Juphoon IP Phone

Published: March 2009 please visit our website at http://www.juphoon.com

uContact BP200 IP Phone User Manual

Juphoon System Software Corporation.
Address: Shangdong international Building 2-910, Jiangdong District, Ningbo, Zhejiang, China.
Tel: +86-574-87287820
Fax: +86-574-87304379
ZIP code: 315041

CONTENTS

LIST OF TABLE

LIST OF FIGURE

1. Definitions

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 BP200, and the phone software version is uContact.P3.07. The document was last updated on April 15, 2009.

3. About BP200

Thank you for choosing uContact BP200! This multimedia IP Phone is a terminal which aims the business environment works with any other standard SIP servers, just acts as a desktop unify communication device.

4. Introduction

As shown in Figure 4-1, the IP Phone's dimension is 70 mm x 215 mm x 195 mm(H x L x W), Its' operating interface includes a handset, a headset, a 2 x 16 text 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 .Details about these keys are in the following table.

Type	Key	Description
Main Keys	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.
	3Way Talk	1. Add a 3rd party to form a 3Way Talk. 2. Change Talk & Hold mode to a 3Way Talk. 3. Accept an incoming call to form a 3Way Talk.
	Delete	Delete a character backward.
	Character selection	Switch input method between digit, lowercase and uppercase. Note: Input method depends on different cases.
	Volume + / -	Adjust the volumes of ringing, handset, headset and hands-free speaker in different scenario.
	Voice Mail	Check the voice message indication
	Hold	1. Put a call on hold or un-hold. 2. Switch between talking and held calls.

		3. Extension usage with 3Way Talk or transfer.
Menu	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.

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.
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. Support headset.
4. 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)
- 3Way Talk
- Automatic Accept Call / Do Not Disturb

Other Functions

- Mute, Volume Adjust & Ring Tone Profile Management

- DTMF (Inbound / Outbound / SIP-Info)
- MWI for Voice Mail
- Call History (Missed Received & Dialed)
- Phone Book (LCD Menu & Web)
- 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 / Manual
- XML-Based Auto-Provision and Auto-Update
- Support for IEEE802.1p/Q tagging (VLAN)
- 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 active accounts while the phone is in idle state. Select the active account you need to dial out by press "Upper" or "Down" key,. 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 3Way Talk 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 "Rediall", and it will call out the number which you latest dial out at once.
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 "OK". 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 "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".
answer the 2nd call	Press "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" to reject the new call. The old one will remain unchanged.
forward the call	See the *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.
using the speaker	Press “Speaker” again..
using the headset	Press “Headset” again..
there are 2 calls and using the handset	Press “Cancel” to end the current call. Hang up the handset to end all calls.
there are 2 calls and using the speaker	Press “Cancel” to end the current call. Press “Speaker” to end all calls.
there are 2 calls and using the headset	Press “Cancel” 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” during a cal. 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”.
you want to resume a call on hold	Press “Hold” again.
there are 2 held calls and you want to talk to one	Select one line and press “Hold”. The phone will unhold the current call and other lines will be held automatically whether it’s held or talking.

Table 5-4 Holding a Call

5.2.6 Transferring a Call

Want to…	Then …
do blind transfer a call	Press “Transfer” and dial the 2nd part number for transferring to.
do attending transfer a call	Press “Hold” and dial the 2nd number, after talk with this party of this number, press “Transfer” to transfer from 1st number to 2nd number.
make transfer between current 2 calls	Press “Transfer” to let these 2 parties to be connected.

Table 5-5 Transferring a Call

5.2.7 Redialing

To redial the most recently dialed number, press “Redial”, and it will dial out the number which you latest dialed 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 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 14 keys (the “Num Keys” 0-9 and the “function keys” F1-F4) 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” or a “function keys”, and press “OK”, the phone will translate this key to a pre-set number.

5.5 Making 3Way Talk Calls

This IP Phone support one 3Way Talk mode: Local 3Way Talk mixing.

When…	Then …
you are in a call and want to invite the 3rd party to build 3Way Talk directly	Press "3Way Talk" ,and dial the 3rd party number.
you are in a call and want to build a 3Way Talk after consult the 3rd party	Press "Hold" and dial the number. After consulting with the 3rd party, press"3Way Talk" to build the 3Way Talk
during a call and receive the 3rd party call	Press "OK" or "1" to accept a new call, as you connect to this party, press "3Way Talk" to build 3Way Talk.

