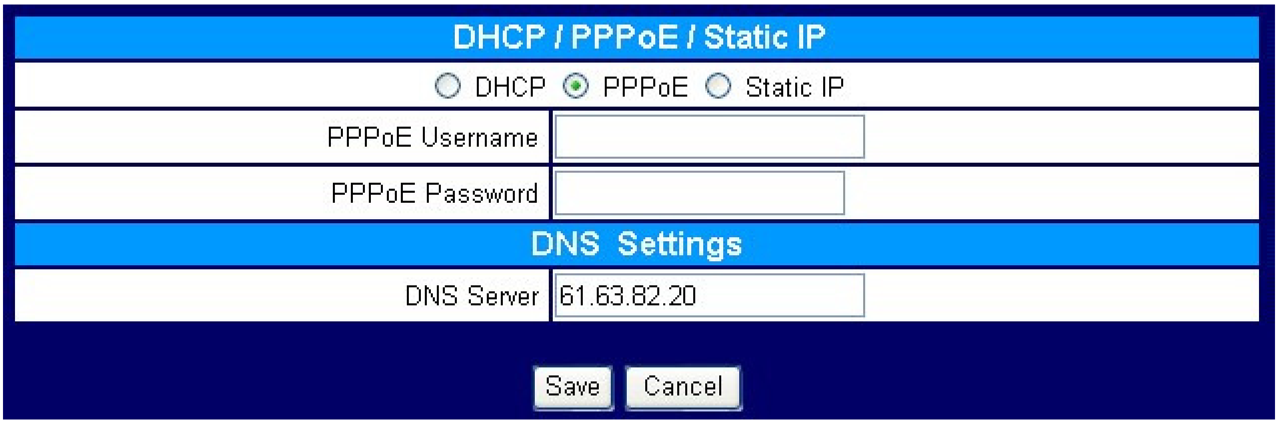
6.4 Network Setting – DHCP



DHCP Server Dynamic Host Configuration Protocol (DHCP) Server address. This IP address information is obtained automatically from your ISP.

DNS Server DNS address provided by your ISP.

6.5 Network Setting – PPPoE



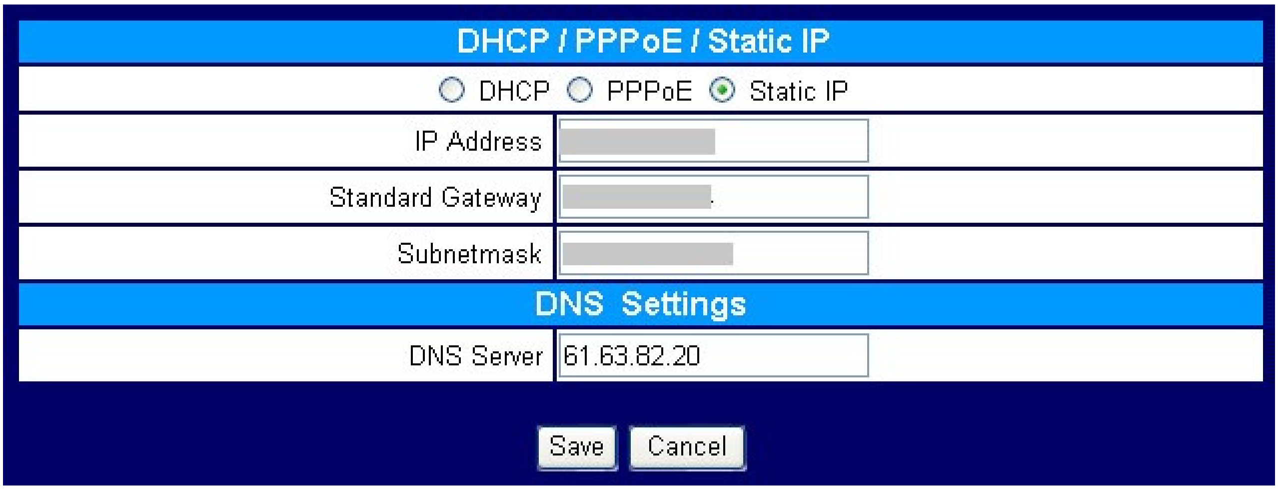
Choose PPPoE if your ISP uses PPPoE. (Most DSL users use PPPoE).

PPPoE ID PPPoE ID/username provided by your ISP.

PPPoE Password PPPoE password.

DNS Server DNS address provided by your ISP.

