MV-374 / MV-378

VoIP GSM Gateway

User Manual

MV-374

MV-378

PulseSupply
pulsesupply.com

【Content】

1. Introduction

MV-374/MV-378 is a 4 / 8 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 4 / 8 calls simultaneously from IP phones to GSM networks and GSM network to IP phone.

2. Function description

2.1 VoIP(SIP)、GSM 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，
*It communicates with other gateway or PC.

3. Parts list

3.1 「MV-374/MV-378」main body

3.2 Power adaptor
MV-374 Output 12V/5A ,Input 100~240V Auto switching
MV-378 Output 12V/9A ,Input 100~240V Auto switching

3.3 Network cable

3.4 Antenna: MV-374:1 pcs / MV-378: 2 pcs

3.5 Rack-mount accessories (compatible with 19“Rack) – option

3.6 User Manual

(3.1) MV-374

(3.1) MV-378

(3.2) MV-374

(3.2) MV-378

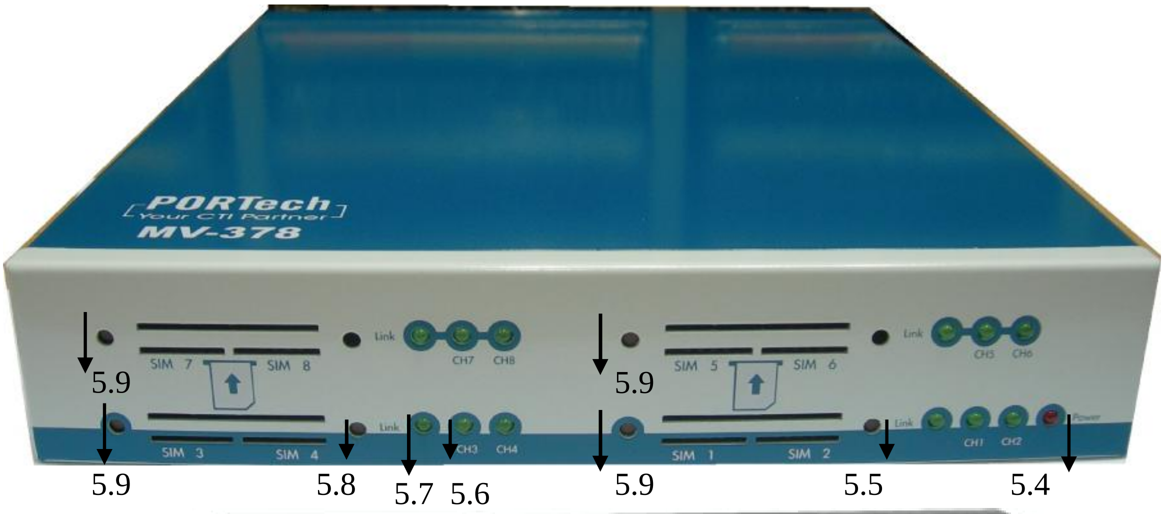
(3.3)

(3.4)

(3.5) option

4. Dimension: 30x28x4 cm

5. Chart of the device



5.1 Antenna：Antenna Connector.

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

5.3 DC 12V：Power input.

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

5.5 IP Reset Button: Press this button about 10 seconds

5.6 CH3：An indicator light of VoIP3

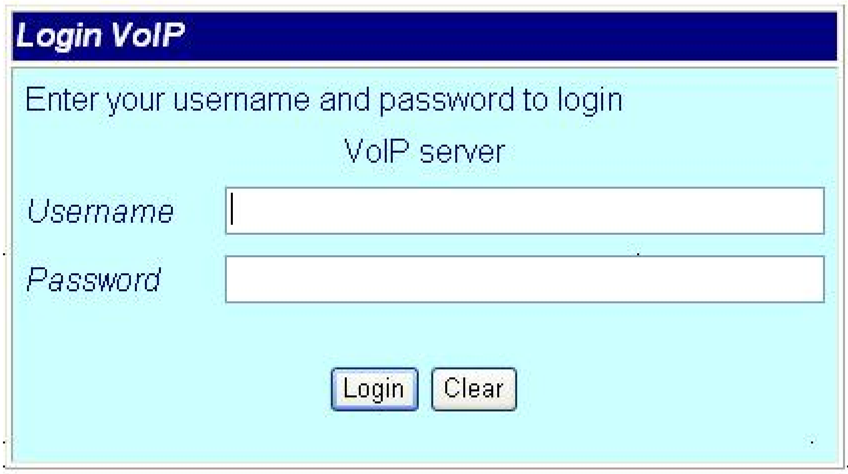
5.7 LINK Indicator：Light up when network is connected.

