

# Mobile VoIP -2

## User Manual



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## 1.Introduction

Mobile VoIP -2 is a 2 channels VoIP GSM Gateway for call termination (VoIP to GSM ) and origination (GSM to VoIP). It is SIP based and compatible with Asterisk. It can enable to make 2 calls simultaneously from IP phones to GSM networks and GSM network to IP phone.

## 2.Function description

2.1 VoIP(SIP) 、 GSM(MOBILE VOIP) conversion.

2.2 50 sets of LAN->MOBILE routes setting , 50 sets of MOBILE->LAN routes setting.

2.3 Voice response for setting and status (dial in from mobile).

2.4 Series connections to save bills.

2.5 Standard SIP(RFC2543,RFC3261) protocol ,  
Communicates with other gateway or PC.

## 3.Parts list

Please check the parts for any missing parts. If do, please contact our agents :

3.1 「 MOBILE VOIP-2 」 main body

3.2 Power adaptor AC-DC (110V AC – 12V DC) or (220V AC – 12V DC)

3.3 Network cable

3.4 Antenna

3.5 User Manual



(1)



(2)



(3)



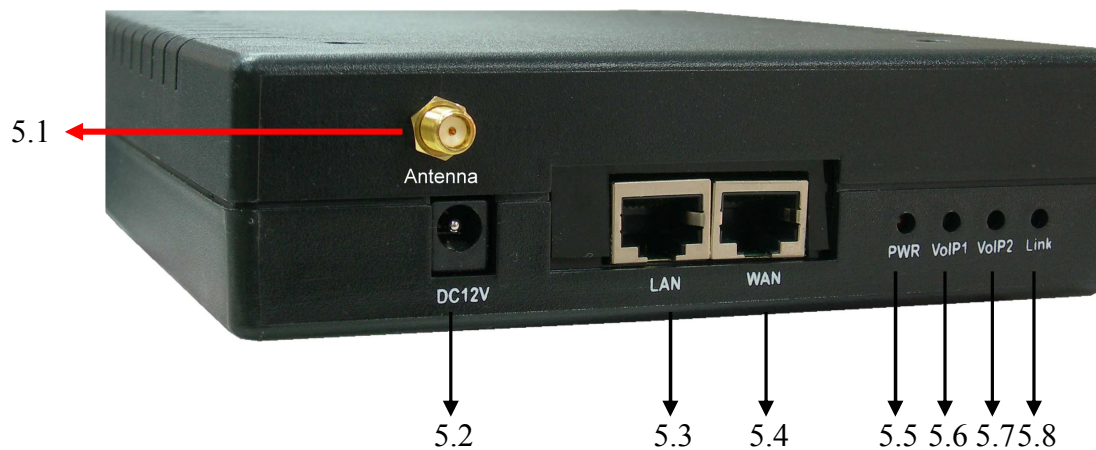
(4)

#### 4.Dimension



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## 5.Chart of the device



5.1 Antenna : Antenna connector.

5.2 DC 12V : Power input.

5.3 LAN : LAN port. It also can be DHCP Server.

5.4 WAN: RJ-45 internet connector , standard RJ-45 socket , connect to HUB.

5.5 PWR (Power LED) : Light up when power is normal.

5.5 VoIP1 : an indicator light of VoIP1

5.6 VoIP2 : an indicator light of VoIP2

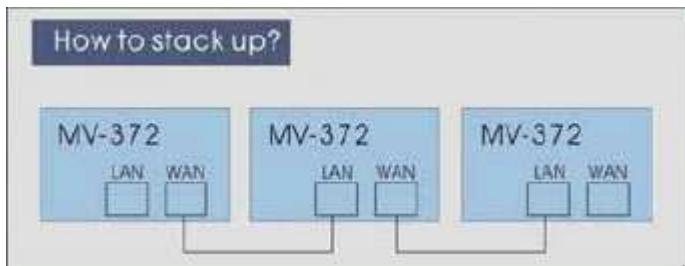
5.8 LINK Indicator : Light up when network is connected.

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## 6.CABLING

6.1 Connect the internet cable from HUB to the 'WAN' connector of the Mobile VoIP-2.

\*If you need to stack up more Mobile VoIP-2, you can stack up as follows.



6.2 Connect the antenna and put it in proper position to get the best signal reception.

6.3 Insert the SIM card from back of the main body. (take the slide off first).

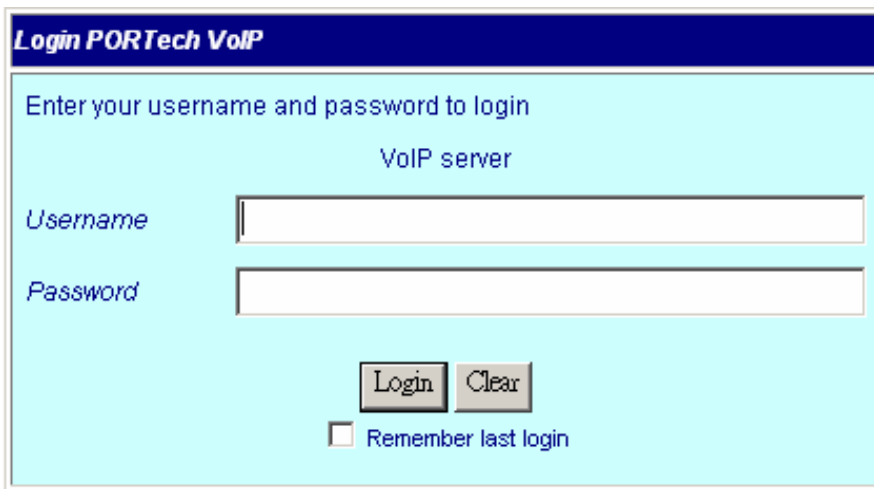


6.4 Connect the power adaptor. The 'POWER' LED should be light up.

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## 7.Web Page Setting

When the IP setting is done, the operator may setup all the rest parameters via web page. Browse the IP address from Internet Explorer (e.g. <http://192.168.0.100>) . The following page shows up :



**Login PORTech VoIP**

Enter your username and password to login

VoIP server

Username

Password

Remember last login

Enter the username and password for authentication. (default username=voip, password=1234). The page follows when the username and password are correct.

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## 8. System Information.

8.1 When you login the web page, you can see the demo system current system information like firmware version, company... etc in this page.

8.2 Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.

The screenshot shows a web interface with a light blue background. On the left is a vertical navigation menu with the following items: **Route**, **Mobile**, **Network**, **SIP Settings**, **NAT Trans.**, **System Auth.**, **Save Change**, **Update**, and **Reboot**. Each item has a right-pointing arrow. The top of the menu has the text **Mobile Voip** in a stylized font. To the right of the menu is the main content area titled **Dual VoIP**. Below the title is the text "This page illustrate the system related information." followed by a horizontal line. Below the line is a table with system information:

Model Name:	VoIP2 GSM:850/900/1800/1900MHz
Firmware Version:	Tue Dec 19 15:59:03 2006.
Codec Version:	Mon Jul 24 10:55:05 2006.



---

## 9. Route

### 9.1 Mobile TO LAN Settings

The operator may assign 50 sets of routing rule to transfer the call incoming from MOBILE to LAN.

Mobile Voip

- Route ▶
- Mobile ▶
- Network ▶
- SIP Settings ▶
- NAT Trans. ▶
- System Auth.
- Save Change
- Update ▶
- Reboot

### Mobile To LAN Table

Page: 1

Item	CID	URL	Select
0	*	*	<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Delete Selected   Delete All   Reset

**Add New**

Position:  (0~49)

CID:  Ex:0911111111, 0911\*, \*

URL:  Ex:192.168.0.1, \*:2St

Add   Reset

The MOBILE VOIP will transfer to the URL according to the caller ID of the Mobile.

