Mobile VoIP -2

User Manual

【Content】

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)

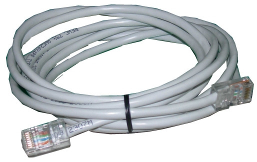
3.3 Network cable

3.4 Antenna

3.5 User Manual

(1)

(2)

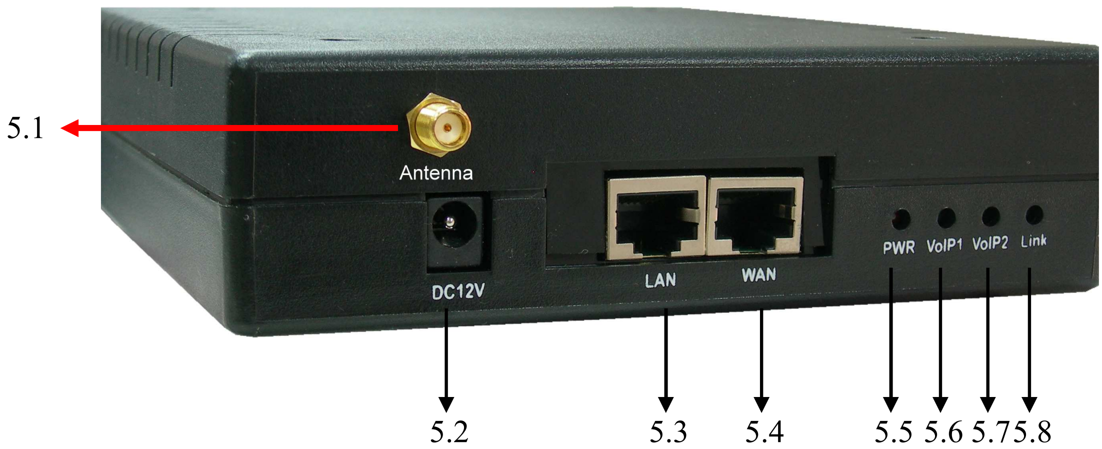


(3)

(4)

4.Dimension

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

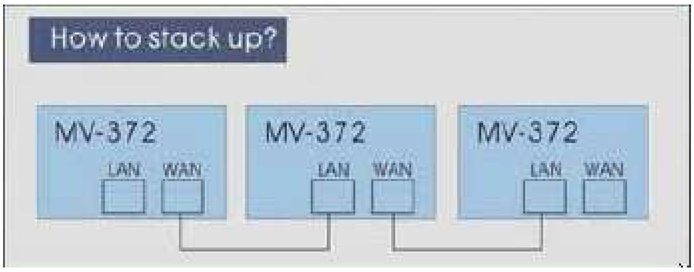
5.6 VoIP2：an indicator light of VoIP2

5.8 LINK Indicator：Light up when network is connected.

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.



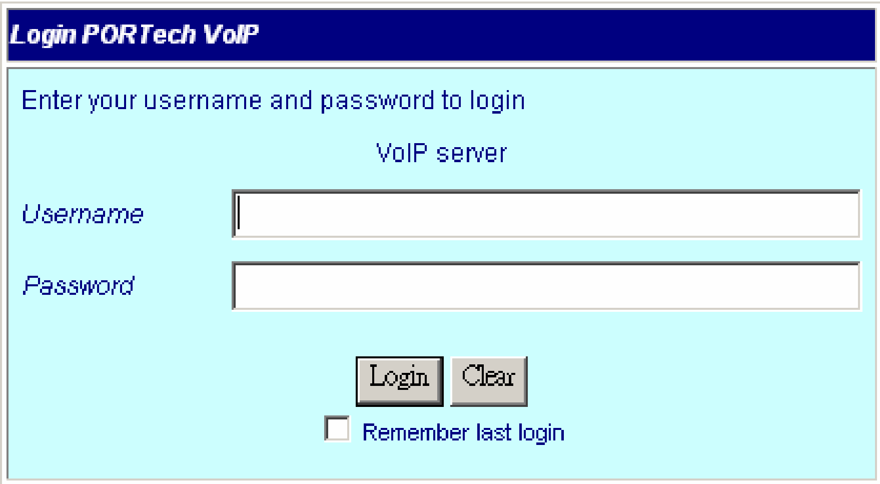
6.2 Connect the antenna and put it in proper position to get the best signal reception.

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.

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：

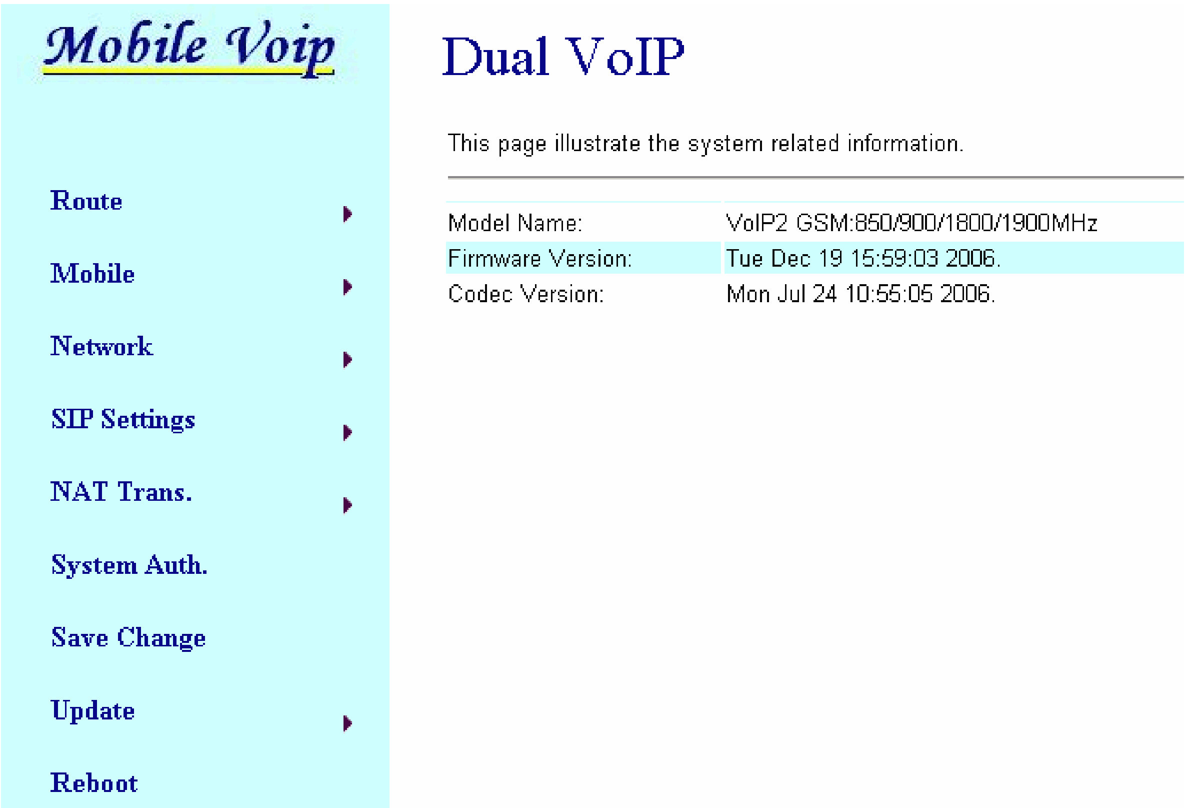


Enter the username and password for authentication. (default username=voip, password=1234). The page follows when the username and password are correct.

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.

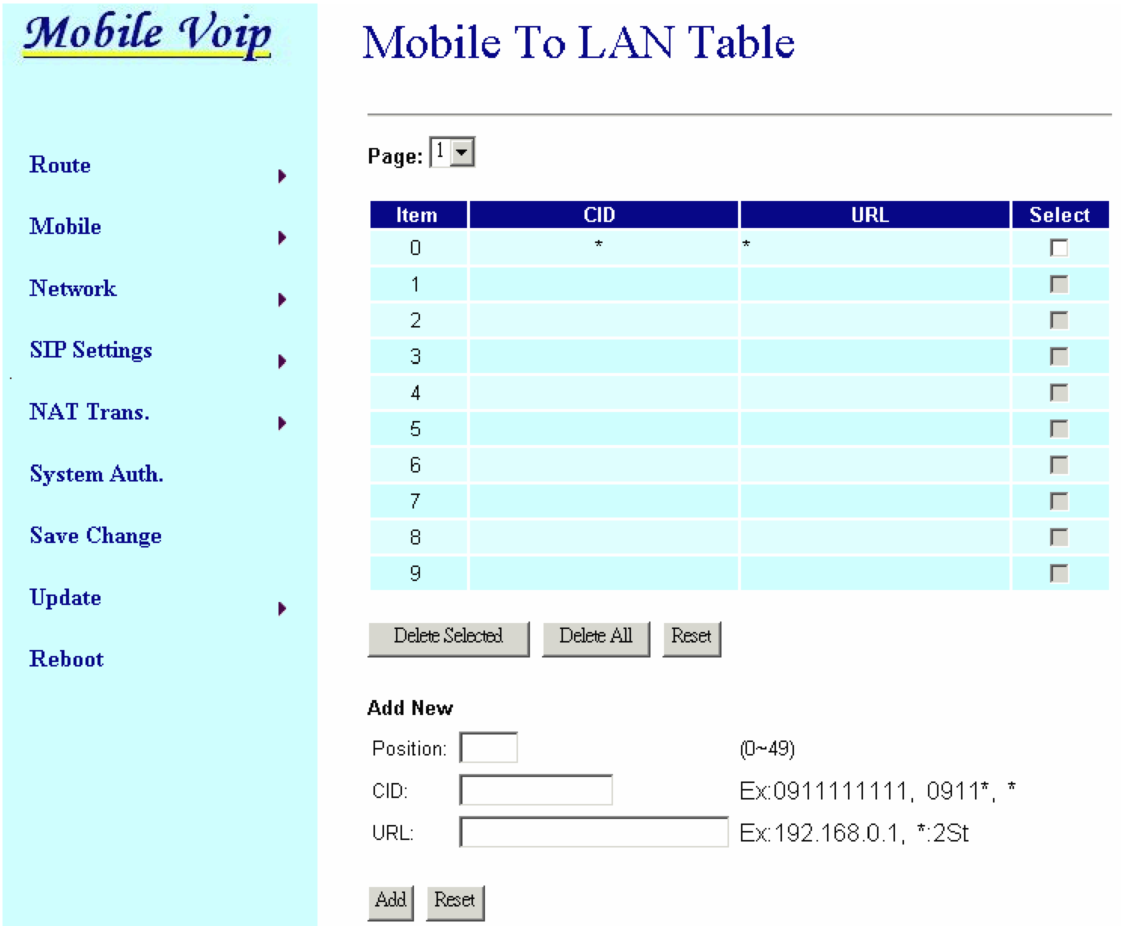
8.2 Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.



