TT32L / TT32U

User Manual

V1.04

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Introduction

Voice over IP (also known as Internet Telephony) is a technology that allows anyone to make a telephone call over the internet environment. This is an operation manual for TraiTel's TT32 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. If you have purchased the model TT32L, then very little configuration is required as it already comes pre-configured for use in TraiTel's network.

Safety Declaration

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

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: TT32)

Ethernet Cable (3 metre)

Power Adaptor (9V DC)

1.2 Phone Specification

Protocol

- IETF SIP (RFC3261)

Network Interface

- RJ45 x 2, 10/100BaseT

LCD Display

- 2 x 16 characters

Key Pad

- 37 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)
- Call Park / Retrieve (RFC3515)
- Direct Station Select (DSS)
- Busy Lamp Field (BLF–RFC4235)
- Call Pickup (Support SIP server required)

Codec

- G.711µ-law (TT32U)
- G.711a-law (TT32U)
- G.729a/b

Phone Functions

- Multi-user (up to 4 SIP accounts)
- One touch dial (up to 11 records)
- Speakerphone communication
- Pre-dial before sending
- Handset / Speakerphone Volume adjustment
- Speed dial (10 records)
- Phone book (200 records)
- Multi-line (up to 12 lines)
- 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 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

IP Assignment

- Static IP
- DHCP
- PPPoE

NAT Traversal

- UPnP
- STUN
- Static port mapping

TCP/IP

- IP/TCP/UDP/DHCP/RTP/ ICMP/HTTP/SNTP/TFTP/DNS

Configuration

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

Firmware Upgrade

- TFTP
- Auto/Manual provisioning system

Power

TT32L

- Adapter
  Input AC 220-240V
  Output DC 9V

Environmental

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

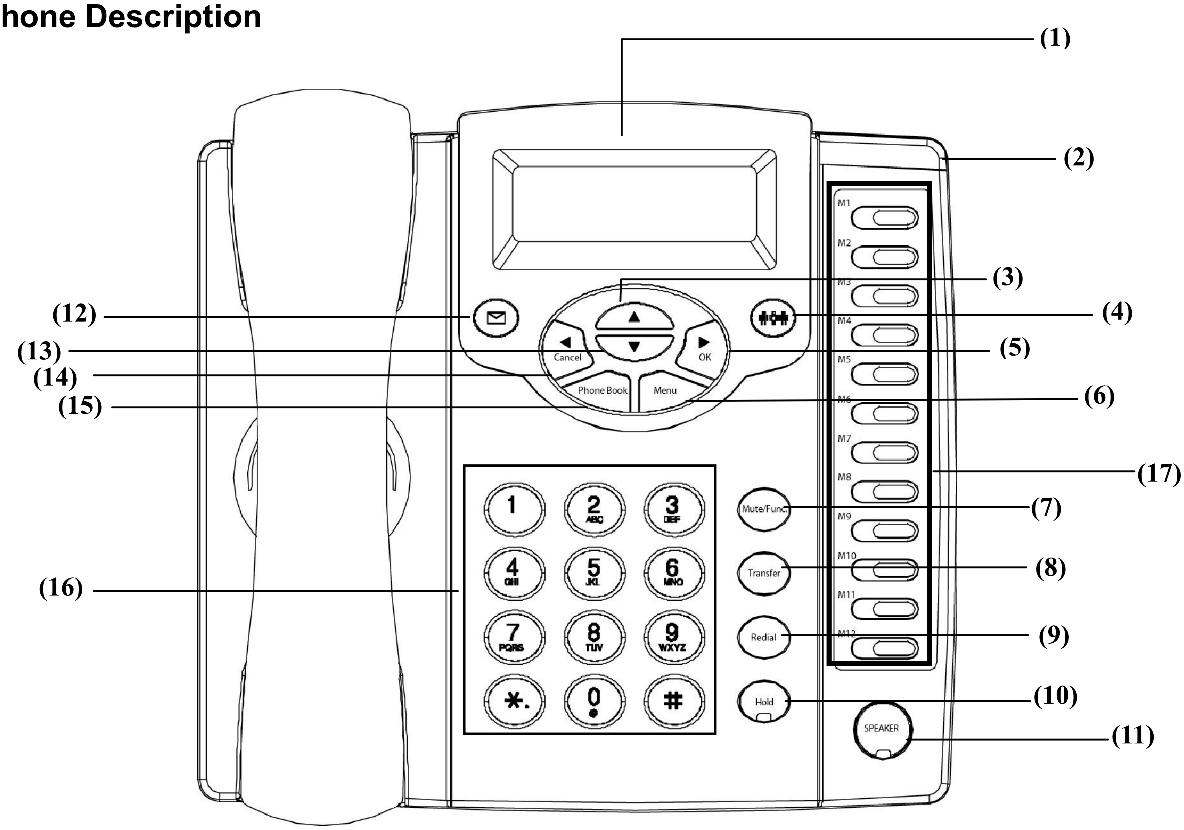
Physical Dimensions

- Size: 200(L) x 220(W) x 100(H) mm
- Wall Mount
- Weight: 860g
- Color: Dark Gray

Certification Compliance

- FCC Part 15 Class B
- CE Class B
- VCCI Class B
- EN60950-1

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
(17)	Local Multi-Line	Switch to different lines

1.4 Key Pad define & Text entry

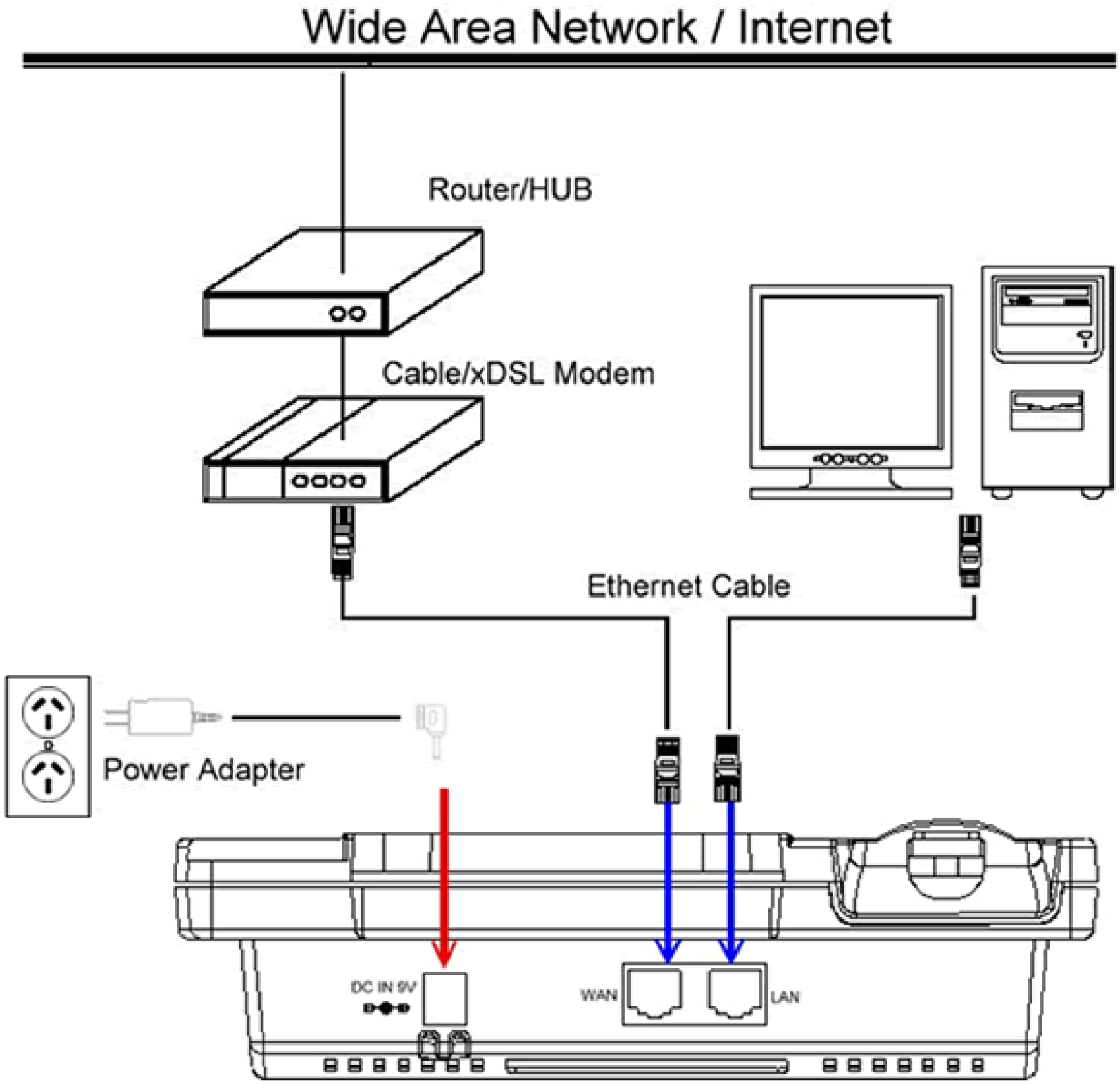
You use alphanumeric characters to enter details into the Phone Book, to create text and e-mail messages. The table below shows the characters that you can enter in the different text modes.