6.6 Network Setting – Static IP

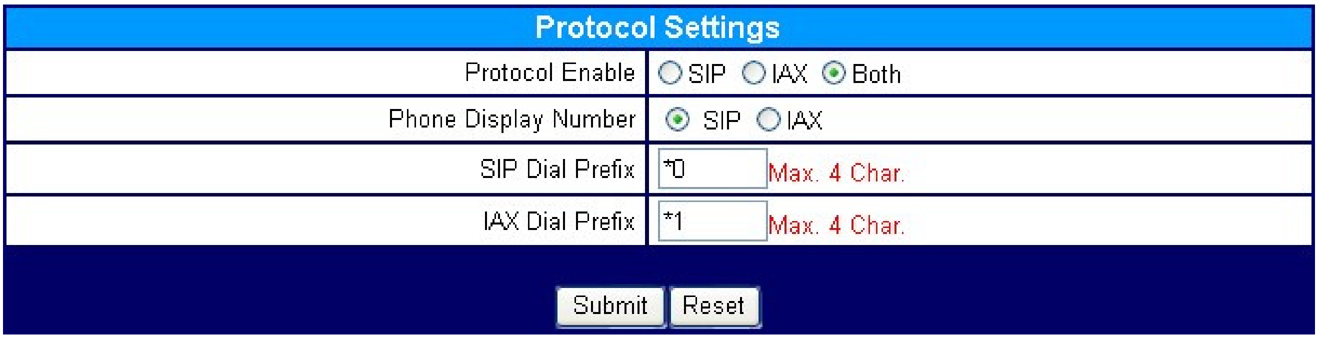


Choose Static IP network setting if all Wide Area Network IP is provided to you by your ISP.

IP Address	IP address assigned to you by your ISP.
Router IP	Router IP address.
Subnet Mask	Subnet mask address.
DNS Server	DNS server address provided by your ISP.

NOTE: RESTART the system for new settings to take effect after you modify the IP address.

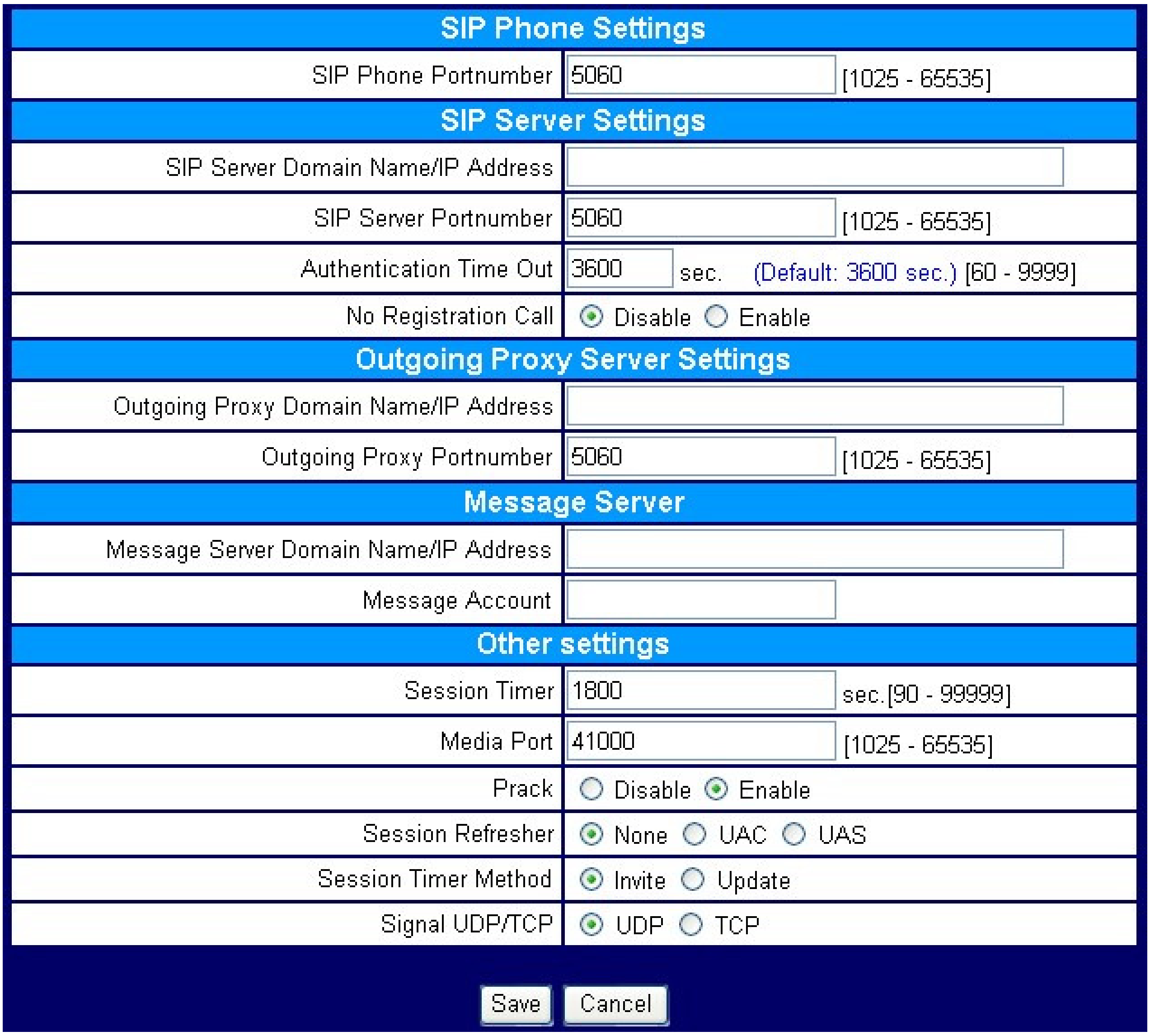
6.7 Protocol Setting



Protocol Enable	Enable SIP, IAX or both.
Phone Display Number	Display SIP or IAX phone number.
SIP Dial Prefix	Dial *0 (Default) to dial SIP number.
IAX Dial Prefix	Dial *1 (Default) to dial IAX number.