5.8 Reboot Button: all channels reboot

5.9 Reboot ch1-2/ch3-4/ch5-6/ch7-8 without power off

6. 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.

7. System Information

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

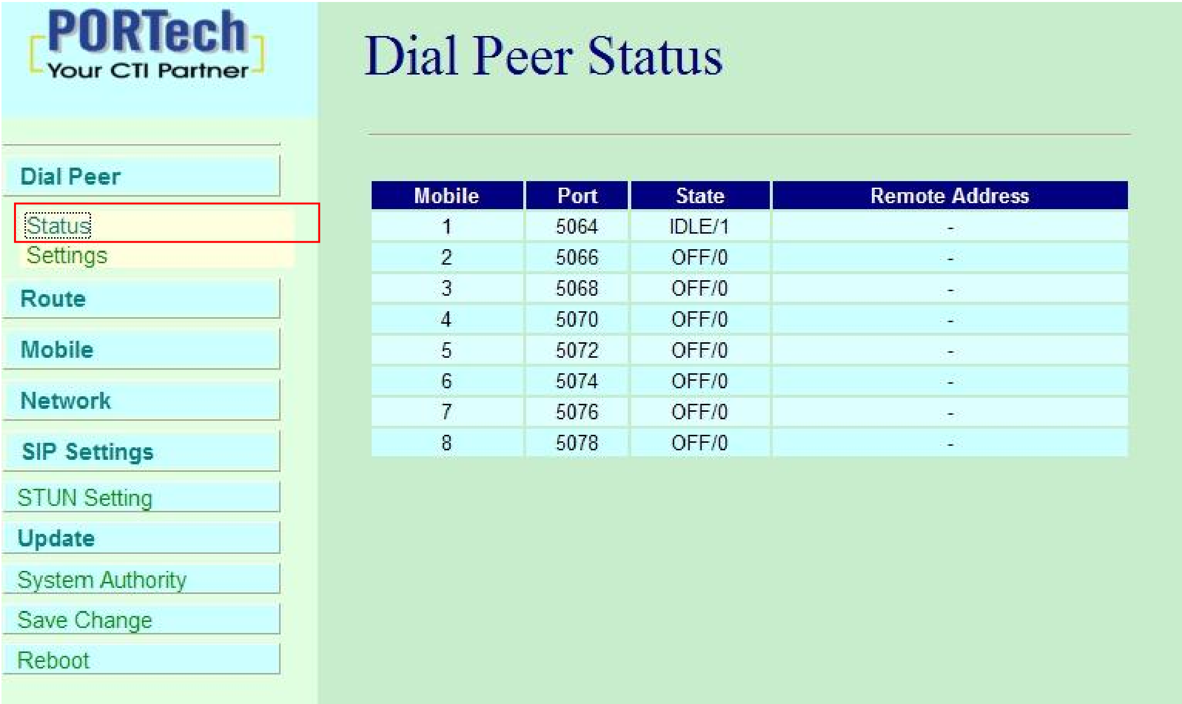
7.2 Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.

8. Dial Peer

8.1 Status

You can check Dial Peer Status here

All the information will be shown on this page.



Mobile	Port	State	Remote Address
1	5064	IDLE/1	-
2	5066	OFF/0	-
3	5068	OFF/0	-
4	5070	OFF/0	-
5	5072	OFF/0	-
6	5074	OFF/0	-
7	5076	OFF/0	-
8	5078	OFF/0	-

Default: Ch1: 5064 Ch2: 5066 Ch3: 5068 Ch4:5070…….