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.



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 automaticlly

(2) Mobile to Lan: *,*

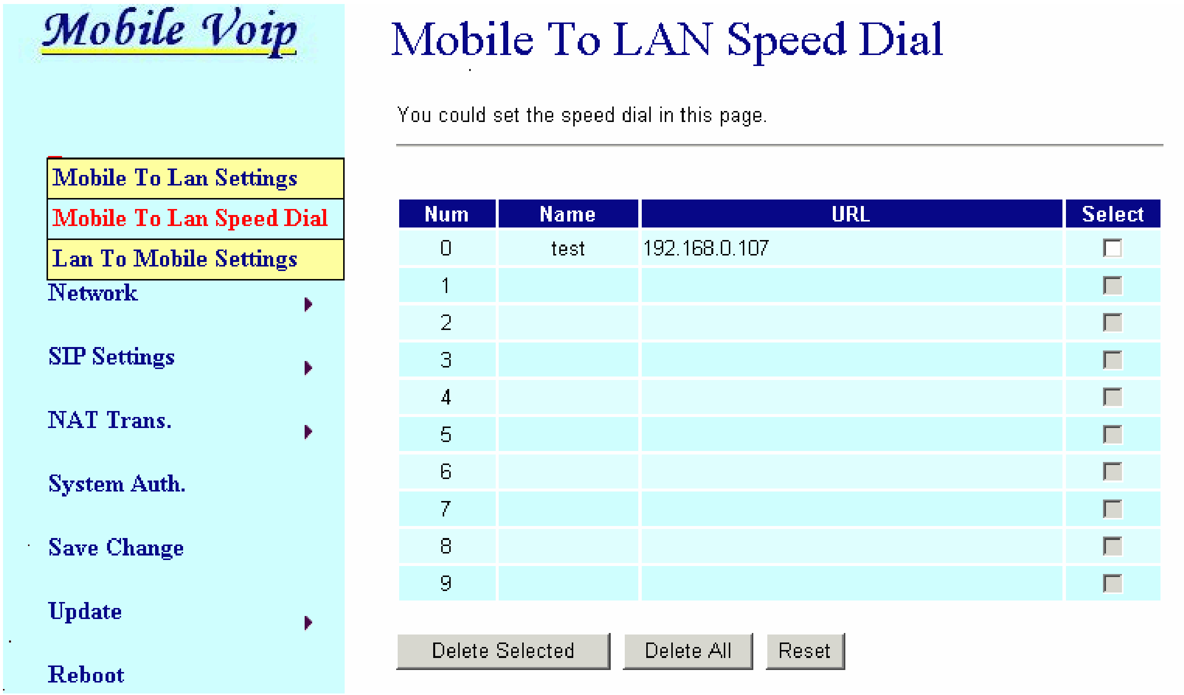
Any caller call the mobile voip’s sim,mobile voip will prompt dial tone.Caller can enter IP or sip extension or phone number.

*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.



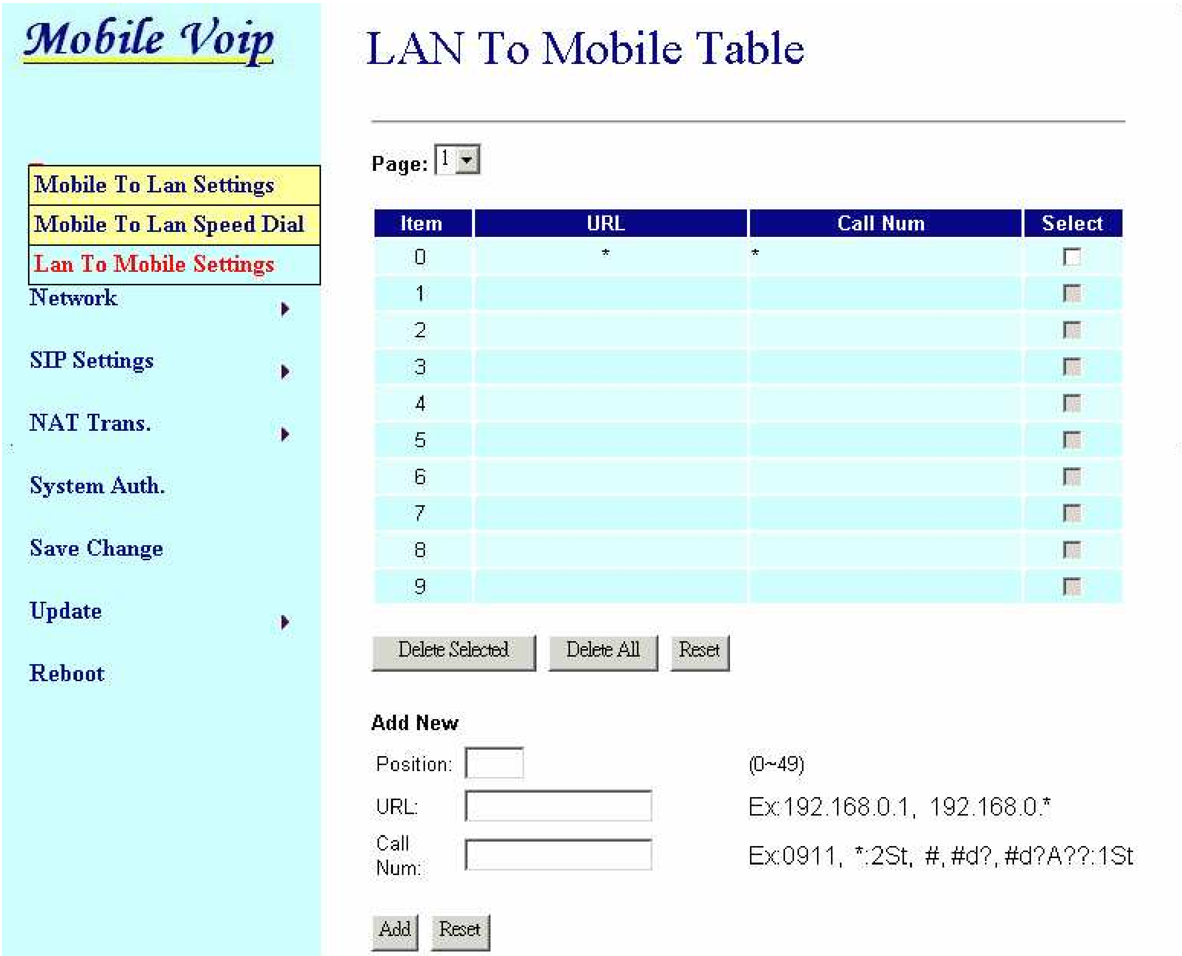
*The call will be answered and prompt dial tone again. When the caller may enter the “Num”, system will connect the “URL” as destination.

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 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.

*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 *,#

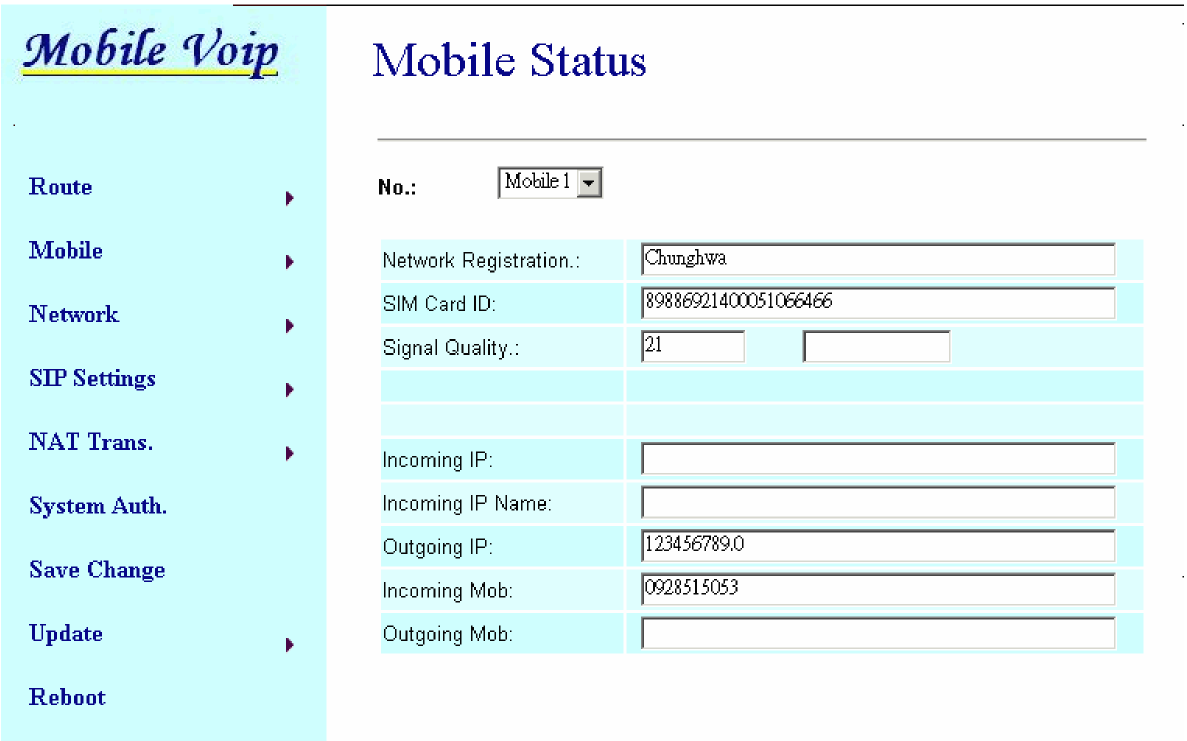
Additionally, two channels Mobile VoIP-2 both need to register proxy server or Asterisk.