Note: Phone rebooting is necessary when any of the above settings is changed

6.8 SIP Setting – SIP Phone Setting, Registrar and Outbound Proxy Server

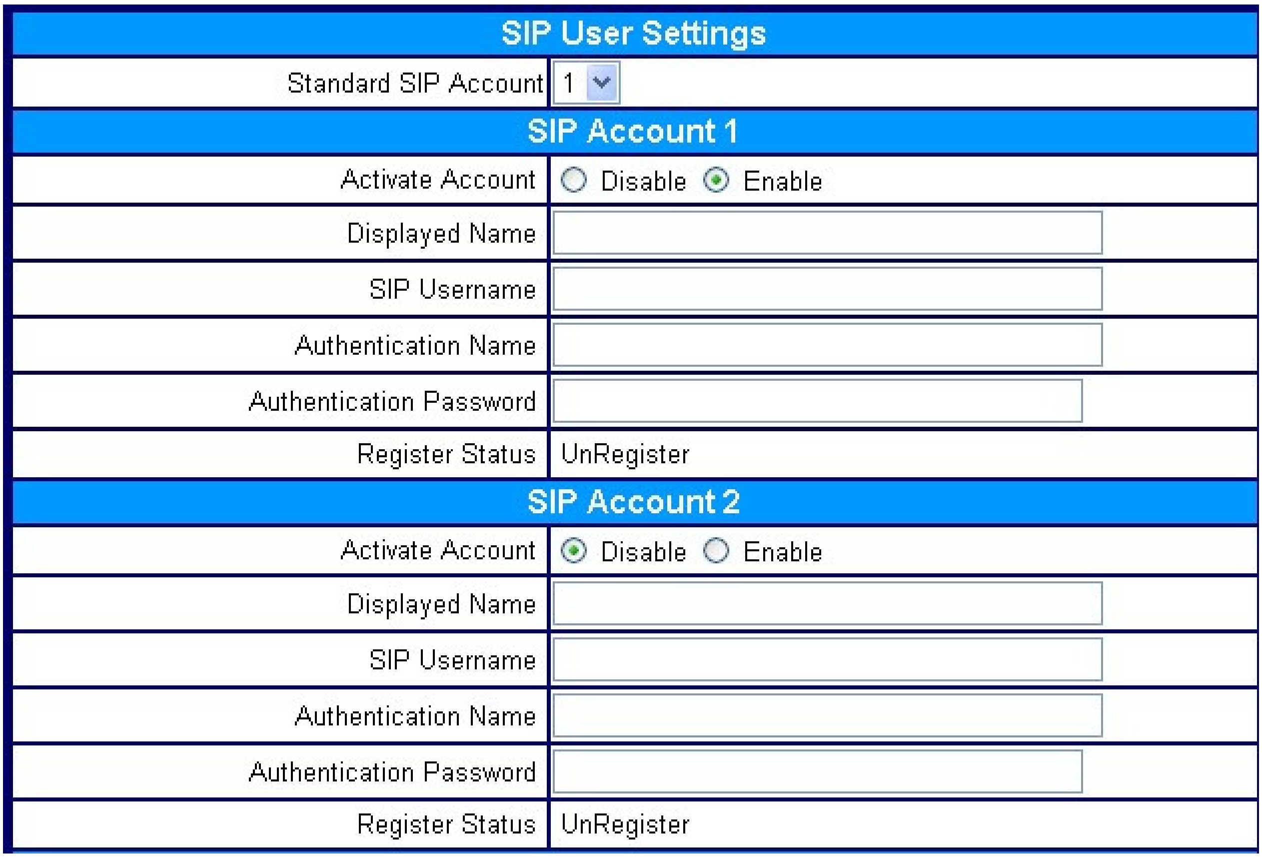


Session Initiation Protocol (SIP) is the most popular Voice over IP standard. It enables two or more people to make phone calls, share multimedia and make multimedia conference over the internet. Please have an administrator setup these settings for you or obtain this information from your SIP service provider.

SIP Phone Port Number	SIP phone port number.
SIP Server Domain Name/IP Address	Registrar server domain name or IP address.
SIP Server Port Number	Registrar server port number.
Authentication Time Out	The time that the phone waits to connect to the SIP server after the user dialed a number. If still not connected, the phone will disconnect and redial.

