MV-370

VoIP GSM Gateway

User Manual

PORTech Communications Inc.

Content

1. INTRODUCTION..1

2. FUNCTIONS... 1

3. THE CONTENTS IN PACKAGE...1

4. DIMENSION AND PANEL DESCRIPTION.. 2

5. ACCESSORY ATTACHMENT...3

6. SETTING AND MANAGING VIA WEB PAGE... 4

7. SYSTEM INFORMATION.. 5

8. ROUTE.. 5

9. MOBILE 10

10. NETWORK... 17

11. SIP SETTING..21

12. NAT TRANS 30

13.SYSTEM AUTH..31

14.SAVE CHANGE.. 32

15.UPDATE... 33

16.REBOOT...35

17. SETTING AND CHECKING VIA IVR... 36

18.SPECIFICATION..38

19. APPLICATIONS.. 39

20. SIMPLE STEPS... 40

21. APPENDIX: SETUP MV-370 WITH ASTERISK.. 41

22.HOW TO SETUP ASTERISK TO RECEIVE CALLER ID FROM 47

1. Introduction

MV-370 series products provide you the best connect solution for heterogeneous network □ including □ WLAN □ GSM or PSTN □ You may use a SIP-protocol VoIP phone or software to connect to the MV-370, then reach this call to the mobile network, and vice versa. With multiple sets of MV-370, you may even build an international call network.

2. Functions

2.1 VoIP (SIP)-GSM conversion.
2.2 VoIP (SIP)-CDMA conversion.
2.3 Voice response for setting and status enquiring. (Dial in GSM numbers of MV-370 to get voice information or to operate.)
2.4 50 sets of LAN->MOBILE routing, and 50 sets of MOBILE->LAN routing.
2.5 Series connections to save bills.
2.6 Standard SIP (RFC2543, RFC3261) protocol to communicate with other gateways or PC.
2.7 settings and managing via web page

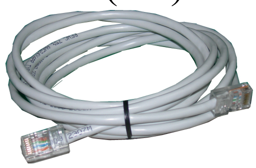
3. The contents in package

3.1 MV-370 main body
3.2 AC-DC Adaptor (110V AC – 12V DC or 220V AC – 12V DC)

3.3 Network cable
3.4 Antenna
3.5 User's Manual

(3.1)

(3.2)

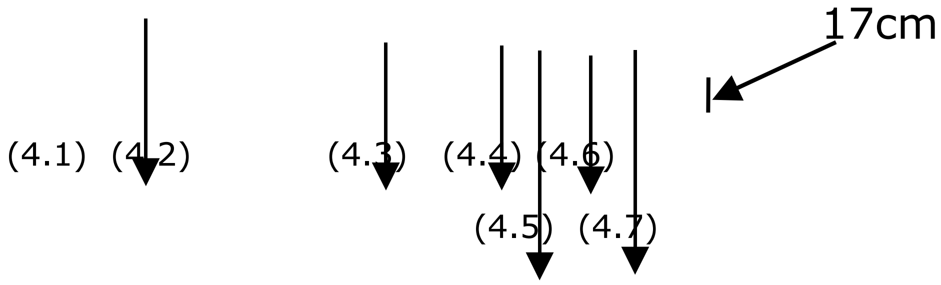


(3.3)

(3.4)

When you receive MV-370 package and find it is damaged or incorrect, please contact your vendor.

4. Dimension and Panel description



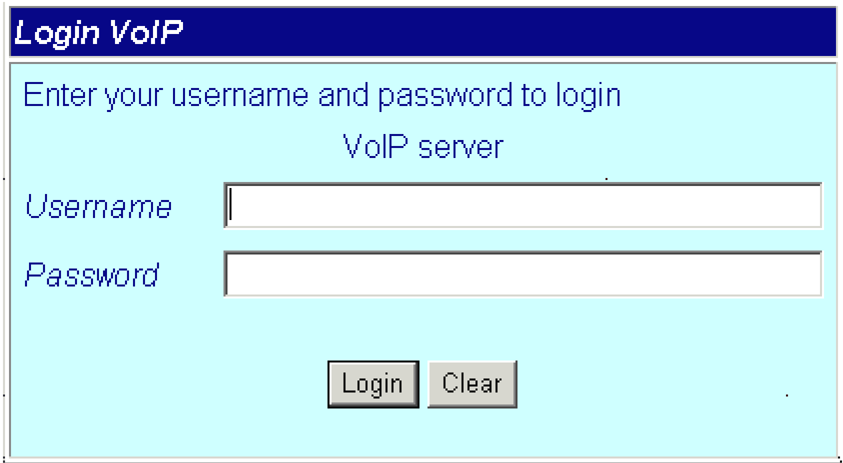
4.1 Antenna：Antenna connector.
4.2 DC 12V：Power socket.
4.3 LAN: Standard RJ-45 socket, connecting to Hub circuit.
4.4 PWR: Power indicator light, red light. Light is on when system's power supply is normal.
4.5 MOBILE: GSM indicator light, green light. Light flashes when GSM status is normal; light turns on constantly when GSM is called.
4.6 LAN: LAN indicator light, green light. Light flashes when Lan is called; light turns off when GSM answered.
4.7 LINK: Link indicator light, green light. Light is on when network is connected correctly.

5. Accessory attachment

5.1 Connect the network cable both to your Hub and to LAN socket of MV-370.
5.2 Connect the antenna and place it in a good receiving location (not too close to the device).
5.3 Insert a SIM card into back of MV-370.
5.4 Plug the adapter in DC 12V socket and PWR socket. The PWR light should turn red at the moment.
5.5 Click reset button 3 sec. MV-370 will restore default IP. Other setting as usual.

6. Setting and managing via web page

The default IP address of MV-370 is http://192.168.0.100. Before accessing the web page, please confirm this address is available in your network.



Enter the default username and password to login.
Default username: voip
Default password: 1234

7. System Information.

7.1 After login, you could see the system information such as: model name, firmware version, codec version name, etc. in this page.

7.2 You could also see the setting table in the left side. Please click on the option you would like to set. The setting methods are indicated as the following chapters, please input the value or select the item according to your situation.

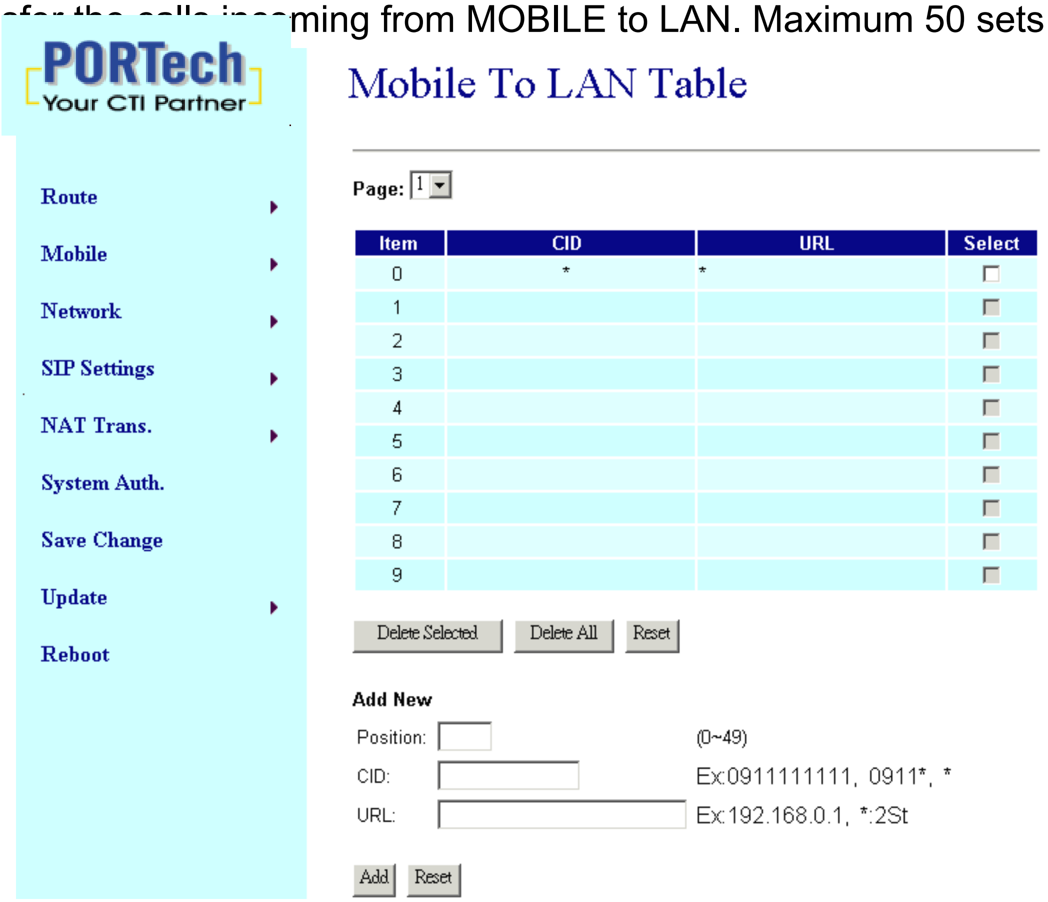
Note:

Please remember to save change whenever you submit any setting. Click "Save Change" then "Save" button, the system will restart and make the changed function/setting operative.

8. Route

8.1 Route/ Mobile to LAN Settings

In this page: Mobile To Lan Table, you could set the routing rules to transfer the calls incoming from MOBILE to LAN. Maximum 50 sets.



When the GSM number of the MV-370 is called, this device transfers the call to URL according to the caller ID of the incoming call.

8.1.1 CID: caller ID, the numbers of incoming call

You could set the CID as the following formats:

(1) The complete number, e.g. 0911111111
(2) The prefix part plus *, e.g. 0911*. This format means any number starting with 0911 will be accepted to transfer.
(3) *, this means any incoming call is accepted to transfer.
(4) N, this means the incoming call without showing its CID is accepted to transfer.

Please note the priority of the routing rules; the CID with more digits gets the priority.

8.1.2 URL：The IP address of destination

You could set the URL as the following formats:

(1) The complete IP address, e.g. 192.168.0.101
(2) The proxy extension numbers
(3) The phone numbers.

Note: If the device has registered at proxy server/Asterisk, you can enter any destination phone number. Also note that in the proxy server/Asterisk, you need to set the route of destination phone number.

(4) Leave it blank or 'N', this mean to refuse to transfer.
(5) *, this means to transfer via 2-stage-dialing. The call will be answered with a prompt dial tone for the caller to press the IP address, proxy extension, or **any phone number** as destination. The caller press the IP address on the phone keys: 192*168*0*101# as 192.168.0.101

8.1.3 Example of Mobile to Lan setting:

(1) Mobile to Lan: 0932*, 0911123456

When the GSM numbers of the device is called, if the caller's prefix numbers are 0932, MV-370 transfers the call to 0911123456, then 0911123456 rings (while available).

Precondition:

a. MV-370 has registered at proxy server/Asterisk

b. The proxy server/Asterisk has the route of "09"

(2) Mobile to Lan: *, *

Any incoming call gets a prompt dial tone; so the caller can enter any IP address, sip extension, or phone number.

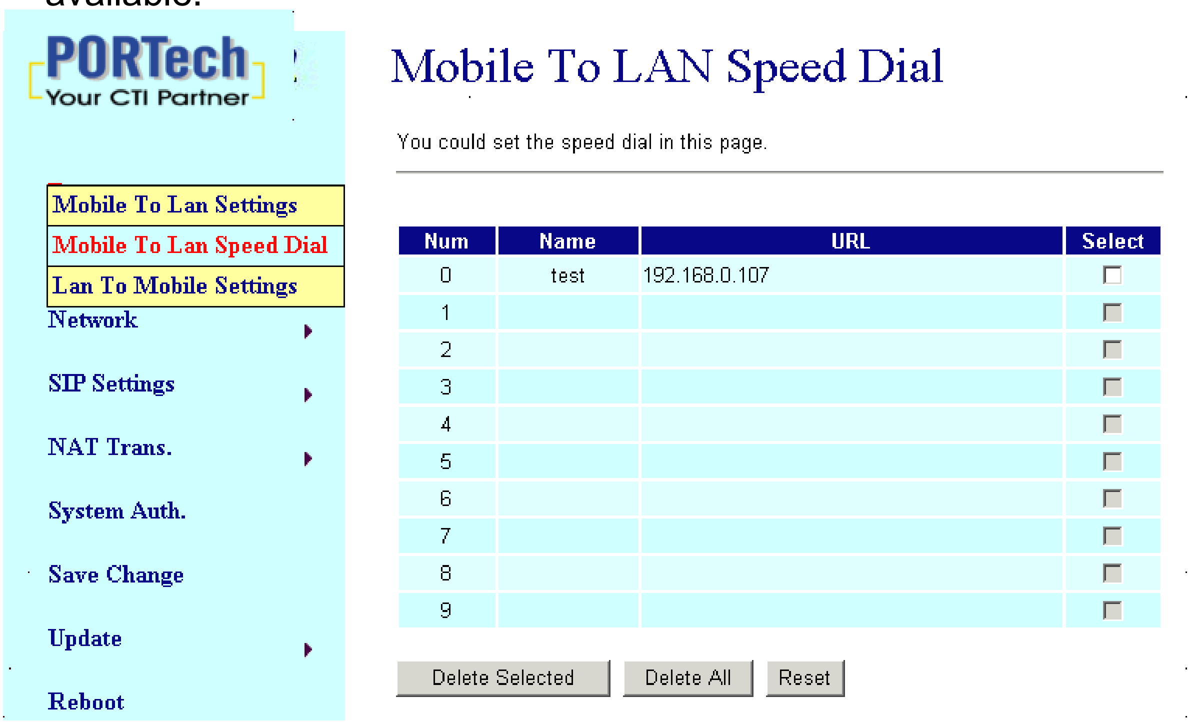
Precondition:

a. SIP extension or phone number needs to register at SIP Proxy Server or Asterisk.

b. Phone number, SIP Proxy Server or Asterisk needs to set the route of destination phone number.

8.2 Route/ Mobile to LAN Speed Dial Settings

When you set both Mobile to LAN Speed Dial Settings and Mobile to LAN settings at the same time, Mobile to LAN Speed Dial Settings gets higher priority. Mobile to Lan setting will be not available.