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)

10.Mobile

10.1 Mobile Status



(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.

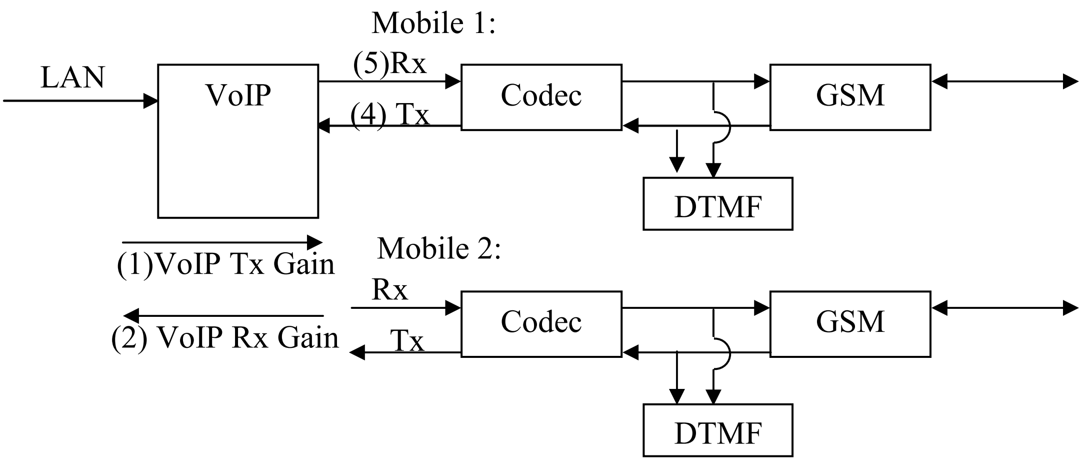
(5)Incoming IP Name: proxy server name

(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



(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

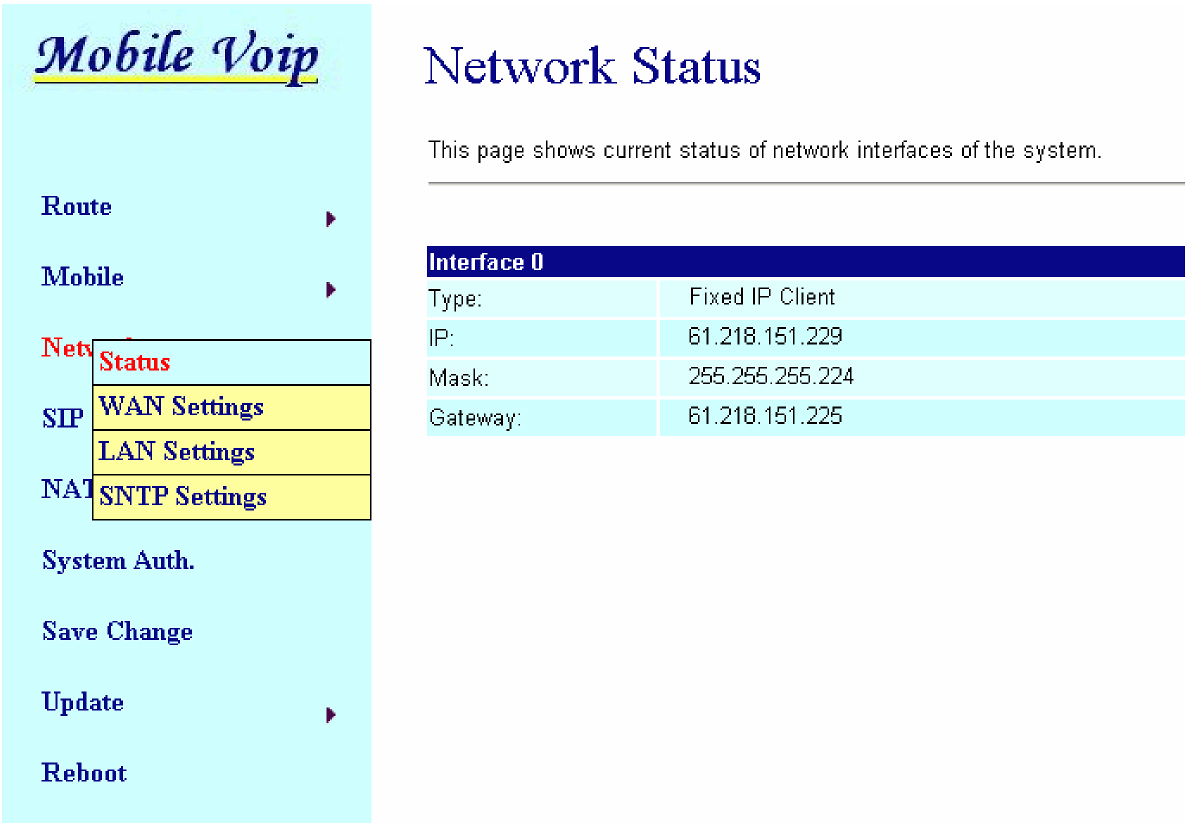
Income : when lan dial out,then connect soon

(10)Band Type:When you buy Quad band,you need to choose your GSM frequency

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.



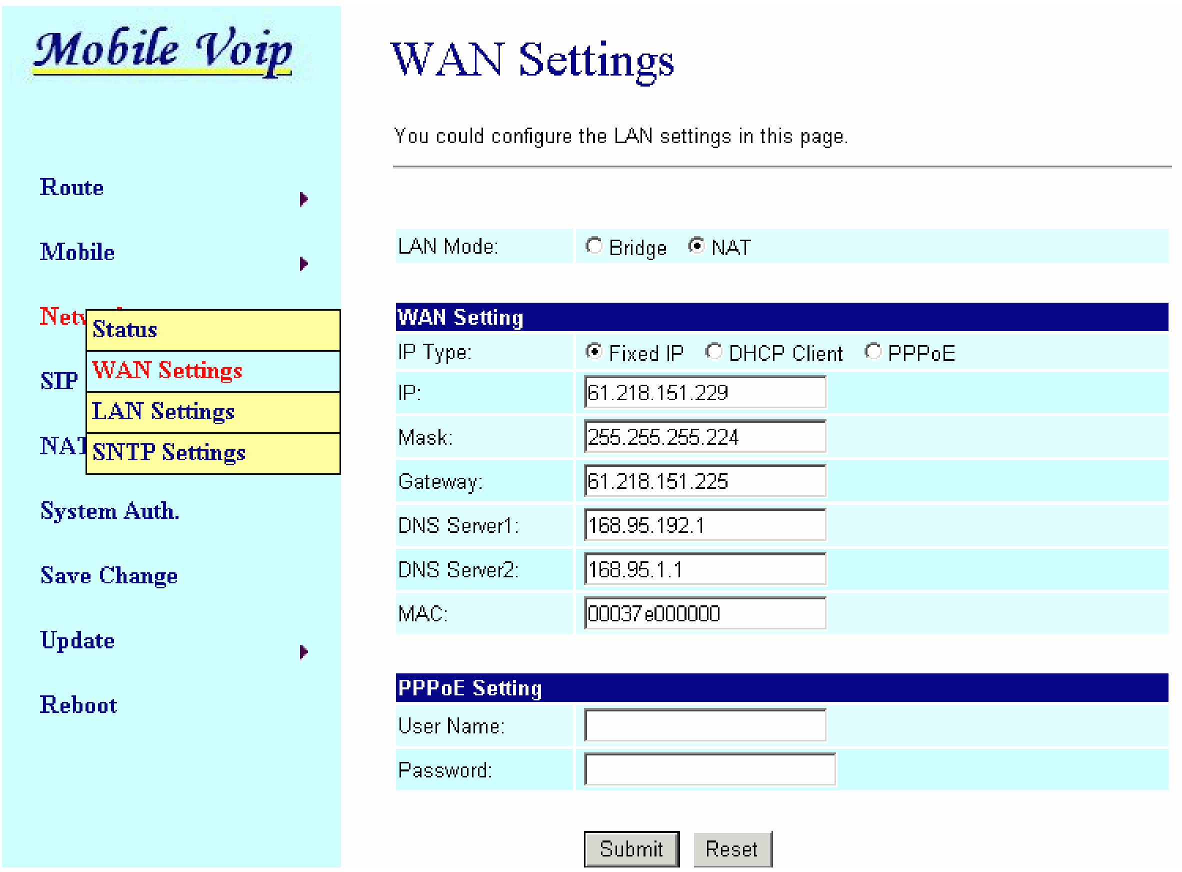
11.2 WAN Settings: You can check the current Network setting in this page.

(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.

