Juphoon IP Phone

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uContact MP100 IP Phone User Manual

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Abbreviation	Description
DND	Do Not Disturb
DTMF	Dual-tone multi-frequency
PSTN	Public Switched Telephone Network
QoS	Quality of Service
SIP	Session Initiation Protocol
STUN	Simple Traversal of UDP through NATs

Table 1-1 Abbreviation

2. Version

The version of the IP Phone described in this document is uContact MP100, and the phone software version is MP100.P3.01.The document was last updated on January 12, 2010.

3. About MP100

Thank you for choosing uContact MP100! This multimedia IP Phone is a terminal which aims the business environment works with IMS server or any other standard SIP servers. With the IM and presence service, it can watch your buddies' status and send instant message to buddies as well; just acts as a desktop unify communication device.

4. Introduction

As shown in Figure, the IP Phone's dimension is 110 mm x 220 mm x 185 mm, the net weight is 696g and total weight is 1017g. Its' operating interface includes a handset, a headset, a 64*128 dot LCD, buttons, and a hands-free MIC.

Figure 4-1 IP Phone

4.1 Keys

There are three types of buttons: main keys, service keys, Menu and soft keys. Details about these keys are in the following table.

Type	Key	Description
	Num Keys	1. Input numbers or letters. 2. Send DTMF signals. 3. Input star (*) as dot (.) or colon (:). 4. Input pound (#) as confirm dial out.
	Speaker	1. In hands-free mode, receive or start a voice call. 2. Switch between hands-free and handset mode. 3. In hands-free mode, end a call. Note: The voice quality in hands-free mode may not be as good as in handset mode in some environment.
	Headset	1. In headset mode, receive or start a voice call. 2. Switch between headset and handset mode. 3. In headset mode, end a call.
	Mute	1. Mute a voice call. 2. Mute the ringing as the phone is idle.
Service Keys	Transfer	1. Transfer the voice call to a 3rd party. 2. Including blind transfer or attend transfer.
	Conference	1. Add a 3rd party to form a conference. 2. Change Talk & Hold mode to a conference. 3. Accept an incoming call to form a conference. 4.invite other party into a conference
	Delete	Delete a character backward.
	Character selection	Switch input method between digit, lowercase and uppercase. Note: Input method depends on different cases.
	Vol+ / -	Adjust the volumes of ringing, handset, headset and hands-free speaker in different scenario.
	Voice Mail	Speed dial for the voice mail number.
	Voice Record	1. Record a call. 2. Play the record. Note: Time length is limited to 60s.
	Hold	1. Put a call on hold or un-hold. 2. Switch between talking and held calls. 3. Extension usage with conference or transfer.

Type	Key	Description
	IM	Open the instant message list page to compose, view, send, forward and delete the IM.
	My status	1. Change my status that will be watched by your subscribed buddies. 2. See the IP Phone status information by shortcut like IP address, current version and etc.
Menu & Softkeys	Menu	1. Enter into menu. 2. Exit from menu.
	Upper	Move focus upper on LCD.
	Down	Move focus down on LCD.
	OK	Confirm your operation or enters into the deeper menu level.
	Cancel	Cancel your operation or go to the up level in menu.
	Softkeys	The softkeys will act as different functions as the different scenario shows prompt on LCD.

Table 4-1 Button Description

4.2 Hardware

The IP Phone hardware platform is based on a powerful 400MHz CPU, that enable the phone provide rich features including IM, presence and multi-language support.
Here are the phone hardware I/O features:

1. 35 keys.
2. 3 ports on the top of IP Phone:
   a) DC 5V port connecting with an AC-DC (Input 100-250VAC 50/60Hz, output 5VDC 1.2A) power adaptor.
   b) WAN port connecting with WAN or LAN by RJ45 cable – this is the main port to access the network.
   c) PC port connecting with PC by RJ45 cable.
3. High resolution 64 x 128 dot matrix LCD with 4 soft keys.
4. Support headset.
5. Built in loudspeaker and supports full duplex hands free talking.

4.3 Software

Call Control

- Basic Call (Voice)
- Caller ID Display
- Redial, Speed Dial, Hotline
- Call From History & Address Book
- Call Hold
- Call Transfer (Blind & Attend)
- Call Waiting
- Call Forward (On Busy / No Answer, Unconditional)
- Conference
- Automatic Accept Call / Do Not Disturb

Other Functions

- Mute, Volume Adjust & Ring Tone Profile Management

- Presence
- Instant Message
- DTMF (Inbound / Outbound / SIP-Info)
- MWI for Voice Mail
- Call History (Missed, Received & Dialed)
- Phone Book (LCD Menu & Web)
- Voice Recording & Voice Mail
- Dial Plan
- Speed Dial
- Multi-Account Management
- Alarm(Once & Repeat)
- SIP Stack and Call Control Logging
- Syslog, Telnet, Ping & Trace route Function

Voice Engine

- Full Duplex Hands-free Talking with Acoustic Echo Cancellation(AEC), Voice Activation Detection(VAD) & Comfort Noise Generation(CNG)
- Volume Adjustment for Ringing, Speaker, Handset & Headset Individually
- Speaker and Headset Mode
- Support for G.711u/a, G.729ab & G.723.1a Codec
- Adaptive Jitter Buffers and Packet Loss Concealment
- Voice Activity Detection and Comfort Noise

Protocol Features

- SIP RFC3261 and its' RFC Extensions
- DNS (A, SRV, NAPTR)
- STUN
- SNMP v2 or TR069
- HTTP, TFTP

Network

- Manual Configuration / DHCP / PPPoE
- Time and Date Synchronization Using NTP / Register
- XML-Based Auto-Provision and Auto-Update
- Support for IEEE802.1p/Q tagging (VLAN), Layer 3 TOS
- Works on Router / Bridge Mode
- NAT Keep Alive
- 3 layers DSCP

4.4 Environment Requirements

Operating temperatures: -10 to 40°C
Storage temperatures: -10 to 40°C
Humidity: 10% to 85%, non-condensing
Storage: The packaged IP Phone shall be placed in dry and airiness environment, and the max number of packages stack is 10.