	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 which 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.

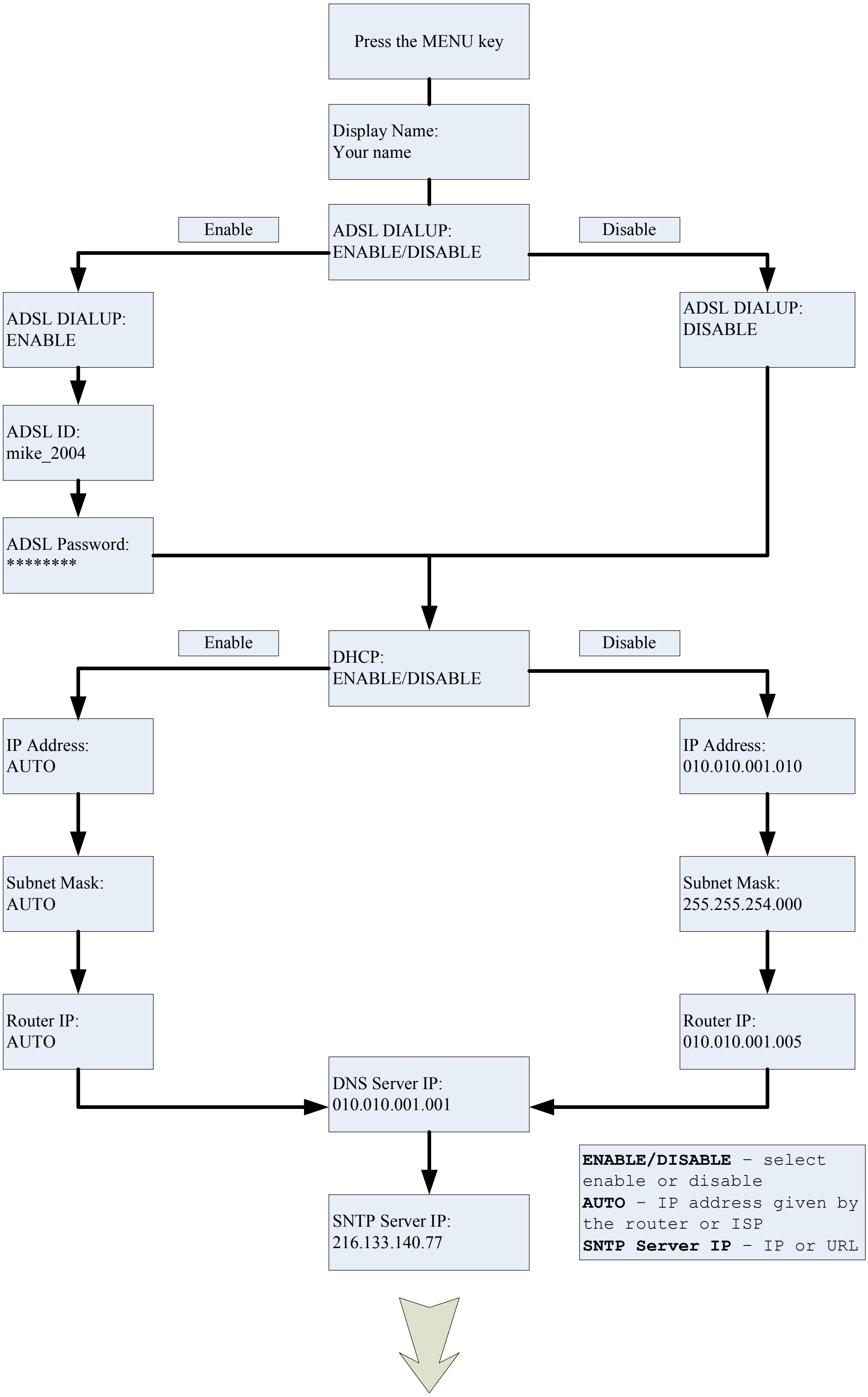
2. Connecting IP Phone

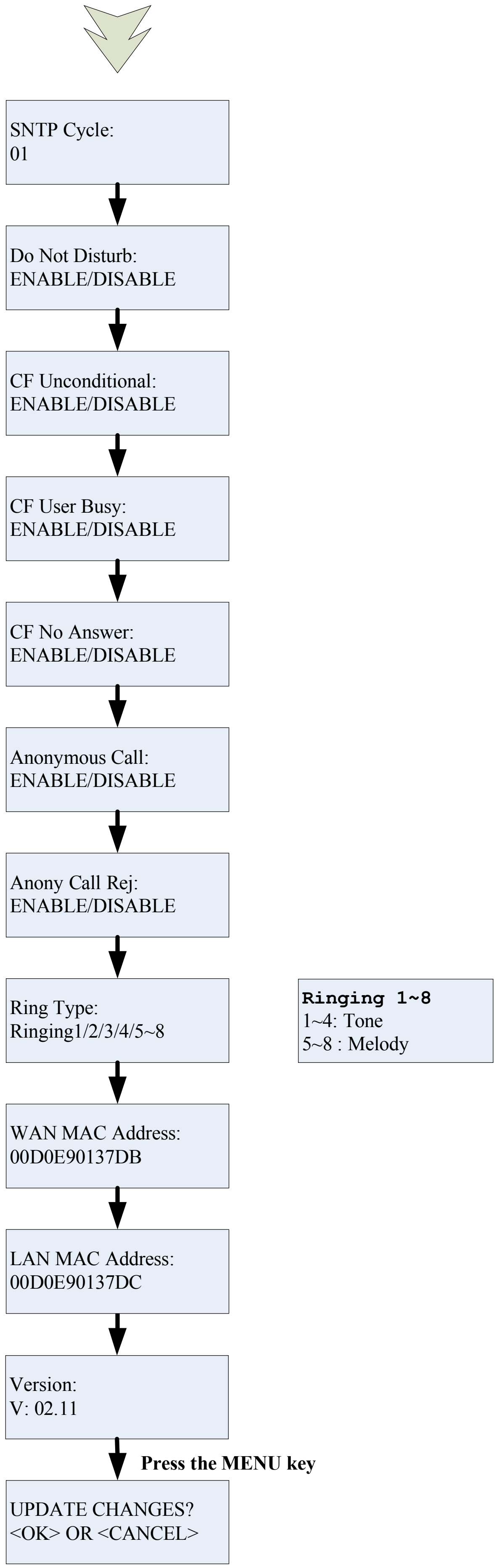
Connect the IP Phone as the following diagram:



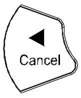
3. Setting up

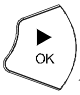
3.1 IP Phone Setup Map



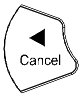

SNTP Cycle:
01
Do Not Disturb:
ENABLE/DISABLE
CF Unconditional:
ENABLE/DISABLE
CF User Busy:
ENABLE/DISABLE
CF No Answer:
ENABLE/DISABLE
Anonymous Call:
ENABLE/DISABLE
Anony Call Rej:
ENABLE/DISABLE
Ring Type:
Ringing1/2/3/4/5~8
Ringing 1~8
1~4: Tone
5~8 : Melody
WAN MAC Address:
00D0E90137DB
LAN MAC Address:
00D0E90137DC
Version:
V: 02.11
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 

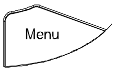
or 

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:
  192.168.001.161
```

- Press
- Subnet mask automatically acquired

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

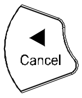
- Press
- Router IP automatically acquired

```
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

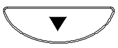
```
Subnet Mask:
 255.255.255.000
```

- Press 
- Enter the router IP address

```
Router IP:
192.168.001.001
```

3.5 DNS Server IP

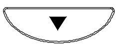
The domain name system (DNS) is the way that Internet domain names are located and translated into Internet Protocol addresses. There is probably a DNS server within close geographic proximity to your ISP that maps the domain names in your Internet requests or forwards them to other servers in the Internet.

- 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.