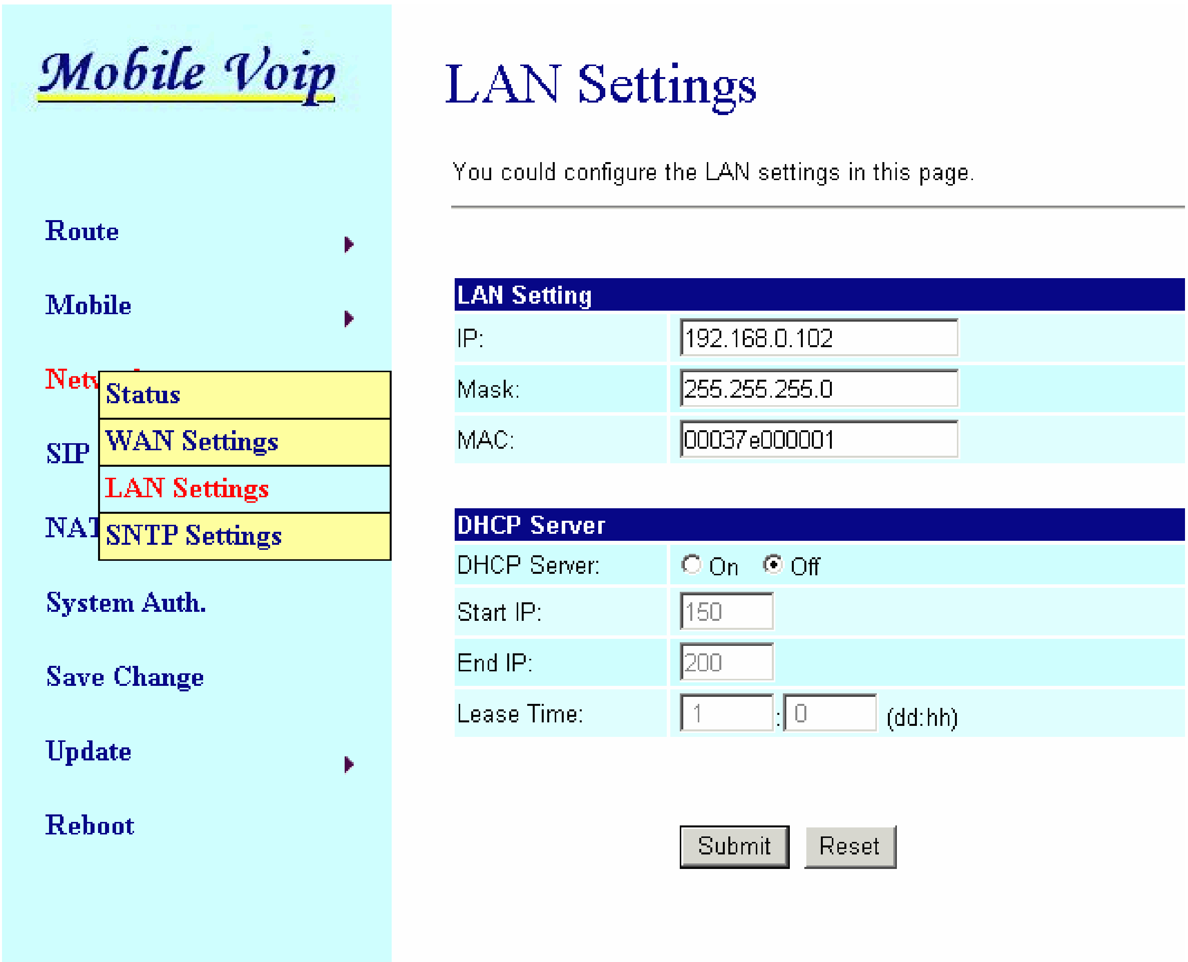
(4) When you finished the setting, please click the Submit button.



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

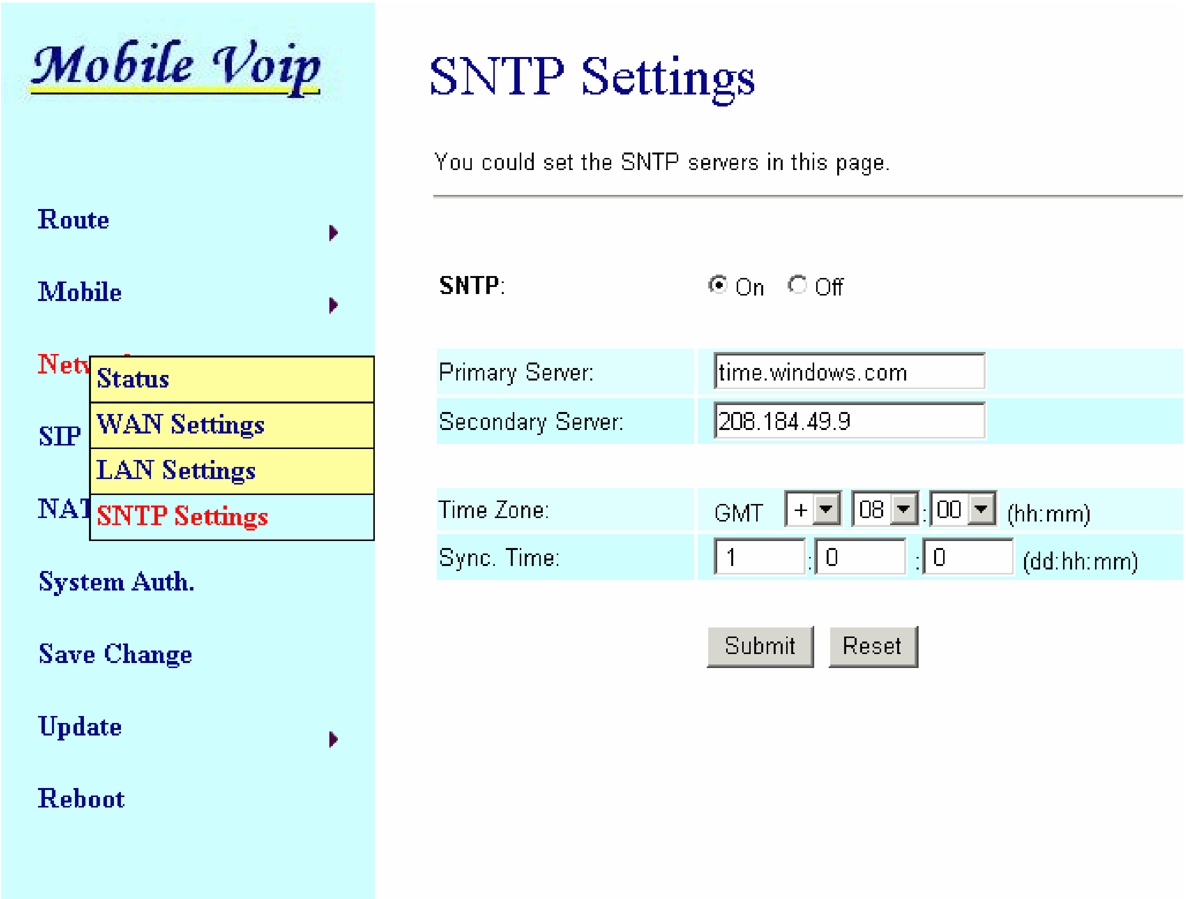
(1) The TCP/IP Configuration item is to setup the WAN port's network environment. You may refer to your current network environment to configure the system properly.

(2)DHCP Server: You may refer to your current network environment to configure the system properly



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.



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.

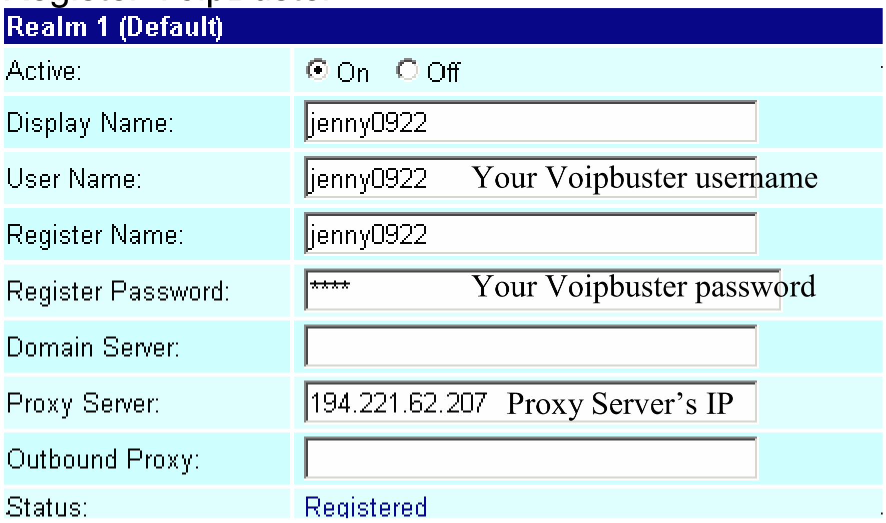
(7) Proxy Server:you need to input the Proxy Server get from your ISP.

(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.