4.5 Installation

Please check to be sure that you have all the following components before installation:

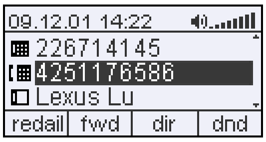
- An AC-DC power adapter
- A RJ45 cable
- An IP environment provided by the phone service provider, which can access to LAN
- A PC connecting to the Internet to perform Web management over the phone.

If the two IP Phones linked to the network are not to set register on, it can be used for peer to peer calls by IP address. If you want to make phone calls through the network by phone number you will have to make sure that you have been given the account and password of the phone by the service provider and the IP Phone is registered.

5. Basic Phone Usage

Basic phone usage includes placing, receiving and forwarding calls. The following sections describe how to perform these basic tasks on your IP Phone.

5.1 Select Registered Account



The IP Phone support multi register totally up to 6! As you see the LCD shows the multi accounts while the phone is in idle state. Select the active account you need to dial out by press “Upper” or “Down” key, besides the selected account it displays the figure . If the account is not set on, it will not list on the LCD, and if the account is not registered on any SIP server, it will show local IP address. The selected account only chose which account is used to *call out*; all other registered accounts can be *called in* at all the time.

NOTE:	The call number for transferring and conferencing is only available on the same selected account registered SIP server

5.2 Making Calls

5.2.1 Placing a Call

The phone supports making calls peer-to-peer or in the domain.

Peer-to-peer: The phone didn’t register on any server. And you should input the IP address as dial number.
Domain: The phone has registered on a server. And you should input the peer’s number like phone number, SIP URI.

NOTE:	See the “Advanced” for how to set the phone to register on a server.

Want to …	Then …
use the handset	Pick up the handset and dial the number.
use the speaker	Press “Speaker” and dial the number.
use the headset	Draw on the headset and press “Headset”, and dial the number.
dial on hook	Press “Num Keys” directly, and pick up the handset.
redial the recently dialed number	Press soft key “redial”. See the “Redialing and Call History” for detail.
dial the number from call history *	See the “Redialing and Call History” for detail.
dial from Address Book *	See the “Address Book” for detail.

use Speed Dial	Press one preset "Num Key" (0 – 9), and press soft key "call". See the "Speed Dial" for detail.
use Hotline *	Pick up the handset or press "Speaker" or press headset; it will call out directly for preset hotline number. See the "Hotline" for detail.
dial a 2nd call	Press "Hold" to hold the current dialog, after hold successfully, press "Num Keys" to dial the 2nd call number and press soft key "call".

Table 5-1 Placing Calls

NOTE: You cannot edit the number before dial out in the cases marked with *. In the rest cases, you can press "OK' after editing, or to be matched by the dial plan to dial out immediately.

5.2.2 Answering a Call

Want to …	Then …
use the handset	Pick up the handset.
use the speaker	Press "Speaker".
use the headset	Draw on the headset and press "Headset".
use the Auto Answer	The phone will automatically answer the call when it receives call. See the "Incoming Call" for how to enable Auto Answer.
reject the call	Press "Cancel" or soft key "reject".
answer the 2nd call	Press "OK" or soft key "ok" to answering the new incoming call. Then the phone will switch the conversation to the new call and put the old one on hold.
reject the 2nd call	Press "Cancel" or soft key "reject" to reject the new call. The old one will remain unchanged.
forward the call	See the Call Forward Directly Setting and Setting Call Forward for detail.

Table 5-2 Answering a Call

5.2.3 Ending a Call

When…	Then …
using the handset	Hang up the handset, or use the soft key "end".
using the speaker	Press "Speaker" again, or use the soft key "end".
using the headset	Press "Headset" again, or use the soft key "end".
there are 2 calls and using the handset	Press "Cancel" or soft key "end" to end the current call. Hang up the handset to end all calls.
there are 2 calls and using the speaker	Press "Cancel" or soft key "end" to end the current call. Press "Speaker" to end all calls.

there are 2 calls and using the headset	Press “Cancel” or soft key “end” to end the current call. Press “Headset” to end all calls.

Table 5-3 Ending a Call

5.2.4 Muting a Call

You can mute the handset, headset or speaker during a call. The mute feature disables the phone’s microphone. It prevents the other party from hearing you, but does not interfere with your ability to hear them.

To mute a call, press the “Mute”, and the phone will display on LCD. To disengage mute, press “Mute” again.

5.2.5 Holding a Call

The hold feature prevents the exchange of the voice data. So you and the other party cannot hear from each other. During some cases, the other party will hear the music sent by the server.

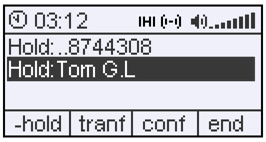
When…	Then …
you want to hold a call	Press “Hold” or soft key “hold”.
you want to resume a call on hold	Press “Hold” again or soft key “-hold”.
there are 2 held calls and you want to talk to one	Select one line and press “Hold” or soft key “-hold”. The phone will un-hold the current call and other lines will be held automatically whether it’s held or talking. It will display like following. 