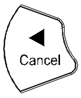
- Press 
- Enter SNTP server IP or URL

```
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 

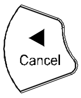
to select "Enable" or "Disable"

```
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 

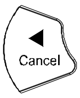
to select "Enable" or "Disable"

```
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 

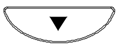
or 

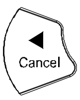
to select "Enable" or "Disable"

```
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"

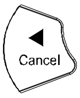
```
CF No Answer:
ENABLE / DISABLE
```

3.11 Anonymous Call

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

- Press 

- Use 

or 

o select "Enable" or "Disable"

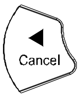
```
Anonymous Call:
ENABLE / DISABLE
```

3.12 Anony Call Rej. (Anonymous Call Rejection)

Reject any anonymous incoming calls.

- Press 

- Use 

or 

to select "Enable" or "Disable"

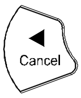
```
Anony Call Rej:
ENABLE / DISABLE
```

3.13 Ringing Type

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

- Press 

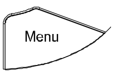
- Use 

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

```
WAN MAC Address:
000FC9017D4A
```

```
LAN MAC Address:
000FC9017D4B
```

3.15 Version

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

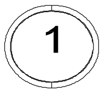
- Press 
- Firmware **version** is displayed on screen

```
Version:
V: 01.20
```

3.16 Language Selection

The VoIP Phone supports 2 languages: English and Japanese.

- Press 

followed by 

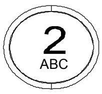

```
Language:
English
```

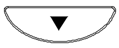
- Use  or  to select the preferred language

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

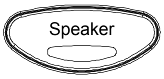
3.18.1 Ringer Volume

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 

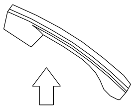
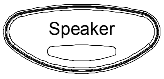
- Press  to increase the speaker volume and  to decrease the speaker volume

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 

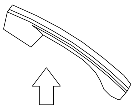
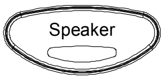
- Dial IP address.

For example: dialing 192.168.0.1

Press OK 

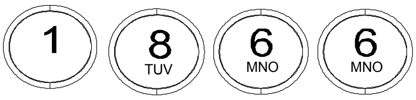
or wait until the timer expires to dial.

4.2 Dialing SIP Number

- Lift handset  or press SPEAKER button 

- Dial SIP Number

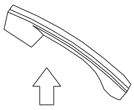
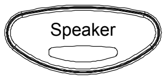
For example: dialing 1866



- Press OK 

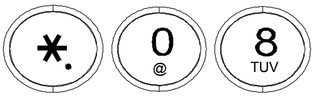
or wait until the timer expires.

4.3 Speed Dialing

- Lift handset  or press SPEAKER button 

- Dial Speed Dial number.

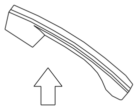
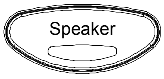
  For example: dialing speed dial number 08,

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 and the line key 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.

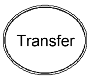
While having a conversation:

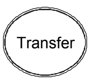
- Press Mute 

You may press Mute key again to resume conversation.

4.7 Call Transfer

While having a conversation:

- Press **Transfer** 

to put the person on the other line on hold.
- Dial the IP address, phone number 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 **TRANSFER** key. Dial the extension/phone number and press the **TRANSFER** key again to transfer the call.

While having a conversation:

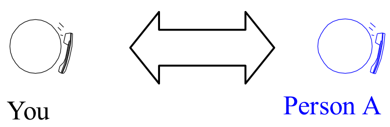
- Press **HOLD** (Press **HOLD** again to resume conversation)

4.10 Call Forward

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

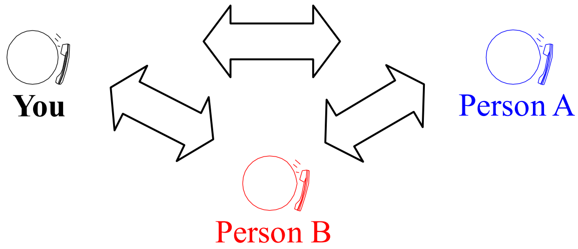
4.11 Three Way Conference

- Pick up the handset and call Person A.



- After Person A pick up the phone, press **Conference** 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 Phone Book to access the phone book.
- Press ▼ to scroll down the list until the name is displayed on the screen.
- Press **OK** ▶ OK to dial.

5.2 Storing a Number

- Press and hold the **PHONE BOOK** key Phone Book until “**Name:**” is displayed on the screen.
- Enter a name then press ▼
- Enter the number that corresponds to the name and press **OK** ▶ OK
- Press **OK** ▶ OK again to save the phonebook.
- Repeat above step to store another phone number.

5.3 Editing a Number

- Press the **PHONE BOOK** key Phone Book to access the phonebook.
- Press ▼ until the name is displayed on the screen.
- Press the **PHONE BOOK** key Phone Book again.

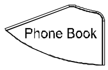
- Select “**Edit**” and press **OK**  to edit.

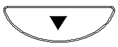
- Enter a new name and press **OK** 

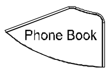
- Enter the new phone number and press **OK** 

- 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.

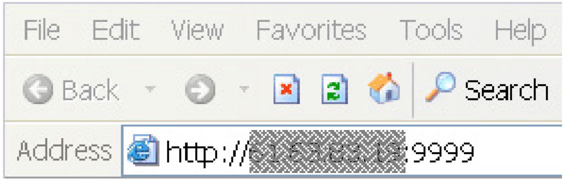
- Press **OK**  again to save the new list on the phonebook.

6 Using the web configuration

The configuration web can be accessed using a web browser.

6.1 Accessing Configuration Menu

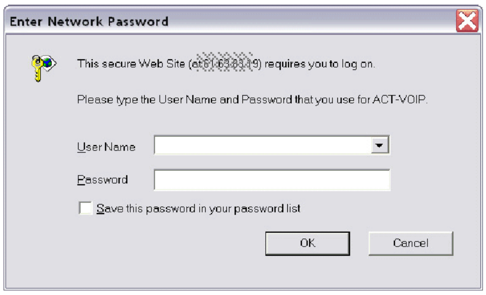
1. Open the web browser (ie. Internet Explorer, Netscape…)
2. Type in the **IP Address** of the phone followed by :9999



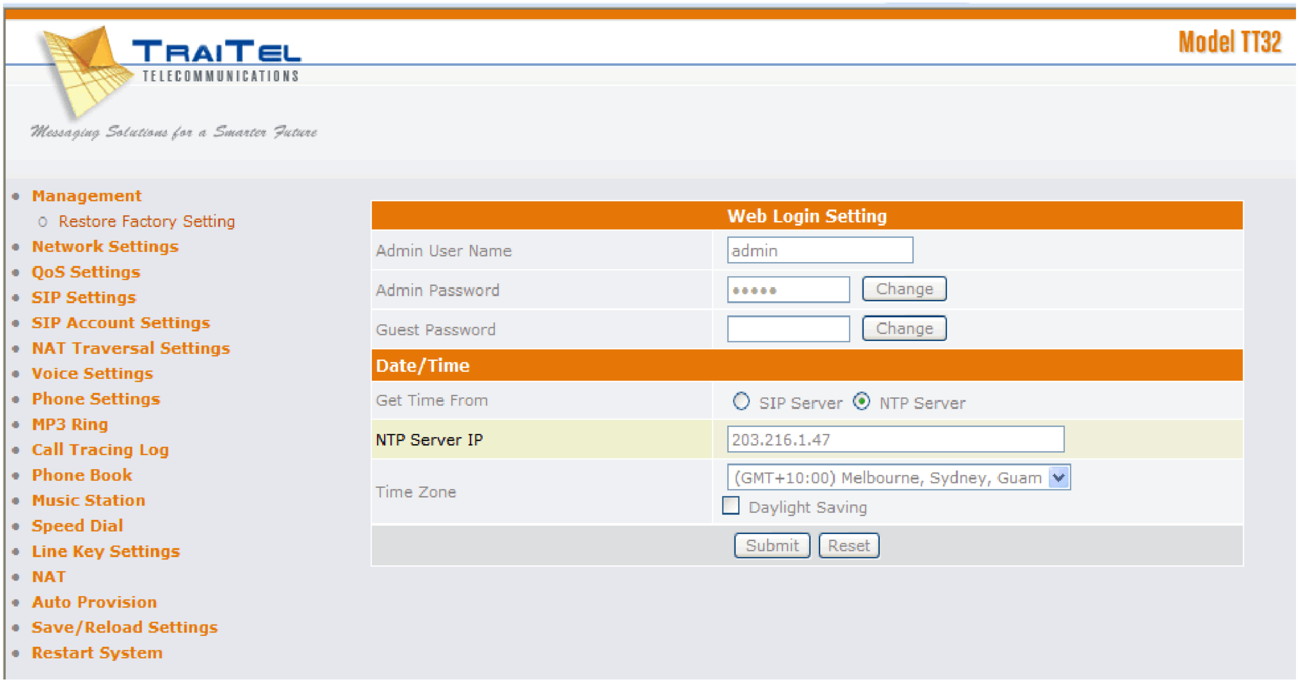
IP address is provided by your Internet Service Provider (ISP). If your ISP supports DHCP, you may obtain the IP address from you phone. Press “ Func.+ 9 ” to get IP address. Also can login from LAN port by http://192.168.15.1:9999 .