Table 5-7 Placing a Local 3Way Talk

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" it will display the voice message waiting indication.

To access voice mail, just call out the voice mail server number (Voice mail server URI). You also can assign phone numbers to the speed dial keys by pre-setting.

5.7 Shortcut Settings

5.7.1 Volume

You can use "Volume + / -" 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 change the different volume 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 LED will light on after the DND is enabled. But you can dial out with no effect.

Want to…	Then ...
enable DND	Press “Do Not Disturb”.
disable DND	Press “Do Not Disturb” again.

Table 5-8 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 5-9 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 Info

Press the key “Info”, you can see the status information, such as IP Address, MAC.

6. Phone Settings

6.1 Operations

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

6.1.1 LCD Menu

Press "Menu" to enter the menu page:

Move the selecting cursor by "Upper" or "Down", use 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".

6.1.2 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:

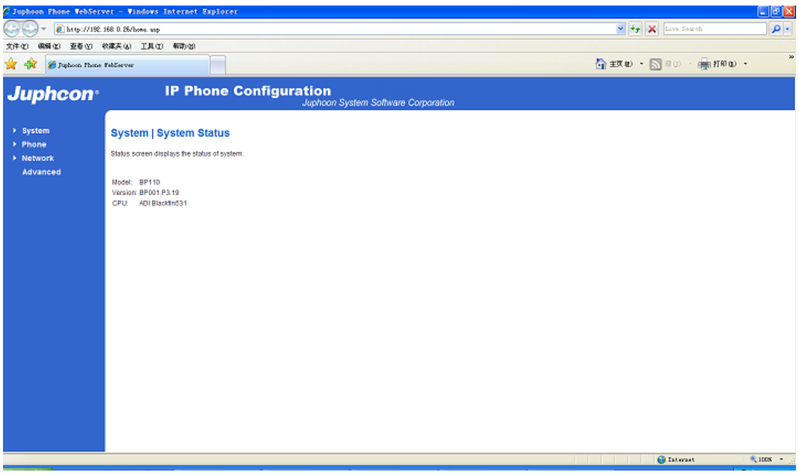

Figure 6-1 Main Page of Web Management

To get the IP Phone's IP address, simply press "*Info*" to find it out.

You must login firstly to set the Advanced setting. The default user's name is "user", the password is "user"; and the administrator's name is "admin", the password is "admin".

6.2 Menu Settings

6.2.1 Directory

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

- **Dial**: Dial out to the contact.
- **New Entry**: Add a new contact.
- **Search**: Search the contact by his name.
- **Edit**: Edit the contact
- **Delete**: Delete the contact.
- **Delete All**: Delete all contact.

6.2.1.1 Dial out to the contact

Just find out the contact which you want to dial, press “OK” and select “Dail”, then press “ok” to dial out to the selected contact. The Phone will dial out directly.

6.2.1.2 New Entry

To add a new contact, press “New Entry”, it’ll entry an edit page to let you input a new contact. After you input name, press “OK”, it will let you input phone number .You can press “OK” to save the new contact or press “cancel” to exit without saving.

6.2.1.3 Search

You can search a contact by his name. For example, you input “B” and press “OK”, it will display contacts which is begin with “B”, such as Bob,

6.2.1.4 Edit the contact

Just find out the contact which you want to edit, press “OK” and select “Edit”, it will let you to edit the selected contact. You can press “OK” to save the new contact or press “cancel” to exit without saving.

6.2.1.5 Delete the contact

You can delete the selected contact just like edit the contact.

6.2.2 Delete all

To delete all contacts, select “Delete All”,.

6.2.3 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.

6.2.3.1 Missed Call

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

- **Dial**: Redial out by the history record.
- **Detail**: See detail of the selected history item.
- **Save**: Save current record, append or modify current phone number as a new contact.
- **Delete**: Delete current selected record.

6.2.3.2 Received Call

Refer the “**Missed Call**”.

6.2.3.3 Dialed Call

Refer the “**Missed Call**”.

6.2.3.4 Delete All

Delete all the history records.