Table 5-4 Holding a Call

NOTE: As more lines are held, it cannot judge who will be transfer, while it can still mix to conf - the soft key "tranf" is hidden. Or in the case it exceed the max number of conference, the conf soft key is hidden too. And there is only one line can be in talk state, un-hold current line will lead other lines be held automatically

5.2.6 Transferring a Call

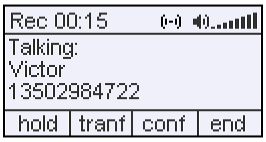
Want to…	Then …
do blind transfer a call	Press “Transfer” or soft key “tranf” and dial the 2^{nd} part number for transferring to.
do attending transfer a call	Press “Hold” or soft key “hold” and dial the 2^{nd} number, after talk with this party of this number, press “Transfer” or soft key “tranf” to transfer from 1^{st} number to 2^{nd} number.
make transfer between	Press “Transfer” or soft key “tranf” to let these 2 parties be

current 2 calls	connected.

Table 5-5 Transferring a Call

5.2.7 Recording a Call

You can record the voice during the call. To record the call, press “Record” during the call. To stop recording, press “Record” again. The phone will display:



The maximum length of recording is about 60 seconds. It will automatically stop when flash space is full or ending the call.

To play the voice record, press “Record” as the phone is in idle, press soft key “play” to play and press soft key “stop” to stop playing.

NOTE:	The new call record will overwrite the old one

5.2.8 Redialing

To redial the most recently dialed number, press soft key “redial”, as the LCD display the call history list, press “dial” again to dial out immediately.

You can also move to the 2nd, 3rd last dialed number by “Upper”, “Down”, press “dial” again to dial out immediately.

NOTE:	You may not redial successfully if you change current active phone account, because that may be another SIP server domain and last dial number is not a valid number

5.3 Call History

Want to…	**Then …**
view call history quickly	Simply press soft key “redial” as the phone in idle, LCD will display dialed call history by time line from last item to the oldest one.
view call history classified by missed / Received / Dialed	Entry the menu by pressing “Menu”, select the item “Call History,” it shows the three classified call history: 1 Missed Call 2 Received Call 3 Dialed Call

Table 5-6 Dial Out from Call History

NOTE: The maximum record number of each class is 75.

5.4 Speed Dial

There are 10 keys (the “Num Keys” 0-9) can used for speed dial. You can assign phone numbers to the speed dial keys by pre-setting.

To dial by the assigned speed dial key, press a “Num Key”, and press “OK” or soft key “call”, the phone will translate this key to a pre-set number.

5.5 Making Conference Calls

This IP Phone support two conference mode:

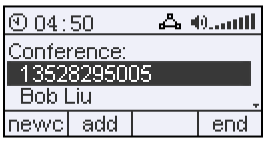
1. Local conference mixing.

2. Conference handled by conference server.

You may set the conference mode by LCD or Web management. If you set the conference mode by *conference server*, you need to set the conference factory URI, which will assign conference resource when you request to build a conference.

5.5.1 Placing a Local Conference

<table>
<tr><th>When…</th><th>Then …</th></tr>
<tr><td>you are in a call and want to invite the 3rd party to build conference directly</td><td>Press “Conf” or soft key “conf” and dial the 3rd party number.</td></tr>
<tr><td>you are in a call and want to build a conference after consult the 3rd party</td><td>Press “Hold” or soft key “hold” and dial the number. After consulting with the 3rd party, press “Conf” or soft key “conf” to build the conference.</td></tr>
<tr><td>during a call and receive the 3rd party call</td><td>Press soft key “ok” to accept a new call, as you connect to this party, press soft key “conf” to build a three way conference.</td></tr>
<tr><td>Invite other party into a conference</td><td>1. Press the soft key “Add” while you are making a conference, and dial the number, the invited party will become a number of the conference as soon as accept.


2. Press the soft key “newc” while you are making a conference, and dial the number. Press the soft key “conf” after the invited party accept.


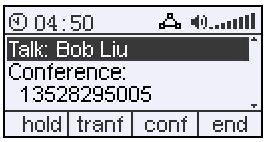

</td></tr>
</table>

Table 5-7 Placing a Local Conference

5.5.2 Placing a Conference by Conference Server

Want to …	Then …
build a conference as you connect with 2 or more participant	Press “Conf” or soft key “conf” to build the conference.
add a 3rd party into the current conference	Press soft key “newc”, and dial the 3rd party number to build a new connect, press “Conf” or soft key “join”.

Table 5-8 Handle a Conference by Conference Server

5.6 Voice Mail

The voice mail system was managed by the service provider (SP). Please ask the phone administrator or SP for the voice mail server number (Voice mail server URI). As you press key “Voice mail” as the phone has not been set the mail number, it will display message to lead you to set the server number, or you can change the server number at LCD menu or on the web management page.

When you receive one or more new voice mail messages, the phone will display following cue .

NOTE:	If there are new messages come in, including voice mail, instant message, the LCD will display the cue . The different is the hard key “Voice mail” LED will twinkle as the message is voice mail, while the instant message will be not. To access voice mail, you can dial out to pre-set voice mail server number by the key “Voice mail”

5.7 Shortcut Settings

5.7.1 Volume

You can use “Vol + / -” to change the volume in following scenarios:

- The phone is idle, set the ringing volume.
- During a call using handset, set the handset volume.
- During a call using speaker, set the louder speaker volume.
- During a call using headset, set the headset volume

The volume setting in these scenarios are separated, you may notice the cue on the LCD will change in different scenarios.