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 (e.g. atomic clock, time server). It is good practice to have all your networked computers synchronized with one server.

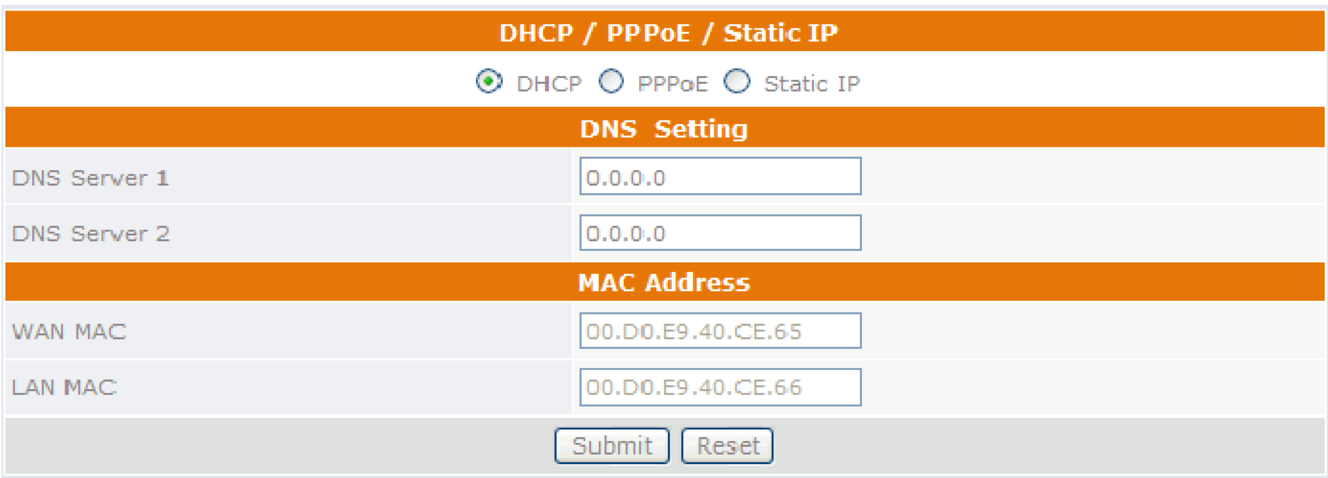
Time Zone Select your time zone. If there is daylight saving in your area, click the check box.

6.3 Management Setting – Restore Factory Setting

Click on "**Management**", Select "**Restore Factory Setting**" and the above screen will display on the screen.

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

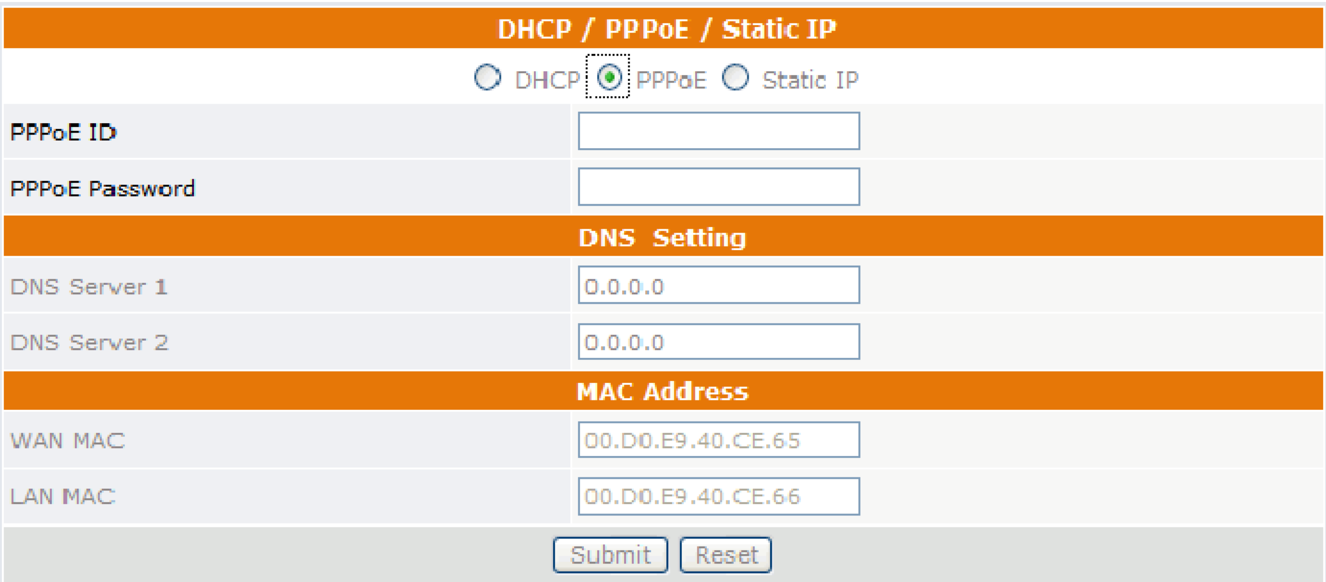
6.4 Network Setting – DHCP



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

DNS Server 1-2 DNS address provided by your ISP.

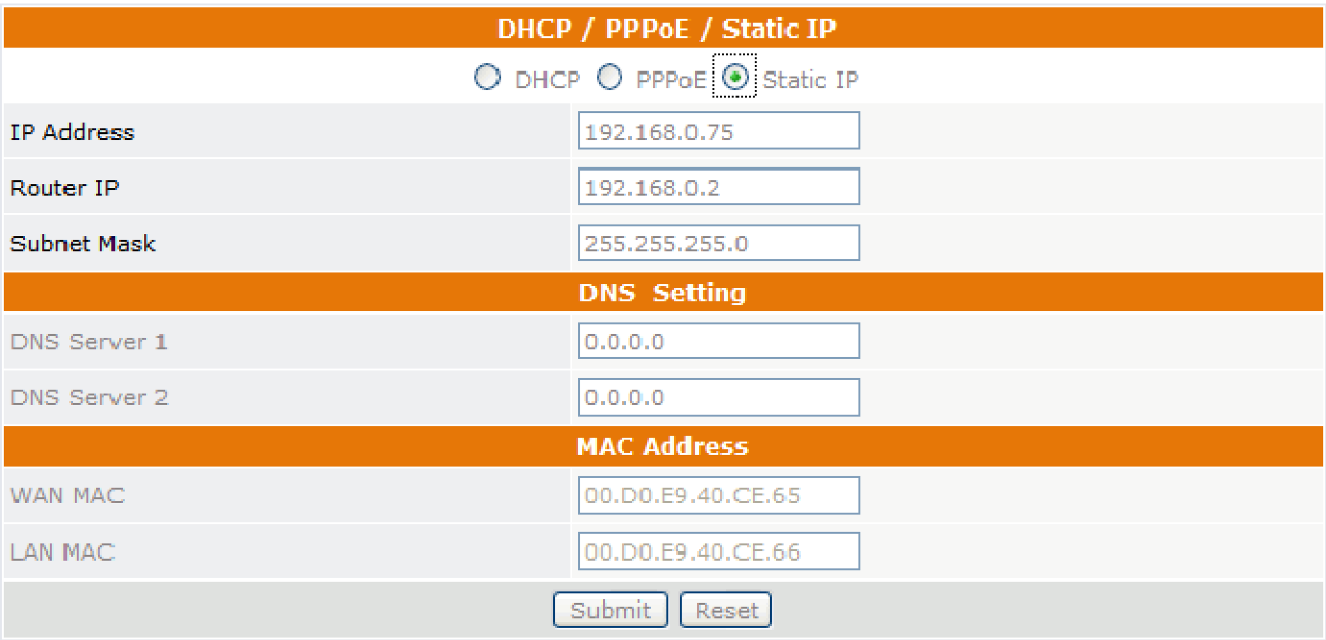
6.5 Network Setting – PPPoE



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 1-2	DNS address provided by your ISP.