No Registration Call	Enable/Disable no registration call.
Outgoing Proxy Domain Name/IP Address	Outbound proxy domain name or IP address.
Outgoing Proxy Port Number	Outbound proxy port number.
Message Server Domain Name/ IP Address	Message server domain name or IP address.
Message Account	Message server port number.
Session Timer	The time interval in which the phone periodically refresh 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	Prack ensures that media information is exchanged and that network checks before connecting the call. 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.
Signal UDP/TCP	Select SIP signal transmission method. Default method is UDP.

6.9 SIP Account Settings



You may have up to 4 accounts. i.e., the IP phone can receive up to four different phone numbers.

Default Account When you dial a number, the default account is used to dial. 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 callee's LCD..

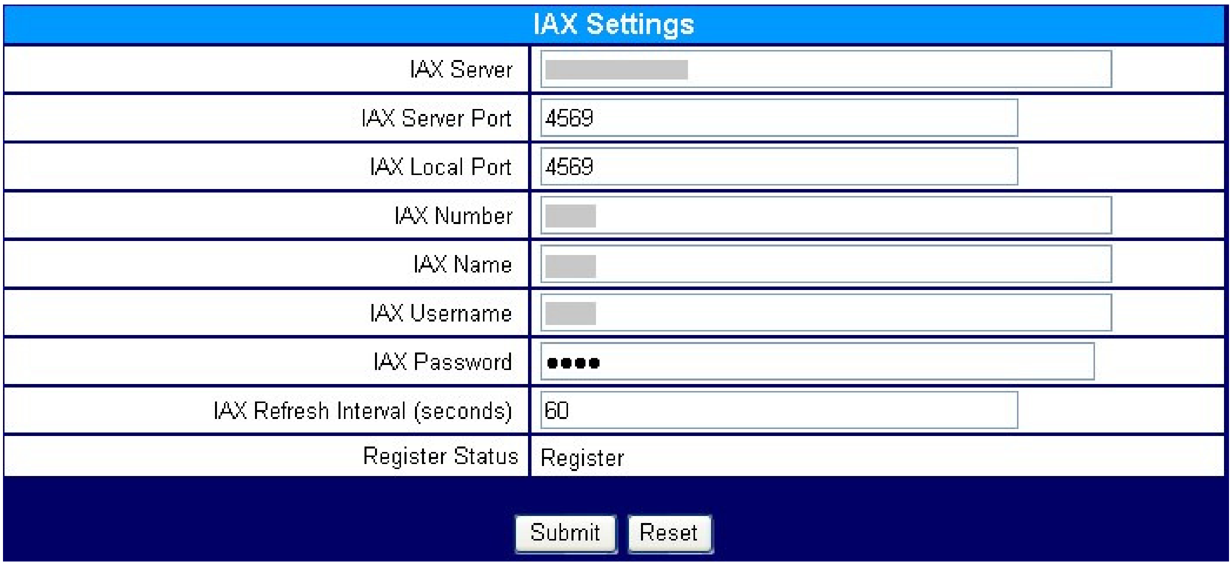
SIP User Name The number in the URI displayed on the LCD for the caller.

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

Authentication Password Password to log into the SIP server.

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

6.10 IAX Settings



IAX Server	The Asterisk server's IP address
IAX Server Port	The port number for the Asterisk server. Default is 4569
IAX Local Port	All7960 supports IAX2 protocol. Normally IAX2 uses Port 4569.
IAX Number / IAX Name / IAX Username / IAX Password	Number, name or password to log into the IAX server.
IAX Refresh Interval (seconds)	The time interval in which the phone periodically refresh IAX sessions by sending repeated INVITE or Update request.
Register Status	Shows register status.

Note: Phone rebooting is necessary when any of the above settings is changed

6.11 STUN&UPnP Settings

NAT traversal is a challenge that all Service Providers looking to deliver public IP-based voice service must solve. The challenge is to provide secure connection to subscribers behind NAT (Network Address Translation) devices and Firewalls. Overcoming this traversal problem will lead to widespread deployment of profitable voice over IP service to any subscriber with a broadband connection. Therefore, this IP Phone implements NAT traversal function for solving the Firewall and NAT traversal problems.

STUN Simple Traversal of User Datagram Protocol through Network Address Translators is a protocol that allows applications to determine the types of NATs and firewalls are in between them and the internet. STUN also provides the ability for applications to determine the public IP addresses allocated to them by the NAT.

STUN Server: Port number Enter STUN domain name or IP address if STUN is enabled.

6.12 STUN&UPnP Settings – Manual Config External IP/Port

External IP/Port Settings	
User Defined External IP/Port	◉ Disable ○ Enable
External IP Address	0.0.0.0
External SIP Port	5060 [1024 - 65535]
External Media Port	41000 [1024 - 65535]

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 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

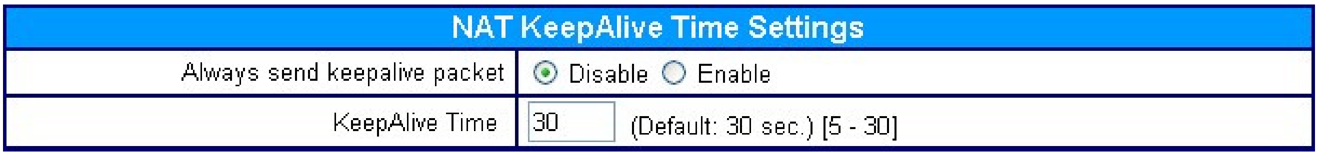
NOTE: It has to be complied with the settings of virtual server of the NAT devices if IP Phone enables the configuration manually.