NOTE:	It cannot set handset / headset / speaker microphone by volume keys, because 2 talking sides change the volume of microphone and speaker will be confused. While the phone administrator can set the concrete volume value mapped to 1-9 volume level on web management page

5.7.2 Do Not Disturb

The Do Not Disturb (DND) feature prevents other people from placing a call to you, the cue will display before the account after the DND is enabled. But you can dial out with no effect.

Want to…	Then …
enable DND	Press soft key “dnd”.

disable DND	Press soft key “-dnd”.

Table 5-9 Do Not Disturb

NOTE:	If you just want to block certain people from placing a call to you, see the “Incoming Call” for more information.

5.7.3 Ring Mute

The Ring Mute function prevents the phone from ringing when received a call.

Want to…	**Then …**
enable Ring Mute	Press “Mute”, and the mute LED will light on.
disable Ring Mute	Press “Mute” again, and the mute LED will be off.

Table 5-10 Ring Mute

NOTE:	As the ring mute is set on, if you pick up the handset, you may notice that the mute LED be off, the reason is the phone is in calling state, and the microphone should be normal instead of mute.

5.7.4 Call Forward Directly Setting

You may set call forward number to a selected account by soft key “fwd” in idle state.

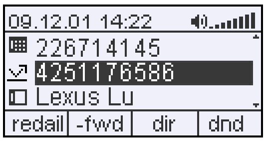
Want to…	**Then …**
set a forward number to a selected account	Press soft key “fwd”, and input forward address; you may change the forward option as *Always / On busy / No answer* by pressing soft key “option”. 
disable forward to the selected account	The account be set forward will display cue . Press soft key “-fwd”, it will goes back to normal. 

Table 5-11 Call Forward Directly Setting

NOTE:	Some SIP server support call forward if the phone is offline, this function is depended on the server and is independent on the terminal.

5.7.5 Directory

Press soft key “dir”, the LCD will display the directory.

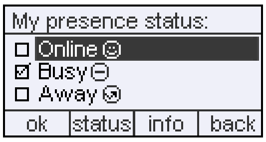
When…	**Then …**
the phone is in idle	Press soft key “dir”, it will display the directory for your invoking (call

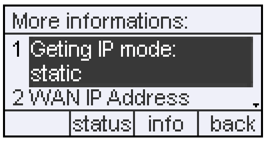
state	out or send IM) or managing (append, edit or delete) the buddies.
the phone is in other state, and it need to invoke directory	Press soft key "dir", it will display the buddy list for your selecting one to use – for conference, transfer, forward IM and etc. You may press soft key "all" or "pres" to view all the buddies or only *presence watched* buddies in the directory. 

Table 5-12 Directory

5.7.6 My Status

Press the key "Status", you can set your presence status, and the concept about presence please refers *Advanced Usage >> Presence*.
Like follows picture, you can press soft key "info" to view more status information.

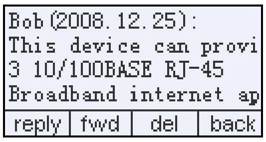




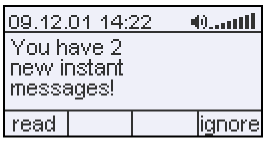
6. Advanced Usage

6.1 Instant Message

The IP Phone support Instant Message (IM), just press "IM" to entry IM page.



The IM feature use SIP Message method and most of the SIP server support this method, or it can not send IM successfully.

When…	**Then …**
it receive a new IM	The IP Phone will display "You have x new instant message" as follows, press soft key "read" to read it; and if you press soft key "ignore" to ignore it. 
you send IM at IM page	Press "IM" to open the IM page, press soft key "new" to edit new instant message for sending. You may notice there are the cues before the items of IM list on the IM page, = not opened IM, = opened IM, = the IM be forwarded, = the IM be replied, = the IM be forward and replied, = the IM on sending, = IM send successfully, = IM send failed.
you send IM at dir	Press soft key "dir" to open the directory page, select a buddy that

When…	Then …
page	you want to send IM, and press soft key “im” to edit the new IM.
you reply or forward an IM as you open an IM	Press soft key “reply” or soft key “fwd” to reply or forward current IM on the opened IM page.

Table 6-1 Instant Message

NOTE:	If the SIP server do not support Message method, or do not transfer this SIP signal, the IM function will not work successfully.

6.2 Presence

The IP Phone support presence that means you can watch your buddies status (Online, Busy or Away) if they accept you.

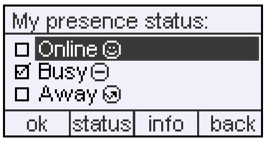
Want to…	Then …
watch a buddy	Press soft key “dir” to entry directory page, move to the buddy who you want to watch it’s presence, press soft key “watch”, the icon beside the buddy will be a circle image . If the buddy accepts your watching request, it will display presence cue. The different cues indicate different presence status or do not be watched: = unknown status, = online status, = busy status, = away status, = static status and not be watched. You may add watching as you append / edit a buddy as well.
set my presence status	Press “My Status”, you can select your current status very easily that shows as follows: 

Table 6-2 Presence

NOTE:	If the SIP server do not support Subscribe / Notify method presence function will not work successfully, if you just watch your buddy the cue will always be .

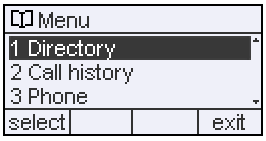
7. Phone Settings

7.1 Operations

There are two ways for setting, one is by LCD menu, and the other is by web management page.

7.1.1 LCD Menu

Press “Menu” to enter the menu page:



Move the selecting cursor by “Upper” or “Down”, use soft key “select” or the key “OK” to select the item you want to operate.
To edit the advanced options, you must login first. The default administrator’s name is “admin”, and password is “admin”.