You can change the ports on SIP Settings/Ports settings

State status:

INIT/0: GSM module is initialing

IDLE/0: GSM module not register

IDLE/1: GSM module registered

BUSY: GSM port is busy

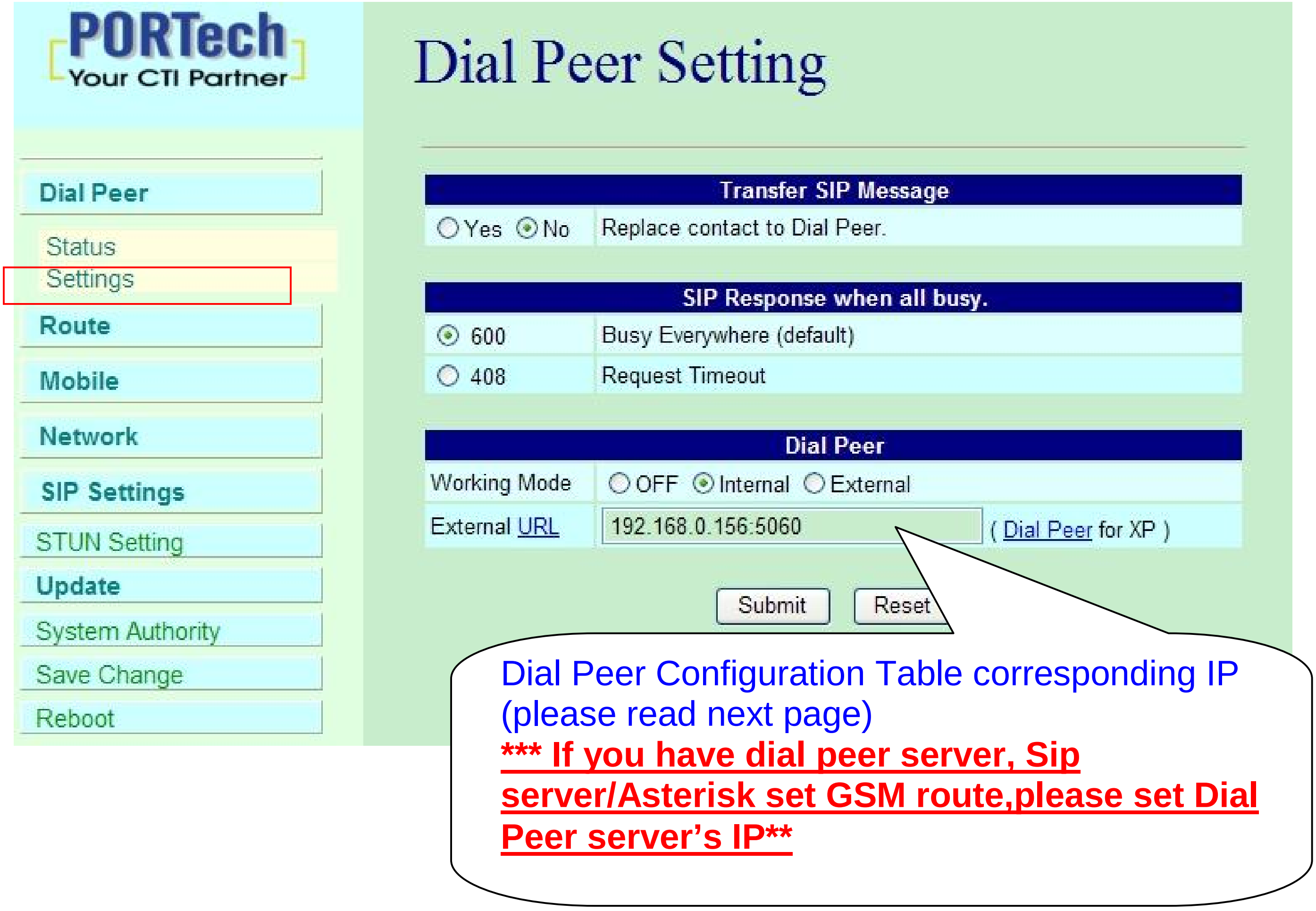
LISTEN: GSM port is engaged

OFF/0: GSM module is out of working

Remote Address:

The IP Address which came from LAN side

8.2 Settings



1. Transfer SIP Message

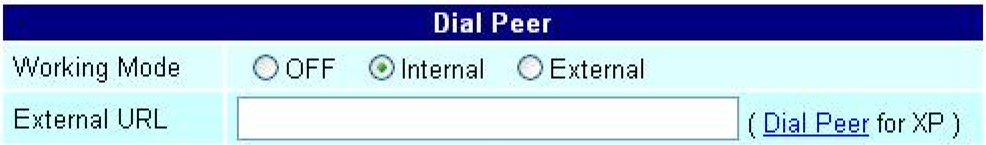
The Replace contact to dial peer: The default is OFF, which won't send the SIP message to corresponding port through Dial Peer.
If ON, all SIP messages will send to corresponding port via Dial Peer.