6.13 STUN&UPnP Settings – UPnP Setting

UPnP Settings	
UPnP	◉ Disable ○ Enable

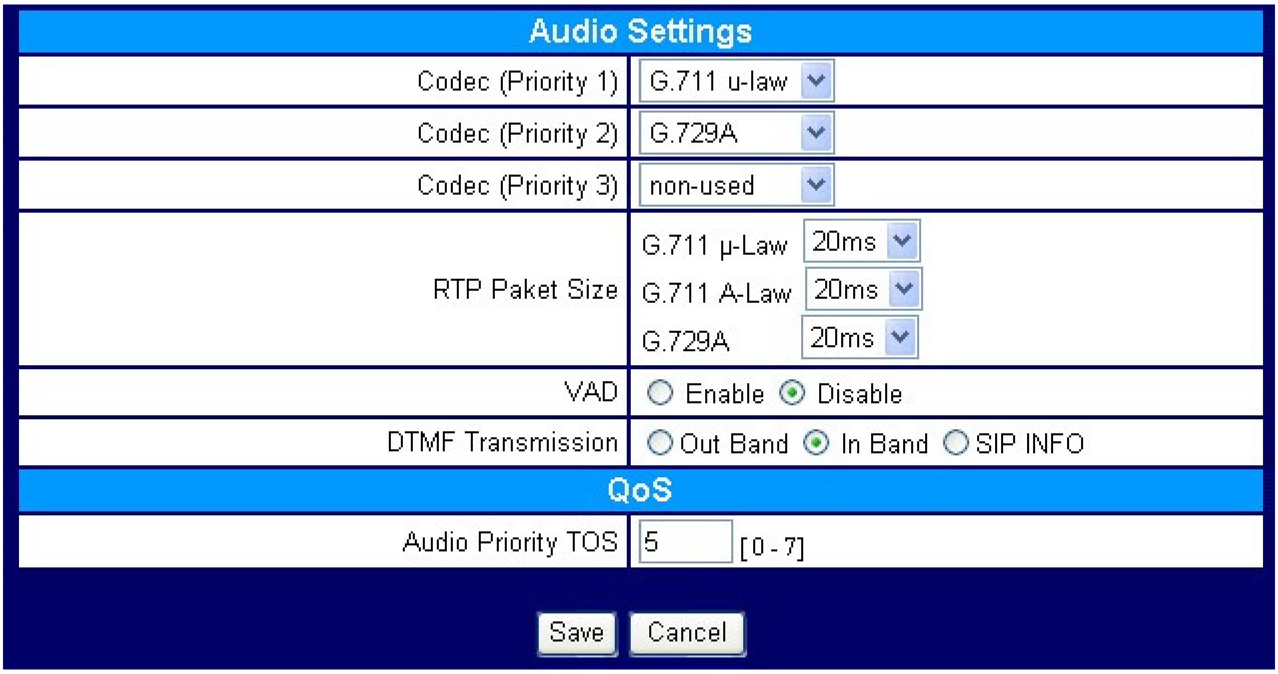
UPnP Enable or disable universal plug and play. Some NAT supports UPnP so STUN is not required and must be disabled.