\*CID :

- (1) may enter the whole number, e.g. 0911111111
- (2) only part of the number (prefix) e.g. 0911\* means any number starting with 0911 will be accepted

---

(3) \* means all numbers can be accepted

(4) N means the calls without the CID

Please note the priority of the rules. The item which has more digits will have higher priority. If the digits are the same, then former one gets the higher priority.

\*URL : The IP address to transfer this call

(1) may enter the whole IP address, e.g. 192.168.0.101 or proxy extension or phone number.

(2) If this field is blank or simply 'N', it means refuse to transfer.

(3) If an '\*' entered, it means 2-stages-dialing. The call will be answered and prompt dial tone again to receive the IP address/sip extension or **any phone number** as the destination. The caller may enter the IP such as 192\*168\*0\*101#.

\*If the device have register proxy server/Asterisk ,you can enter any destination phone number. Please note the proxy server/Asterisk need to set the route of destination phone number.

Example:

(1) Mobile to Lan: 0932\*,0911123456

Mobile VoIP have register proxy server/Asterisk

The proxy server/Asterisk have the route "09"

When the caller's prefix number is 0932, Mobile VoIP will connect 0911123456 automaticly

(2) Mobile to Lan: \*,\*

Any caller call the mobile voip's sim, mobile voip will prompt dial tone. Caller can enter IP or sip extension or phone number.

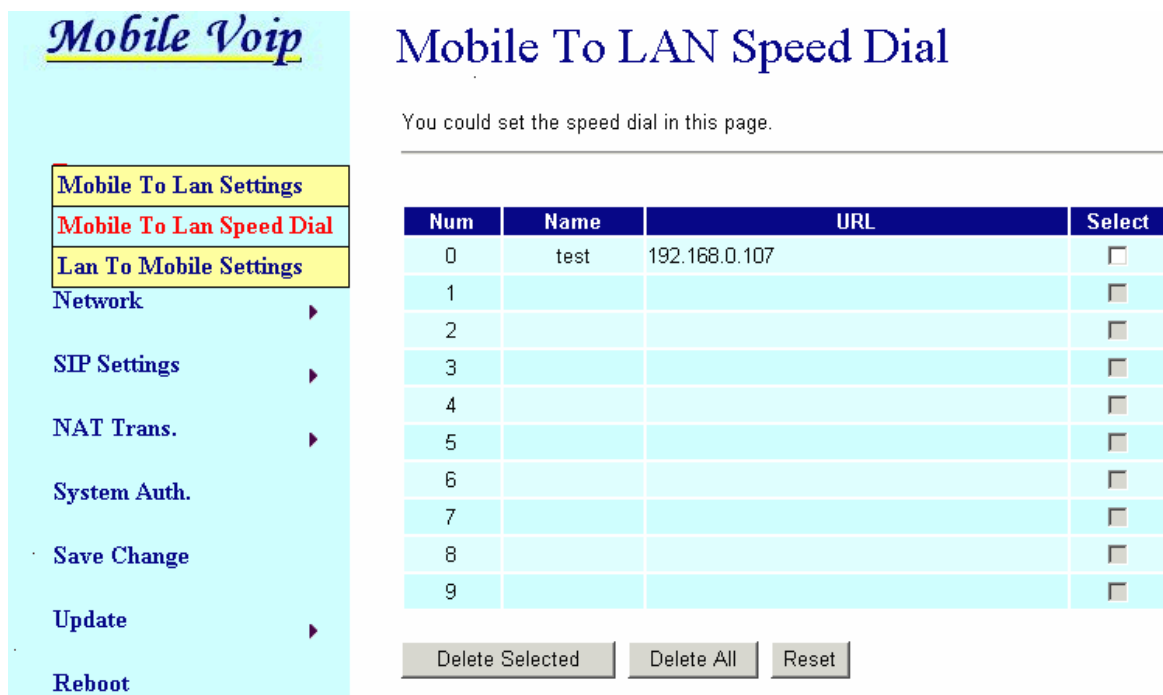
\*sip extension or phone number both need to register SIP Proxy Server or Asterisk.

\*Phone number, SIP Proxy Server or Asterisk need to set the route of this phone number.

---

## 9.2 Mobile to LAN Speed Dial Settings

When you set Mobile to LAN Speed Dial Settings and Mobile to LAN at the same time, Mobile VoIP-2 will give priority to Mobile to LAN Speed Dial Settings.



The screenshot shows the 'Mobile Voip' configuration interface. On the left is a navigation menu with options: Mobile To Lan Settings, Mobile To Lan Speed Dial (highlighted in red), Lan To Mobile Settings, Network, SIP Settings, NAT Trans., System Auth., Save Change, Update, and Reboot. The main content area is titled 'Mobile To LAN Speed Dial' and includes the instruction: 'You could set the speed dial in this page.' Below this is a table with columns: Num, Name, URL, and Select. The table contains one row with Num: 0, Name: test, URL: 192.168.0.107, and a select checkbox. At the bottom of the table are three buttons: 'Delete Selected', 'Delete All', and 'Reset'.

Num	Name	URL	Select
0	test	192.168.0.107	<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

\*The call will be answered and prompt dial tone again. When the caller may enter the “Num”, system will connect the “URL” as destination.

E.g Num:0 Name:test URL:192.168.0.107

When the caller hear dial tone and enter 0, system will connect 192.168.0.107

### 9.3 LAN to Mobile Settings

The operator may assign 50 sets of routing rule to transfer the call incoming from LAN to MOBILE.

The screenshot shows a web-based configuration interface for 'Mobile Voip'. On the left is a navigation menu with options: Mobile To Lan Settings, Mobile To Lan Speed Dial, Lan To Mobile Settings (highlighted in red), Network, SIP Settings, NAT Trans., System Auth., Save Change, Update, and Reboot. The main area is titled 'LAN To Mobile Table' and features a table with 10 rows (Item 0-9). The first row (Item 0) has '\*' in both the URL and Call Num columns. Below the table are buttons for 'Delete Selected', 'Delete All', and 'Reset'. An 'Add New' section contains input fields for Position (0-49), URL (with examples like 192.168.0.1), and Call Num (with examples like 0911, \*2St, #, #d?, #d?A??:1St), along with 'Add' and 'Reset' buttons.

Item	URL	Call Num	Select
0	*	*	<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

The MOBILE VOIP will transfer to the mobile number according to the incoming URL

\*URL : The IP address of the incoming call.

may enter the whole IP address, e.g. 192.168.0.101 or proxy server's extension. If a simple "\*" is entered, means no restriction for the incoming IP address.

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\*Call Num :

- 1.may enter the whole number, e.g. 0911111111
- 2.a simple "\*"means 2-stages-dialing. The call will be answered and prompt dial tone again to receive the called number as the destination, e.g. 0911111111 or 0911111111#
- 3.#['d'n]['a'ppp] for one-stage dialing  
[...] is option  
'd'n means to delete the beginning n codes,  
'a'ppp means to add 'ppp' in front.  
for example #d2a09 means one-stage dialing,  
delete the first 2 codes from your destination number,  
then add 09 in front as the new destination number.

**Example:**

Lan to Mobile: \*, #

- (1)Mobile VoIP and Lan Phone both need to register proxy server or Asterisk.
- (2)Proxy server/asterisk set the route that the prefix of destination number
- (3)When you dial any destination phone number from lan phone,Mobile VoIP will connect this call auto.