6.2.4 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.
- **Clock**: The date time settings.
- **Alarm**: The alarm settings.
- **Ring**: Selected the different ring as the phone encounter the different event.

6.2.4.1 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.

6.2.4.2 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.

6.2.4.3 Voice Mail

You can set the voice mail message subscribe on or off, and you can get your voice mail by calling out the voice mail server number (Voice Mail Server URI). You also can assign phone numbers to the speed dial keys by pre-setting.

6.2.4.4 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.

6.2.4.5 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 on LCD 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.

6.2.4.6 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 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.

6.2.5 Network

This submenu contains the network settings.

- **General Setting**: The general network settings including IP mode, WAN port(main) and PC port (secondary) settings.
- **QoS**: The Quality of Service settings.
- **NAT Traversal**: The NAT traversal settings.

6.2.5.1 General Setting

6.2.5.1.1 IP Mode Setting

The IP Phone has 2 RJ45 ports, the one is WAN port, and the other is PC 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.

[Bridge mode]

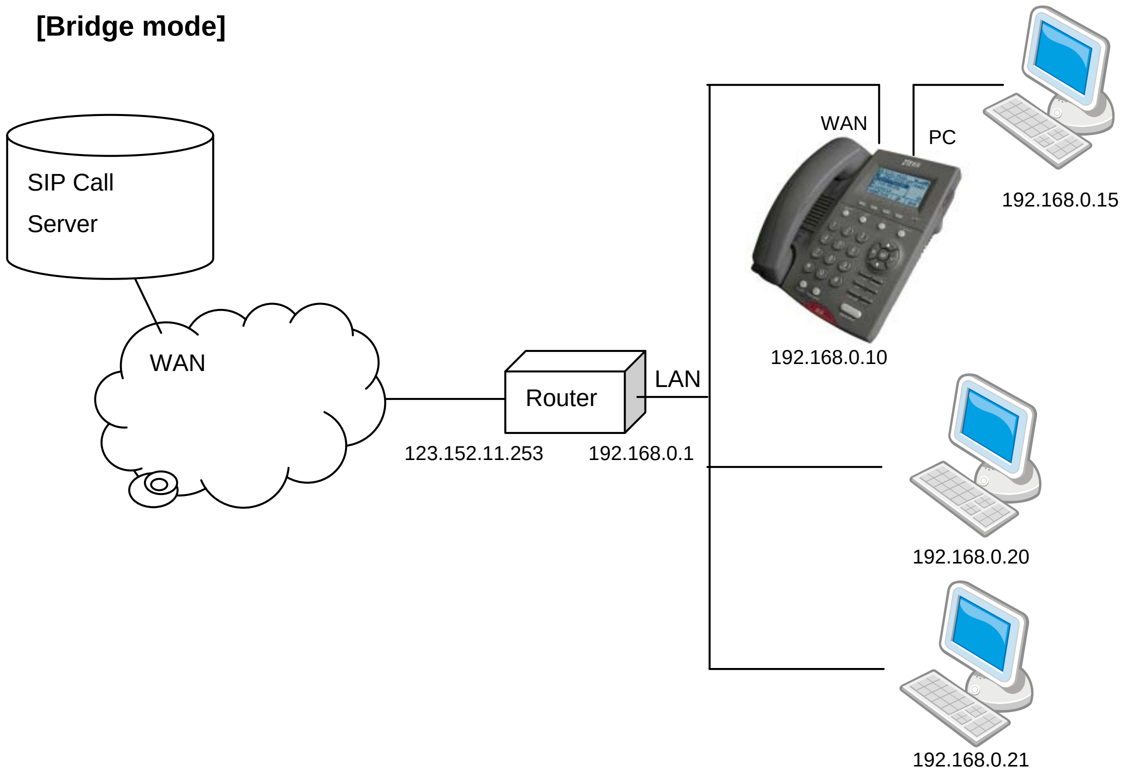

Figure 6-2 Bridge Mode Scenario

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]