7.1.2 Web Management

Beside the menu, you can also set the phone through web browser. Open the browser (like IE©, Firefox© or Chrome©), and input the IP address of the phone. It will display like:

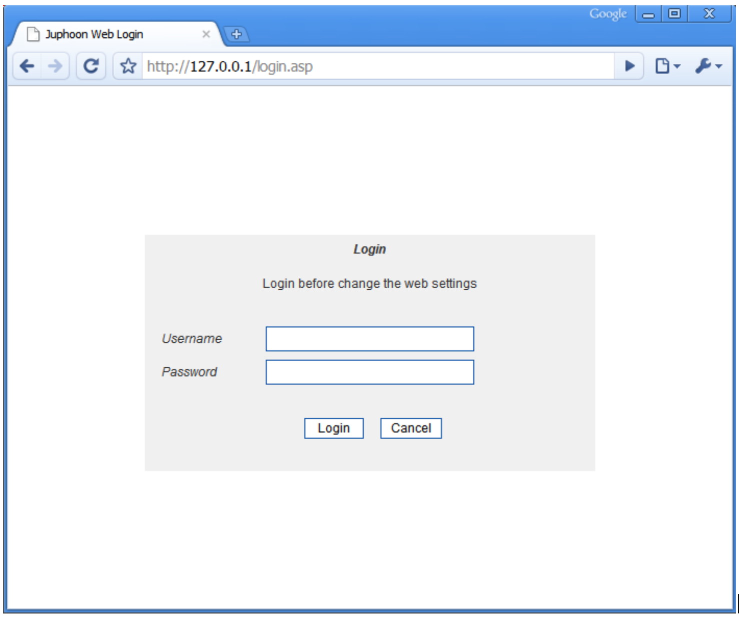


Figure 7-1 Main Page of Web Management

To get the IP Phone's IP address, simply press "Status", and press soft key "info", move the info item to *Device IP Address (Int0 IP Address)* to find it out.

You must login firstly to use web management. The default user's name is "user", the password is "user"; and the administrator's name is "admin", the password is "admin".

7.2 Menu Settings

7.2.1 Directory

You can use Directory to manage the buddy (contact) information.

- **Dial**: Dial out to the contact.
- **Add**: Add a new contact.
- **Back**: Return to the previous page.
- **IM**: Send IM to the contact.
- **Edit**: Edit the contact
- **Watch / -watch**: Watch or not watch the buddy's presence.
- **Del**: Delete the contact.
- **More**: Switch the function of soft key.

7.2.1.1 Dial out to the contact

Press soft key "dial" to dial out to the selected contact. The Phone will dial out directly or if there are more than one phone numbers in the contact, it will ask you to select one to dial.

7.2.1.2 Send IM to the contact

Press soft key "im" to send IM to the selected contact. The Phone will open IM editing page to let you compose an IM or if there are more than one phone numbers in the contact, it will ask you to select one to send IM.

7.2.1.3 Add a new contact

To add a new contact, press soft key "add", it'll entry an edit page to let you edit a new contact. You can input name, work phone, mobile phone and set presence watch option on the page. You can press soft key "save" to save the new contact or press "cancel" to exit without saving.

7.2.1.4 Edit the contact

You can edit the selected contact by pressing soft key "edit".

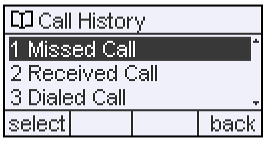
7.2.1.5 *Delete the contact*

You can delete the selected contact by pressing soft key “del”.

7.2.2 Call History

Call History arranges the call history in 3 types: Missed, Dialed and Received, and you may delete all the history by one action.

- **Missed Call**: History of missed calls.
- **Received Call**: History of received calls.
- **Dialed Call**: History of dialed out calls.
- **Delete All**: Delete all call history.



7.2.2.1 *Missed Call*

You can press “Upper” and “Down” to move the selection item. Press soft key for more actions:

- **Dial**: Redial out by the history record.
- **Del**: Delete current selected record.\
- **Back**: Return to the previous page.
- **Edit**: Edit current record, append or modify current phone number as a new contact.
- **IM**: Send IM by the history record phone number.
- **Detail**: See detail of the selected history item.
- **More**: Switch the function of soft key.

7.2.2.2 *Received Call*

Refer the “**Missed Call**”.

7.2.2.3 *Dialed Call*

Refer the “**Missed Call**”.

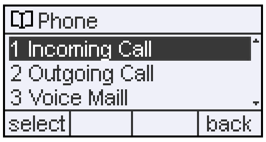
7.2.2.4 *Delete All*

Delete all the history records.

7.2.3 Phone

This submenu contains the information related with the phone.

- **Incoming Call**: The phone's behavior when received a call.
- **Outgoing Call**: The phone's behavior when dial out a call.
- **Voice Mail**: Set the voice mail server number.
- **Clock**: The date time settings.
- **Alarm**: The alarm settings.
- **Ring**: Selected the different ring as the phone encounter the different event.



7.2.3.1 Incoming Call

1. Status

You can select one from 3 types:

- **Normal**
- **Do Not Disturb**: The phone will automatically reject the call, while you can still call out.
- **Auto Accept Call**: The phone will automatically accept the call.

2. Block User

It manages the black list of the phone number from which the phone will ignore the call.

7.2.3.2 Outgoing Call

1. Status

You can select one from 2 types:

- **Normal**
- **Hotline**: The phone will automatically dial out the hotline number when you pick up the handset or press “Speaker” or “Headset”.

2. Hotline

You can set the hotline number in this submenu.