**Example of Application:**

When you call the ch.1 Mobile VoIP-2 gsm number,it will provide dial tone and you enter a destination number.

Then ch.2 Mobile VoIP-2 will dial this number and connect.

ch.1 Mobile VoIP-2: mobile to lan set route table \*,\*

ch.2 Mobile VoIP-2:lan to mobile set route table \*,#

Additionally, two channels Mobile VoIP-2 both need to register proxy server or Asterisk.

And proxy server/asterisk set the route that the prefix of destination number dial out from ch.2 Mobile VoIP-2.

\*The channel 2 Mobile VoIP-2's ip: the first ip + :5062 (e.g http://192.168.0.100:5062)

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## 10.Mobile

### 10.1 Mobile Status

<b>Mobile Voip</b>		<b>Mobile Status</b>	
<b>Route</b>	▶	<b>No.:</b>	Mobile 1 ▼
<b>Mobile</b>	▶	Network Registration.:	Chunghwa
<b>Network</b>	▶	SIM Card ID:	89886921400051066466
<b>SIP Settings</b>	▶	Signal Quality.:	21
<b>NAT Trans.</b>	▶	Incoming IP:	
<b>System Auth.</b>	▶	Incoming IP Name:	
<b>Save Change</b>		Outgoing IP:	123456789.0
<b>Update</b>	▶	Incoming Mob:	0928515053
<b>Reboot</b>		Outgoing Mob:	

- (1)Network Registration : The telecom carrier which the SIM card been registered.
- (2)SIM Card ID : SIM card ID.
- (3)Signal Quality : Signal quality.
- (4)Incoming IP : The IP address of the last incoming call from LAN.
- (5)Incoming IP Name: proxy server name
- (6)Outgoing IP : The IP address of the last outgoing call to LAN.
- (7)Incoming Mob : The caller ID of the last incoming call from MOBILE.
- (8)Outgoing Mob: The called number of the last outgoing call to MOBILE.

## 10.2 Mobile Setting

Mobile Voip

Route ▶

Mobile ▶

Network ▶

SIP Settings ▶

NAT Trans. ▶

System Auth. ▶

Save Change

Update ▶

Reboot

### Mobile Setting

You could set the volume of your phone in this page.

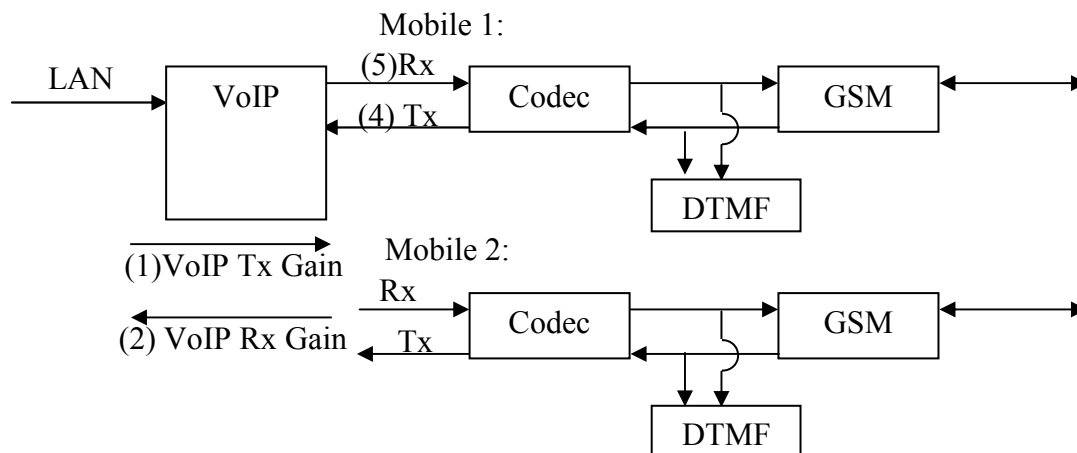
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(1) VoIP Tx Gain:  (0~12)    (2) VoIP Rx Gain:  (0~15)

(3) LAN Dialtone Gain:  (0~12)

Mobile 1	
(4) CODEC Tx Gain: <input type="text" value="7"/> (0~7)	(5) CODEC Rx Gain: <input type="text" value="7"/> (0~7)
(6) Caller ID	<input checked="" type="radio"/> Clid <input type="radio"/> Fix (SIP User)
(7) Presentation CLIR	<input checked="" type="radio"/> Suppression <input type="radio"/> Invocation
(8) Mobile PIN Code: On <input type="checkbox"/>	Code: <input type="text"/> Confirmed: <input type="text"/>
(9) LAN Answer Mode	<input checked="" type="radio"/> Answered <input type="radio"/> Alerted <input type="radio"/> Income
(10) Band Type: <input type="text" value="900/1800 MHz"/>	

Mobile 2	
CODEC Tx Gain: <input type="text" value="7"/> (0~7)	CODEC Rx Gain: <input type="text" value="7"/> (0~7)
Caller ID	<input checked="" type="radio"/> Clid <input type="radio"/> Fix (SIP User)
Presentation CLIR	<input checked="" type="radio"/> Suppression <input type="radio"/> Invocation
Mobile PIN Code: On <input type="checkbox"/>	Code: <input type="text"/> Confirmed: <input type="text"/>
LAN Answer Mode	<input checked="" type="radio"/> Answered <input type="radio"/> Alerted <input type="radio"/> Income
Band Type: <input type="text" value="900/1800 MHz"/>	



- 
- 
- (3)LAN Dialtone Gain: DTMF Reciver is not good,you can adjust gain down.
- (4)CODEC Tx Gain: as above
- (5)CODEC Rx Gain: as above
- (6)Caller ID: You may select to display the Caller ID from GSM incoming call, or fixed SIP user name.
- (7)Presentation CLIR : If you need to block the Caller Id for call termination,please choose Suppression
- (8)Mobile PIN Code:If you need to unlock pin code via MOBILE VOIP,you can click "On" and enter pin code.
- (9)LAN Answer Mode:
- Answered : when mobile answer,then connect the call
  - Alerted : when the mobile is ringing back tone,then connect the call
  - Income : when lan dial out,then connect soon
- (10)Band Type:When you buy Quad band,you need to choose your GSM frequency