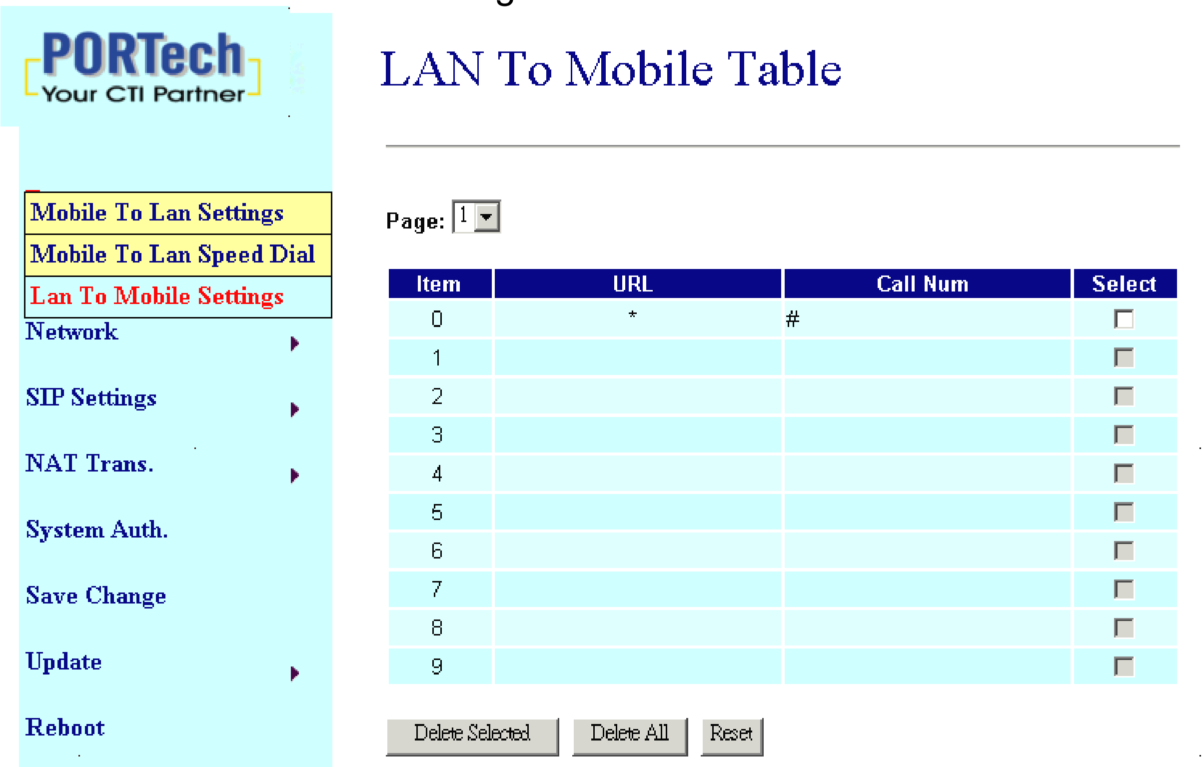


The call is answered with a prompt dial tone for the caller to press the "Num", and then the device connects the "URL" as destination.

Example: after you call the GSM number of the device and hear a dial tone, you press 0, then the lan phone of IP address: 192.168.0.107 rings.

8.3 Route/ LAN to Mobile Settings

In this page: Lan To Mobile table, you could set the routing rules to transfer the calls incoming from Lan to Mobile. Maximum 50 sets.



When the Lan of the MV-370 is called, this device transfers the call to Call Num according to the URL of the incoming call.

8.3.1 URL: The IP address or proxy extension numbers of the incoming call.

You could set the URL as the following formats:

(1) The complete IP address, e.g. 192.168.0.101

(2) The proxy extension numbers, e.g. 103

(3) Part of an IP address plus *, e.g. 192.168.0.*. This means the IP address starting with 192.168.0 would be accepted to transfer,

(4) Part of the proxy extension numbers plus, e.g. 10*. This means the extension numbers starting with 10 would be accepted to transfer.

8.3.2 Call Num: the phone numbers of destination.

You could set the Call Num as the following formats:

(1) The complete number, e.g. 0911111111

(2) *, this means to transfer via 2-stage-dialing. The call will be answered with a prompt dial tone for the caller to press the destination phone numbers, e.g. 0911111111

(3) #, this allow the caller with lan phone dial directly the destination numbers.

Precondition:

(1) MV-370 and incoming lan Phone are both registered at proxy server or Asterisk.

(2) Proxy server/asterisk has set the routing rules to assign specific prefix of numbers to be transferred from MV-370.

(3) Lan to Mobile routing sets: *, #

Usage: You could dial on your lan phone call any destination number with prefix of “09”. When your lan phone and MV-370 had registered and “09” prefix is setted the routing rules at proxy server or Asterisk.

(4) #['d'n]['a'ppp], this means to do the above routing, and to modify the numbers.

Note: 'd'n means to delete the number of prefix,

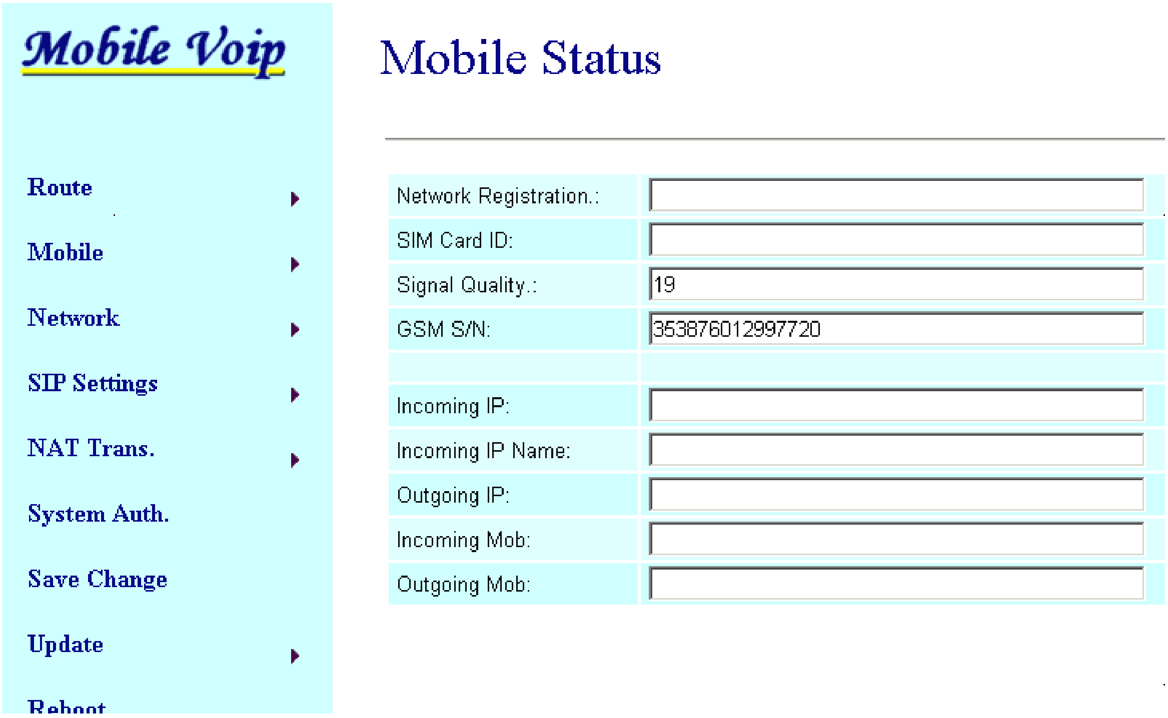
'a'ppp means to add 'ppp' prefix.

E.g. #d2a09 means to call the registered numbers via one-stage-dialing. The numbers are modified to: delete 2 digits of prefix of the original numbers, then add 09 to be new prefix of the destination numbers.

9. Mobile

9.1 Mobile/ Mobile Status

In this page: Mobile Status, you could get the information of your GSM network and the latest operation.



(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)GSM S/N : IMEI Number

(5)Incoming IP：The IP address of the last incoming call from LAN.

(6)Incoming IP Name: proxy server name

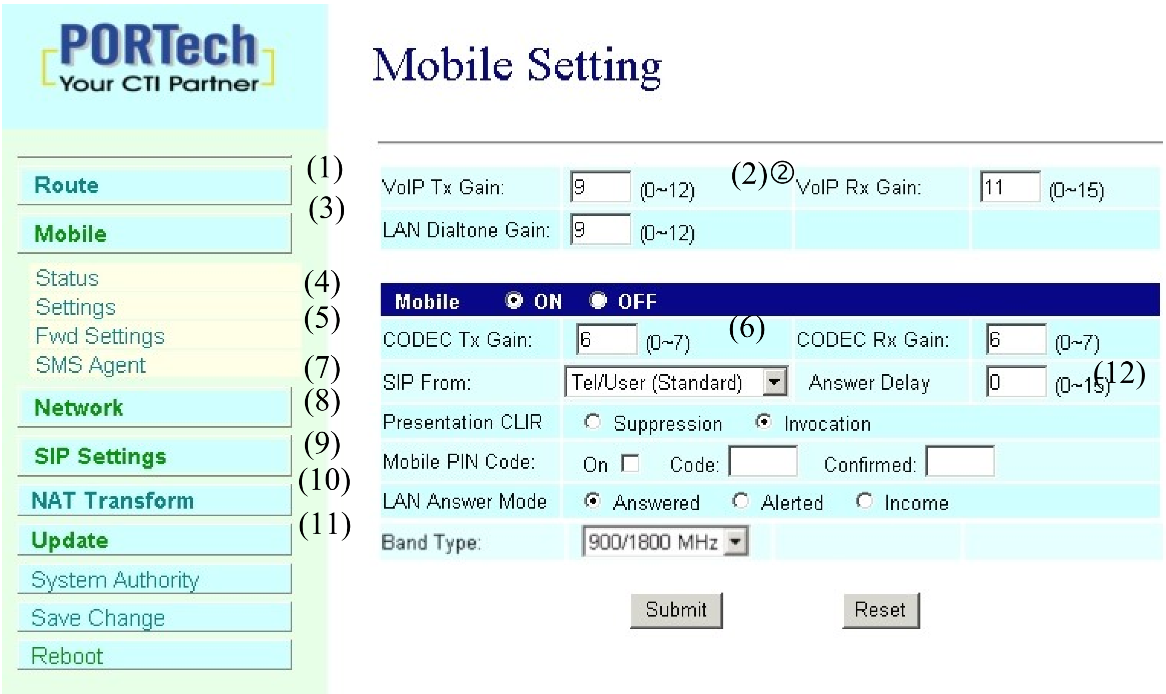
(7)Outgoing IP：The IP address of the last outgoing call to LAN.

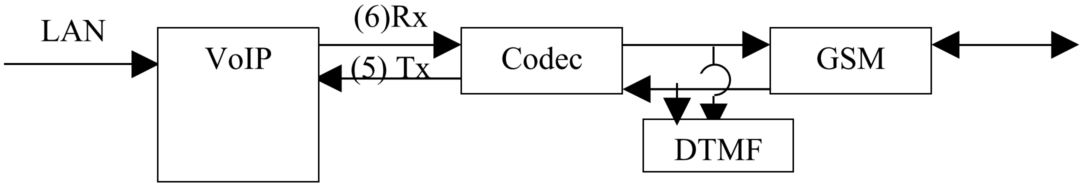
(8)Incoming Mob：The caller ID of the last incoming call from MOBILE.

(9)Outgoing Mob：The called number of the last outgoing call to MOBILE.

9.2 Mobile/ Mobile Setting

In this page: Mobile setting, you could adjust the parameter and click on the option to fit your need. **You could leave those default value before you had tried the complete operation of this device.**





(3)LAN Dialtone Gain: DTMF Reciver is not good,you can adjust gain down.

(4) ON/Off: If you use this channel,please click on. Otherwise,please click off.

(5)CODEC Tx Gain: as above
(6)CODEC Rx Gain: as above
(7) SIP From: Caller ID transfer

- Tel/User(Standard): If you need to register to Asterisk and proxy server,please choose this option. And how to transfer the caller ID to LAN,please refer 22.How to setup Asterisk to receive Caller ID from MV-370 (page 43)

  MV-370-2 will send the message as follows in the Packet.

  From: " caller number " <sip:3001@192.168.0.228>;tag=51088abb

- Tel/Tel :

  MV-370 will send the message as follows in the Packet.

  From: "caller number" <sip: caller number @192.168.0.228>;tag=6ac93f7c

  □Please note:If you choose this option,please don't register to Asterisk and proxy server. Please only fill proxy server ip and choose Active: on (else field empty) in sip setting/service demain

- User/Tel

  MV-370 will send the message as follows in the Packet.