6.14 STUN&UPnP Settings – NAT Keep Alive Time Settings



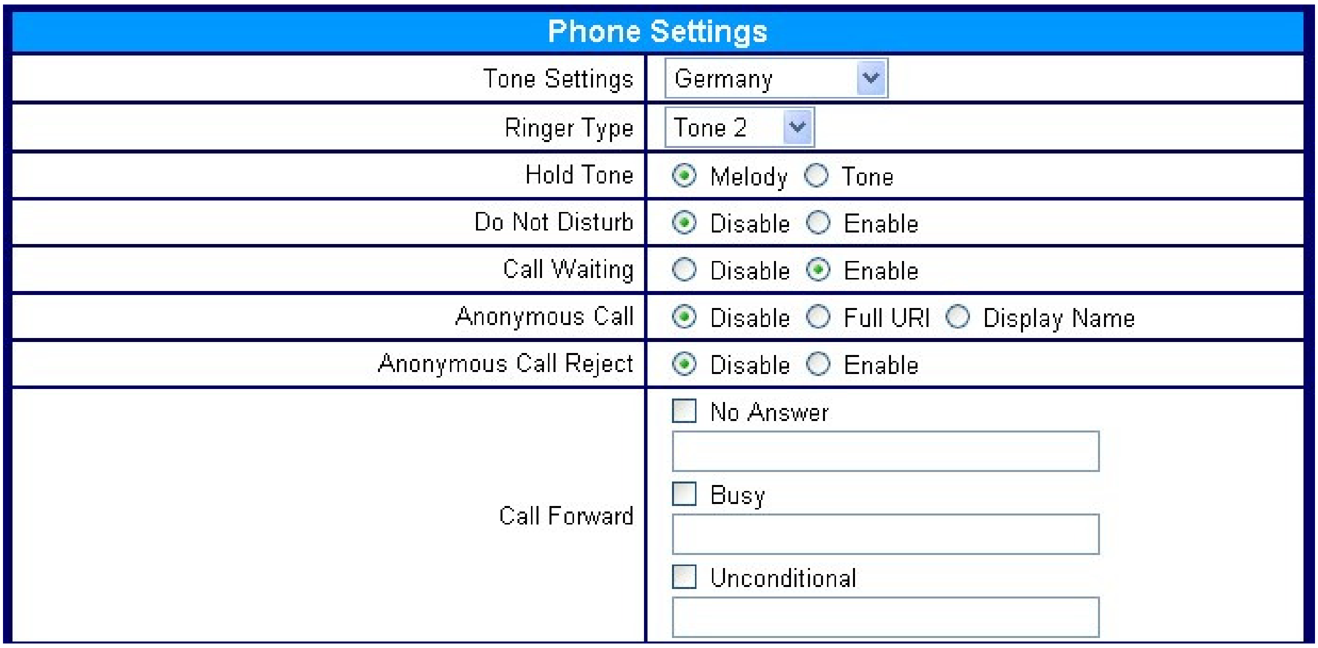
Always send keepalive packet	Enable or disable to send keepalive packet always.
KeepAlive Time	The time interval that the IP phone always sends the keepalive packet in order to ensure NAT works properly.

6.15 Audio Setting & QoS



Codec	Voice Compression Algorithm priority settings. Select from the most used codec to the least used codec.
RTP Packet Size	Real-Time Transfer Protocol (RTP) packet length.
VAD	Enable or disable VAD.
DTMF Transmission	Select the tone method for IP phone.
Audio Priority TOS	Sets the type of service (Layer 3) for this Internet datagram.

6.16 Phone Settings



Tone Setting	Select the tone for particular country
Ringer Type	Select the type of ring (1 to 8).
Hold Tone	Select melody or tone when HOLD key is pressed.
Do Not Disturb	Reject all incoming calls.
Call Waiting	Enable or disable 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, only user name is displayed on the receiver's phone when the user makes a phone call. 3. When Display Name is selected, only name is displayed on the receiver's phone when the user makes a phone call.
Anonymous Call Reject	Select Enable to reject anonymous calls.
Call Forward	1. Click No Answer to enable call forward to another number when no one answers the phone after 180s (default). The timer can be changed from 0-600s. Refer to section 7.14 to change the timer. 2. Click Busy to enable call forward to another number when user is busy on the phone. 3. Click Unconditional to transfer all incoming calls to another number. Enter the call forward number on the text box.

6.17 Phone Settings – Auto Dial

Activate	Enable/ Disable auto dial.
Phone Number	The phone number of auto dial.

6.18 Phone Settings – Timer

Timer		
NTP Recycle Timer	1	hour [1 - 24] Time Server Adjustment Interval
Inter Digit Timer	5	sec. [0 - 60] 0: Disable
Originating Not Accept Timer	180	sec. [0 - 60] 0: Disable
Incoming No Answer Timer	180	sec. [0 - 60] 0: Disable
Hold Recall Timer	180	sec. [0 - 60] 0: Disable
Auto Speaker Off Timer	30	sec. [0 - 60] 0: Disable

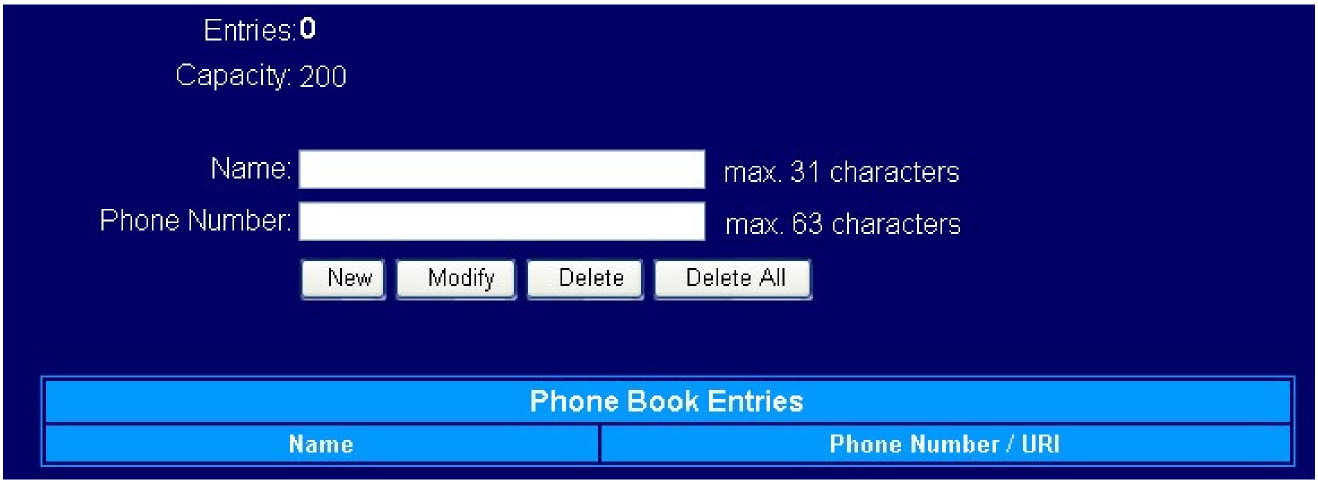
NTP Recycle Timer	NTP recycle time.
Inter Digit Timer	The time interval that the IP phone waits to detect the end of DTMF digits. No more digits are accepted after this period and the phone begins to dial.
Originating Not Accept Timer	The time interval that the caller's phone waits to establish a call. 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 time interval that the caller is put on hold before the phone automatically disconnect.
Auto Speaker Off Timer	The time interval that the speaker phone is on before turning off automatically (due to inactivity).