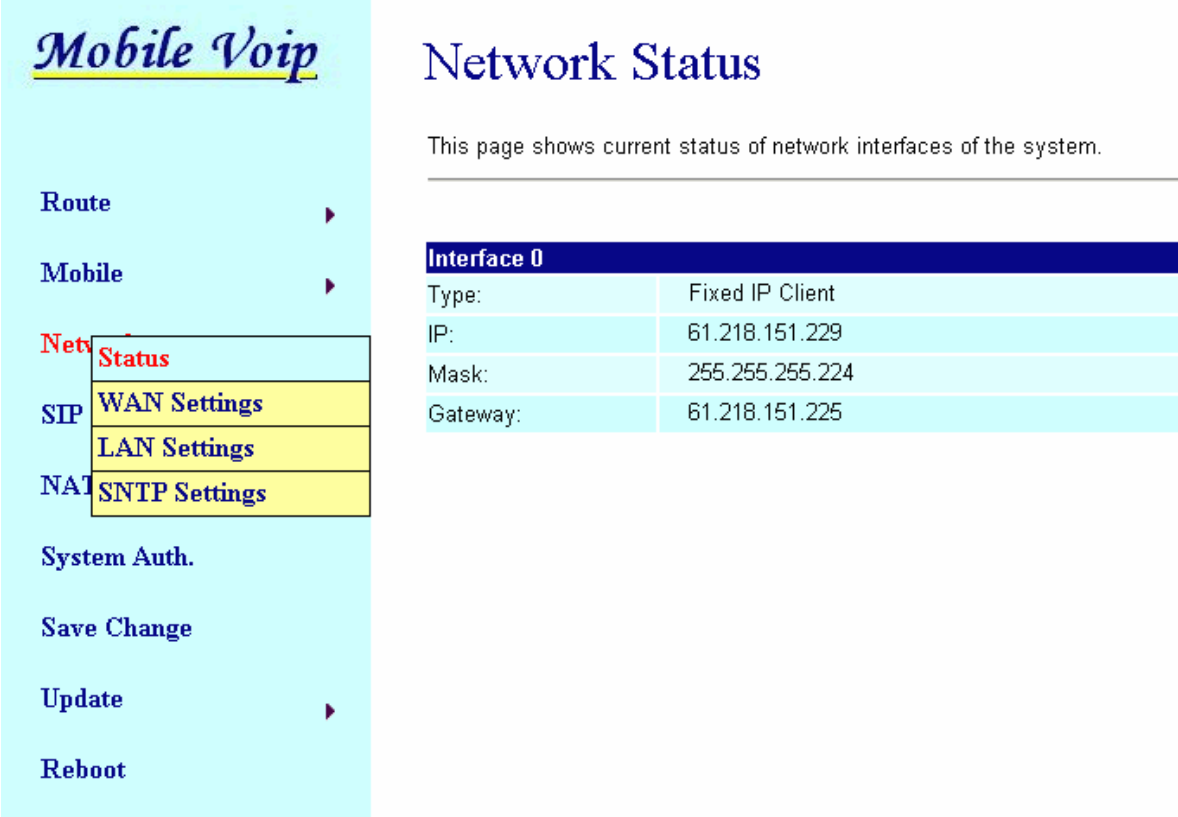


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## 11. Network

In Network you can check the Network status, configure the WLAN Settings , LAN Setting and SNTP settings.

11.1 Network Status: You can check the current Network setting in this page.



**Mobile Voip**

- Route
- Mobile
- Network Status**
- WAN Settings
- LAN Settings
- Sntp Settings
- System Auth.
- Save Change
- Update
- Reboot

### Network Status

This page shows current status of network interfaces of the system.

Interface 0	
Type:	Fixed IP Client
IP:	61.218.151.229
Mask:	255.255.255.224
Gateway:	61.218.151.225

11.2 WAN Settings: You can check the current Network setting in this page.

- (1) The TCP/IP Configuration item is to setup the WAN port's network environment. You may refer to your current network environment to configure the system properly.
- (2) The PPPoE Configuration item is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.
- (3) The Bridge Item is to setuo the system Bridge mode Enable/Disable. If you set the Bridge On, then the two Fast Ethernet ports will be transparent.
- (4) When you finished the setting, please click the Submit button.

**Mobile Voip**

Route

Mobile

Network

- Status
- WAN Settings**
- LAN Settings
- SNTP Settings

SIP

NAT

System Auth.

Save Change

Update

Reboot

## WAN Settings

You could configure the LAN settings in this page.

LAN Mode:  Bridge  NAT

### WAN Setting

IP Type:  Fixed IP  DHCP Client  PPPoE

IP:

Mask:

Gateway:

DNS Server1:

DNS Server2:

MAC:

### PPPoE Setting

User Name:

Password:

11.3 LAN Settings: You can check the current Network setting in this page.

- (1) The TCP/IP Configuration item is to setup the WAN port's network environment. You may refer to your current network environment to configure the system properly.
- (2)DHCP Server: You may refer to your current network environment to configure the system properly

**Mobile Voip**

- Route ▶
- Mobile ▶
- Network
  - Status
  - WAN Settings
  - LAN Settings**
  - SNTP Settings
- System Auth.
- Save Change
- Update ▶
- Reboot

## LAN Settings

You could configure the LAN settings in this page.

---

LAN Setting	
IP:	192.168.0.102
Mask:	255.255.255.0
MAC:	00037e000001

DHCP Server	
DHCP Server:	<input type="radio"/> On <input checked="" type="radio"/> Off
Start IP:	150
End IP:	200
Lease Time:	1 : 0 (dd:hh)

## 11.4 SNTP Settings:

SNTP Setting function: you can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.

### Mobile Voip

- Route
- Mobile
- Network
  - Status
  - SIP
    - WAN Settings
    - LAN Settings
    - SNTP Settings**
  - NAT
- System Auth.
- Save Change
- Update
- Reboot

## SNTP Settings

You could set the SNTP servers in this page.

---

**SNTP:**  On  Off

Primary Server:	<input type="text" value="time.windows.com"/>
Secondary Server:	<input type="text" value="208.184.49.9"/>
Time Zone:	GMT <input type="text" value="+"/> <input type="text" value="08"/> : <input type="text" value="00"/> (hh:mm)
Sync. Time:	<input type="text" value="1"/> : <input type="text" value="0"/> : <input type="text" value="0"/> (dd:hh:mm)

---

## 12.SIP Setting

In SIP Setting you can setup the Service Domain,Port Settings,Codec Settings,RTP setting,RPort Setting and Other SettingS. If the VoIP service is provided by ISP,you need to setup the related informations correctly then you can register to SIP Proxy Server correctly.

12.1 In Servcie Domain Function you need to input the account and the related informations in this page,please refer to your ISP Provider. You can register three SIP accounts . You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from the tree SIP account.

First you need to click Active to enable the Service Domain,then you can input the following items.

- (1)No.,: choose Mobile 1 or Mobile 2
- (2) Display name: you can input the name you want to display.
- (3) User name: you need to input the User Name get from your ISP.
- (4) Register Name: you need to input the Register Name get from your ISP.
- (5) Register Password: you need to input the Register Password get from ISP.
- (6) Domain Server:you need to input the Domain Server get from your ISP.
- (7) Proxy Server:you need to input the Proxy Server get from your ISP.
- (8) Outbound Proxy: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information,then you can skip this item.
- (9) You can see the Register Status in the Status item.
- (10) When you finished the setting,please click the Submit button.  
Remember to click "Save Charge"

## Mobile Voip

Route

Mobile

Network

SIP Settings

NAT Trans.

System Auth.

Save Change

Update

Reboot

## Service Domain Settings

You could set information of service domains in this page.

No.:

### Realm 1 (Default)

Active:  On  Off

Display Name:

User Name:

Register Name:

Register Password:

Domain Server:

Proxy Server:

Outbound Proxy:

Status: Registered

Example:

Register VoipBuster

### Realm 1 (Default)

Active:  On  Off

Display Name:

User Name:  Your Voipbuster username

Register Name:

Register Password:  Your Voipbuster password

Domain Server:

Proxy Server:  Proxy Server's IP

Outbound Proxy:

Status: Registered

## 12.2 Port Setting

You can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTP port setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.

**Mobile Voip**

- Route ▶
- Mobile ▶
- Network ▶
- SIP Settings**
  - Service Domain
  - Port Settings**
  - Codec Settings
  - Codec ID Setting
  - DTMF Setting
  - RPort Setting
  - Other Settings
- NAT
- System
- Save
- Update
- Reboot

### Port Settings

You could set the port number in this page.