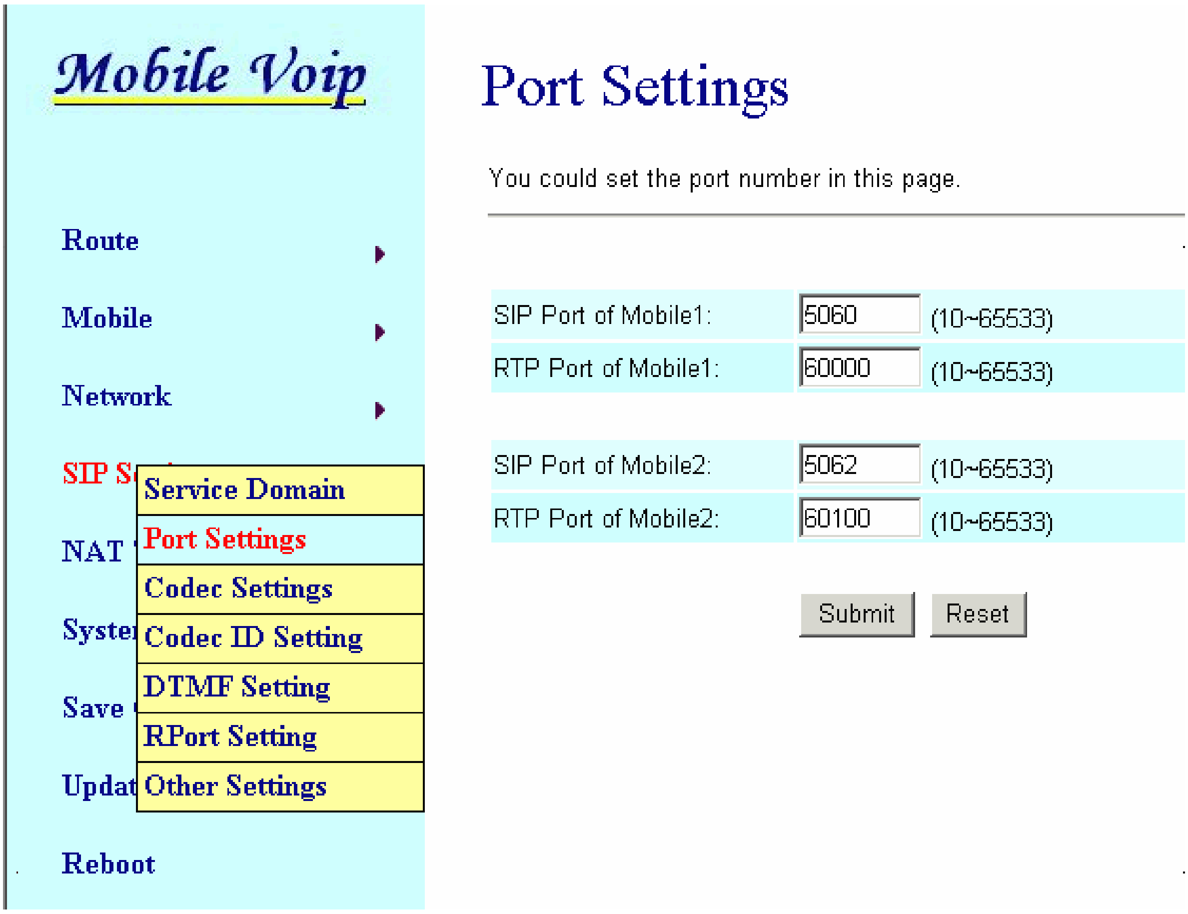
(10) When you finished the setting,please click the Submit button. Remember to click “Save Charge”

Example:
Register VoipBuster



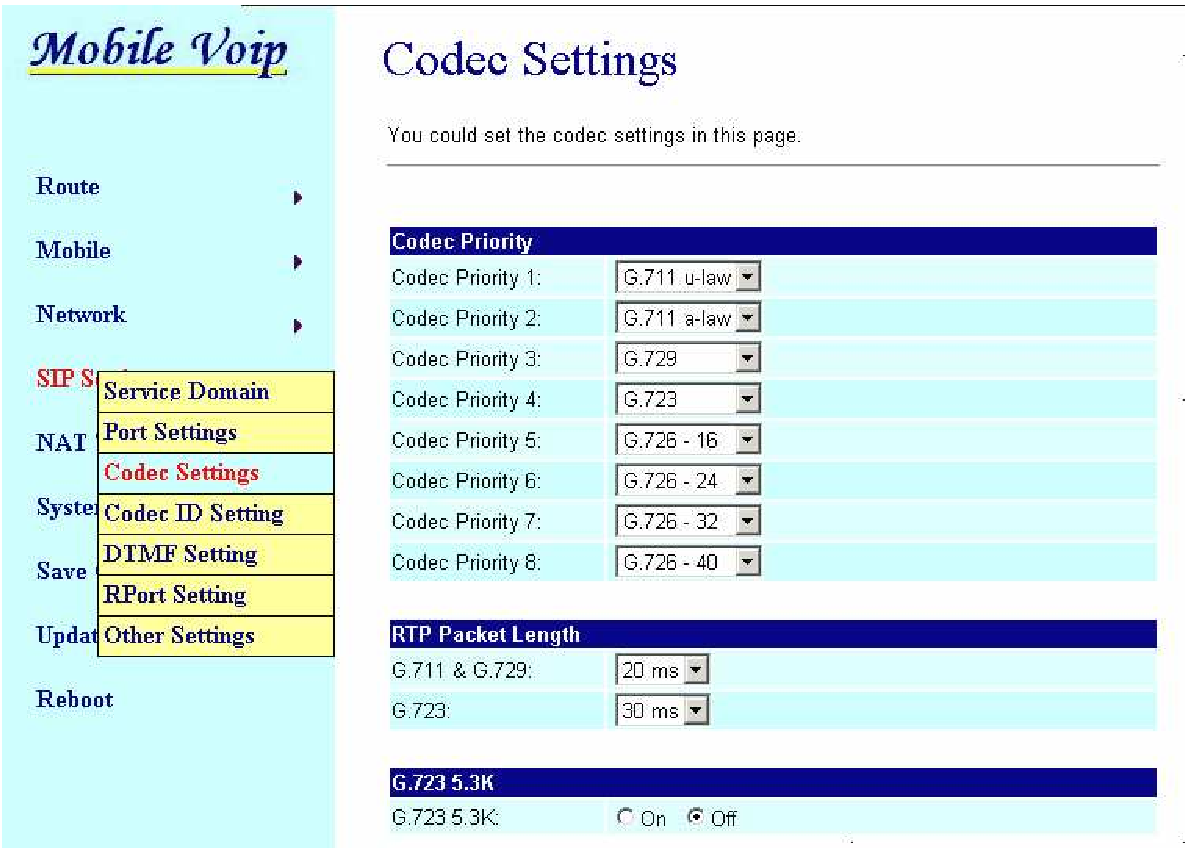
12.2 Port Setting

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



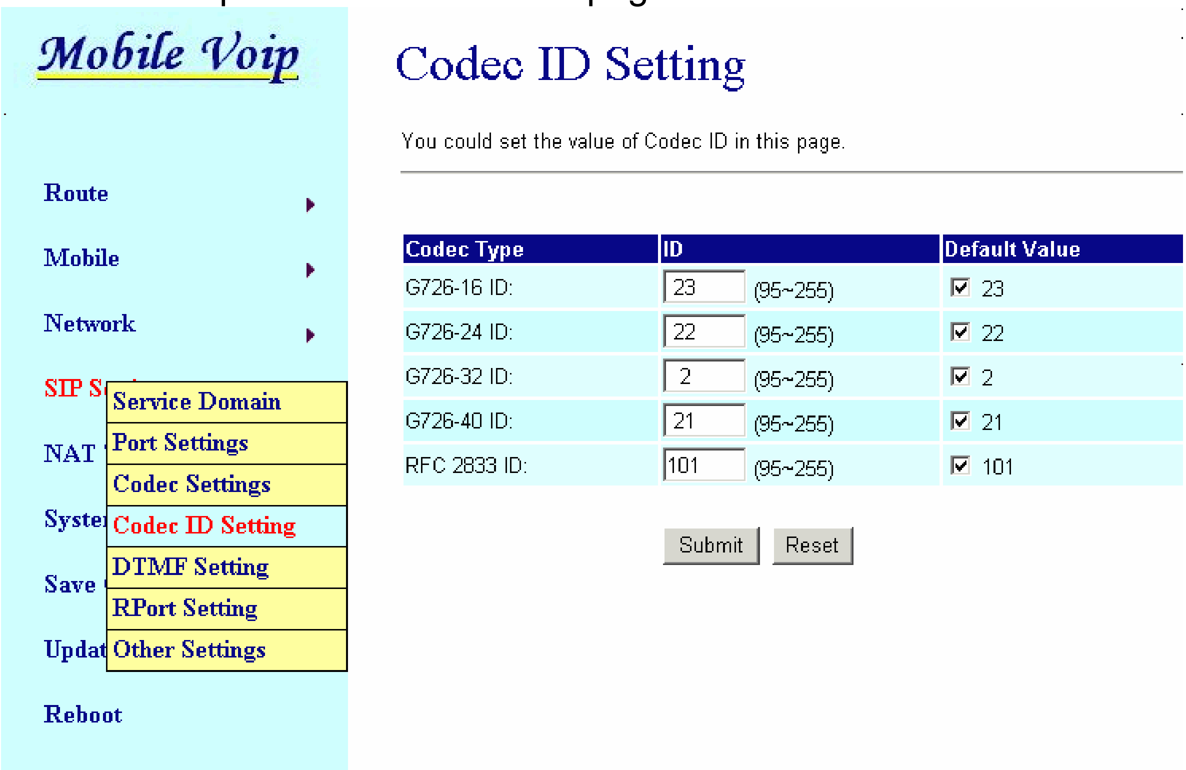
12.3 Codec Settings:

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.