  From: " Username " <sip: caller number @192.168.0.228>;tag=7f130947

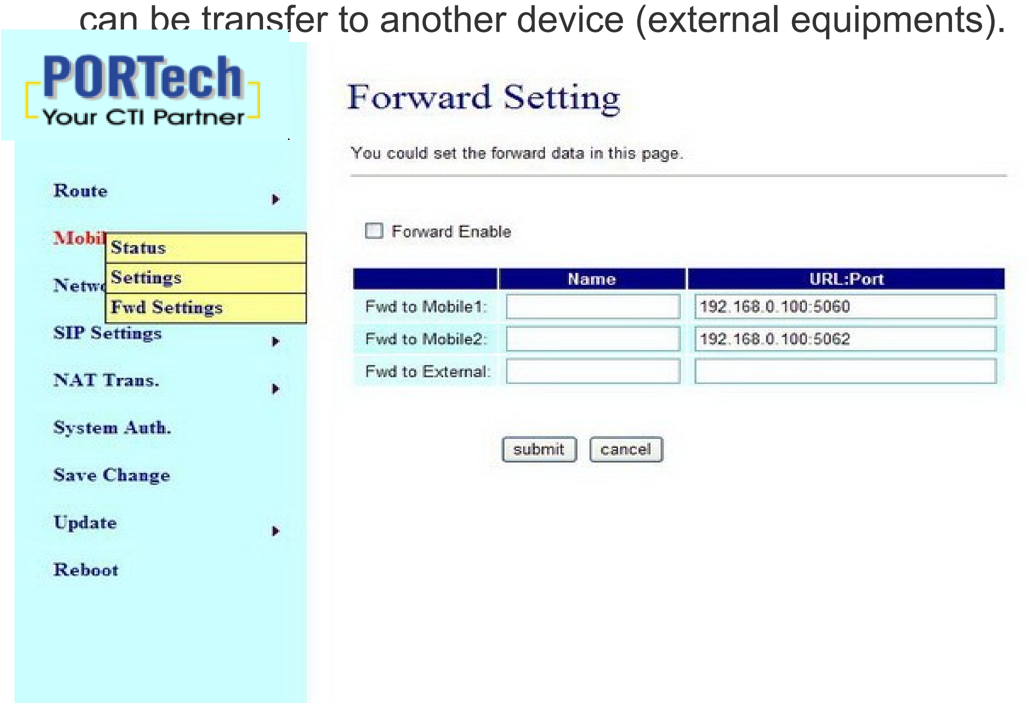
  □If you choose this option,please don't register to Asterisk and proxy server. Please only fill proxy server ip,Username and choose Active: on (else field empty) in sip setting/service demain

(8)Presentation CLIR : If you need to block the Caller Id for call termination,please choose Suppression
(9)Mobile PIN Code:If you need to unlock pin code via MV-370,you can click "On" and enter pin code.
(10)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
(11)Band Type:When you buy Quad band,you need to choose your GSM frequency

(12)Answer Delay: Delay for incoming call when the ring.

9.3 Mobile / Forward Setting :

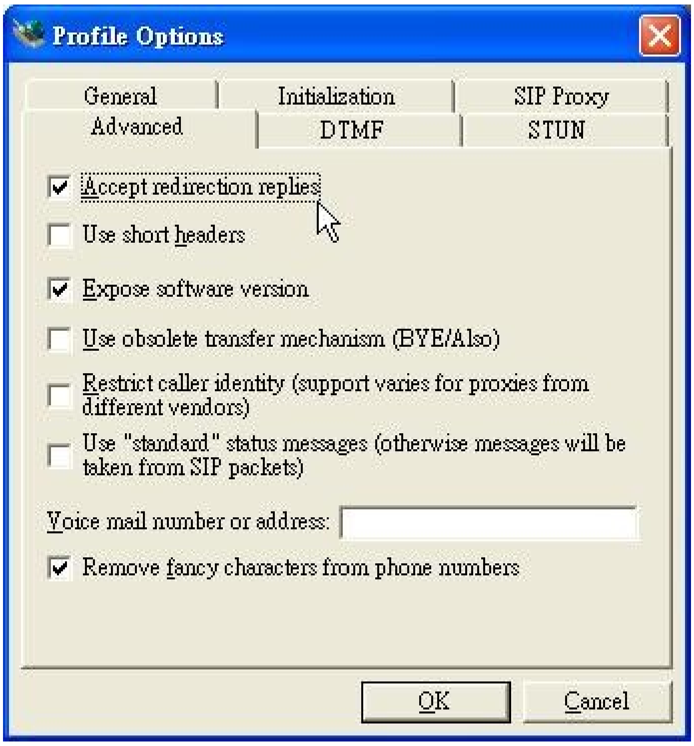
When the first route are busying, SIP can transfer phone call to another free route.   When the device are busying, the phone call can be transfer to another device (external equipments).



* "Forward Enable" is not motivate on Defualt value.

So please, mark "Forward Enable" this blank to motivate this function.

Take SJ Phone for example: Profiles -> Edit -> Advanced -> Accept redirection replies (Turn on the "Forward Enable", therefore the SJ Phone can designate a port which are free to use.)



Name	URL:Port
Fwd to Mobile1:	
Fwd to Mobile2:	
Fwd to External:	

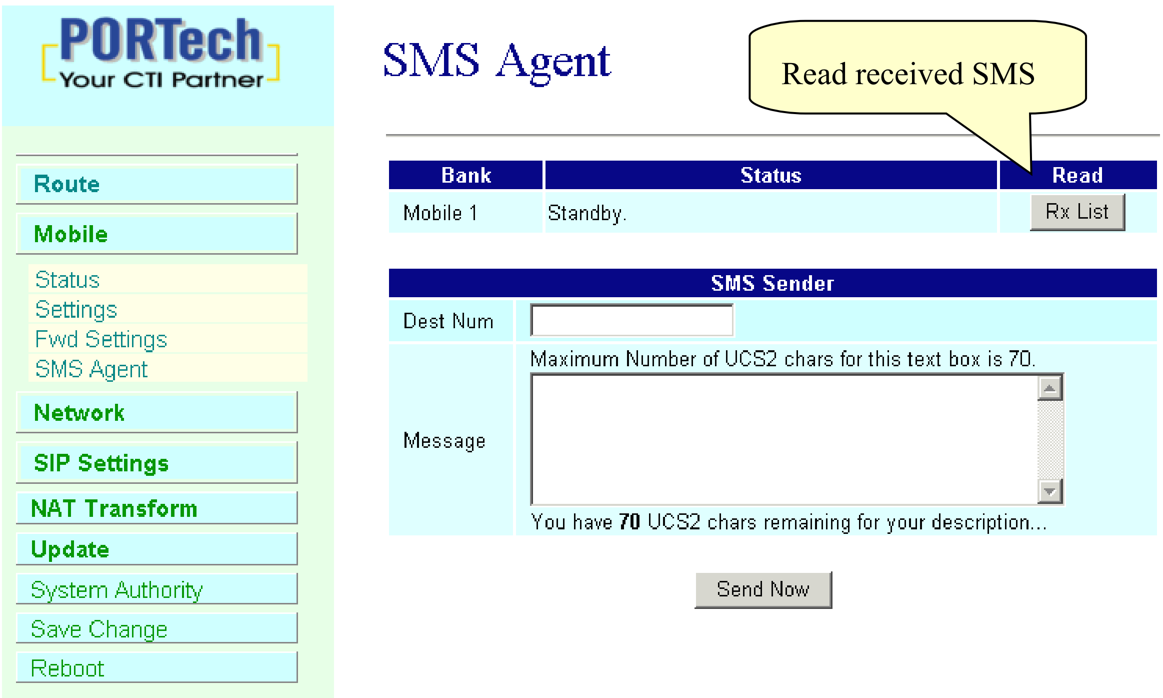
The Explanation of Picture:

Fwd to Mobile1:192.168.0.100 : 5060, it means when 5062 Port are busying, SJ Phone can transfer the call to 5060 Port (192.168.0.100).

Fwd to Mobile2:192.168.0.100 : 5062, it means when 5060 Port are busying, SJ Phone can transfer the call to 5062 Port (192.168.0.100).

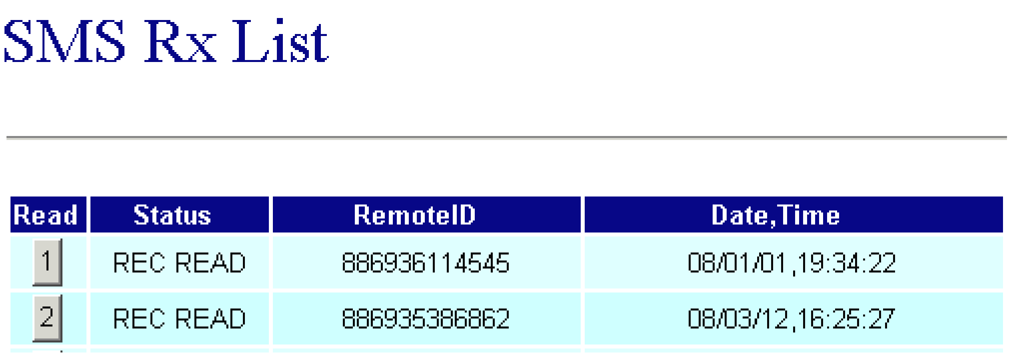
- If both 5060 port and 5062 port are busying at same time, you can set up "Fwd to External", then you can transfer the phone call to another designate device.

9.4 Mobile / SMS Agent :

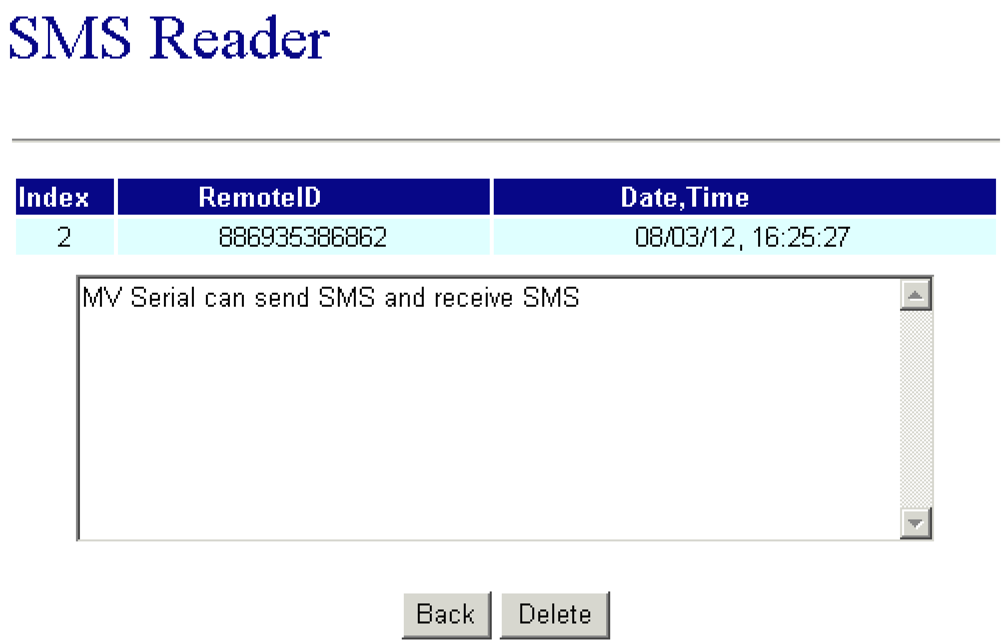


(1) Rx List: Read received SMS
(2) Dest Num: the Receiver's phone number
(3) Message: Please fill the message that want to send to receiver.

When you click Rx List, you can view all received SMS as follows.



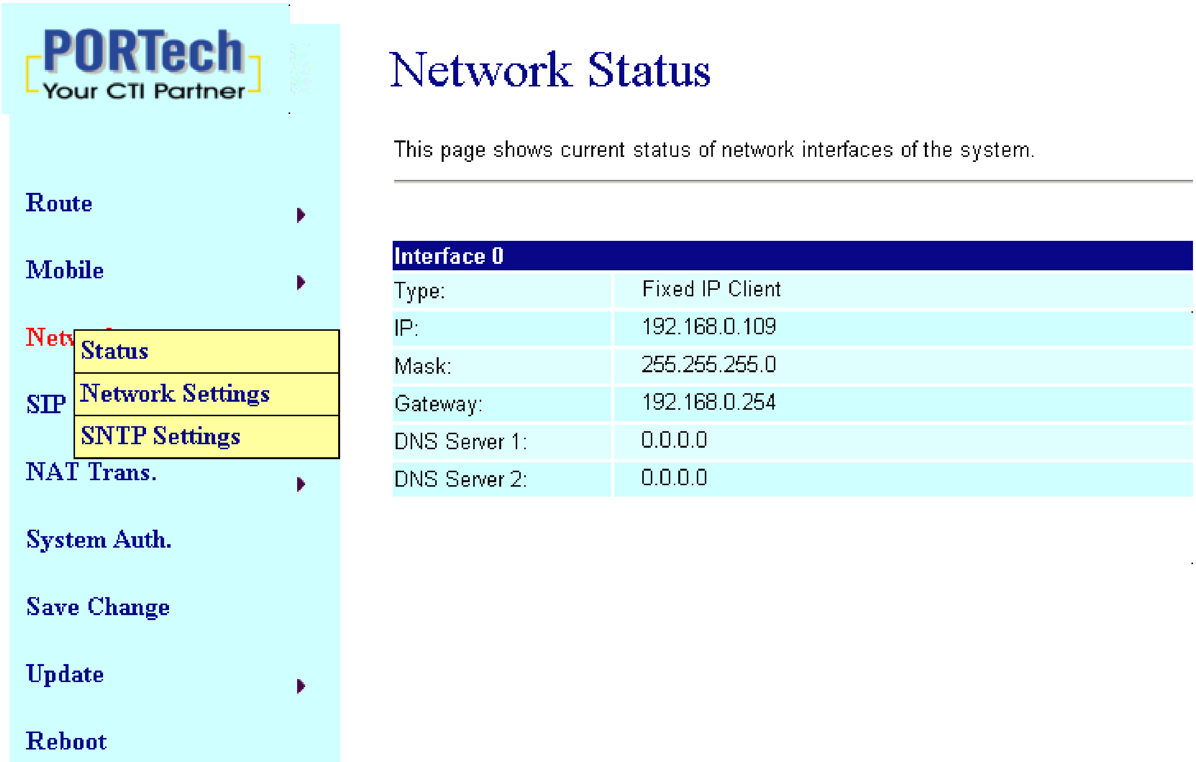
Click the serial no,you can view message as follows.