SIP Port of Mobile1:	<input type="text" value="5060"/>	(10~65533)
RTP Port of Mobile1:	<input type="text" value="60000"/>	(10~65533)
SIP Port of Mobile2:	<input type="text" value="5062"/>	(10~65533)
RTP Port of Mobile2:	<input type="text" value="60100"/>	(10~65533)

### 12.3 Codec Settings:

You can setup the Codec priority, RTP packet length in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.

## Mobile Voip

- Route ▶
- Mobile ▶
- Network ▶
- SIP S**
  - Service Domain
- NAT**
  - Port Settings
  - Codec Settings**
- System**
  - Codec ID Setting
- Save**
  - DTMF Setting
  - RPort Setting
- Update**
  - Other Settings
- Reboot

## Codec Settings

You could set the codec settings in this page.

Codec Priority	
Codec Priority 1:	G.711 u-law ▼
Codec Priority 2:	G.711 a-law ▼
Codec Priority 3:	G.729 ▼
Codec Priority 4:	G.723 ▼
Codec Priority 5:	G.726 - 16 ▼
Codec Priority 6:	G.726 - 24 ▼
Codec Priority 7:	G.726 - 32 ▼
Codec Priority 8:	G.726 - 40 ▼

RTP Packet Length	
G.711 & G.729:	20 ms ▼
G.723:	30 ms ▼

G.723 5.3K	
G.723 5.3K:	<input type="radio"/> On <input checked="" type="radio"/> Off



## 12.4 Codec ID Setting

You can setup the Codec ID in this page.

**Mobile Voip**

- Route
- Mobile
- Network
- SIP Settings**
  - Service Domain
  - Port Settings
  - Codec Settings
  - Codec ID Setting**
  - DTMF Setting
  - RPort Setting
  - Other Settings
- Save
- Update
- Reboot

### Codec ID Setting

You could set the value of Codec ID in this page.

Codec Type	ID	Default Value
G726-16 ID:	<input type="text" value="23"/> (95~255)	<input checked="" type="checkbox"/> 23
G726-24 ID:	<input type="text" value="22"/> (95~255)	<input checked="" type="checkbox"/> 22
G726-32 ID:	<input type="text" value="2"/> (95~255)	<input checked="" type="checkbox"/> 2
G726-40 ID:	<input type="text" value="21"/> (95~255)	<input checked="" type="checkbox"/> 21
RFC 2833 ID:	<input type="text" value="101"/> (95~255)	<input checked="" type="checkbox"/> 101

---

## 12.5 DTMF Setting

You can setup the DTMF Setting in this page.

**Mobile Voip**

- Route
- Mobile
- Network
- SIP Settings**
  - Service Domain
  - Port Settings
  - Codec Settings
  - Codec ID Setting
  - DTMF Setting**
  - RPort Setting
  - Other Settings
- NAT
- System
- Save
- Update
- Reboot

### DTMF Setting

You could set the DTMF setting in this page.

2833

Inband DTMF

Send DTMF SIP Info

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---

## 12.6 RPort Function:

You can setup the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

**Mobile Voip**

Route ▶

Mobile ▶

Network ▶

SIP S

NAT

System

Save

Update

Reboot

**RPort Setting**

You could enable/disable the RPort setting in this page.

RPort of Mobile1:  On  Off

RPort of Mobile2:  On  Off

Submit Reset

## 12.7 Other Settings

Other Settings: you can setup the Hold by RFC and QoS in this page. To change these settings, please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.

<u>Mobile Voip</u>		Other Settings	
Route ▶		You could set other settings in this page.	
Mobile ▶			
Network ▶			
SIP Settings		Hold by RFC of Mobile1: <input type="radio"/> On <input checked="" type="radio"/> Off	
NAT		Hold by RFC of Mobile2: <input type="radio"/> On <input checked="" type="radio"/> Off	
System		Voice QoS: <input type="text" value="40"/> (0~63)	
Save		SIP QoS: <input type="text" value="40"/> (0~63)	
Update		SIP Expire Time: <input type="text" value="300"/> (60~86400 sec)	
Reboot		<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

---

## 13. NAT Trans

In NAT Trans. you can setup STUN and uPnP function. These functions can help your VoIP device working properly behind NAT.

13.1 STUN Setting: you can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your VoIP device working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

**Mobile Voip**

- Route
- Mobile
- Network
- SIP Settings
- NAT Trans** **STUN Setting**
- System Auth.
- Save Change
- Update
- Reboot

### STUN Setting

You could set the IP of STUN server in this page.

---

STUN of Mobile1:	<input type="radio"/> On <input checked="" type="radio"/> Off
STUN of Mobile2:	<input type="radio"/> On <input checked="" type="radio"/> Off
STUN Server:	<input type="text" value="stun.xten.com"/>
STUN Port:	<input type="text" value="3478"/> (1024~65535)

---

## 14. System Auth.

In System Authority you can change your login name and password.

The screenshot shows a web interface for 'Mobile Voip' with a sidebar menu and a 'System Authority' configuration section. The sidebar menu includes: Route, Mobile, Network, SIP Settings, NAT Trans., System Auth. (highlighted in red), Save Change, Update, and Reboot. The 'System Authority' section contains a heading, a descriptive text, and three input fields for 'New username:', 'New password:', and 'Confirmed password:'. Below the fields are 'Submit' and 'Reset' buttons.

<u>Mobile Voip</u>	
Route	▶
Mobile	▶
Network	▶
SIP Settings	▶
NAT Trans.	▶
<b>System Auth.</b>	
Save Change	
Update	▶
Reboot	

### System Authority

You could change the login username/password in this page.

New username:	<input type="text"/>
New password:	<input type="password"/>
Confirmed password:	<input type="password"/>

---

## 15. Save Change

In Save Change you can save the changes you have done. If you want to use new setting in the VoIP system, You have to click the Save button. After you click the Save button, the system will automatically restart and the new setting will effect.



The screenshot shows a web interface for 'Mobile Voip'. On the left is a light blue sidebar menu with the following items: 'Route', 'Mobile', 'Network', 'SIP Settings', 'NAT Trans.', 'System Auth.', 'Save Change' (highlighted in red), 'Update', and 'Reboot'. Each item has a right-pointing arrow. The main content area has a white background with the title 'Save Changes' in blue. Below the title is a horizontal line and the text 'You have to save changes to effect them.' Below this is a 'Save Changes:' label followed by a grey 'Save' button.

---

## 16.Update

In Update you can update the system's firmware to the new one or do the factory reset to let the system back to default setting.

### 16.1 Update firmware

- (1) In New Firmware function you can update new firmware via HTTP in this page. You can upgrade the firmware by the following steps:
- (2) Select the firmware code type, Risc code.
- (3) Click the "Browse" button in the right side of the File Location or you can type the correct path and the filename in File Location blank.
- (4) Select the correct file you want to download to the system then click the Update button.

The screenshot shows the 'Update Firmware' page of a 'Mobile Voip' system. On the left is a navigation menu with options: Route, Mobile, Network, SIP Settings, NAT Trans., System Auth., Save Change, Update (highlighted), and Reboot. The 'Update' menu item is expanded to show 'New Firmware' (highlighted) and 'Default Settings'. The main content area is titled 'Update Firmware' and contains the text 'You could update the newest firmware.' Below this, there are two sections: 'HTTP' and 'TFTP'. The 'HTTP' section has a 'Code Type' dropdown menu set to 'Risc' and a 'File Location' input field with a '瀏覽...' (Browse) button. The 'TFTP' section has a 'TFTP Server' input field containing '192.168.1.250'. At the bottom of the form are 'Update' and 'Reset' buttons.