6.19 System Log

No.	Trace Log
000	!8 my_malloc: 118D6008 total use(73111) = 4084 + 69027
001	!8 my_free: 118D6008 total use(4084) = 73111 - 69027
002	!8 my_malloc: 107F4118 total use(4340) = 4084 + 256
003	!8 my_malloc: 118D6008 total use(70041) = 4340 + 65701
004	!8 my_free: 118D6008 total use(4340) = 70041 - 65701
005	!8 my_malloc: 118D6008 total use(70056) = 4340 + 65716
006	!8 my_free: 118D6008 total use(4340) = 70056 - 65716
007	!8 my_free: 107F4008 total use(4084) = 4340 - 256

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

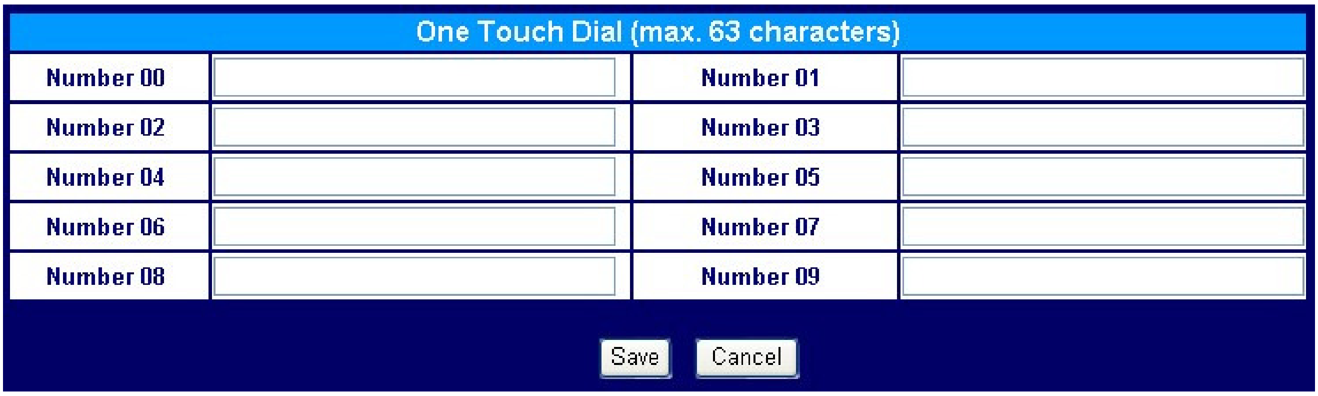
6.20 Phone Book



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.
Phone Number	Phone number that corresponds to the name.