6.6 Network Setting – Static IP

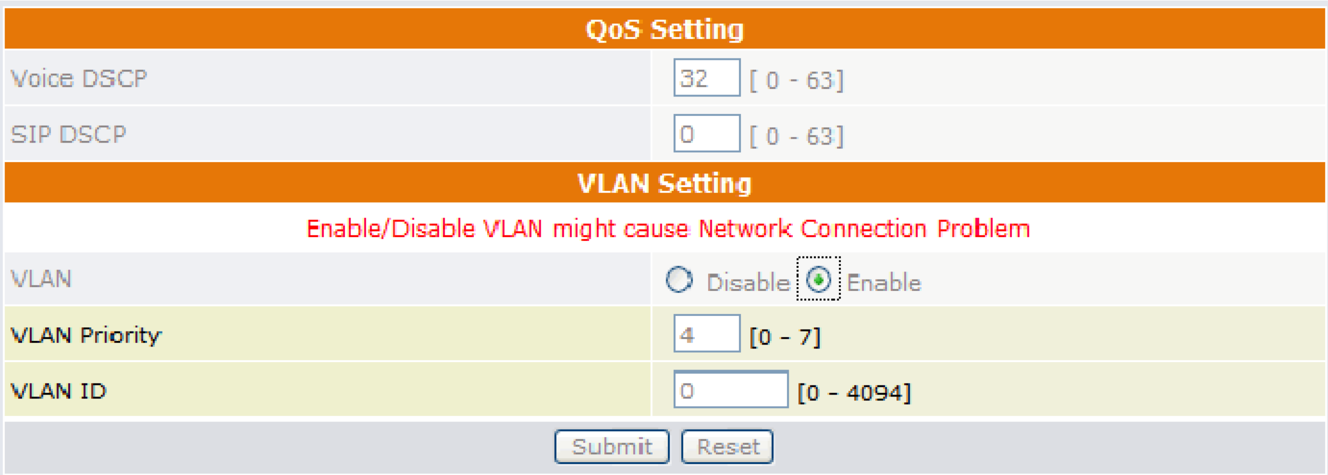


Choose Static IP network setting if all Wide Area Network IP is provided to you by your ISP.

IP Address	IP address assigned to you by your ISP.
Router IP	Router IP address.
Subnet Mask	Subnet mask.
DNS Server 1-2	DNS server address provided by your ISP.

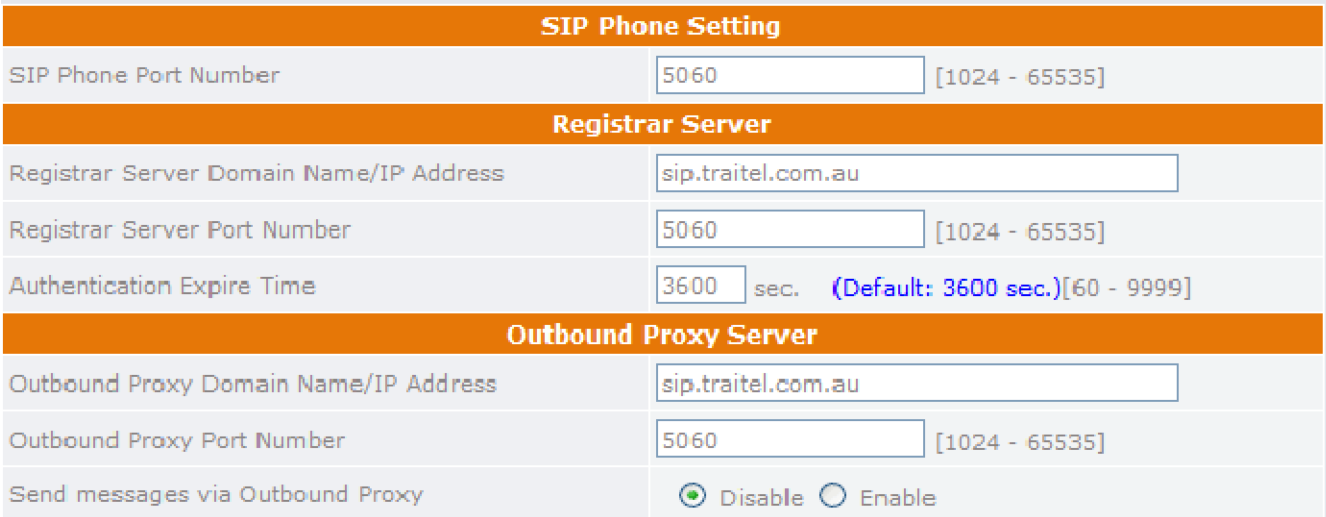
NOTE: RESTART the system for new settings to take effect after you modify the IP address.

6.7 QoS Setting



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	Support eight classes for prioritization on VLAN.
VLAN ID	The identification of VLAN.

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 listening port.
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 messages via Outbound Proxy	Select Enable to send all SIP requests through Outbound Proxy.

6.9 Message Server

Message Server	
MWI Message Server Domain Name/IP Address	
MWI Message Server Port Number	5060 [1024 - 65535]
MWI Message Subscribe Expire Time	3600 sec. (Default: 3600 sec.)[60 - 9999]
Voice Message Account	

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

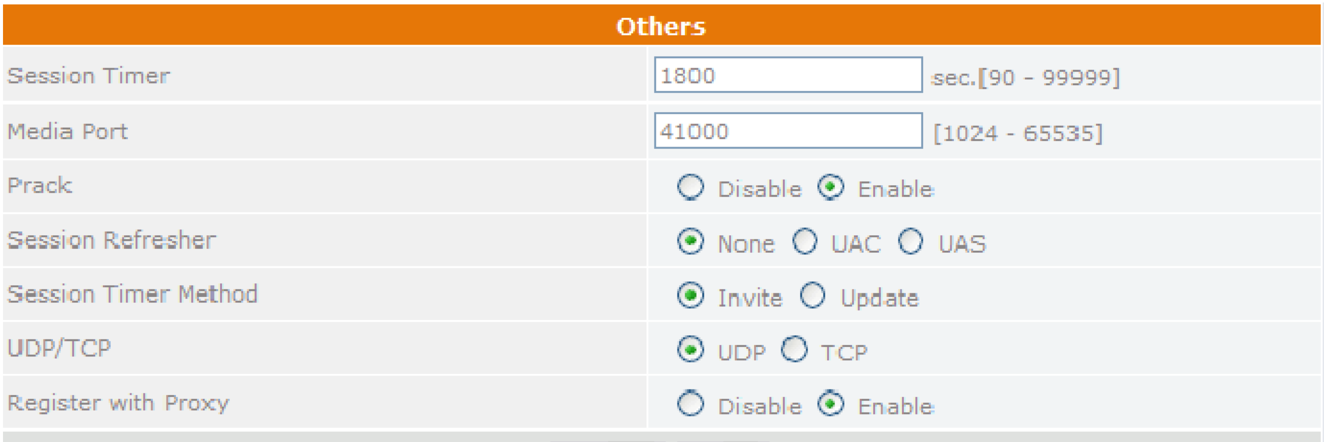
6.10 Park Server & Presence Server

Park Server	
Park Server Domain Name/IP Address	
Park Account	
Presence Server	
Presence Server Domain Name/IP Address	

Park Server Domain Name / IP Address	Park server host name or IP address.
Park Account	The number of the parking area on Park server
Presence Server Domain Name / IP Address	Presence server host name or IP address.

The settings which are described as above are corresponding to section 6.23 Line Key Settings.

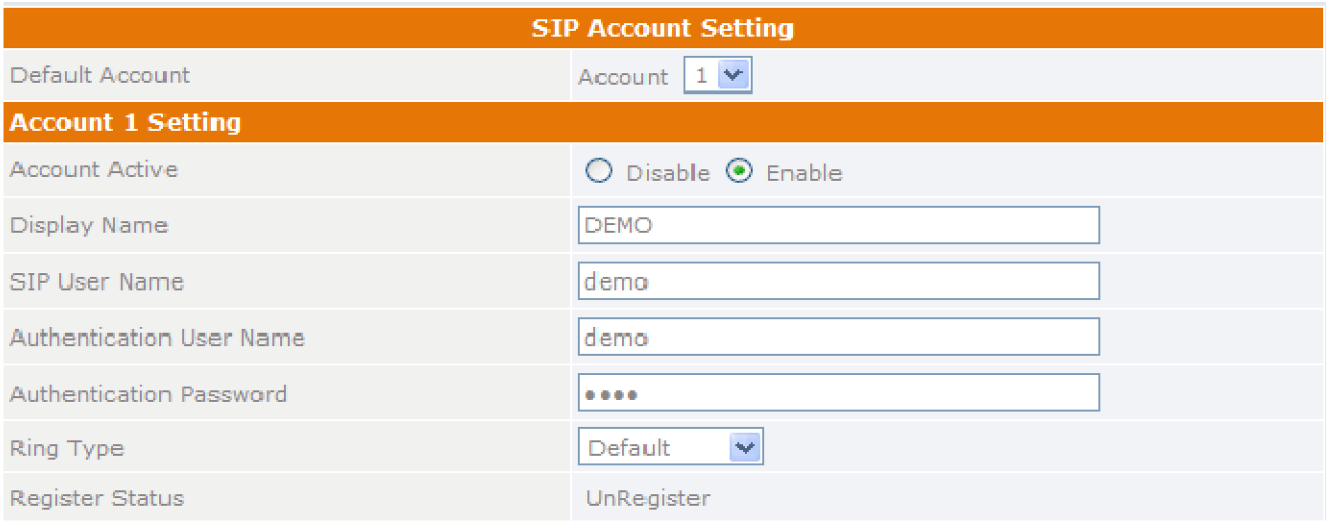
6.11 SIP Setting – Others



This section is for network administrators.