10. Network

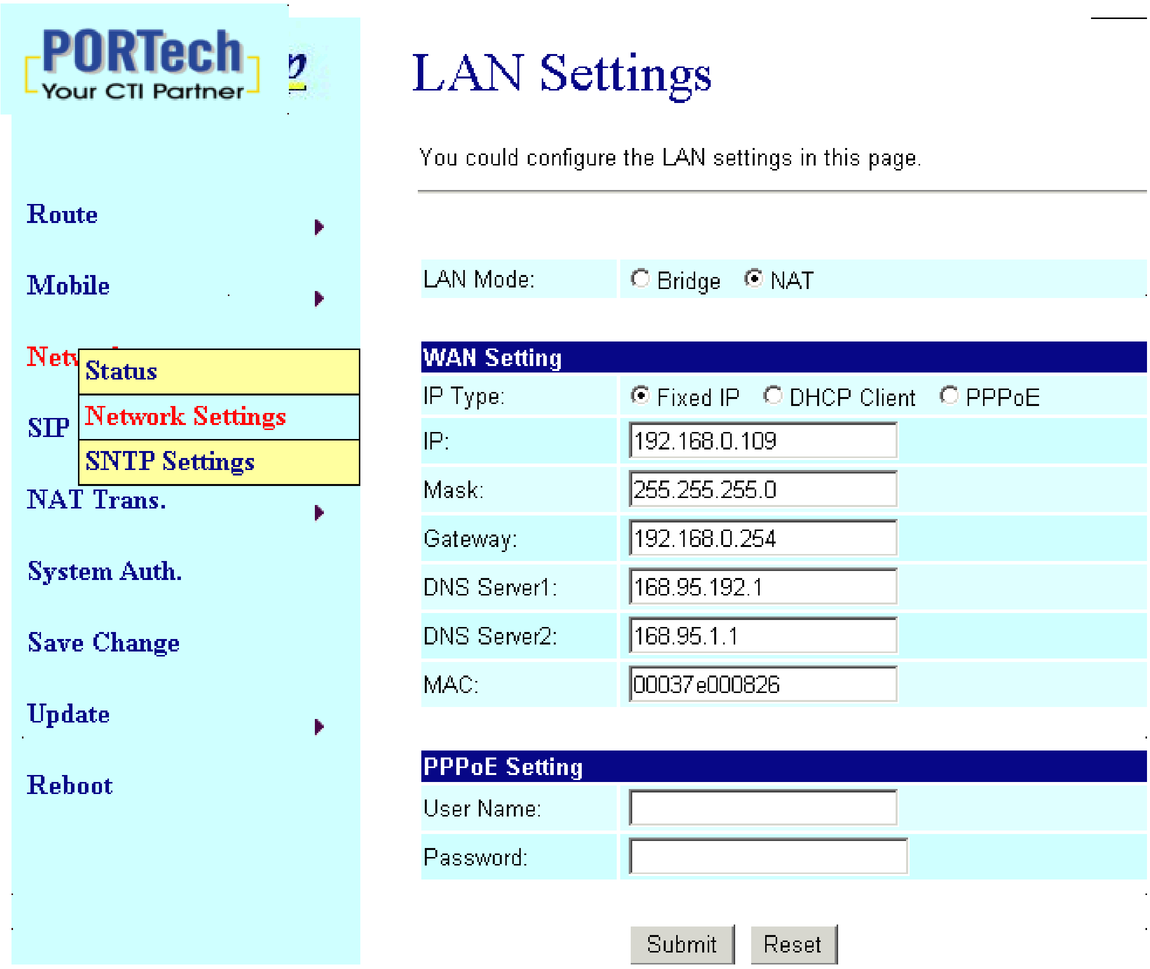
In Network, you could check the Network status; configure the WLAN Settings, LAN Settings and SNTP settings.

10.1 Network/ Status/ Network Status: information of current Network in this page.



10.2 Network/ Network Settings/ Lan Settings: You can check the current Network setting in this page.

The default IP is 192.168.0.100; you could change it to any available IP address, or select different IP type to suit your environment.



(1) LAN Mode: select NAT

(2) Fixed IP: 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.

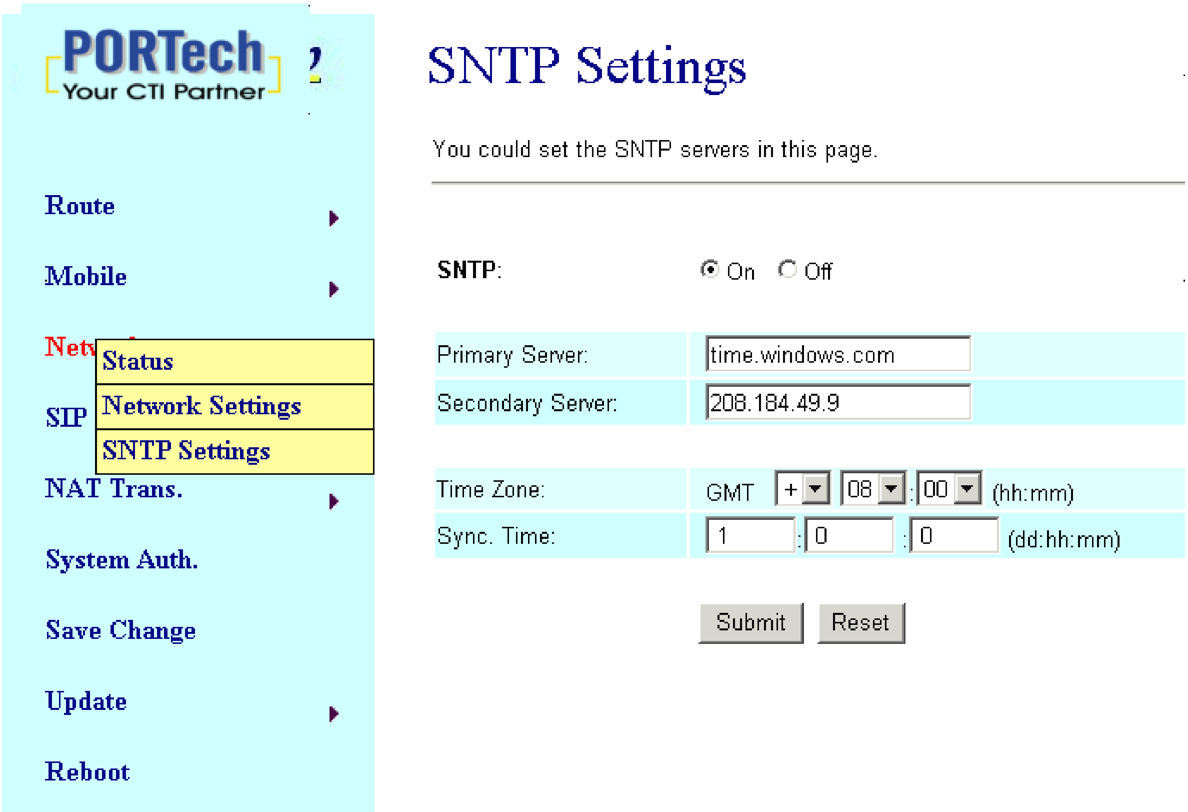
(3) DHCP client: you could refer to your current network environment to configure the system properly

(4) PPPoE: If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.

(5) After you input or modify the value, click the Submit button.

10.3 Network/ SNTP Settings:

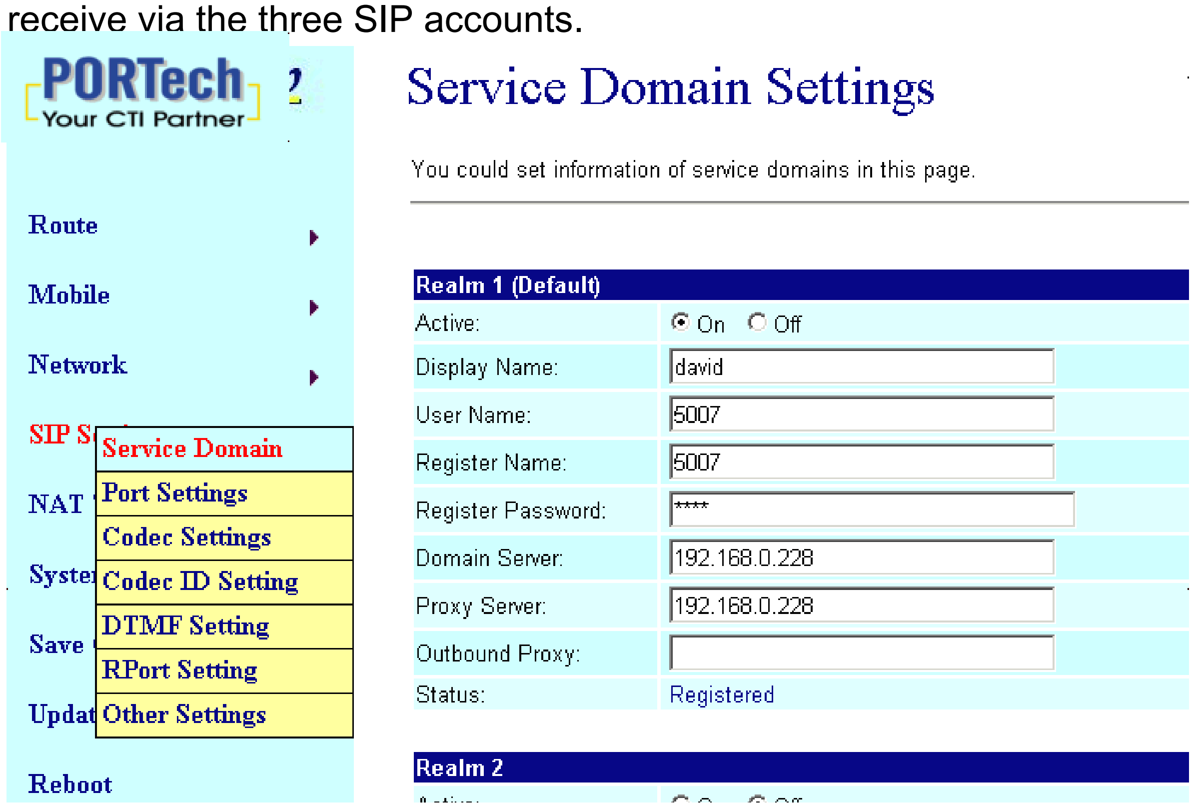
You could select “On” to give SNTP function to this device.
Input the primary and secondary IP Address of SNTP Server to get the date/time information. Also you could set the Time Zone according to your location; and set the time to synchronize. After setting, remember to click the Submit button.



11. SIP Setting

If you need, you could setup the Service Domain, Port Settings, Codec Settings, RTP setting, RPort Setting and Other Settings.
If ISP provides the VoIP service, you need to input the related information correctly to register at SIP Proxy Server.

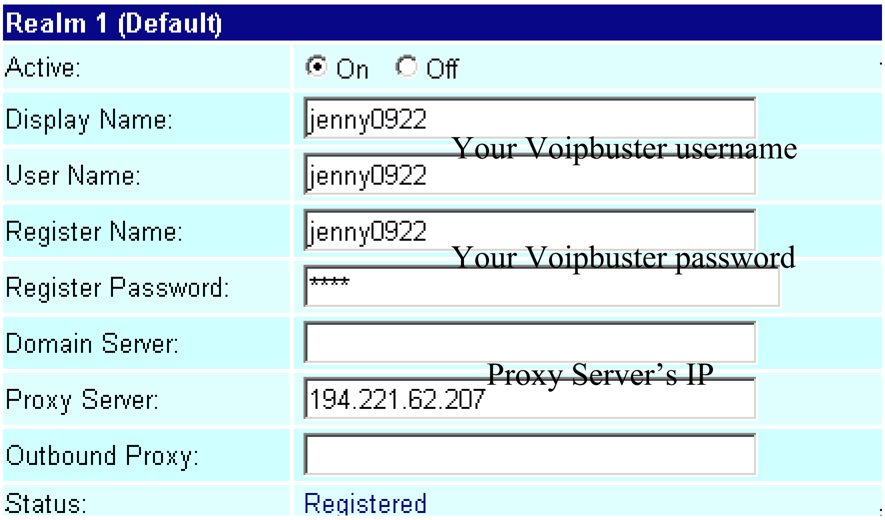
11.1 SIP Setting/ Service Domain:
In this page, you should input the data refer to your ISP. Maximum is 3 accounts (Realm 1 to 3). You could dial out via first SIP account, and receive via the three SIP accounts.



(1) Active: click “On” to enable the function in Service Domain, then input the following items.
(2) Display name: input the name you would like to display.
(3) User name: input your user name in ISP.
(4) Register Name: input your register name in ISP.
(5) Register Password: input your password in ISP.

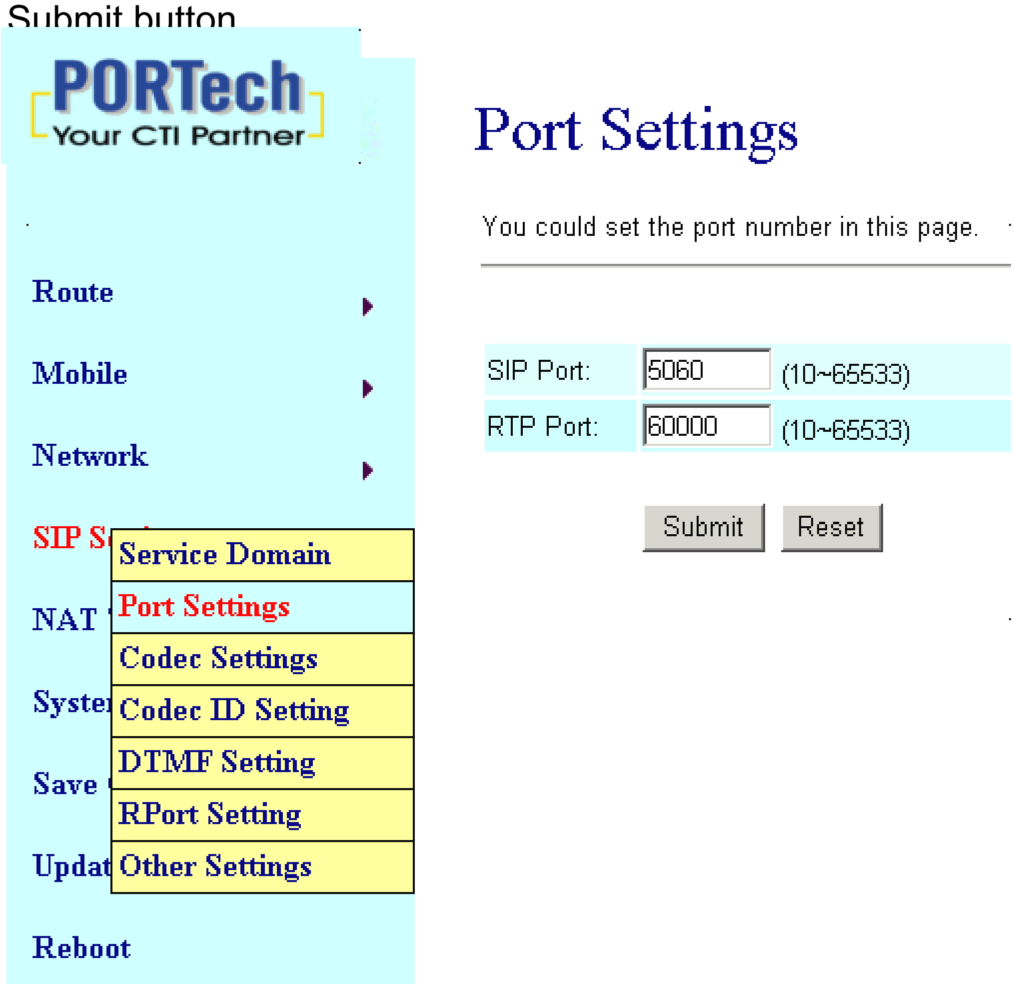
(6) Domain Server: input the Domain Server IP address.
(7) Proxy Server: input the Proxy Server IP address.
(8) Outbound Proxy: input the Outbound Proxy IP address. If your ISP does not provide the information, you could skip this item.
(9) After setting, click the Submit button. Remember to click "Save Charge"
(10) You can see the Register Status in the Status item.