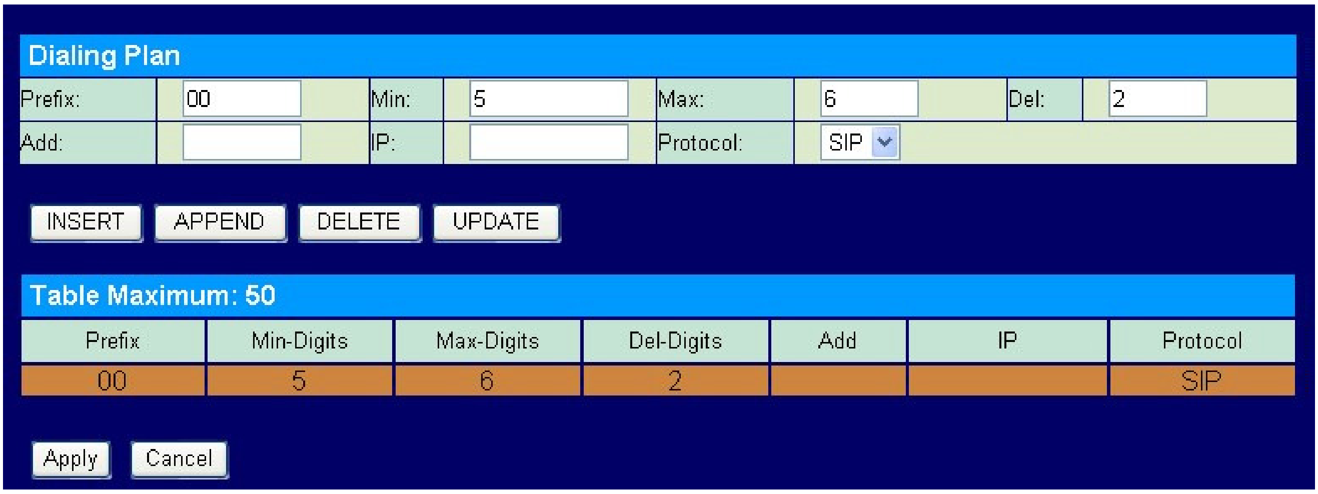
6.21 One Touch Dial



One Touch dial numbers can be accessed from the IP phone.

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

6.22 Dialing Plan



Click on **Dialing Plan** and the above screen will be displayed.

Prefix	Numbers defined in this field will be inserted at the beginning of the dialing pattern. Maximum input length is 6 digits.
Min.	Minimum digits user can key in.
Max.	Maximum digits user can key in.
Del.	Number of digit defined in this field will be removed from the dialing pattern. For example, if we dialed 81352109378 and the delete digit is 2, then the actual dialed number is 352109378. First 2 digits are removed. Maximum delete digit is 3 digits.

Add — Numbers in this field are added at the beginning of the dialing pattern. For example, if 001 is in this field, the number dialed is 001+the rest of the numbers. The input length is limited to 6 digits.

IP — Remote side gateway IP addresses. When the prefix number is matched, this call will go to the gateway with this IP address

Protocol — Choose the dialing plan for SIP or IAX.

[Insert] — Insert a record where the current record is located (Current record is marked as different color).

[Append] — Add a new record to the bottom of the list.

[Delete] — Delete a record.

[Update] — Modify the value of the selected record in the LATT.

6.23 Factory Settings

Click on **Factory Settings** and the above screen will be displayed.

Factory Settings — Restores all the phone-settings back to factory default settings. You will also loose all entries in the phonebook!

6.24 Restart System

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

6.25 Support

Contact information.

7. Trouble Shooting

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 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 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	1. Unplug the power cord from the IP phone. Wait 2 seconds and plug the power cord back in the IP phone. Press and hold MENU key. The system should bypass boot up and go straight into phone setup menu. Modify the phone setting and make sure you save it before you exit.
TFTP firmware update method	1. Make sure you have a TFTP application installed on your PC. 2. Run Command Line and go to the directory which you have placed the firmware file. 3. Type in “tftp –i [IP of AlI7960] put [firmware file name]”. For example, tftp –i 193.169.80.125 put act_sip.gz 4. Firmware upgrading starts and it might take up to 5 minutes. 5. Firmware upgrading finishes after the automatic reboot.

8. CE Declaration

EC – Declaration of conformity for ALL7960

This equipment conforms with the requirements of the Council Directive 89/336/EEC on the approximation of the laws of the member states relating to Radio and Telecommunication Terminal Equipment and the mutual recognition of their conformity.

The safety advice in the documentation accompanying the products shall be obeyed. The conformity to the above directive is indicated by the CE sign on the device.
The ALLNET ALL7960 conforms to the **European Council Directives 89/336/EEC.** This equipment meets the following conformance standards:

EN 55022: 1998/A1: 2000/A2: 2003 Class B
EN 61000-3-2: 2000/A1: 2001, EN 61000-3-3: 1995/A1: 2001 and EN 55024: 1998/A1: 2001/A2: 2003 (IEC 61000-4-2: 1995/A2: 2000, IEC 61000-4-3: 2002, IEC 61000-4-4: 1995/A2:2001, IEC 61000-4-5: 1995/A1: 2000 , IEC61000-4-6: 1996/A1: 2000, IEC61000-4-8: 1993/A1: 2000, IEC 61000-4-11: 1994/A1:2000

This equipment is intended to be operated in all countries.
This declaration is made by

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and can be downloaded from http://www.allnet.de/ce-certificates/ .

9. GPL_Licensed Declaration

DISCLAIMER FOR ALLNET-PRODUCTS WHICH CONTAIN GPL-LICENSED SOFTWARE

The Allnet-product you have bought is using Linux software and parts of the netfilter/iptables-project. (www.netfilter.org) , which both are licensed using the GPL (General Public License, available for download at www.gnu.org/home.de.html).

You can download the complete source-code of this product at www.allnet.de in the download-area for this model.

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Attention:
Please be aware that any modifications of the firmware for this device will void the warranty. All modifications will be on your own risk and, in case of any damage to the device, Allnet can not repair or replace the device free of charge.