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	A SIP method which is applied to the condition of acknowledging to the 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 “Set messages via Outbound Proxy” is enabled, all the SIP requests including Register will be sent through Outbound Proxy. Enable the option will against the rule and send SIP Register directly to the Registrar as described in section 6.8.

6.12 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 LCD of called party.
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.
Ringer Type	Eight types of tone and melody can be selected for the specified account
Register Status	Displays if the current phone is registered or unregistered with SIP server.

6.13 NAT Traversal Settings – STUN Server Setting

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 Server Setting	
STUN	○ Disable ⦿ Enable
STUN Domain Name/IP Address	stun.traitel.com.au

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 Domain Name/IP Address Enter STUN domain name or IP address if STUN is enabled.

6.14 NAT Traversal Settings – Manual Config External IP/Port

Manual Config External IP/Port	
User Defined External IP/Port	⦿ Disable ○ Enable
External IP Address	○ Manual Set 0.0.0.0 ⦿ Use Stun get External IP Address ○ Use UPNP get External IP Address
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.

6.15 NAT Traversal Settings – UPnP Setting

UPnP Setting	
UPnP	◉ Disable ○ Enable

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

6.16 NAT Traversal Settings – NAT Keep Alive Time Settings

NAT KeepAlive Time Settings	
Always send keepalive packet	◉ Disable ○ Enable
KeepAlive Time	30 (Default: 30 sec.) [5 - 30]

Always send keepalive packet Enable or disable to keep the channel which is created for SIP signaling alive.

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

6.17 Voice Setting

Voice Setting	
Codec (Priority 1)	G.729A
Codec (Priority 2)	non-used
Codec (Priority 3)	non-used
RTP Packet Length	G.711 μ-Law 20ms G.711 A-Law 20ms G.729A 20ms
VAD	○ On ◉ Off
DTMF Method	◉ Out Band ○ In Band ○ SIP INFO
Payload Type	96 [96 - 127]

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

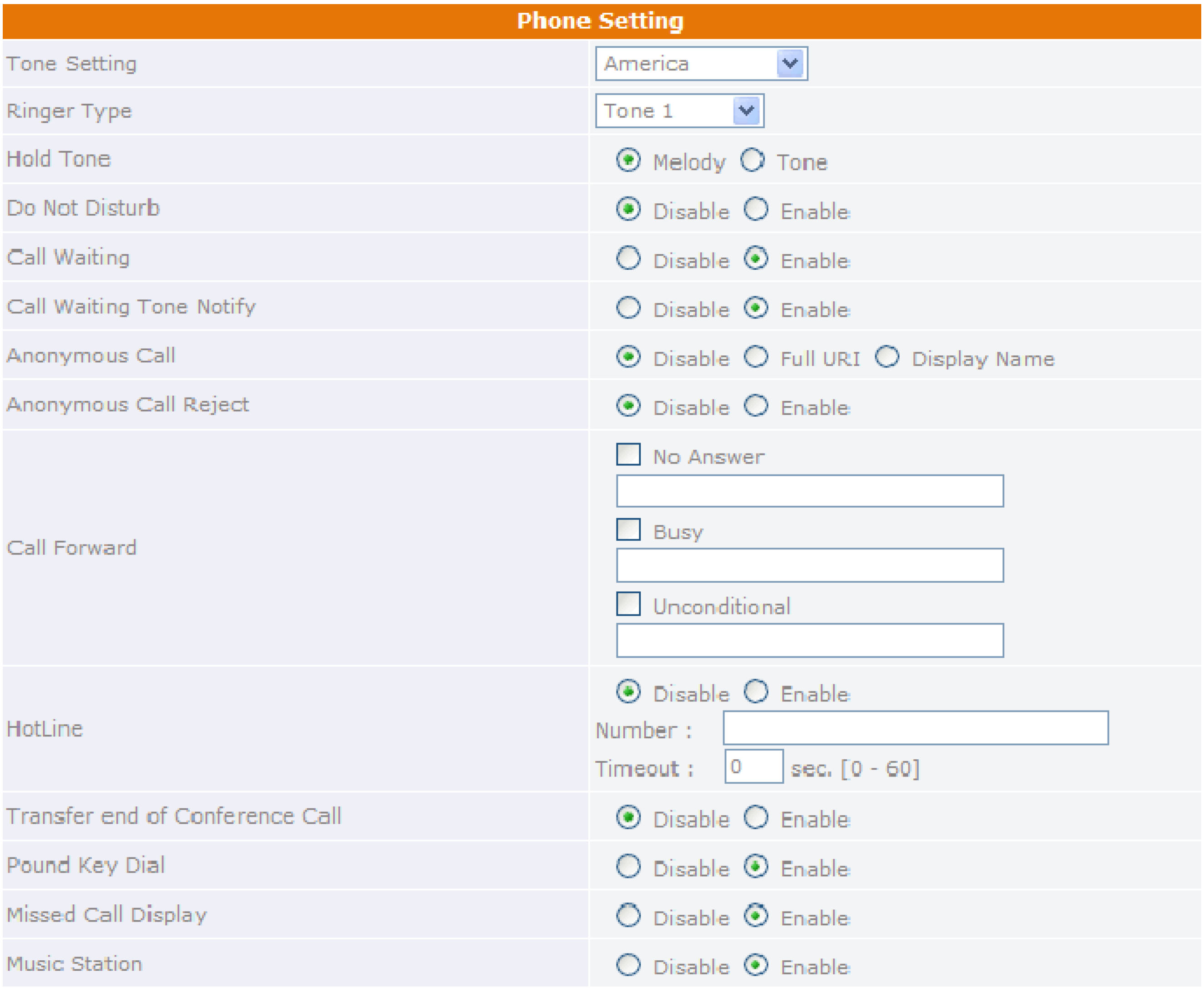
RTP Packet Length The payload size for each RTP packet.

VAD Support VAD 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 Setting the payload type for the Out Band DTMF (Default is 101).

6.18 Phone Setting



Tone Setting Select the tone for particular country

Ringer Type Select the type of ring (Tone 1 ~ 4 and Melody 5 ~ 8).

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.

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 display name is replaced by "Anonymous" when the user makes a phone call. It may display "Anonymous" or nothing on the receiver's phone.
4. Note: This function is disabled when the phone is used within TraiTel's network.

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 6.19 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.

Transfer end of Conference Call

Make a call transfer after the phone leave a Conference call. The phone must be the initiator of the conference call.

Pound Key Dial

Enable or disable Pound key Dial. Pound Key (#) can be defined as a <send> key.

Miss Call Display

Enable or disable to display miss calls on the LCD.