Example:
Register VoipBuster



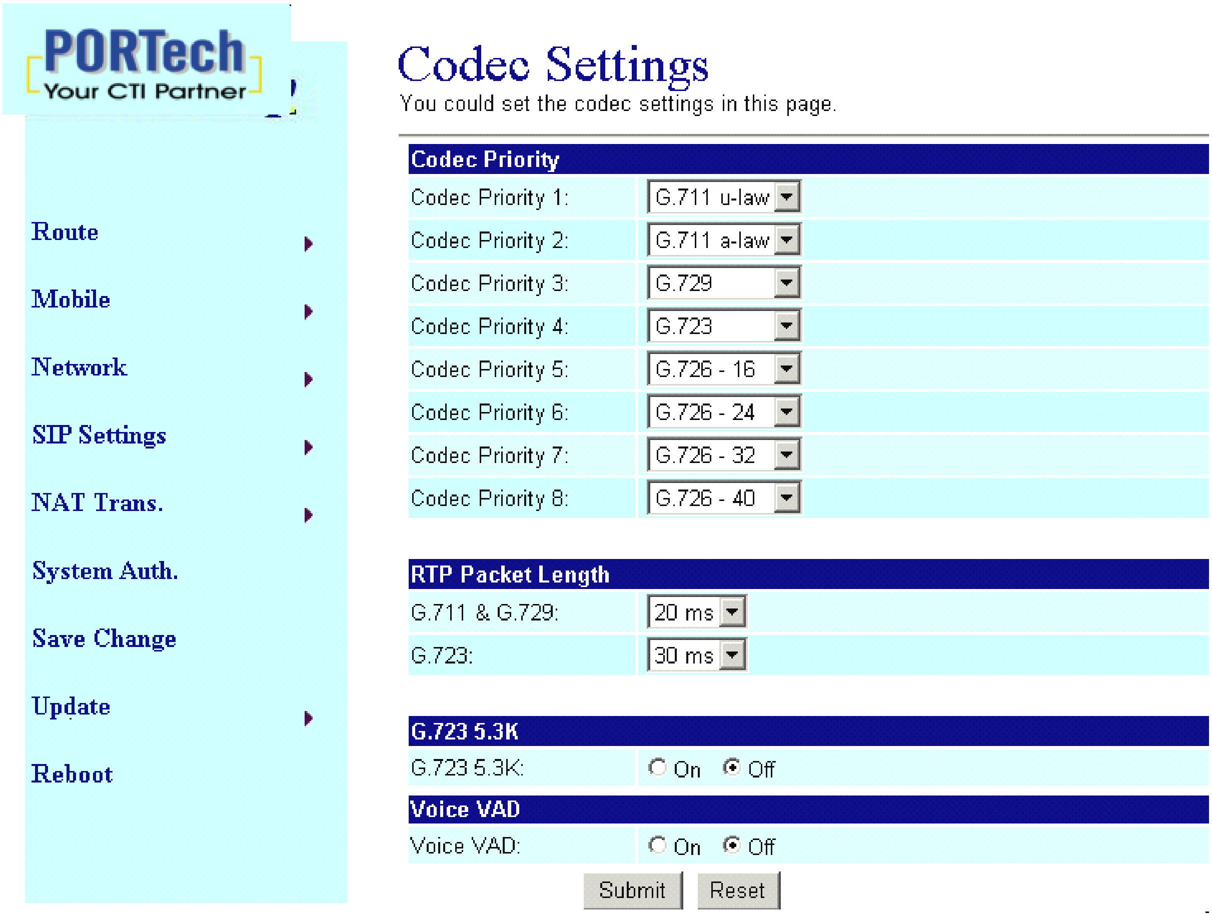
11.2 Port Setting

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. After setting, remember to click the Submit button



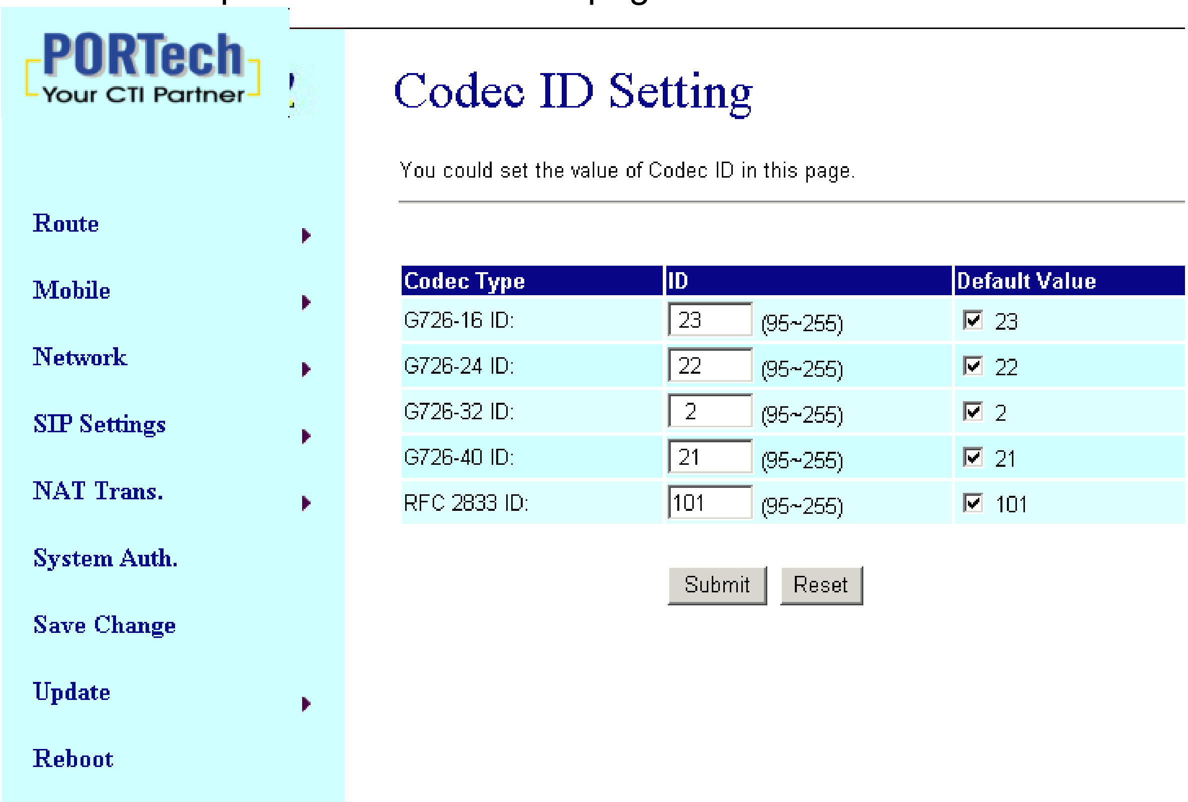
11.3 Codec Settings:

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



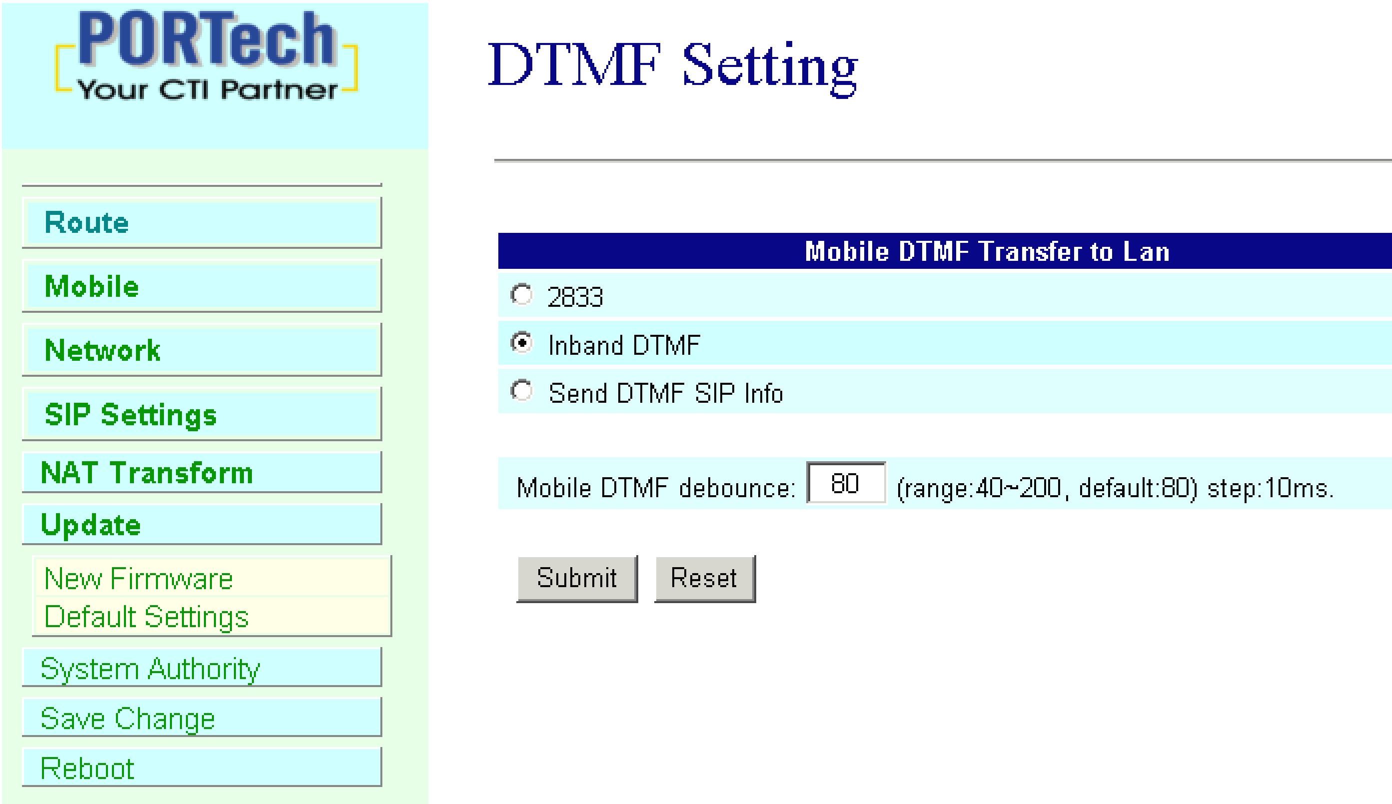
11.4 Codec ID Setting

You can setup the Codec ID in this page.



11.5 DTMF Setting

You can setup the DTMF Setting in this page.

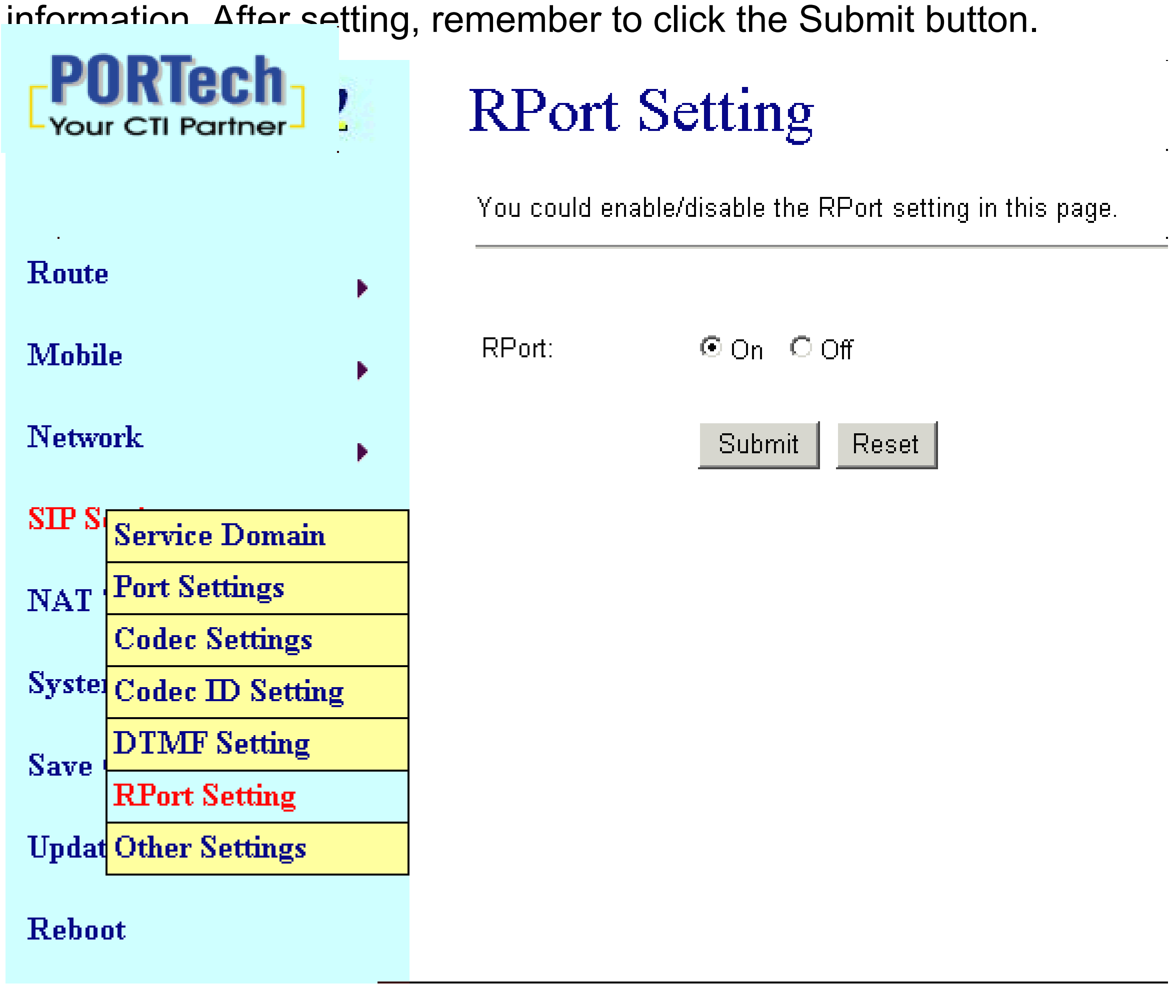


Note:

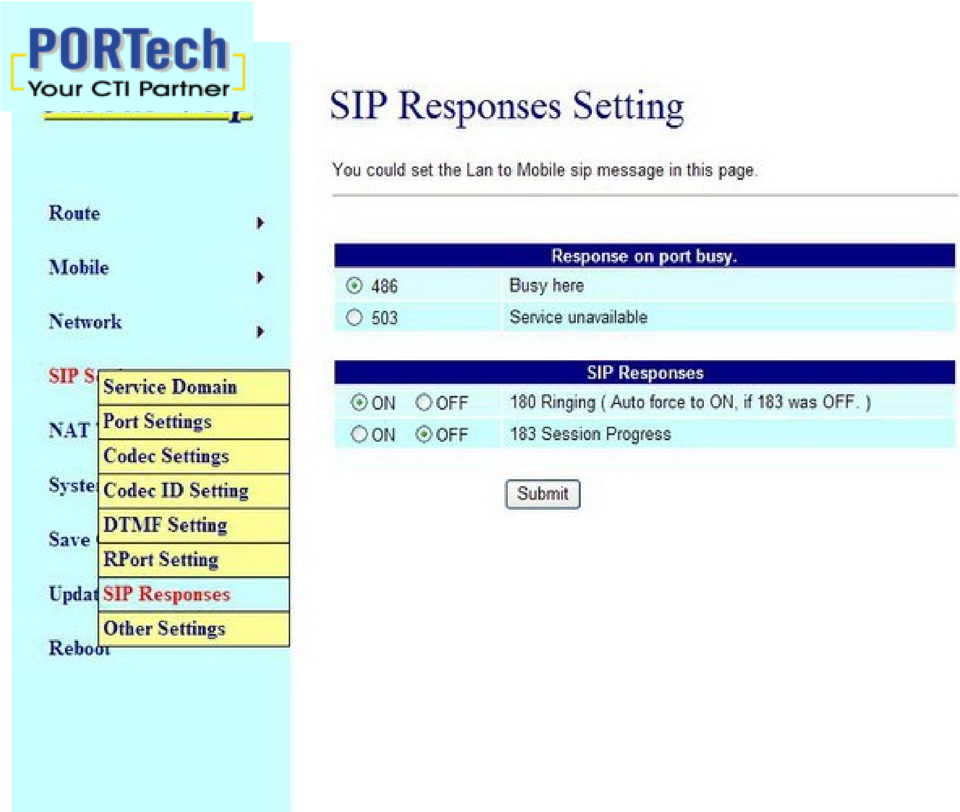
If this device has registered at SIP Proxy Server/Asterisk, please select "2833". If not, please select "Inband DTMF".

11.6 RPort Setting:

You can setup the RPort Enable/Disable according to your ISP information. After setting, remember to click the Submit button.



11.7 SIP Setting: SIP Responses



11.7.1 486(busy here), 503(Service unavailable): When Device are busying, you can select 486 or 505 to response to SIP.

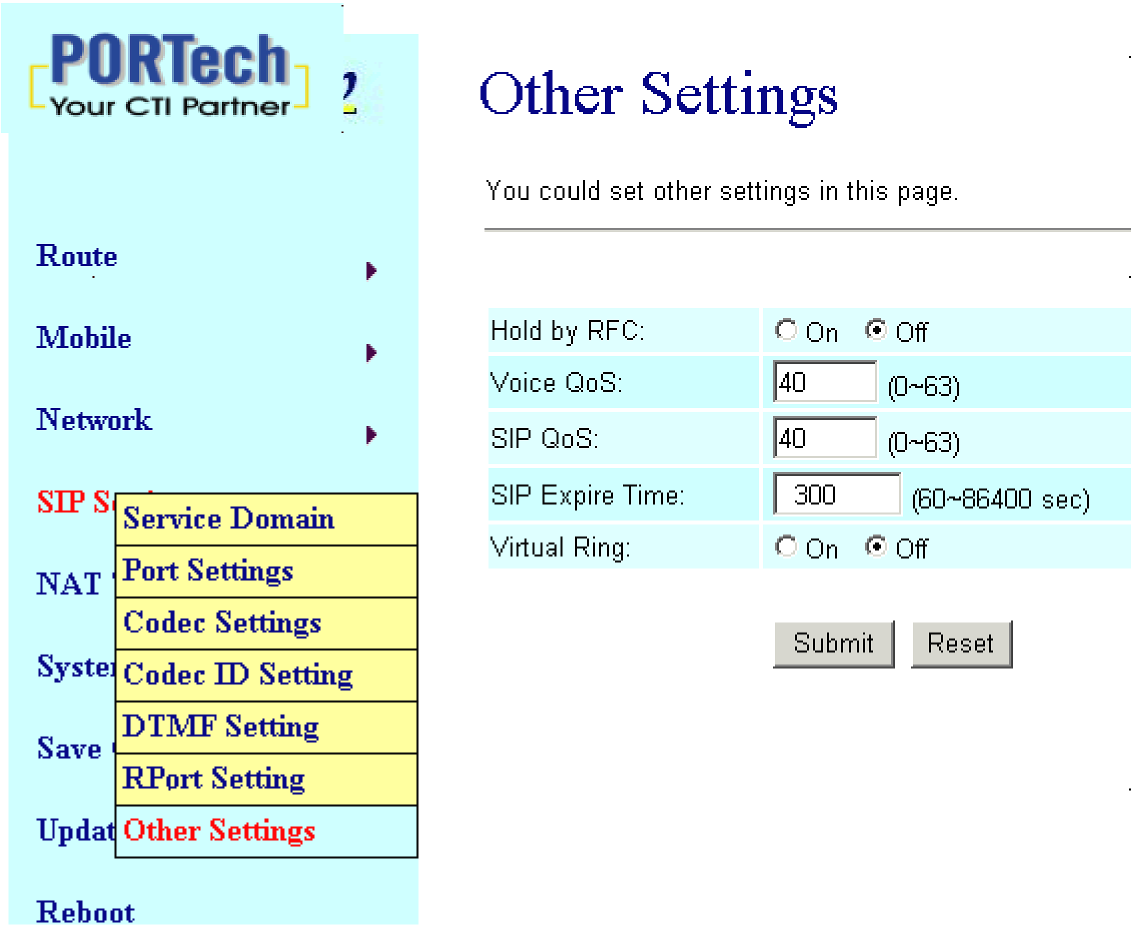
11.7.2 180 Ring on/off: LAN TO MOBILE two stage dialing can be turn off, therefore there will be no the Ring Back Tone, all the phone call will be transferred to Voice-Attempt directly. (For this function, 183 must be turn on)

11.7.3 183(Session Progress)-->[It means"on progressing"] : When you turn 183 on, it means you can hear Voice-Attempt while GMS side are busying. We recommend you to turn this on if you use SIP Proxy.

11.8 Other Settings

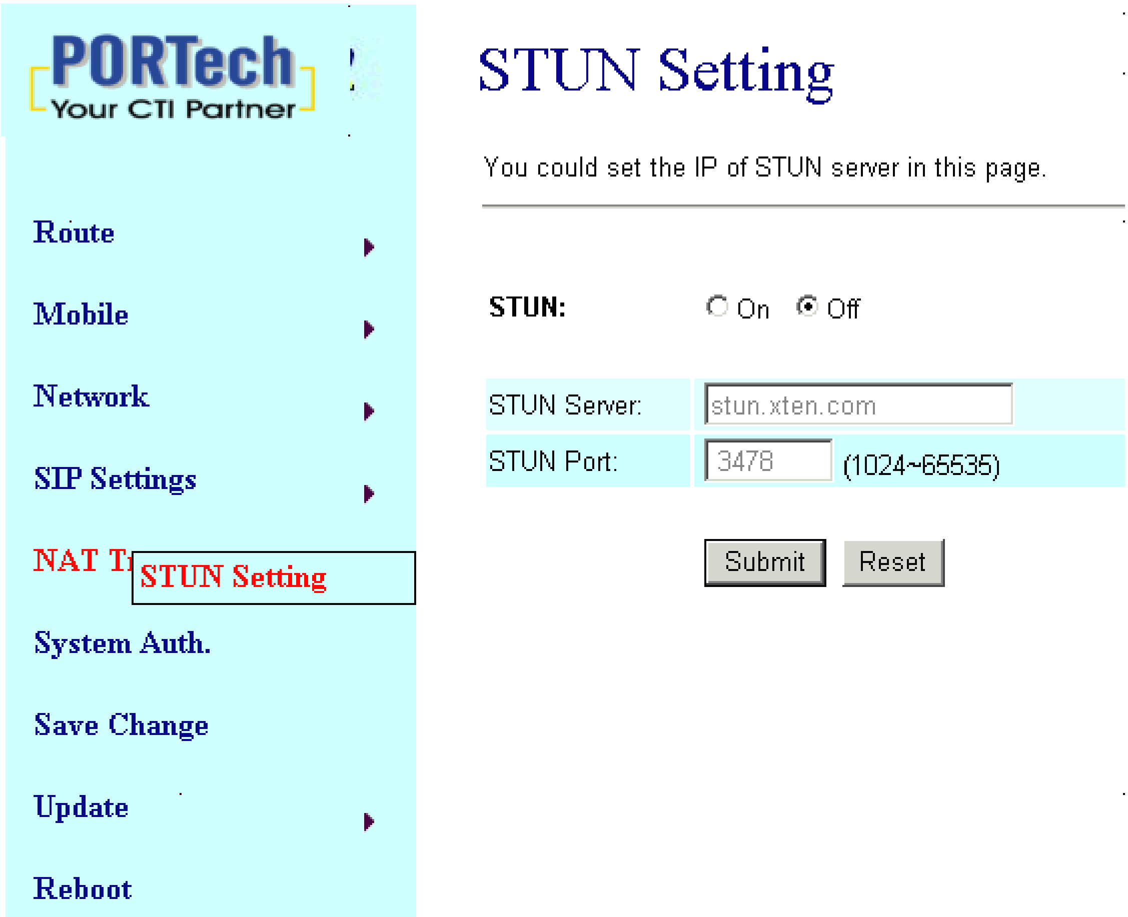
You could setup the RFC and QoS according to your ISP information. After setting, remember to click the Submit button.

The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.



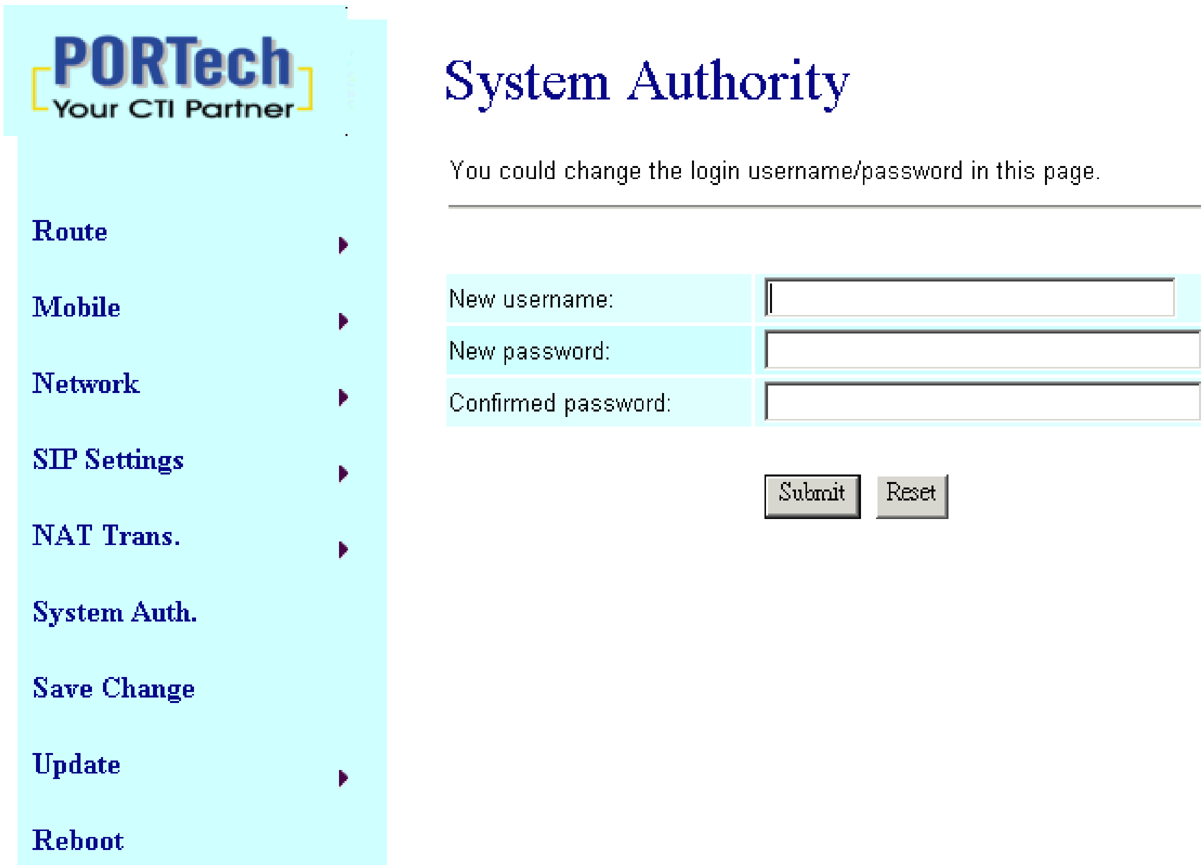
12. NAT Trans

In this page: NAT Trans./ STUN, you could setup the STUN Enable/Disable and STUN Server IP address. This function helps your VoIP device work properly behind NAT. Change these settings according to your ISP information. After setting, remember to click the Submit button.