7.2.3.3 Voice Mail

You can set the voice mail message subscribe on or off, and you can also pre-set the voice mail number for one pressing dial out to the mail server.

7.2.3.4 Conference

You can set the conference mode and the conference number in the mode of conference made by server.

7.2.3.5 Clock

1. Status

You can select one from 3 methods by which the phone get the date time:

- **Manual**: Set by user.
- **NTP**: Get time from NTP Server.
- **Register**: Get time from REGISTER SIP message.

2. Time Zone

You can select the time zone from the list.

3. NTP Server

Set the address of NTP server. You should set the Status to NTP first.

4. Time

Set current date time. You should set Status to Manual first.

7.2.3.6 Alarm

There are 4 alarms you can set. The set options:

- **Status**: Set the alarm on or off.
- **Alarm Type**: Set the alarm as a repeatable or one time alarm. You can set repeat alarm by the circle of a week, just select the *Monday, Tuesday, Wednesday, Thursday, Friday, Saturday, Sunday* to repeat.
- **Alarm Words**: Set the alarm words that will displays as the alarm is trigged.
- **Alarm Date**: Set the alarm date; notice, the alarm date is only available as the alarm type is *one time* alarm.
- **Alarm Time**: Set the alarm time.
- **Tone**: Select the alarm tone from the ring tone list.
-

7.2.3.7 Ring

You can select the IP phone ring tone as your favor tones.

- **Incoming call**: The ring will play as there is a new incoming call.
- **Message indication**: The ring will play as a new IM or voice mail comes.
- **Error Warning**: The ring will play as an error occurs.

You may upload your customized ring tone via web site, and select as your own ring.

7.2.4 Speed Key

You can set speed key name and phone number, there are 10 speed keys for you.

7.2.5 Network

This submenu contains the network settings.

- **General Setting**: The general network settings including IP mode, WAN port (main port) and PC port (secondary port) settings.

- **QoS**: The Quality of Service settings.
- **NAT Traversal**: The NAT traversal settings.

7.2.5.1 General Setting

- **WAN port Setting**

[IP]

1. IP Type

You can select one from 3 methods by which the phone get the IP parameters:

- **Manual**: Set IP address by user.
- **DHCP**: Get IP address from DHCP server.
- **PPPoE**: Get IP address from PPPoE server.

2. IP Address

Edit or view the IP address of the phone (WAN port). You can only edit the IP address if current IP getting type is *Manual*, or if you get IP by *DHCP/PPPoE*, you can only view the current IP address.

3. Net Mask

Edit or view the net mask. As the same as IP address, you can only edit the Net Mask if current IP getting type is *Manual*.

4. Gateway

Edit or view the gateway to access WAN. As the same as IP address, you can only edit the gateway if current IP getting type is *Manual*.

5. MAC

View the MAC address of WAN port. You cannot change this value.

[DNS]

You can change following values.

1. DNS Type

You can select one from 2 methods by which the phone get the DNS parameters:

- **Manual**: Set DNS server by user.
- **Auto**: Get DNS server from DHCP server or PPPoE server.

2. Primary DNS

Set the IP address of the primary DNS server. This parameter only editable as the DNS type is set by *Manual*.

3. Secondary DNS

Set the IP address of the secondary DNS server. This parameter only editable as the DNS type is set by *Manual*.

[PPPoE]

Set the IP Type to PPPoE firstly or you cannot change following values.

1. PPPoE Name

Set the PPPoE authorized name.

2. PPPoE Password

Set the PPPoE authorized password.

3. PPPoE ISP

Set the PPPoE ISP name.

4. Retry Period

Set the PPPoE retry period time length when connecting.

- **PC port Setting**

[Mode Setting]

The IP Phone has 2 RJ45 ports, the one is WAN port (main port) the other is PC port (secondary port). And the 2 ports can work in 2 modes: *Bridge* and *Router*.

If the phone be set as *Bridge* mode, the phone will act as a simple switch, which will transfer the IP data within a same sub-network without router; the WAN port is as an upper link of the switch, and the PC port share the LAN to other devices like PC. If the phone be set as *router* mode, the phone will act as a simple router; the WAN port is as an upper link of the router, and the PC port is owned by the LAN and is the gateway for the phone.

To understand what's the meaning of the *Bridge* mode and *Router* mode. Here are two scenarios to explain the *Bridge* mode and *Router* mode, while the real net work may be different with here.

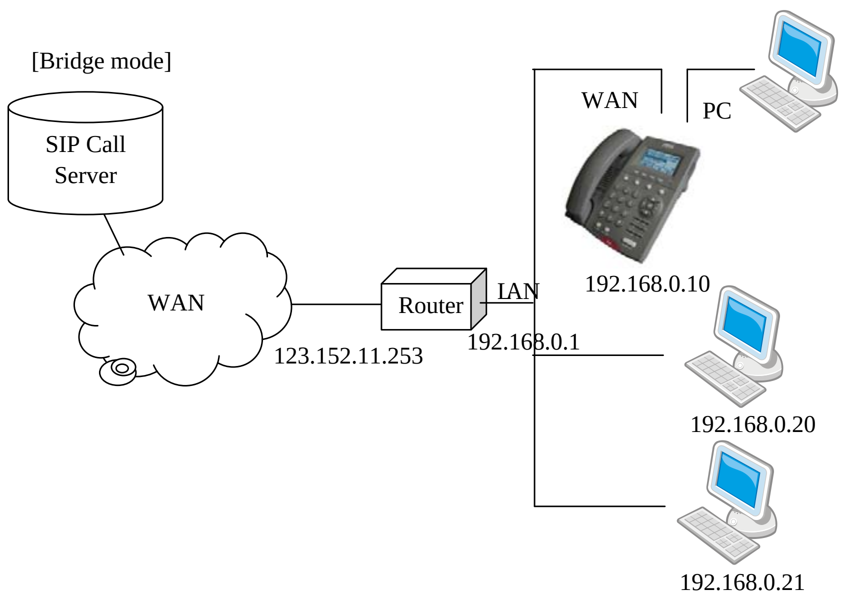


Figure 7-2 Bridge Mode Scenario

Bridge mode: as you seen in the figure, the IP Phone only has one IP address (192.168.0.10) in *Bridge* mode, and it will transfer the IP data to the connected PC within the same sub-network – 192.168.0.15, it just works as a one port switch.