6.19 Phone Setting – Timer

Timer	
NTP Recycle Timer	1 hour [1 - 24] Network Time Adjustment Period
Inter Digit Timer	3 sec. [0 - 60] 0: Disable
Originating Not Accept Timer	180 sec. [0 - 600] 0: Disable
Incoming No Answer Timer	180 sec. [0 - 600] 0: Disable
Hold Recall Timer	180 sec. [0 - 600] 0: Disable
Auto Speaker Off Timer	30 sec. [0 - 600] 0: Disable

NTP Recycle Timer

The time interval that the IP phone synchronize with 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

The time interval that the caller's phone waits to establish a call. If the

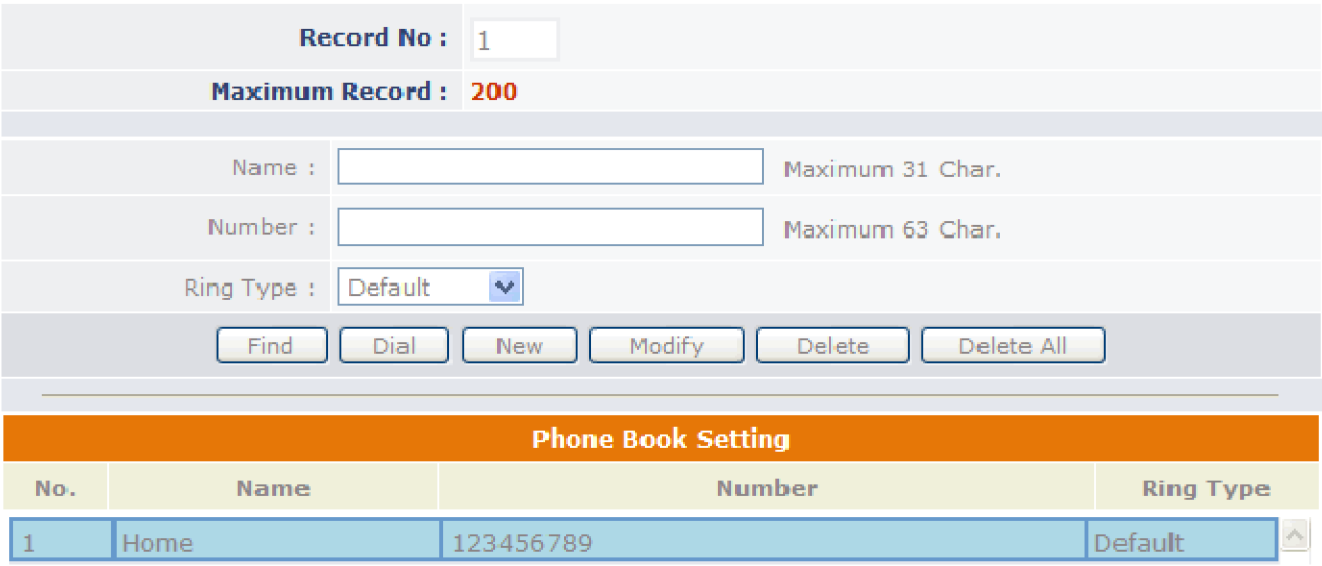
Timer	receiver fails to answer the phone during this time interval, the caller's phone will automatically disconnect.
Incoming No Answer Timer	The time interval that the receiver's phone will ring. If the receiver fails to answer the phone during this time interval, the phone will automatically disconnect.
Hold Recall Timer	The time interval that the call party which is put on held by the phone recalls.
Auto Speaker Off Timer	The time interval that the speaker phone is on before turning off automatically (due to inactivity).

6.20 Call Tracing Log

No.	Trace Log
000	stun_list_check: 192.168.0.75:41001 ---> 58.6.37.250:41001
001	stun_list_check: 192.168.0.75:41003 ---> 58.6.37.250:41003
002	stun_list_check: 192.168.0.75:5060 ---> 58.6.37.250:5060
003	stun_list_check: 192.168.0.75:41000 ---> 58.6.37.250:41000
004	stun_list_check: 192.168.0.75:41002 ---> 58.6.37.250:41002
005	stun_list_check: 192.168.0.75:41001 ---> 58.6.37.250:41001
006	stun_list_check: 192.168.0.75:41003 ---> 58.6.37.250:41003
007	stun_list_check: 192.168.0.75:3478 ---> 58.6.37.250:3478

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

6.21 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.

6.22 Speed Dial

Speed Dial Setting (Maximum 63 Char.)			
Number 00		Number 01	
Number 02		Number 03	
Number 04		Number 05	
Number 06		Number 07	
Number 08		Number 09	

Speed dial numbers can be accessed from the IP phone.

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

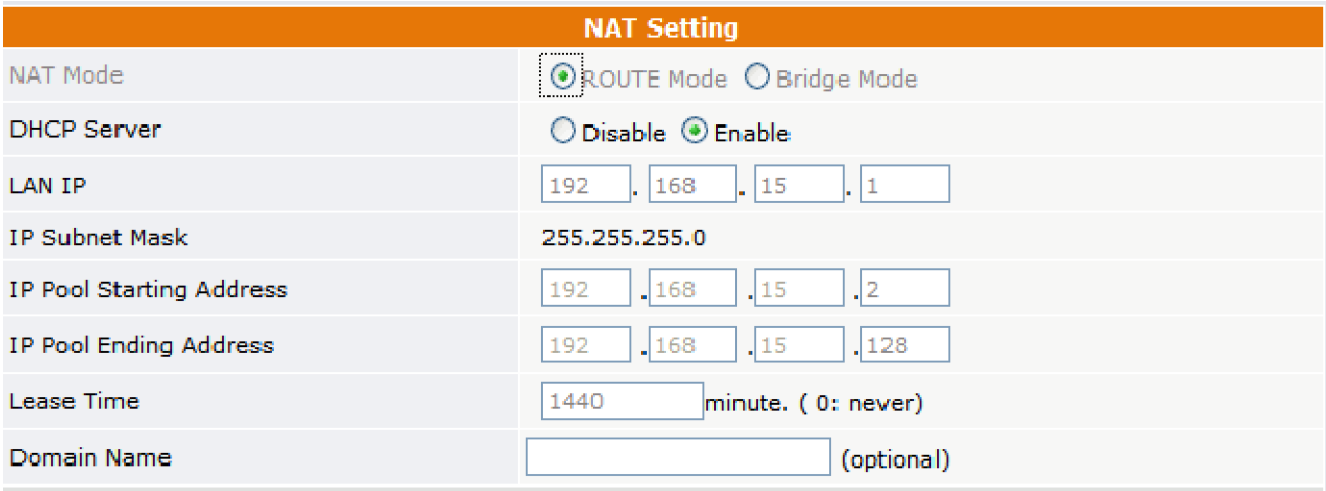
6.23 Line Key Settings

M2 Setting	
Type	◉ Line ○ Park ○ One Touch Dial ○ Extension
M3 Setting	
Type	◉ Line ○ Park ○ One Touch Dial ○ Extension
M4 Setting	
Type	◉ Line ○ Park ○ One Touch Dial ○ Extension
M5 Setting	
Type	◉ Line ○ Park ○ One Touch Dial ○ Extension

Type	Four types to the programmable keys can be selected. Default is "Line"
Park Number	The phone number of the parking area that is corresponding to "Park".
Phone Number	The phone number of the destination which can be called by one-touch-dial that is corresponding to "One Touch Dial".
Monitor Number	The phone number of the monitored extension that is corresponding to "Extension".

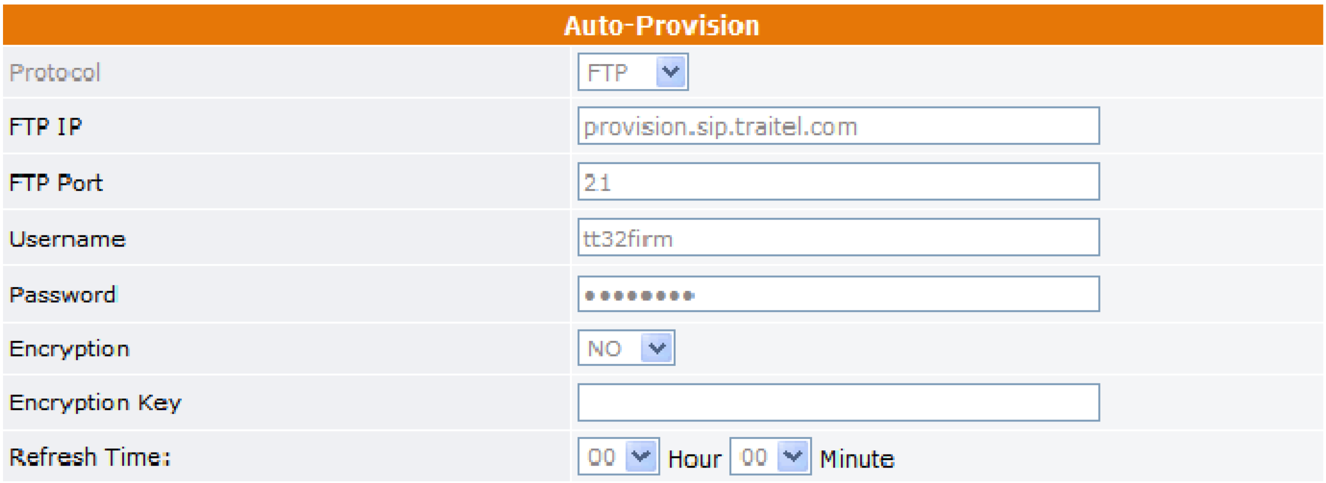
- Park: It is an advanced feature to park the active call in the parking area which is a special extension on Park server. The phones which have been assigned to monitor the parking area can retrieve calls if there are calls on parked. The Park server is generally co-located with SIP proxy.
- Extension: It is an advanced feature called "DSS/BLF". It watches the specified extension by receiving the notification of status from Presence server, which is generally co-located with SIP proxy and shows the status by LED indicator. The pre-configured key can be treated as the representative of the watched extension. It can be used to call the extension directly and pick up calls of the extension by pressing the key.