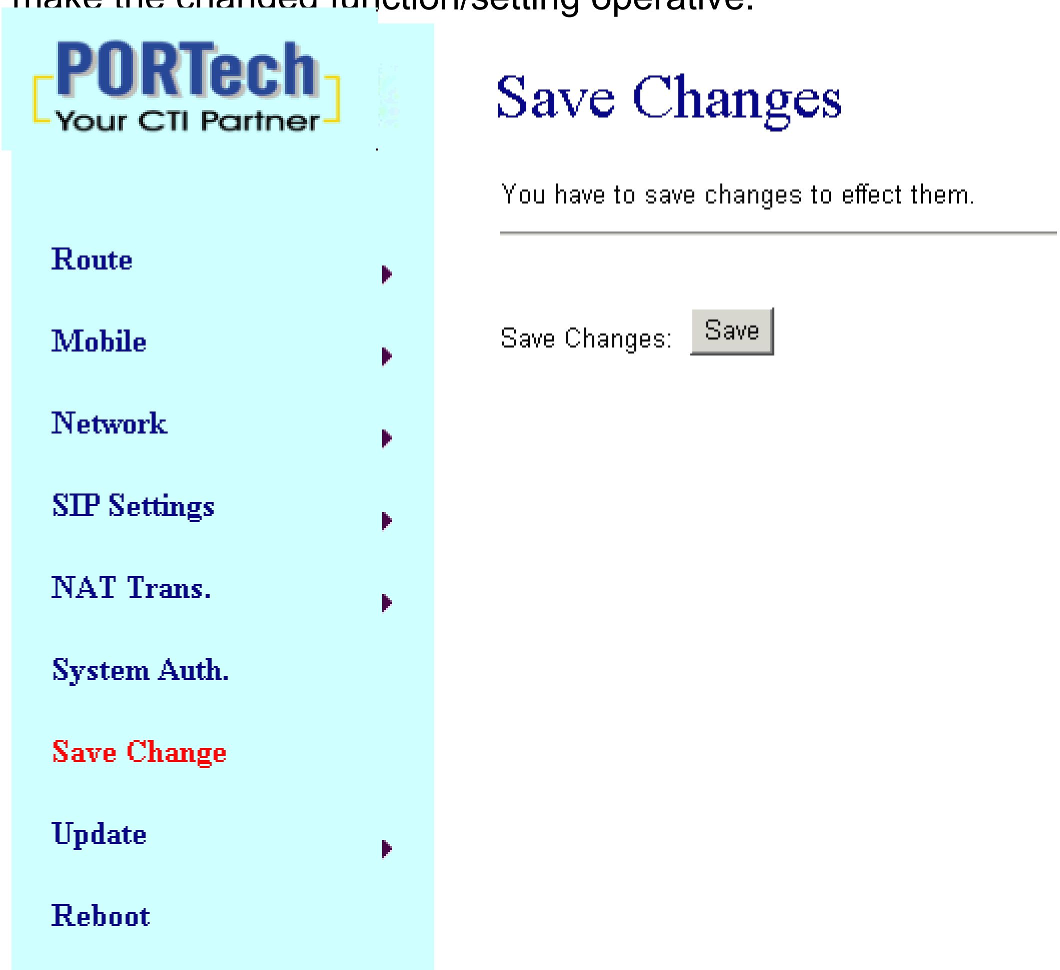
13.System Auth.

In this page: System Authority, you could change your login name and password.



14.Save Change

Please remember this step whenever you submit any setting.
Click “Save Change” then “Save” button, the system will restart and make the changed function/setting operative.

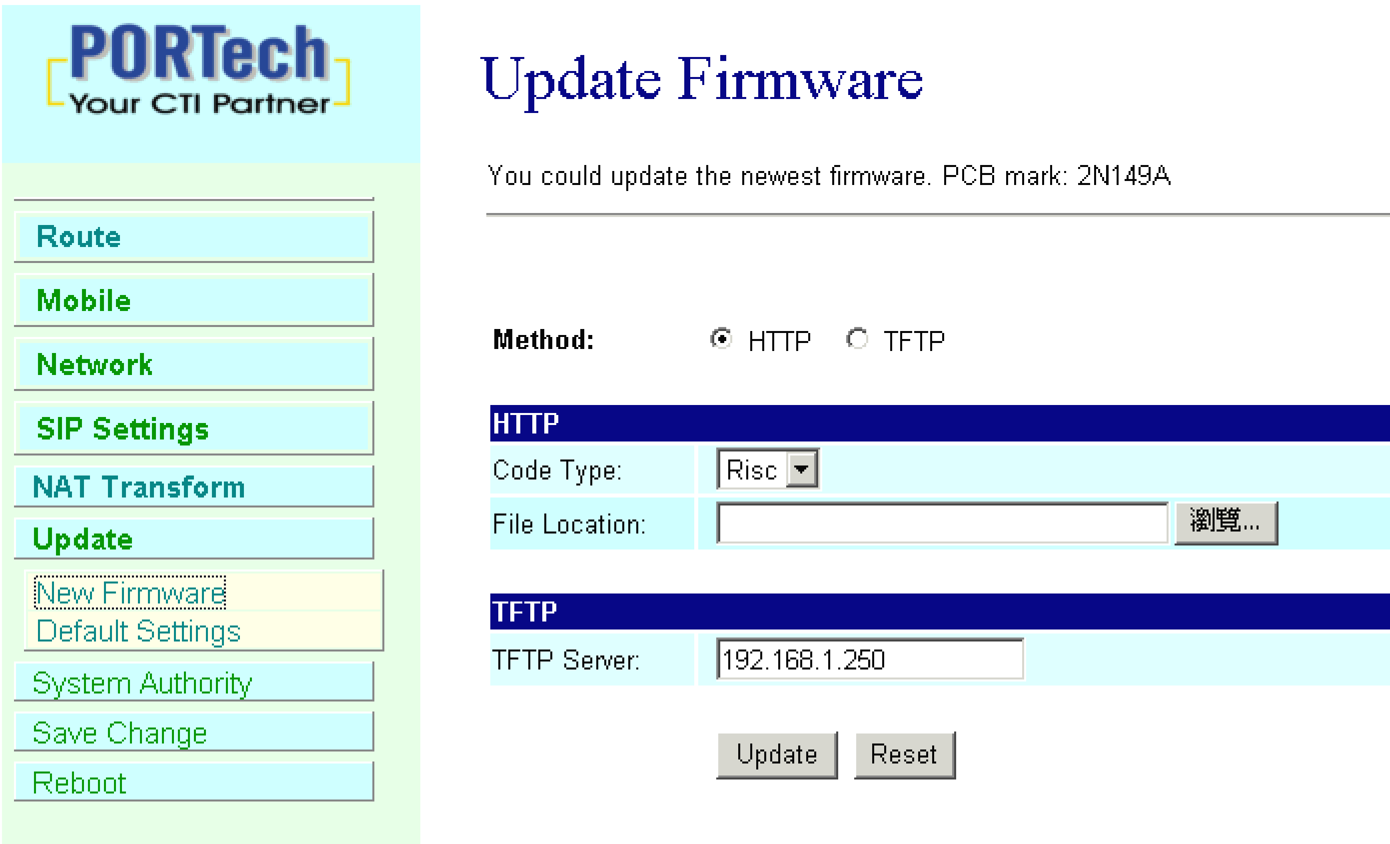


15.Update

Here you could update the latest firmware and restore the default settings.

15.1 Update/ New Firmware/ Update Firmware
Download the latest firmware, then

(1) Method: select "HTTP"
(2) Code Type: select "Risc".
(3) File Location: Click the "Browse" button in the right side of the File Location for the file.
Please note: no need to unzip the firmware file.
(4) Click "Update", it takes few minutes to generate new firmware.

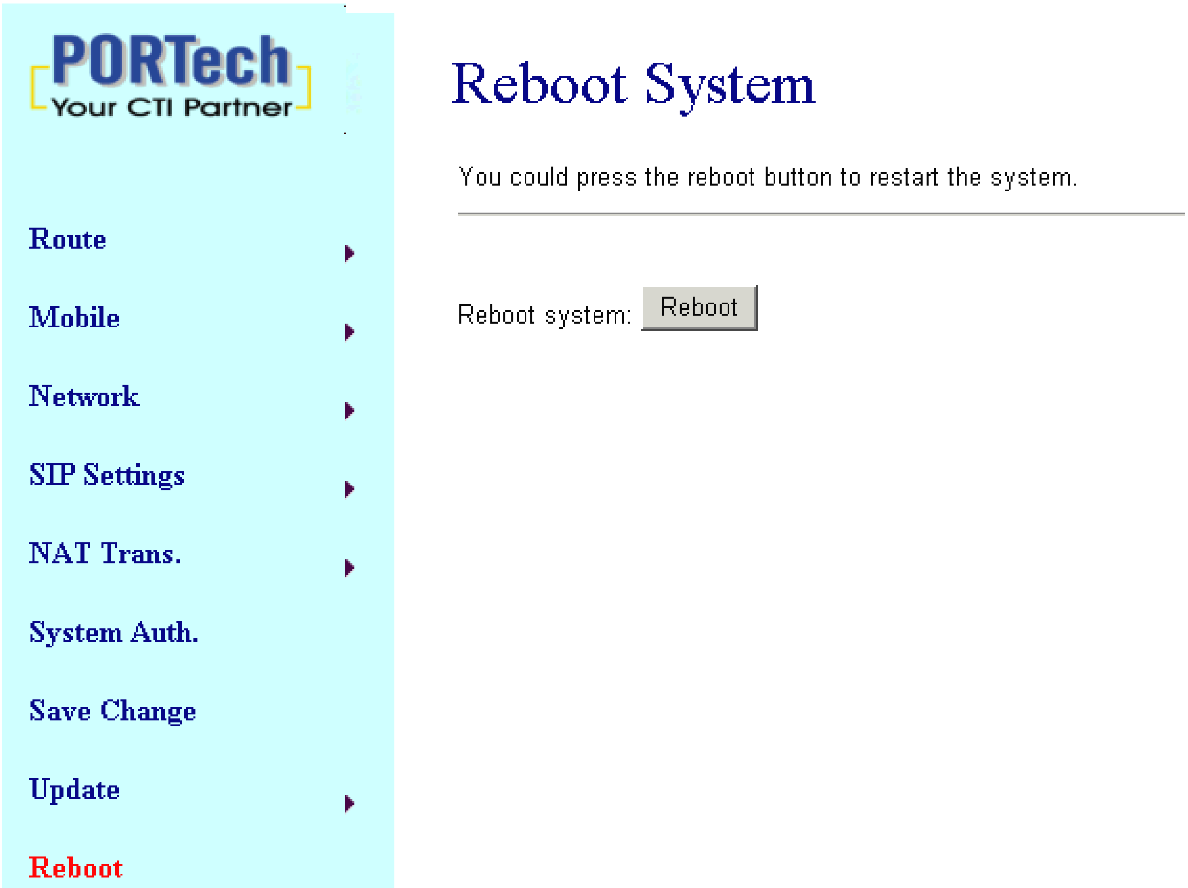


15.2 Restore Default Settings

In this page: Update/ Default Settings, you could restore the factory default settings to the system. Click the Restore button, then the system returns to default IP http://192.168.0.100 (the other settings e.g SIP setting, mac address remains), and automatically restart.

16.Reboot

In this page, you could click the Reboot button to restart the system.



17. Setting and checking via IVR

User could get or set some parameters of the system by dialing in the mobile numbers of the device. The status or result is reported via voice response system. In the first 20 seconds after power-on (when only Mobile light flash), you could dial its mobile numbers. When you hear the dial tone, press the following codes to set or check the device.

Item	Function	Code	Action
1	Reboot	#195#	Reboot the device
2	Factory Reset	#198#	Return to default settings
3	Check IP Address	#120#	IVR announces the current IP address. Default: 192.168.0.100
4	Check IP Type	#121#	IVR announces DHCP is on or off. Default: off
5	Check Network Mask	#123#	IVR announces the current network mask. Default: 255.255.255.0
6	Check Gateway IP Address	#124#	IVR announces the current gateway IP address. Default: 192.168.0.254
7	Check Primary DNS Server	#125#	IVR announces the current setting in the Primary DNS field. Default: 192.168.0.1
8	Check Firmware Version	#128#	IVR announces the version of the firmware.
9	Set as DHCP client	#111#	The system is changed to DHCP Client type
10	Set Static IP Address	#112xxx*xxx*xxx*xxx#	DHCP is disable and system is changed to static IP type. Enter IP address using numbers on the telephone keypad. 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 keypad. 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 keypad. 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 keypad. Use the * (star) key when entering a decimal point.

18.Specification

18.1 Protocols

SIP (RFC2543, RFC3261)

18.2 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

18.3 Codec

G.711 u-Law

G.711 a-Law

G.723.1 (5.3k)

G.723.1 (6.3k)

G.729A

G.729A/B

18.4 Voice Quality

VAD

CNG

AEC, LEC

Packet loss

18.5 GSM (MV-370)

Dual BAND: 900/1800 MHZ

Tri BAND: 900/1800/1900 MHZ

Quad BAND: 900/1800/1900/850 MHZ

19. Applications

1.Connect to VoipBuster

a). Register VoipBuster account at Service Domain.

b). Route setting: Mobile to Lan set: *,*

When you call in GSM number of MV-370, you can enter destination number that will dial out from VoipBuster. (Landline is free, GSM rate is cheap)

2.How to apply 2 sets of MV-370?