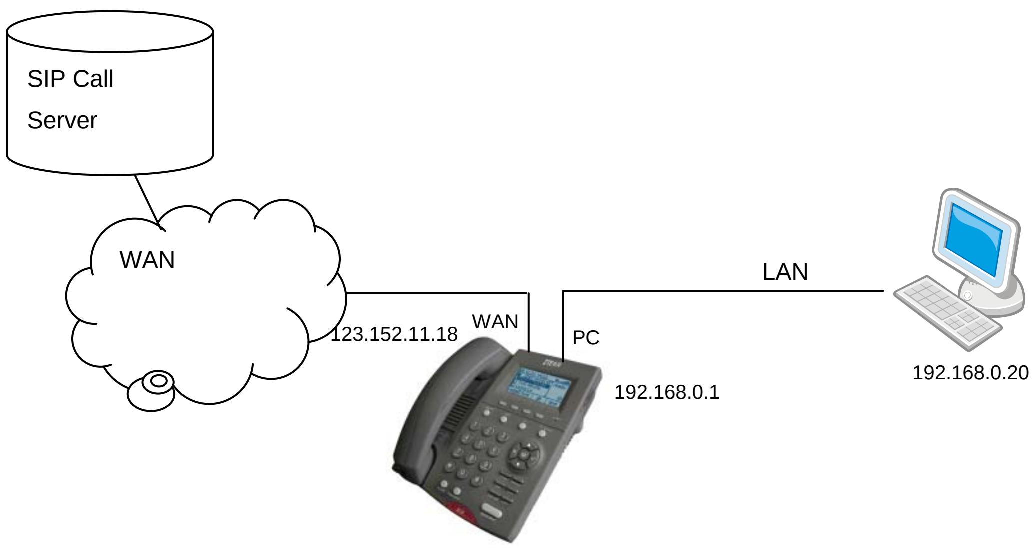

Figure 6-3 Router Mode Scenario

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.

6.2.5.1.2 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

   The PPPoE authorized name.

2. PPPoE Password

   The PPPoE authorized password.

3. PPPoE ISP

   The PPPoE ISP name.

4. Retry Period

   The PPPoE retry period time length when connecting.

6.2.5.1.3 PC port setting

The PC port setting is only needed in case of the *IP Mode* is *Router*. You don't need set anything about PC port when the *IP Mode* is *Bridge*.

[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.

6.2.5.2 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.

6.2.5.3 NAT Traversal

6.2.5.3.1 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.

6.2.5.3.2 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.

6.2.6 Advanced

- **Account**: Set the account information.
- **Media**: Set parameters related with media.
- **SIP**: Set the SIP parameters – only set on web browser.
- **Auto Provision**: Set the provision.
- **Local Upgrade**: Download and upload the factory setting and upload firmware – only set on web browser.
- **Password**: Set the login password.
- **Misc**: Miscellaneous

6.2.6.1 Account

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

- **Status**
- **Proxy**
- **Registrar**
- **User**
- **Call Forward**
- **Codec**

6.2.6.1.1 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.

6.2.6.1.2 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. Address / Secondary Address
   Set the address of the proxy.
3. Realm
   Set the realm of the proxy.
4. Port / Secondary Port
   Set the port of the proxy.
5. Protocol
   You can select UDP or TCP as the transport protocol.

6.2.6.1.3 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. Address
   Set the address of the SIP register server.
3. Realm
   Set the realm of the register server.
4. Port
   Set the port of the register server.
5. Protocol
   You can select UDP or TCP as the transport protocol.

6.2.6.1.4 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.

6.2.6.1.5 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.

6.2.6.1.6 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 "up" or "down" to change the priority order.

6.2.6.2 Media

It contains following in this submenu:

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

6.2.6.2.1 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.

6.2.6.2.2 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.

6.2.6.2.3 RTP

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

6.2.6.2.4 RTCP

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

6.2.6.3 SIP

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

6.2.6.4 Auto Provision

The Auto Provision feature can setup the phone automatically. Set the auto provision server URL and the other parameters:

- **Provision Enable**: Set auto provision on or off.
- **HTTP URL**: Set the provision server address with file path – http URL
- **HTTP Port**: Set the provision server port
- **User Confirmation**: If it set on, it will display message on LCD to indicate there needs some auto upgrade, asking you to confirm or cancel.
- **Check Scheduling**: The IP Phone will check with provision server by the scheduling timer as you selected.

6.2.6.5 Local Upgrade

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 firmware.

Waning:	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.

6.2.6.6 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”.

6.2.6.7 Misc

- **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.