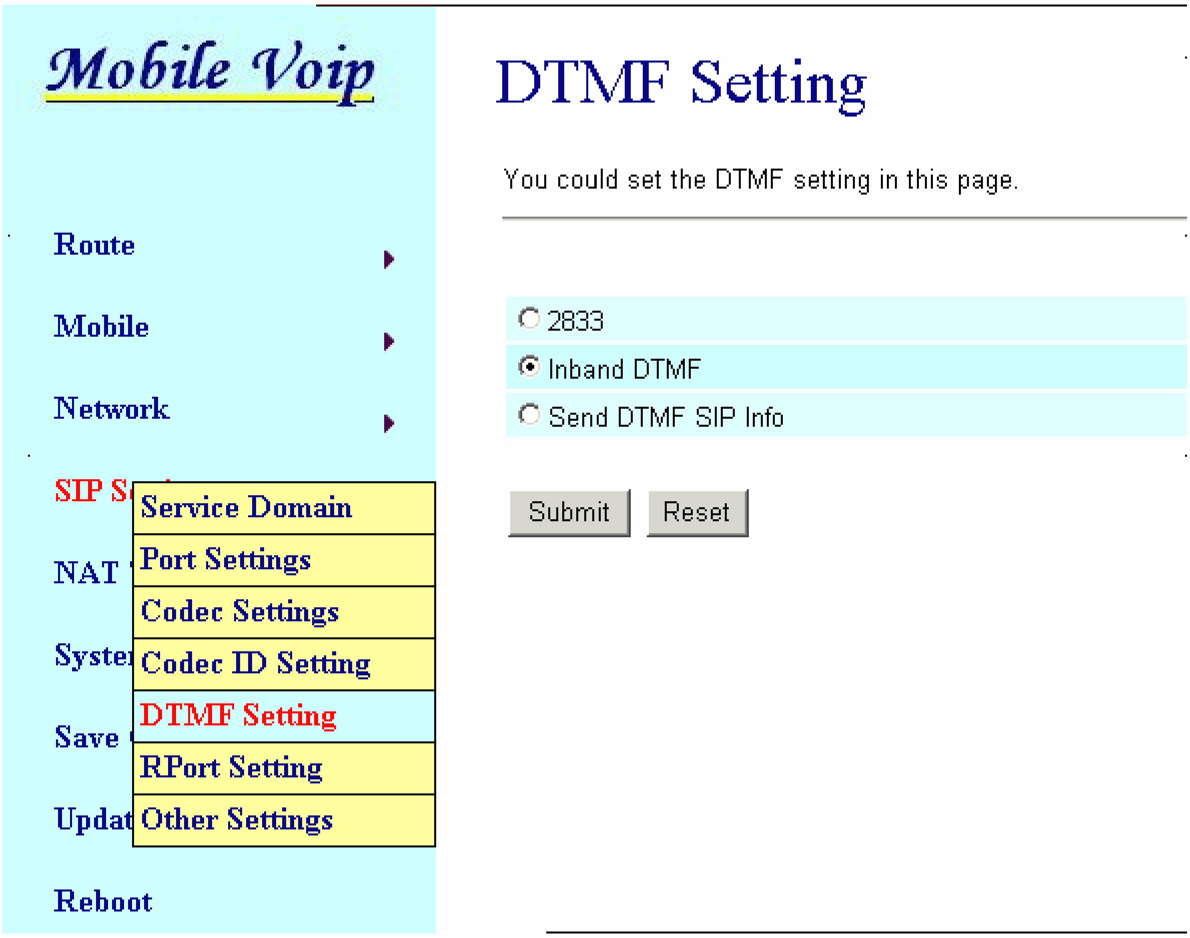
12.4 Codec ID Setting

You can setup the Codec ID in this page.



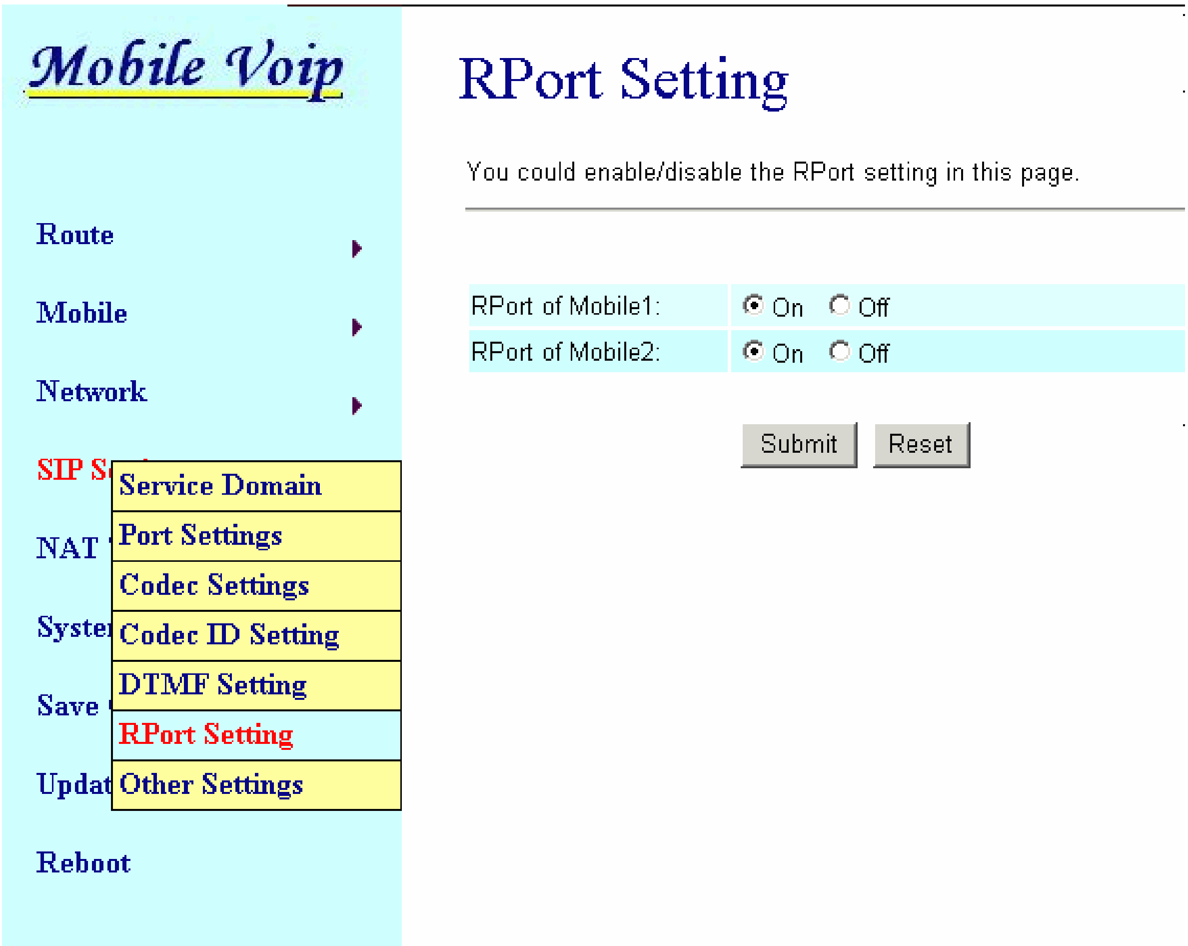
12.5 DTMF Setting

You can setup the DTMF Setting in this page.



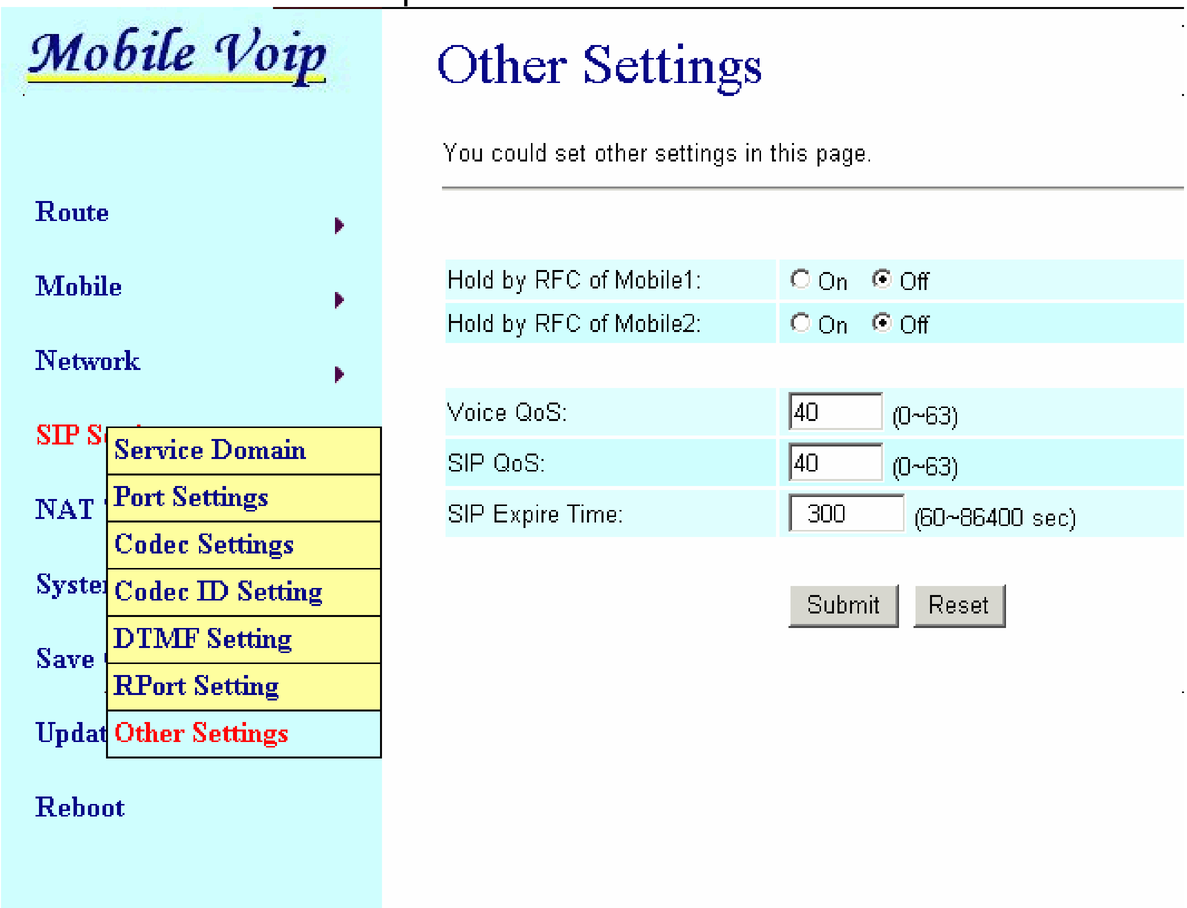
12.6 RPort Function:

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.