[Router mode]

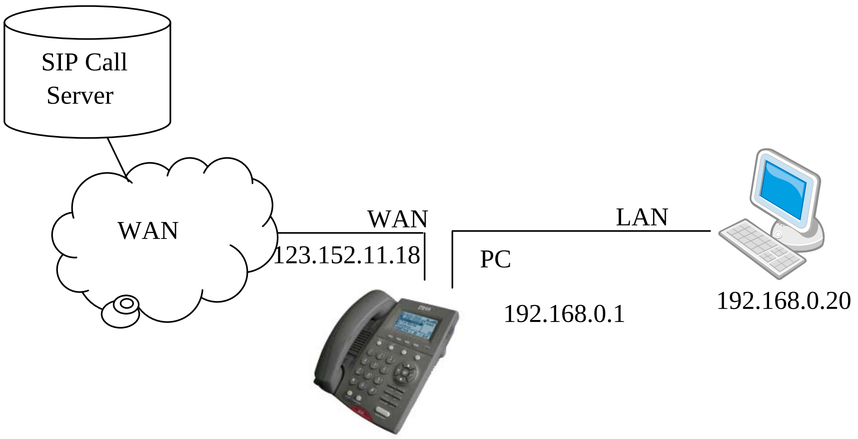


Figure 7-3 Router Mode Scenario

Router mode: the IP Phone has two IP addresses when it works in *Router* mode, the one IP address (123.152.11.18) is as a net end point on WAN, the other IP address (192.168.0.1) is as the LAN gateway for PC (192.168.0.20) to access WAN.

[LAN Setting]

You can set PC port IP parameters as the subnet gateway:

- **LAN IP Address**: Set LAN IP address as PC's gateway.
- **Subnet Mask**: Set the current subnet mask.
- **MAC**: View PC port MAC address.

[DHCP Server]

The PC port can provide DHCP service when it acts as the subnet gateway, the PC connects with the IP Phone will get IP address automatically if PC DHCP client is running.

- **Status**: Set DHCP service on or off; notice if there are more than one DHCP server in the LAN, it may conflict to assigning IP address.
- **From / To**: Set the auto assigning IP address range, from IP address1 to IP address2. For instance, the address1 = 192.168.0.21, the address2 = 192.168.0.30, the attempting assigned IP address will be 10 addresses.
- **Address Rent Duration**: The client will re-request the IP address after the duration time is expired. The default time is 120 minutes.

7.2.5.2 QoS

1. VLAN

Virtual LAN arranges a separate LAN from the existing LAN, that enables the IP Phone in VLAN insure the voice media transmit quality without the disturbing of the other IP data.

- **Status**: You can select on or off to enable or disable VLAN.
- **VID**: Set the VLAN ID.
- User Priority: Set the priority of the VLAN.

NOTE:	Please do not set the VLAN if you don't understand its' concept, or you may not access the IP Phone if the PC is not at the same VLAN ID.

2. DSCP

DSCP is another QoS setting, means Differentiated Services Code Point, it will set the TOS value on the IP data header, and the network device will take different transmit strategy for different service. The default DSCP value is 0.

7.2.5.3 NAT Traversal

- **STUN**

The STUN feature is to traversal NAT between LAN and WAN.

- **Status**: You can select on or off to enable or disable STUN feature.
- **Address**: Set the address of the STUN server.
- **Port**: Set the port of the STUN server.

NOTE:	Please do not set STUN if you don't understand what you are doing, or the IP Phone may not work normally.

- **NAT Keep Alive**

The NAT Keep Alive feature is to keep the binding of the address and port of the phone with the router.

- **Status**: You can select on or off to enable or disable NAT Keep Alive feature.
- **Interval Time**: Set the IP Phone send keeping alive package by the interval time.

7.2.6 Advanced

- **Account**: Set the account information.
- **Presence**: Set the presence information
- **Media**: Set parameters related with media.
- **SIP**: Set the SIP parameters – only set on web browser.
- **Local Upgrade / Factory setting**: Download and upload the factory setting and upload system file – only set on web browser. You only can save the current setting to factory setting and restore the settings by menu.
- **Password**: Set the login password.
- **Volume**: Set volume for handset, handset Mic, Speaker, Speaker Mic, Ringer. – only set on web browser.
- **Miscellaneous**: Miscellaneous

7.2.6.1 Account

The phone can support registering 6 accounts at the same time. Each account contains the same options as following.

- **Status**
- **Subscribe Registration**
- **Use Tel URI**
- **Incoming Call Display**
- **Proxy**
- **Registrar**
- **User**
- **Call Forward**
- **Codec**

- **Status**

You can select on or off to enable or disable this account setting. To use the phone to make calls, you must at least enable one account.

- **Subscribe Registration**

You can enable or disable the function of subscribe registration.