2. SIP Response when all busy

Both 600 and 408 are SIP message, that user can select the corresponding response while all ports are busy.
The Default is 600

8.2.1 Dial Peer

Lan to mobile *,#: Dial peer software will look for available channel to dial out.



Working Mode: OFF → To disable dial peer, so MV-378 will working under one IP and 8 ports

Internal → To motivate dial peer, so MV-378 will working under one IP and one Port.

Mode: calls will come to dial peer, and dial peer will route calls to idle channels.

> **E.g SIP Server sends call to MV-378 IP: 5060**
> when the first port is busy, MV-378 will use the second port to dial out…and so forth.

External → MV-378/MV-374 will be controlled by external dial peer program.

External URL → External dial peer program's IP address and port number.

Edit DialPeer.ini (External Dial Peer)

[Window]
Xpos=512
Ypos=252
Width=471
Height=399
[Info]
Total=16

Total ip / port

[VoipIP]
1=192.168.0.100
2=192.168.0.100
3=192.168.0.100
4=192.168.0.100
5=192.168.0.100
6=192.168.0.100
7=192.168.0.100
8=192.168.0.100

The first MV-378

9=192.168.0.110
10=192.168.0.110
11=192.168.0.110
12=192.168.0.110
13=192.168.0.110
14=192.168.0.110
15=192.168.0.110
16=192.168.0.110