---

## 16.2 Restore Default Settings

Default Setting you can restore the system to factory default in this page. You can just click the Restore button, then the system will restore to default and automatically restart again.

**Mobile Voip**

- Route ▶
- Mobile ▶
- Network ▶
- SIP Settings ▶
- NAT Trans. ▶
- System Auth.
- Save Change
- Update
  - New Firmware
- Reboot
  - Default Settings

### Restore Default Settings

You could click the restore button to restore the factory settings.

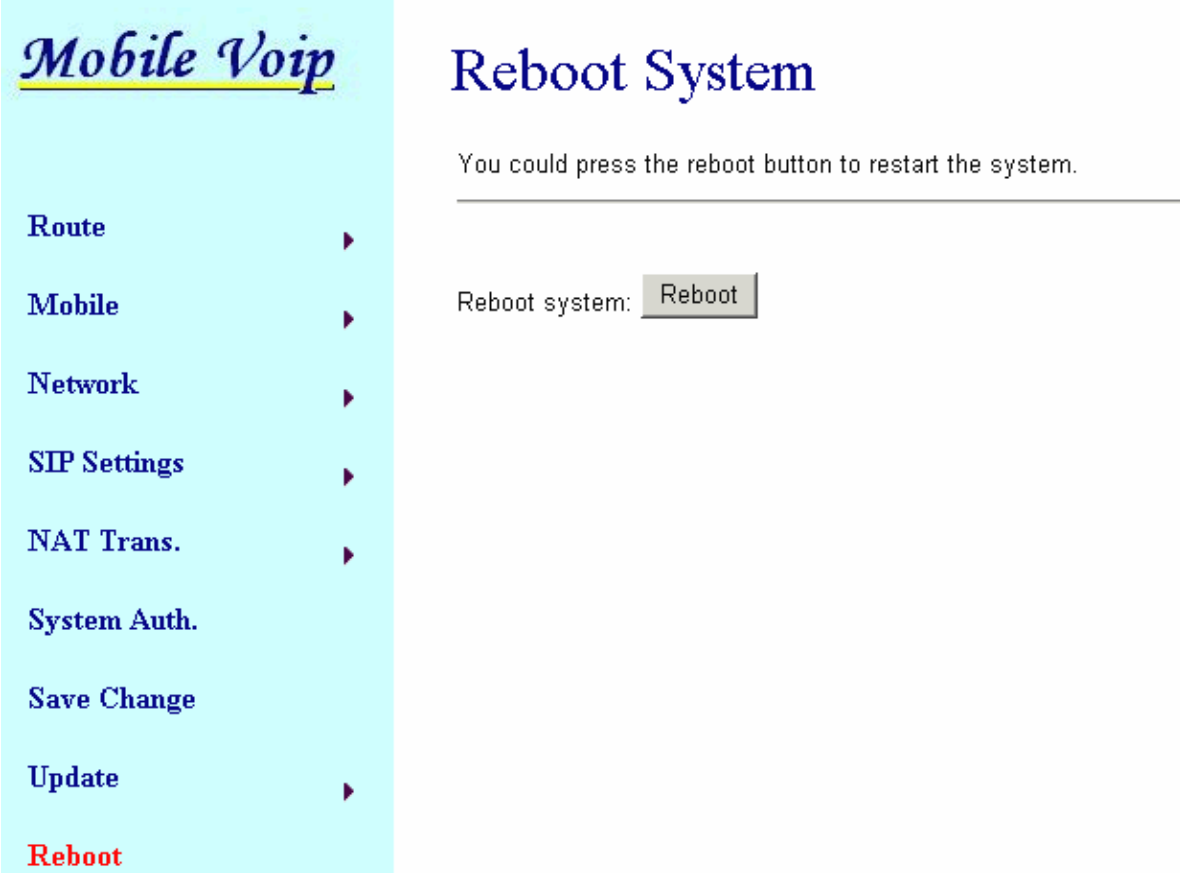
---

Restore default settings:

---

## 17.Reboot

Reboot function you can restart the system. If you want to restart the system, you can just click the Reboor button, then the system will automatically.



The screenshot shows a web interface for 'Mobile Voip'. On the left is a light blue sidebar menu with the following items: 'Route', 'Mobile', 'Network', 'SIP Settings', 'NAT Trans.', 'System Auth.', 'Save Change', 'Update', and 'Reboot' (highlighted in red). The main content area is titled 'Reboot System' and contains the text 'You could press the reboot button to restart the system.' followed by a horizontal line. Below the line, it says 'Reboot system:' followed by a grey 'Reboot' button.

---

---

## 18. IP Setting

The operator can setup or query the network parameters by dialing in the mobile number which it SIM card has been put in the main body. The status or result is response by voice. In the first 20 seconds after power-on, the VoIP GSM Gateway enters the IP setting mode. The operator may dial in the mobile number during this period to set or query the network parameters.

Item	IVR Action	IVR Menu Choice	Notes
1	Reboot	#195#	After you hear "Option Successful," hang-up. Unit will reboot automatically.
2	Factory Reset	#198#	System will automatically Reboot.WARNING: ALL User-Changeable" NONDEFAULT SETTINGS WILL BE LOST! This will include network and service provider data.
3	Check IP Address	#120#	IVR will announce the current IP address , Default : 192.168.0.100
4	Check IP Type	#121#	IVR will announce if DHCP is enabled or disabled. default : OFF
5	Check Network Mask	#123#	IVR will announce the current network mask.Default : 255.255.255.0
6	Check Gateway IP Address	#124#	IVR will announce the current gateway IP address, Default : 192.168.0.254
7	Check Primary	#125#	IVR will announce the current

	DNS Server		setting in the Primary DNS field. Default : 192.168.0.1
8	Check Firmware Version	#128#	IVR will announce the version of the firmware running
9	Set as DHCP client	#111#	The system will change to DHCP Client type
10	Set Static IP Address	#112xxx*xxx*xxx*xxx#	DHCP will be disabled and system will change to the Static IP type. Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.
11	Set Network Mask	#113xxx*xxx*xxx*xxx#	Must set Static IP first. Enter value using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.
12	Set Gateway IP Address	#114xxx*xxx*xxx*xxx#	Must set Static IP first. Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.
13	Set Primary DNS Server	#115xxx*xxx*xxx*xxx#	Must set Static IP first. Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.

---

## **19.Specification**

### 19.1 Protocols

SIP (RFC2543,RFC3261)

### 19.2 TCP/IP

IP/TCP/UDP/RTP/RTCP/

CMP/ARP/RARP/SNTP

DHCP/DNS Client

IEEE802.1P/Q

ToS/DiffServ

NAT Traversal

STUN

uPnP

IP Assignment

Static IP

DHCP

PPPoE

### 19.3 Codec

G.711 u-Law

G.711 a-Law

G.723.1 (5.3k)

G.723.1 (6.3k)