6.24 NAT



Select NAT mode to ROUTE Mode or Bridge Mode.

6.25 Auto Provision



Protocol Support FTP and HTTP for downloading firmware and configuration automatically. Default is NO to disable the function.

HTTP IP The IP address of HTTP server.

HTTP Port The port number of HTTP server. Default is TCP:80.

FTP IP The IP address of FTP server.

FTP Port The port number of FTP server. Default is TCP:21.

Username The username required by the auto provision system for authorization.

Password The password required by the auto provision system for authorization.

Refresh Interval (hr) The time interval to connect with the auto provision system for checking the update. Default is 168 hrs (7 days).

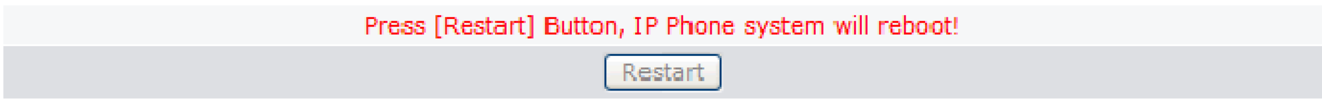
OS Release Version The current version of the operating system that has installed on the IP phone.

AP Release Version The current version of the SIP application that has installed on the IP phone.

DATA Release Version The current version of the package which includes the required files except OS and AP that have installed on the IP phone.

Note: For detail of the application, please refer to the specification of Auto Provision.

6.26 Restart System



Click **Restart** to reboot the system.

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.
How to update Firmware?	1. IP Phone automatically updates firmware when it powers up (while connected to the internet) if auto-provisioning is available.
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.
Why isn't my firmware updating?	1. Your IP phone automatically detects for new firmware when you unplug the power. If a new version is available the phone will automatically update the firmware.

	2. Check if FTP address is correct. 3. Check with your supplier if firmware filename is correct.
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.
Why do I get "Can't Upgrade Now" screen when I click [Submit] in the configuration menu?	1. Make sure you exit setting mode (phonebook, menu, speed dial…) before you click [Submit] in the configuration menu.

Appendix A: Wall Mount Installation

This appendix herein illustrates the installation step by step if you would like to mount the TT32L/TT32U on the wall. Please print 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.

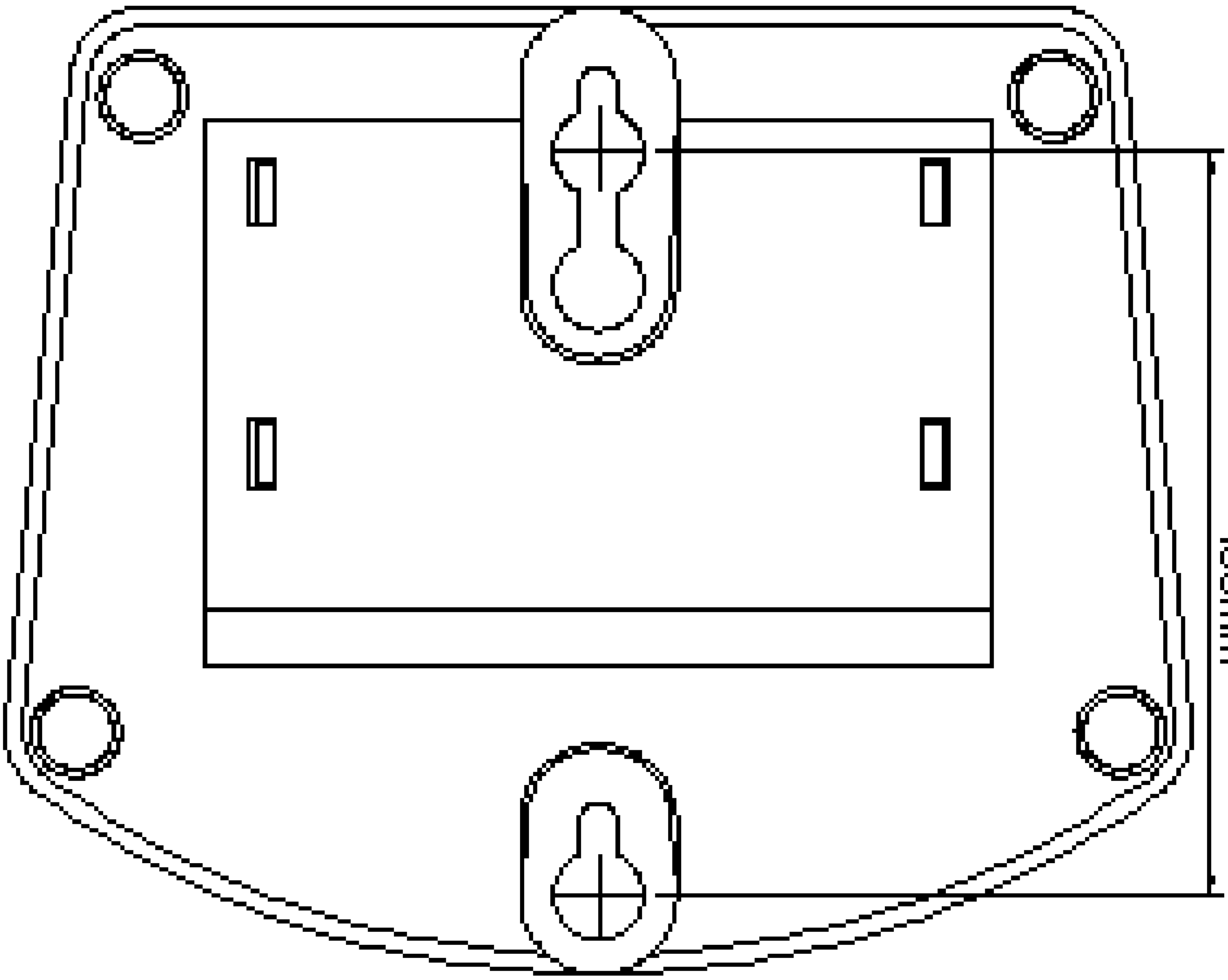


Figure A1

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

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

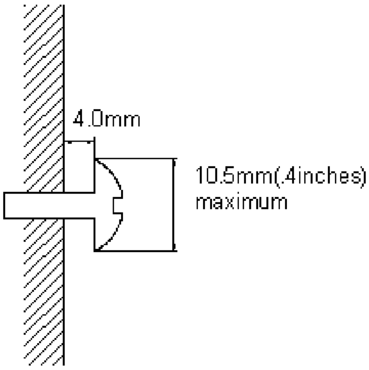


Figure A2

3. Place the mount on the wall as 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 entity of TT32L/TT32U on the wall mount as Figure A4.

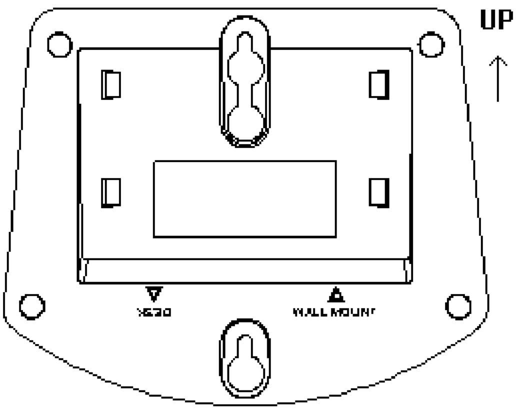


Figure A3

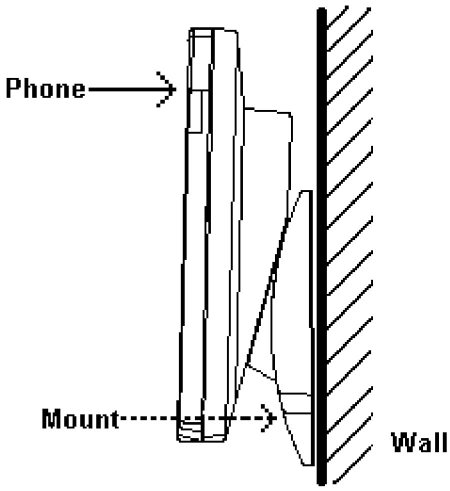


Figure A4