- **Use Tel URI**

You can enable or disable the use of Tel URI,

- **Incoming Call display**

You can set the field to display when there is a call coming.

- **Proxy**

In some cases, to work properly, the phone must send the SIP message to the proxy.

1. Status

   You can select on or off to enable or disable this feature.
2. Realm

   Set the realm of the proxy.
3. Protocol

   You can select UDP / TCP / TLS as the transport protocol.
4. Primary Proxy

   Set the details about the primary proxy including the primary address and the UDP / TCP / TLS port.
5. Secondary Proxy

   Set the details about the secondary proxy including the secondary address and the UDP / TCP / TLS port.

6. Address / Secondary Address

Set the address of the proxy.

- **Registrar**

In some cases, to work properly, the phone must register on a SIP server before making calls.

1. Status

You can select on or off to enable or disable this feature.

2. Realm

Set the realm of the register server.

3. Protocol

You can select UDP / TCP / TLS as the transport protocol.

4. Address

Set the address of the SIP register server.

5. UDP / TCP / TLS Port

Set the port of the register server.

- **User**

1. User Name

Set the user name in SIP message.

2. Display Name

Set the display name in SIP message.

3. Authorization Name

Set the authorized name to register on SIP server.

4. Authorization Password

Set the authorized password to register on SIP server

- **Call Forward**

If you enable this feature, the phone will automatically forward the received call to another phone number under certain cases. The phone supports 3 cases:

  - **Unconditional**: The received call will always be forwarded.
  - **On Busy**: The received call will be forwarded when you are during a call.
  - **On No Answer**: The received call will be forwarded when you didn't answer it in 30 seconds.

For each case, there are 2 options:

1. Status

You can select on or off to enable or disable this feature in the corresponding case.

2. Forward Uri

Set the phone number or the URI to forward to.

NOTE:	If Unconditional is enabled, DND and AAC option will be ignored. If On Busy is enabled, the received call will be forwarded when DND is enabled or user reject the call.

> You can also enable or disable forward by shortcut setting, see Shortcut Setting for more information

- **Codec**

You can configure the codec information assigned to the phone. First select codec to use in Codec List, and then arrange the priority order of them.

1. Codec List

   There are 4 codec you can select: PCMU, PCMA, G.729ab, iLBC.

2. Priority

   You can select one codec by “Upper” or “Down” keys and press soft key “up” or “down” to change the priority order.

7.2.6.2 Presence

You need to set presence enable if you want to watch buddies state in real time.

1. Status

You can enable or disable the presence feature.

2. Active Presence User

You can select one of the accounts to enable the presence feature.

3. Presence type

There are 2 setting types of presence:

- **P2P**: Peer to Peer mode, that only need the server support subscribe / notify SIP method and retransmission.
- **Event list**: Event list mode is more efficient for the IP network, and it needs the server support XCAP, and you need to ask administrator for the XCAP server information.

If you want to operate the presence state releasing and watching, please refer the chapter 6 *advance usage* >> *presence*.

4. XCAP Auth Name

Set the XCAP auth name.

5. XCAP Auth Password

Set the XCAP auth password.

6. XCAP Root

Set the XCAP root.

7. XCAP User ID

Set the XCAP user ID.

8. Proxy Address

Set the aggregation proxy address.

9. Proxy Port

Set the Aggregation proxy port.

10. Proxy Host

Set the aggregation proxy host name.

11. RIs-Service URI

Set the Rls-Service URI template.

7.2.6.3 *Media*

It contains following in this submenu:

- **DTMF**: Dual-tone multi-frequency.
- **RTP**: Set the port range for RTP.
- **RTCP**: Setting with RTCP.
- **SRTP**: Set SRTP Key.
- **Side Tone**: Set the voice effect during a call.

- **DTMF**

DTMF, Dual-tone multi-frequency, is used for telephone signaling over the line in the voice-frequency band to the call server.

1. DTMF Type

You can select one of 3 methods to carry the DTMF signal:

- **Inband**: Mix the signal into the voice band.
- **Outband**: Carry the signal in RTP packet (RFC2833).
- **SIP Info**: Carry the signal by SIP INFO message.

2. DTMF Payload

You can set the payload value of the RTP packet of DTMF signal in Outband mode.

- **RTP**

In following options, you can set the port range from Start Port to End Port which the RTP used.

- **RTCP**

You can select on or off to enable or disable RTCP alone with the voice transmission.

- **SRTP**

You can set SRTP Key.

- **Side Tone**

If the side tone is set, the voice will be transferred from mic to earphone in real time with attenuation – the effect sounds like a traditional phone.

7.2.6.4 *SIP*

You can set the SIP parameters if you understand SIP protocol, or please do not modify these settings.

7.2.6.5 *Local Upgrade / Factory Setting*

You can upgrade by the web management page on PC, you can restore factory settings and save current settings to factory setting; and download the factory setting file to PC and upload the factory setting file to IP Phone; and upload the file system upgrade package, kernel image file, file system image file.

NOTE:	The error may occur when you download or upload those files. Please make sure you are understand what you are doing especially upload the firmware, wrong version of the firmware may lead not being able to roll back. If you want upload several files, you'd better upload one by one.

7.2.6.6 Password

- **Operator Password**: Set the operator login password, the login name is fixed as "admin".
- **User Password**: Set the user login password, the login name is fixed as "user".

7.2.6.7 Miscellaneous

- **Watch Dog**: The watch dog can reboot the phone when it went to wrong.
- **Log Level**: Set the different log level as *None*, *Critical*, *Detail* or *All*.
- **Syslog**: Set the Syslog server address that will be sent Syslog to. More information please refers RFC3164.