G.729A

G.729A/B

### 19.4 Voice Quality

VAD

CNG

---

AEC, LEC

Packet loss

19.5 GSM (MOBILE VOIP)

Dual BAND: 900/1800 MHZ

Tri BAND: 900/1800/1900 MHZ

Quad BAND: 900/1800/1900/850 MHZ

## 20. Setup Mobile VoIP-2 with Asterisk

Test version

trixbox-2.2

SIP Softphone

- SJPhone 1.60.289a
- X-Lite 1105x

Modify file

- Add the following setting to/etc/asterisk/sip.conf

[1000]

type=friend

secret=1000

qualify=yes

nat=yes

host=dynamic

canreinvite=no

context=internal

[1001]

type=friend

secret=1001

qualify=yes

---

```
nat=yes
host=dynamic
canreinvite=no
context=internal
```

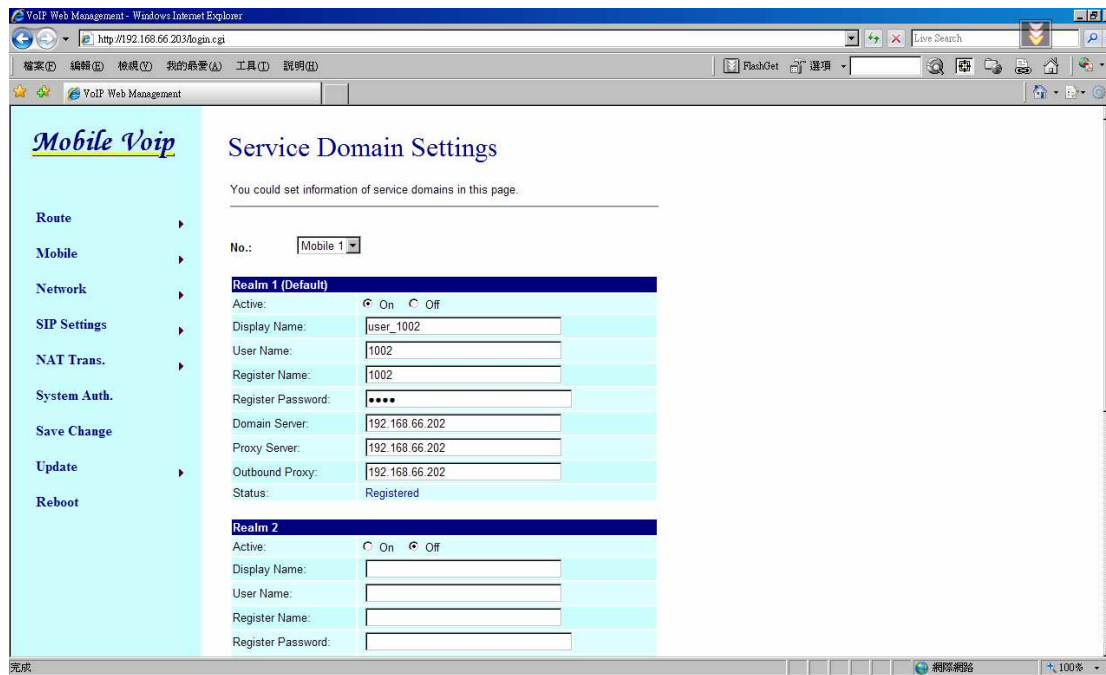
```
[1002]
type=friend
secret=1002
qualify=yes
nat=yes
host=dynamic
canreinvite=no
context=internal
```

- Add the following setting to `/etc/asterisk/extensions.conf`

```
[internal]
exten => 1000,1,Dial(SIP/1000)
exten => 1001,1,Dial(SIP/1001)
exten => 1002,1,Dial(SIP/1002)
```

**configure:**

```
trixbox-2.2: address=192.168.66.202:5060
SJPhone: address=192.168.66.145:5060; username=1000,
displayname=user_1000
X-Lite: address=192.168.66.145:7331; username=1001, displayname=user_1001
MOBILE VOIP-2: address=192.168.66.203:5060; username=1002,
displayname=user_1002
```



test1

pstn → call 0928492911(mobile number) → MOBILE VOIP-2 → hear the second dial tone, call SoftPhone's number → SoftPhone → show pstn caller id

This Is X-Lite receiving packet, red word is pstn number. Test ok.

```
INVITE sip:1001@192.168.66.145:7331 SIP/2.0
Via: SIP/2.0/UDP 192.168.66.202:5060;branch=z9hG4bK3d0bbaf7;rport
From: "035678238" <sip:1002@192.168.66.202>;tag=as580472a7
To: <sip:1001@192.168.66.145:7331>
Contact: <sip:1002@192.168.66.202>
Call-ID: 20fa417265e6a26d0b0aae4f551f06f3@192.168.66.202
CSeq: 102 INVITE
User-Agent: Asterisk PBX
Max-Forwards: 70
Date: Tue, 22 May 2007 02:50:37 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
```



---

Content-Type: application/sdp  
Content-Length: 242

v=0  
o=root 2737 2737 IN IP4 192.168.66.202  
s=session  
c=IN IP4 192.168.66.202  
t=0 0  
m=audio 15852 RTP/AVP 0 8 101  
a=rtpmap:0 PCMU/8000  
a=rtpmap:8 PCMA/8000  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-16  
a=silenceSupp:off - - - -

SIP/2.0 200 Ok  
Via: SIP/2.0/UDP 192.168.66.202:5060;branch=z9hG4bK3d0bbaf7;rport  
From: "035678238" <sip:1002@192.168.66.202>;tag=as580472a7  
To: <sip:1001@192.168.66.145:7331>;tag=677373503  
Contact: <sip:1001@192.168.66.145:7331>  
Call-ID: 20fa417265e6a26d0b0aae4f551f06f3@192.168.66.202  
CSeq: 102 INVITE  
Content-Type: application/sdp  
Server: X-Lite release 1105x  
Content-Length: 254

v=0  
o=1001 4804366 4807851 IN IP4 192.168.66.145  
s=X-Lite  
c=IN IP4 192.168.66.145  
t=0 0  
m=audio 8000 RTP/AVP 0 8 3 101  
a=rtpmap:0 pcmu/8000

---

a=rtpmap:8 pcma/8000  
a=rtpmap:3 gsm/8000  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
a=sendrecv

test 2

SoftPhone → call 1002 → MOBILE VOIP-2 → hear second dial tone and call pstn →  
pstn answer → show caller id-mobile number 0928492911

This Is X-Lite receiving packet. Test ok.

INVITE sip:1002@192.168.66.202 SIP/2.0  
Via: SIP/2.0/UDP  
192.168.66.145:7331;rport;branch=z9hG4bK4C4315351FC84CA582D14FB8C25F  
C3BF  
From: user\_1001 <sip:1001@192.168.66.202:7331>;tag=1121869743  
To: <sip:1002@192.168.66.202>  
Contact: <sip:1001@192.168.66.145:7331>  
Call-ID: F4B32CA6-1835-4E68-941A-C685B39C43FF@192.168.66.145  
CSeq: 63148 INVITE  
Proxy-Authorization: Digest  
username="1001",realm="asterisk",nonce="0d3b2879",response="8aaaaa5b5ad53  
654bf0a2ab0fa9bb118",uri="sip:1002@192.168.66.202",algorithm=MD5  
Max-Forwards: 70  
Content-Type: application/sdp  
User-Agent: X-Lite release 1105x  
Content-Length: 254

v=0  
o=1001 5111461 5111501 IN IP4 192.168.66.145  
s=X-Lite

---

c=IN IP4 192.168.66.145  
t=0 0  
m=audio 8000 RTP/AVP 0 8 3 101  
a=rtpmap:0 pcmu/8000  
a=rtpmap:8 pcma/8000  
a=rtpmap:3 gsm/8000  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
a=sendrecv