12.7 Other Settings

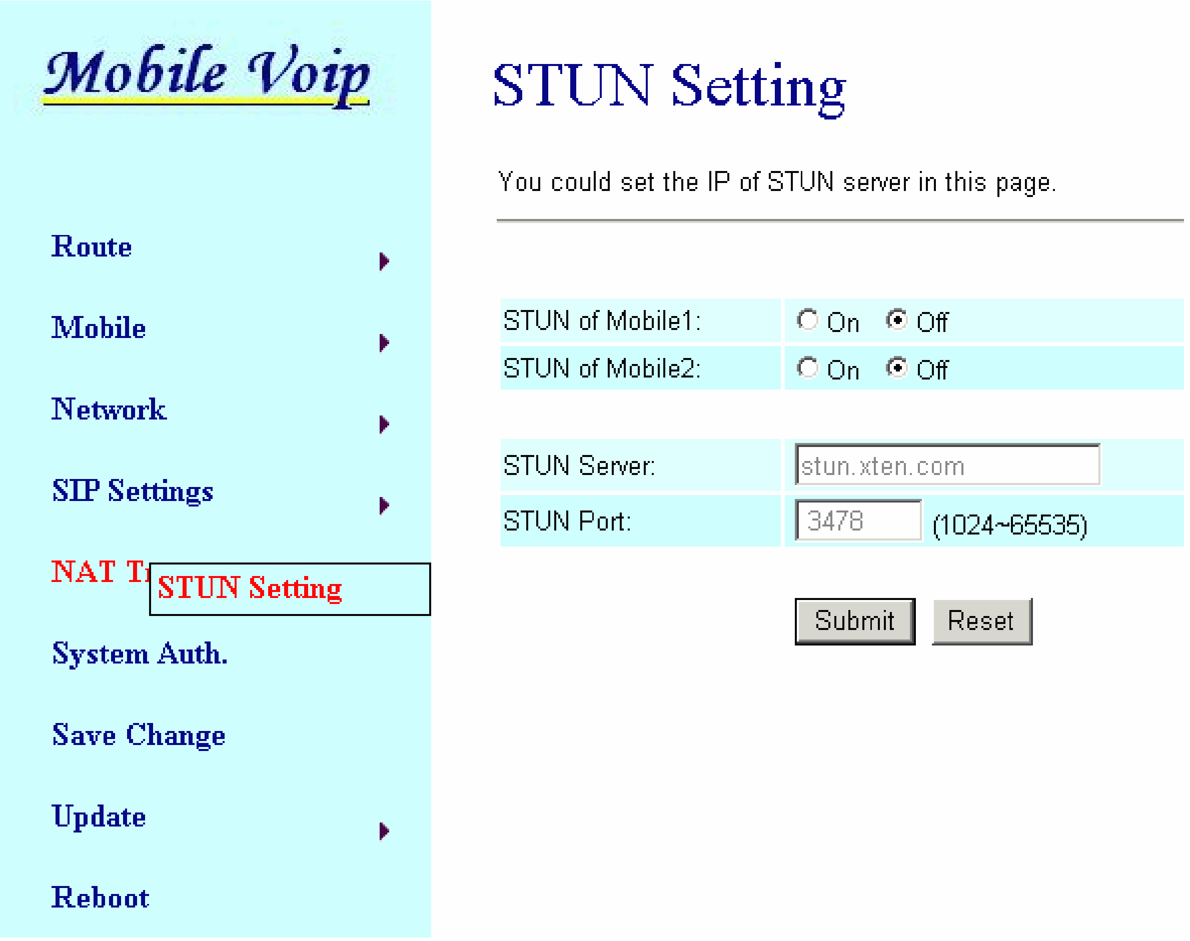
Other Settings: you can setup the Hold by RFC and QoS in this page. To change these settings. please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice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



13. NAT Trans

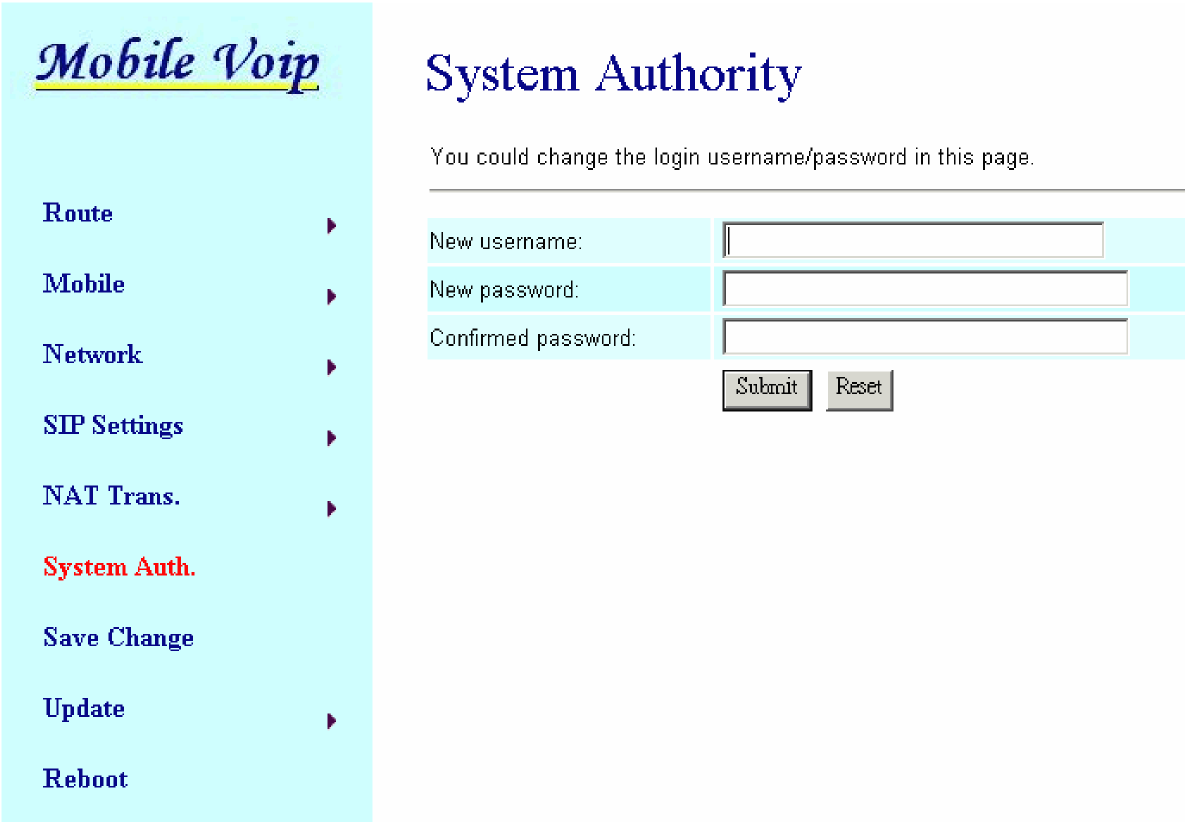
In NAT Trans. you can setup STUN and uPnP function. These functions can help your VoIP device working properly behind NAT.

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.



14.System Auth.

In System Authority you can change your login name and 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.

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.

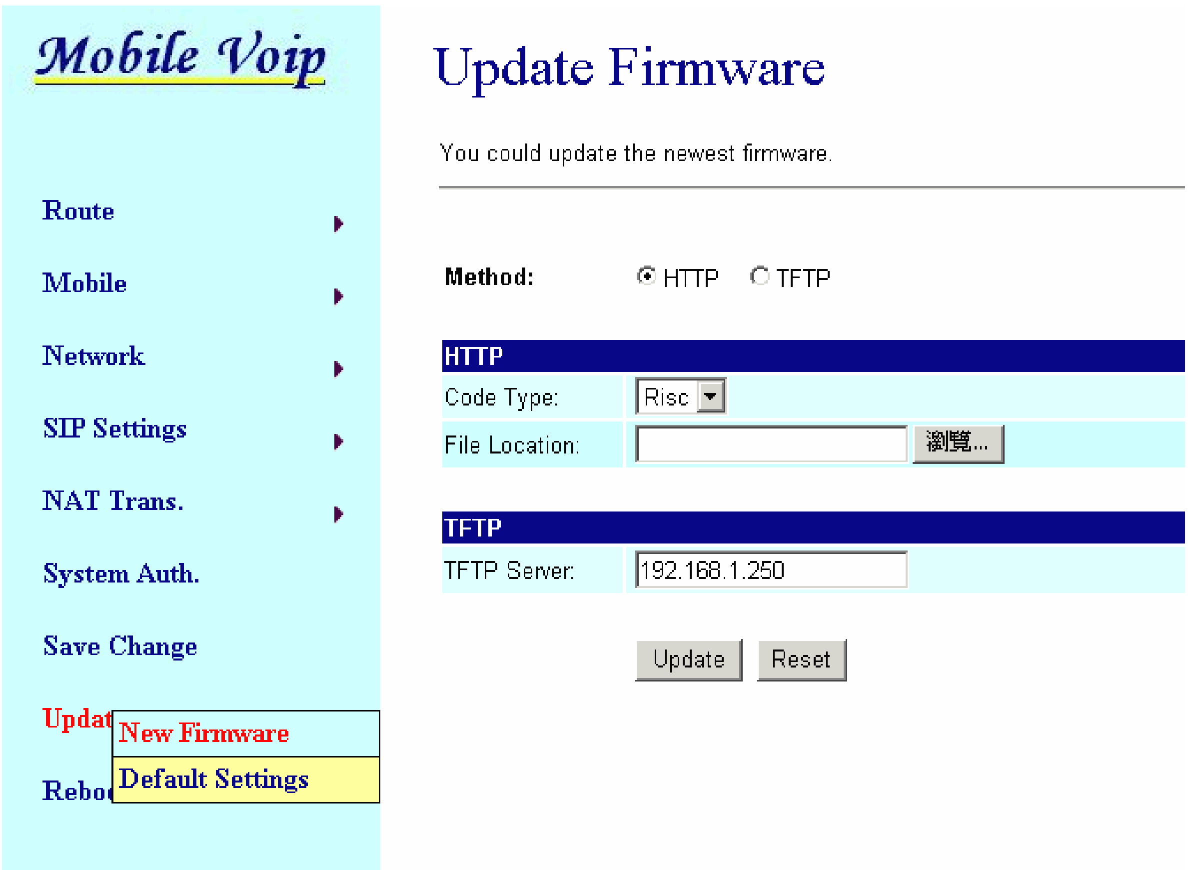
16.1 Update firmware

(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.