(1) When you call the no.1 MV-370 gsm number, it will provide dial tone and you enter a destination number. Then no.2 MV-370 will dial this number and connect.

Step 1:no.1 MV-370: mobile to lan set route table *,*

Step 2:no.2 MV-370:lan to mobile set route table *,#

Step 3:Additionally, two pcs MV-370 both need to register proxy server.

Step 4:And proxy server set the route that the prefix of destination number to dial out from no.2 MV-370.

(2) When you call the no.1 MV-370 gsm number,no.2 MV-370 will dial this specific number and connect

Step 1:no.1 MV-370: mobile to lan set route table *, specific destination number

Step 2:no.2 MV-370:lan to mobile set route table *,#

Step 3:Additionally, two pcs MV-370 both need to register proxy server.

Step 4:And proxy server set the route that the prefix of destination number to dial out from no.2 MV-370.

20. 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, MV-370 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, MV-370 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, MV-370 will provide dial tone and you can enter mobile number.	
(2)	*, specific mobile number	
	when lan phone call in, MV-370 will connect with the specific mobile number auto.	
(3)	*,#--->It is 1 stage dialing	
	When lan phone and MV-370 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 MV-370

All changes both need to click "save and change"

21. Appendix: Setup MV-370 with Asterisk

21.1 Usage

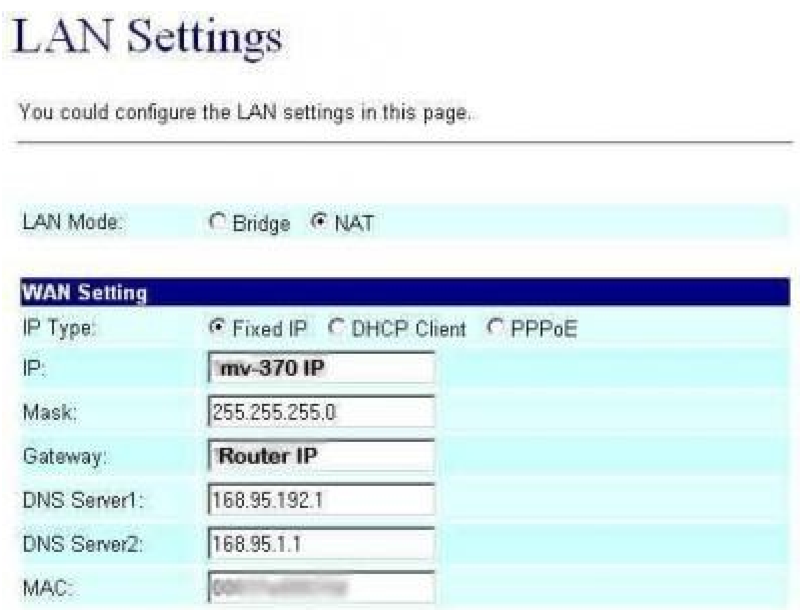
A typical usage of such a gateway is to be able to give a call with your normal mobile to any destination at voip cost :

Your mobile <----*gsm network*----> MV-370 <--*lan*--> Asterisk <--*internet*--> VOIP provider <--*whatever*--> landline

To do such a call, you just call your MV-370 number (it has its own simcard), then you get an invitation tone, then you dial the number which is handled by Asterisk.
If you have some special deals with your mobile operator, like free special number, you can call your MV-370 for free.
You can then call all around the world from your mobile at voip cost :-)

21.2 MV-370 Configuration
Once you've configured everything in the box, one good advice is to unplug the power and to restart it. By this way you should have all the parameters taken into account.

To have the MV-370 to work with Asterisk, you need first to configure the box.
Here are some screen shots showing all the important parameters.
You have to note that in all the configuration process, the MV-370 is considered as extension '103' of the IPBX.
In **Bold** are the parameters depending on your installation



LAN To Mobile Table

¡@

Page: 1

Item	URL	Call Num	Select
0	your asterisk IP	#	
1			
2			
3			
4			
5			
6			
7			
8			
9			

Here the '#' is important to avoid the two stage dialing when you give a call from Asterisk to GSM.

Mobile To LAN Table

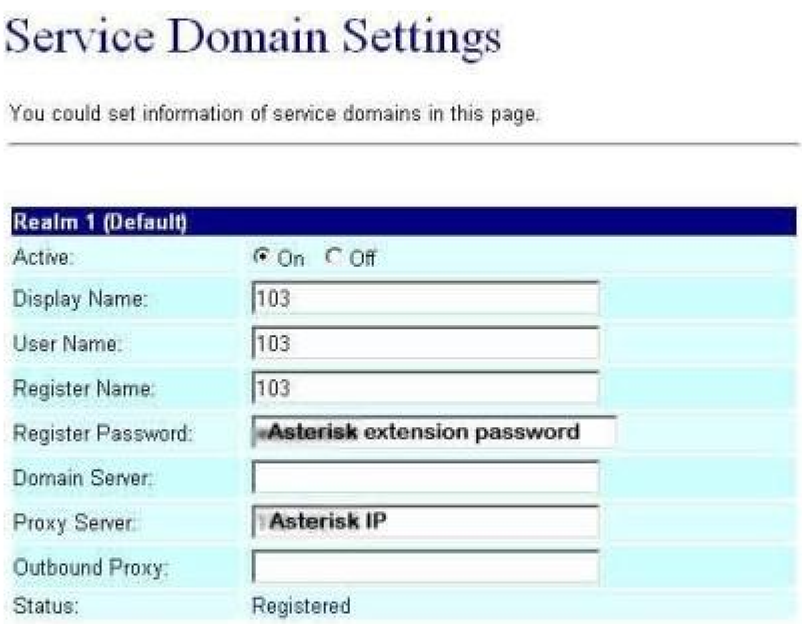
¡@

Page: 1

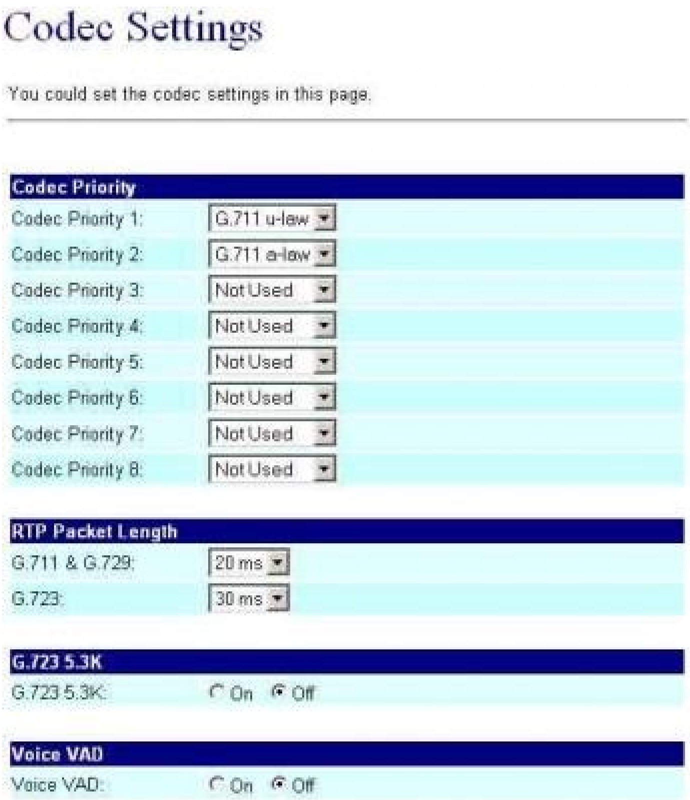
Item	CID	URL	Select
0	authorised mobile n°	103	
1	another authorised n°	103	
2			
3			
4			
5			
6			
7			
8			
9			

The mobile number you give in that page are the authorised mobile

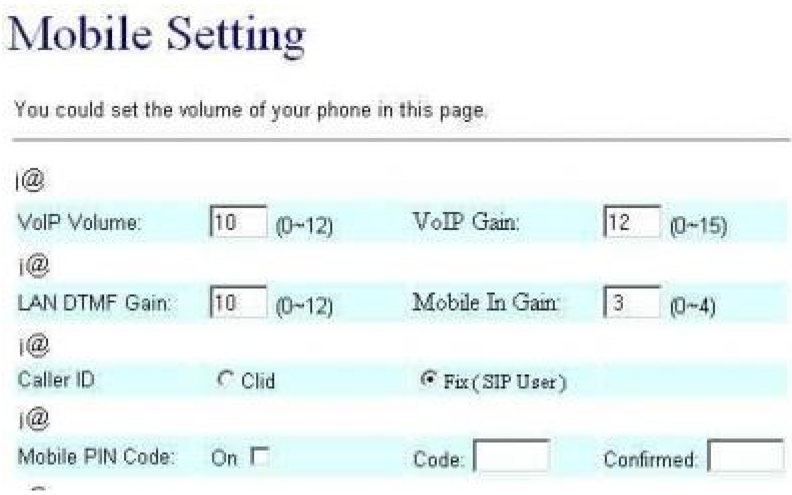
which can call GSM to Asterisk.
These mobile number must be defined as your GSM provider displays the number.
If you don't know how it is displayed, just give a call to the box and check the number given in the 'Incoming Mob' field of the 'Mobile Status' page.
Any number which is not in that list won't have acces to the LAN side, so to Asterisk.
If you want to allow any number, just set '*' in that field ... but beware of the bill ;-)



Once Asterisk configuration is made, you should get 'Registered' on the Realm1.



It is very important to use only u-law or a-law as all DTMF is inband. So if you want to be able to do some DISA when you call from GSM to Asterisk, it has to be one of these 2 codecs.



These settings seem to be ok, just adjust ...

21.3 Antenna position
Another important thing is to properly place the provided antenna.
If your gsm reception is good, you should get around 18 or 19 as Signal Quality in the "Mobile Status" page.
With that level of signal quality, your audio quality will be very good.
On the other end,the signal quality down to 11, audio becomes very jerky.
So, maximum signal quality = maximum audio quality.

21.4 Asterisk configuration
Once the MV-370 is set, you have to configure Asterisk.
On that side, you have to setup files as follow :

21.5 sip.conf
; GSM VOIP Gateway MV-370
[103]
type=friend
username=103
fromuser=103
regexten=103 ; When they register, create extension 401
secret=xxxxxxx ; Asterisk extension password
context=gateway ; Incoming calls context
dtmfmode=inband ; Very important for DISA to work
call-limit=1 ; Limit to 1 call max
callerid=GSM Gateway <103>
host=dynamic
nat=no ; Gateway is not behind a NAT router
canreinvite=no ; Typically set to NO if behind NAT
insecure=very
qualify=yes
disallow=all

allow=ulaw ; prefered codec for DTMF detection
allow=alaw

21.6 extensions.conf

```
; ******* GSM Gateway incoming calls **********
[gateway]
exten => _103,1,Answer()
exten => _103,2,DigitTimeout(3) ; give enough time to do second stage dialing
exten => _103,3,ResponseTimeout(5)
exten => _103,4,DISA(no-password|outgoing) ; here 'outgoing' is the normal context to deal with the dial plan

[outgoing]
...
; example of LAN to GSM call
; call the MV-370 sim card mail box thru GSM
exten => _888,1,SetCallerID("xxxxxxxxxx")
exten => _888,2,Dial(SIP/${EXTEN}@103,60,r)
exten => _888,3,Hangup()
```

22.How to setup Asterisk to receive Caller ID from

Test version
trixbox-2.2

SIP Softphone

- SJPhone 1.60.289a
- X-Lite 1105x

Modify file

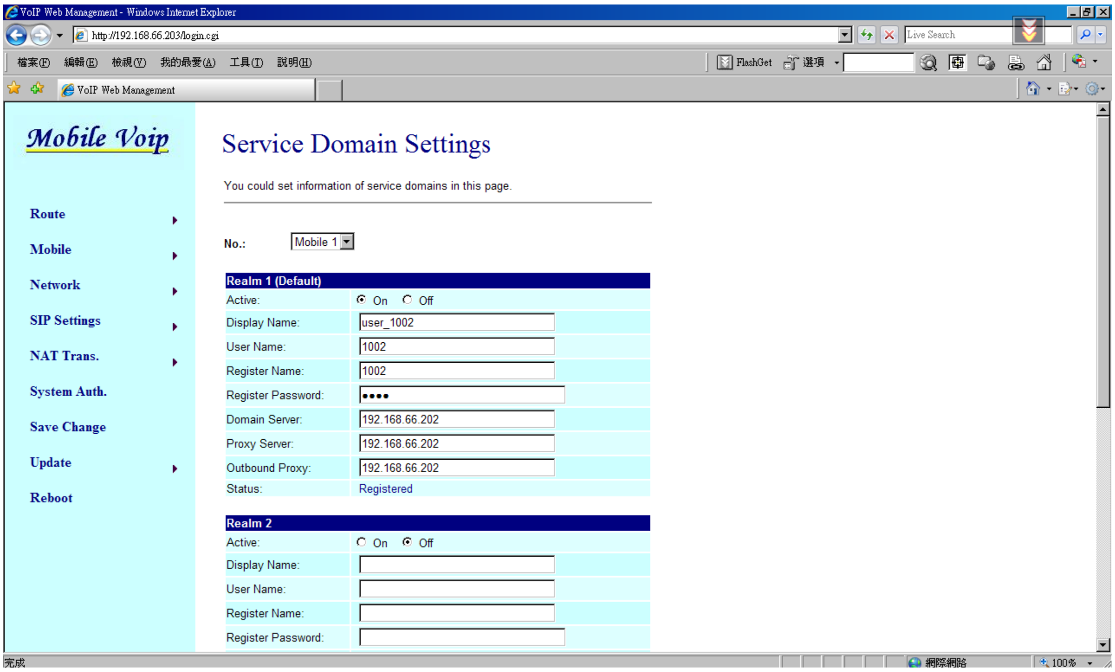
- Add the following setting to/etc/asterisk/sip.conf

```
[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
MV-370: address=192.168.66.203:5060; username=1002, displayname=user_1002



test1
pstn → call 0928492911(mobile number) → MV-370 → 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 → MV-370 → 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=z9hG4bK4C4315351FC84CA582D14FB8C25FC3BF
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="8aaaaa5b5ad53654bf0a2ab0fa9bb118",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