SIP/2.0 200 OK  
Via: SIP/2.0/UDP  
192.168.66.145:7331;branch=z9hG4bK4C4315351FC84CA582D14FB8C25FC3BF  
;received=192.168.66.145;rport=7331  
From: user\_1001 <sip:1001@192.168.66.202:7331>;tag=1121869743  
To: <sip:1002@192.168.66.202>;tag=as2a2fbf98  
Call-ID: F4B32CA6-1835-4E68-941A-C685B39C43FF@192.168.66.145  
CSeq: 63148 INVITE  
User-Agent: Asterisk PBX  
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY  
Contact: <sip:1002@192.168.66.202>  
Content-Type: application/sdp  
Content-Length: 242

v=0  
o=root 2737 2737 IN IP4 192.168.66.202  
s=session  
c=IN IP4 192.168.66.202  
t=0 0  
m=audio 13798 RTP/AVP 0 8 101  
a=rtpmap:0 PCMU/8000  
a=rtpmap:8 PCMA/8000  
a=rtpmap:101 telephone-event/8000

---

---

a=fmtp:101 0-16  
a=silenceSupp:off - - - -

register issue

The packet date from Asterisk as follows.  
Please note, user\_1002's display name don't appear  
So the website's Display Name is not available

<-- SIP read from 192.168.66.203:5060:  
REGISTER sip:192.168.66.202 SIP/2.0  
Via: SIP/2.0/UDP  
192.168.66.203:5060;rport;branch=z9hG4bK590e92b551233a10a0ae71944c19b5  
aa  
From: <sip:1002@192.168.66.202>;tag=4e36d8f1  
To: <sip:1002@192.168.66.202>  
Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203  
Contact: <sip:1002@192.168.66.203:5060>  
CSeq: 10 REGISTER  
Expires: 300  
Authorization: Digest  
username="1002",realm="asterisk",nonce="3ca93a1e",response="4d39ccb0dae64  
bb2f1341e9896ac1ea7",uri="sip:192.168.66.202",algorithm=MD5  
User-Agent: CMI CM5K  
Content-Length: 0

--- (11 headers 0 lines) ---

Using latest REGISTER request as basis request  
Sending to 192.168.66.203 : 5060 (NAT)  
Transmitting (NAT) to 192.168.66.203:5060:  
SIP/2.0 100 Trying  
Via: SIP/2.0/UDP  
192.168.66.203:5060;branch=z9hG4bK590e92b551233a10a0ae71944c19b5aa;rec

---

eived=192.168.66.203;rport=5060  
From: <sip:1002@192.168.66.202>;tag=4e36d8f1  
To: <sip:1002@192.168.66.202>  
Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203  
CSeq: 10 REGISTER  
User-Agent: Asterisk PBX  
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY  
Contact: <sip:1002@192.168.66.202>  
Content-Length: 0

---

Transmitting (NAT) to 192.168.66.203:5060:  
SIP/2.0 401 Unauthorized  
Via: SIP/2.0/UDP  
192.168.66.203:5060;branch=z9hG4bK590e92b551233a10a0ae71944c19b5aa;received=192.168.66.203;rport=5060  
From: <sip:1002@192.168.66.202>;tag=4e36d8f1  
To: <sip:1002@192.168.66.202>;tag=as13a32ae8  
Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203  
CSeq: 10 REGISTER  
User-Agent: Asterisk PBX  
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY  
WWW-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="5def9231"  
Content-Length: 0

---

Scheduling destruction of call  
'7e45b773130f1fc945efcee502f84042@192.168.66.203' in 15000 ms  
asterisk1\*CLI>  
<-- SIP read from 192.168.66.203:5060:  
REGISTER sip:192.168.66.202 SIP/2.0

---

Via: SIP/2.0/UDP  
192.168.66.203:5060;rport;branch=z9hG4bK672fa67f59c2223275f5ee286d27597a  
From: <sip:1002@192.168.66.202>;tag=4e36d8f1  
To: <sip:1002@192.168.66.202>  
Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203  
Contact: <sip:1002@192.168.66.203:5060>  
CSeq: 11 REGISTER  
Expires: 300  
Authorization: Digest  
username="1002",realm="asterisk",nonce="5def9231",response="046a412f4e7ed4  
e98fd507416994a80a",uri="sip:192.168.66.202",algorithm=MD5  
User-Agent: CMI CM5K  
Content-Length: 0

--- (11 headers 0 lines) ---

Using latest REGISTER request as basis request  
Sending to 192.168.66.203 : 5060 (NAT)  
Transmitting (NAT) to 192.168.66.203:5060:  
SIP/2.0 100 Trying  
Via: SIP/2.0/UDP  
192.168.66.203:5060;branch=z9hG4bK672fa67f59c2223275f5ee286d27597a;received=192.168.66.203;rport=5060  
From: <sip:1002@192.168.66.202>;tag=4e36d8f1  
To: <sip:1002@192.168.66.202>  
Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203  
CSeq: 11 REGISTER  
User-Agent: Asterisk PBX  
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY  
Contact: <sip:1002@192.168.66.202>  
Content-Length: 0

---

---

---

12 headers, 0 lines

Reliably Transmitting (NAT) to 192.168.66.203:5060:

OPTIONS sip:1002@192.168.66.203:5060 SIP/2.0

Via: SIP/2.0/UDP 192.168.66.202:5060;branch=z9hG4bK7b92dd8a;rport

From: "Unknown" <sip:Unknown@192.168.66.202>;tag=as5dee3942

To: <sip:1002@192.168.66.203:5060>

Contact: <sip:Unknown@192.168.66.202>

Call-ID: 5ebc2211278e2cb7699911ad39454d4e@192.168.66.202

CSeq: 102 OPTIONS

User-Agent: Asterisk PBX

Max-Forwards: 70

Date: Tue, 22 May 2007 03:11:54 GMT

Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY

Content-Length: 0

---

Transmitting (NAT) to 192.168.66.203:5060:

SIP/2.0 200 OK

Via: SIP/2.0/UDP

192.168.66.203:5060;branch=z9hG4bK672fa67f59c2223275f5ee286d27597a;received=192.168.66.203;rport=5060

From: <sip:1002@192.168.66.202>;tag=4e36d8f1

To: <sip:1002@192.168.66.202>;tag=as13a32ae8

Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203

CSeq: 11 REGISTER

User-Agent: Asterisk PBX

Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY

Expires: 300

Contact: <sip:1002@192.168.66.203:5060>;expires=300

Date: Tue, 22 May 2007 03:11:54 GMT

Content-Length: 0

---

## 21. Simple Steps

Step 1. Change the Network setting if you need (Network/network setting)

Step 2. Register SIP proxy Server or Asterisk or VoipBuster if you need  
(sip setting/service domain)

Step 3. Set Route ( **request** )

<b>mobile to lan:</b>	
(1)	*,* --->it is two stage dialing.
	when mobile call in,MOBILE VOIP-2 will provide dial tone and you can enter ip or asterisk extension or phone number.
	* If you want to enter phone number,please note your asterisk need to have route of destination number.
(2)	*, specific extension or IP or phone number
	when mobile call in,MOBILE VOIP-2 will connect with this specific extension or IP or phone number auto
	* If you want to set specific phone number,please note your asterisk need to have route of destination number.
<b>Lan to Mobile:</b>	
(1)	*,* --->it is two stage dialing.
	when lan phone call in,MOBILE VOIP-2 will provide dial tone and you can enter mobile number.
(2)	*, specific mobile number
	when lan phone call in,MOBILE VOIP-2 will connect with the specific mobile number auto.
(3)	*,#--->It is 1 stage dialing
	When lan phone and MOBILE VOIP-2 both register Asterisk, you can dial any destination number from lan phone directly.
	* Please note:Asterisk need to set route of destination number that dial out from MOBILE VOIP-2

\* All changes both need to click "save and change"