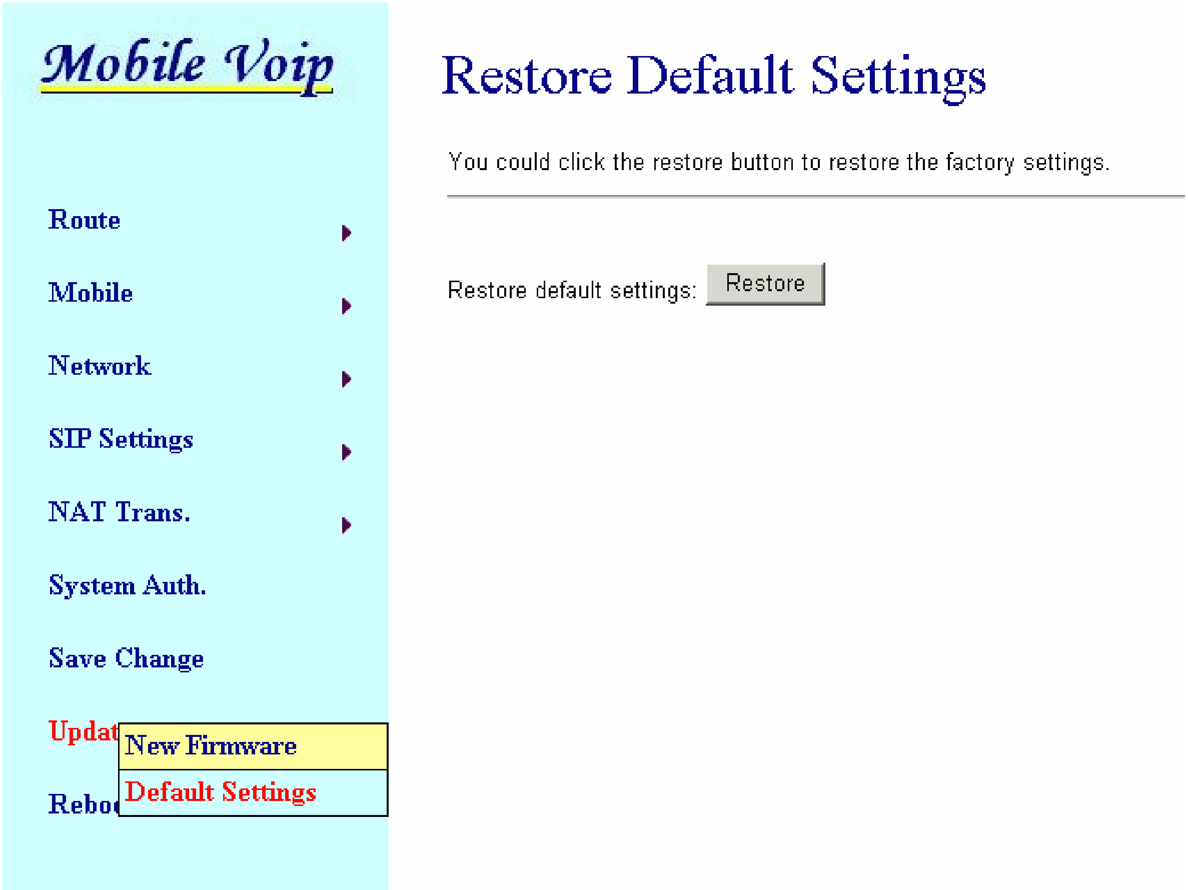
(3)Click the "Browse" button in the right side of the File Location or you can type the correct path and the filename in File Location blank.

(4)Select the correct file you want to download to the system then click the Update button.



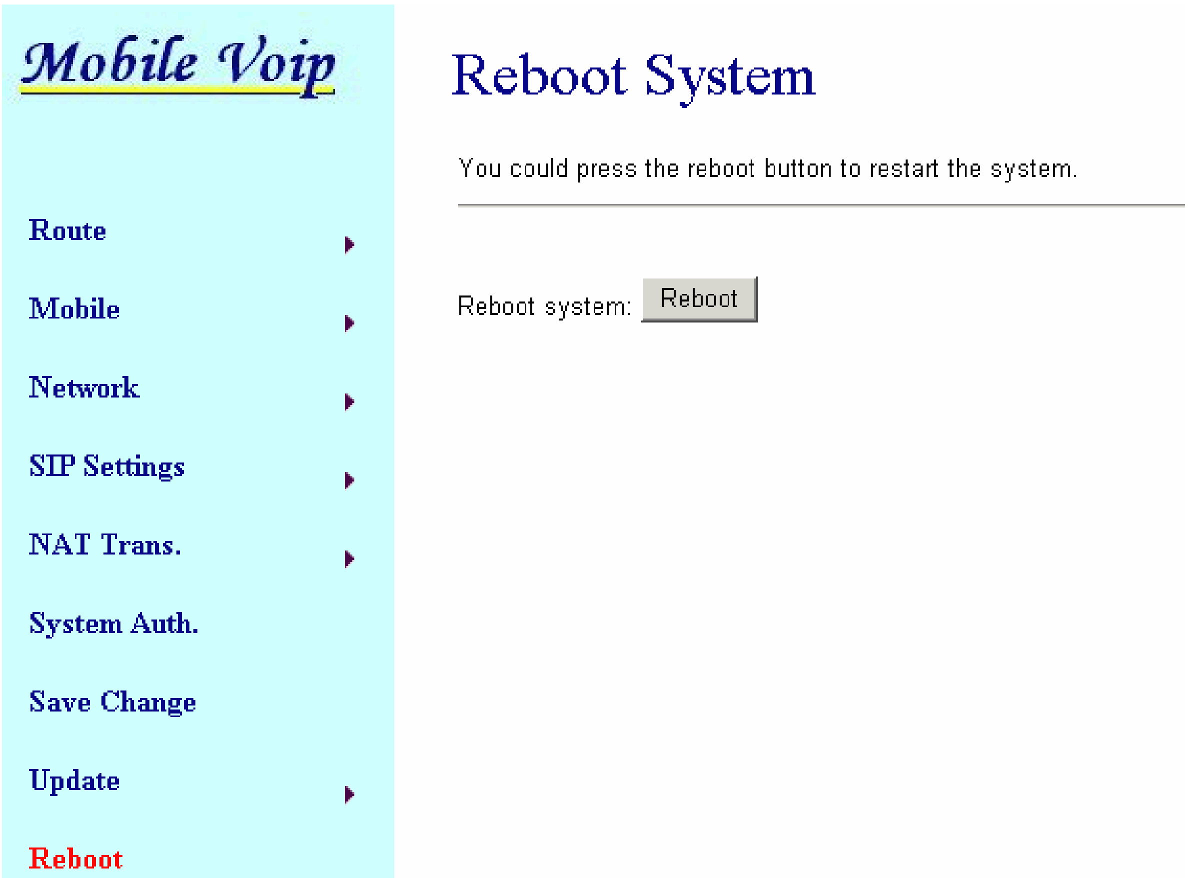
16.2 Restore Default Settings

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.



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.



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 in 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

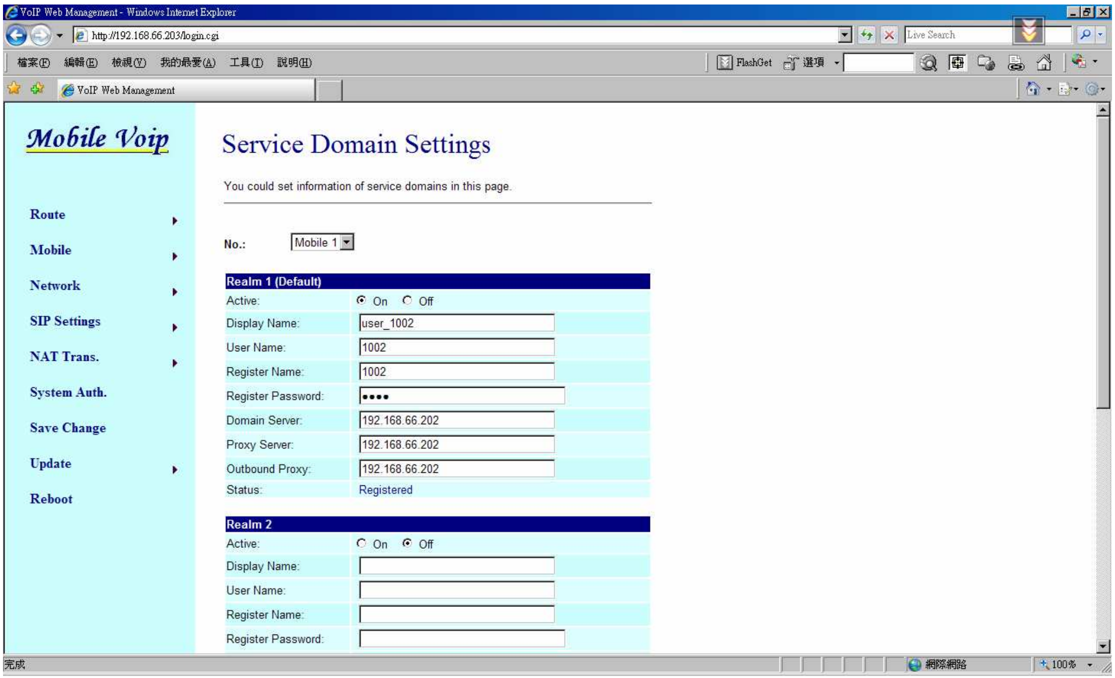
nat=yes
host=dynamic
canreinvite=no
context=internal

[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;rec
eived=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;recei
ved=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;recei
ved=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**)

mobile to lan:		
(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.
Lan to Mobile:		
(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"