The second MV-378

[SipPort]
1=5060
2=5062
3=5064
4=5066
5=5068
6=5070
7=5072
8=5074

The first MV-378

9=5060
10=5062
11=5064
12=5066
13=5068
14=5070
15=5072
16=5074

The second MV-378

[RtpPort]
1=60000
2=60002
3=60004
4=60006
5=60008
6=60010
7=60012
8=60014

The first MV-378

9=60000
10=60002
11=60004
12=60006
13=60008
14=60010
15=60012
16=60014

The second MV-378

[PtcPort]
1=40000
2=40000
3=40008
4=40008
5=40016
6=40016
7=40024
8=40024

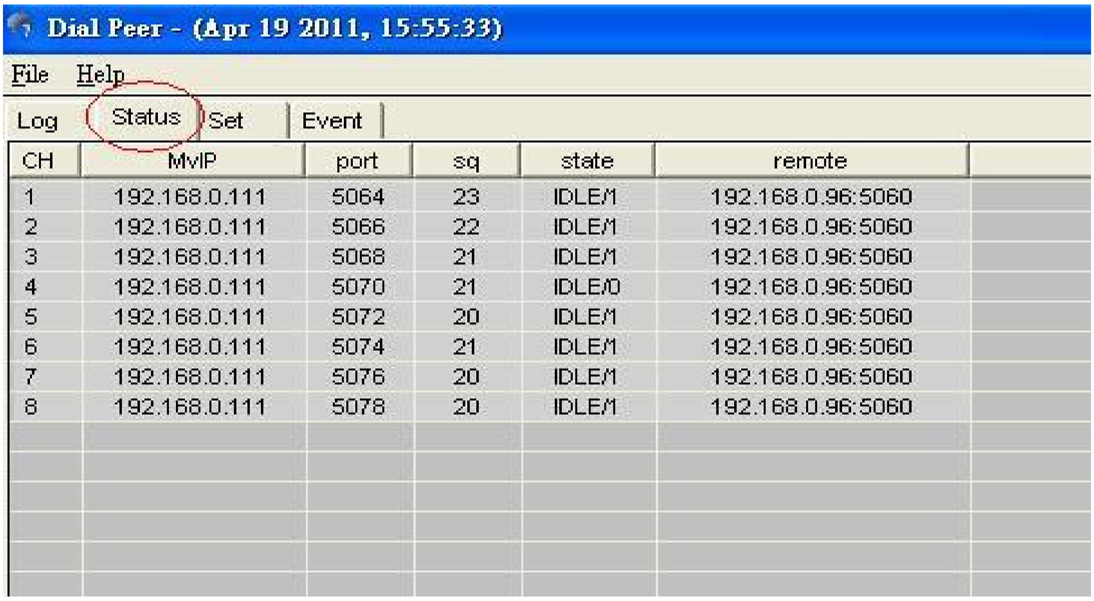
The first MV-378

9=40000
10=40000
11=40008
12=40008
13=40016
14=40016
15=40024
16=40024

The second MV-378

External Dial Peer Log

You can check the Statue here



CH	MvIP	port	sq	state	remote
1	192.168.0.111	5064	23	IDLE/1	192.168.0.96:5060
2	192.168.0.111	5066	22	IDLE/1	192.168.0.96:5060
3	192.168.0.111	5068	21	IDLE/1	192.168.0.96:5060
4	192.168.0.111	5070	21	IDLE/0	192.168.0.96:5060
5	192.168.0.111	5072	20	IDLE/1	192.168.0.96:5060
6	192.168.0.111	5074	21	IDLE/1	192.168.0.96:5060
7	192.168.0.111	5076	20	IDLE/1	192.168.0.96:5060
8	192.168.0.111	5078	20	IDLE/1	192.168.0.96:5060

1. CH: The number for GSM port of MV-37X
2. MvIP: The IP address of MV-37X for Dial Peer connection
3. Port: The corresponding port for MV-37X
4. Sq: Signal Quality for MV-37X GSM Port:
5. State: The GSM Port Sate status

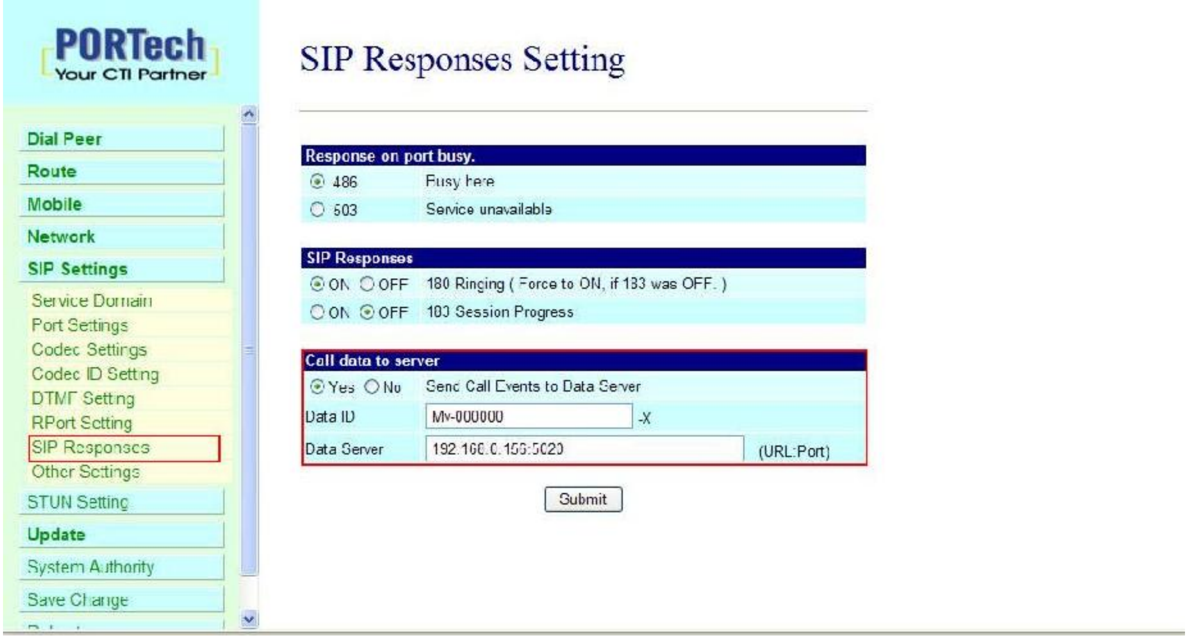
   INIT/1: GSM module is initialing
   IDLE/0: GSM module is not register
   IDLE/1: GSM module is registered
   BUSY: GSM Port is busy
   LISTEN: GSM port is engaged
   OFF/0: GSM module is out of working
6. Remote: The VoIP Sender’s IP

8.3 Call Data to Server (CDR)

MV can provide Call Detail Record (CDR) for traffic and accounting management. User need to download external Dial Peer software on PC and can monitor traffic.

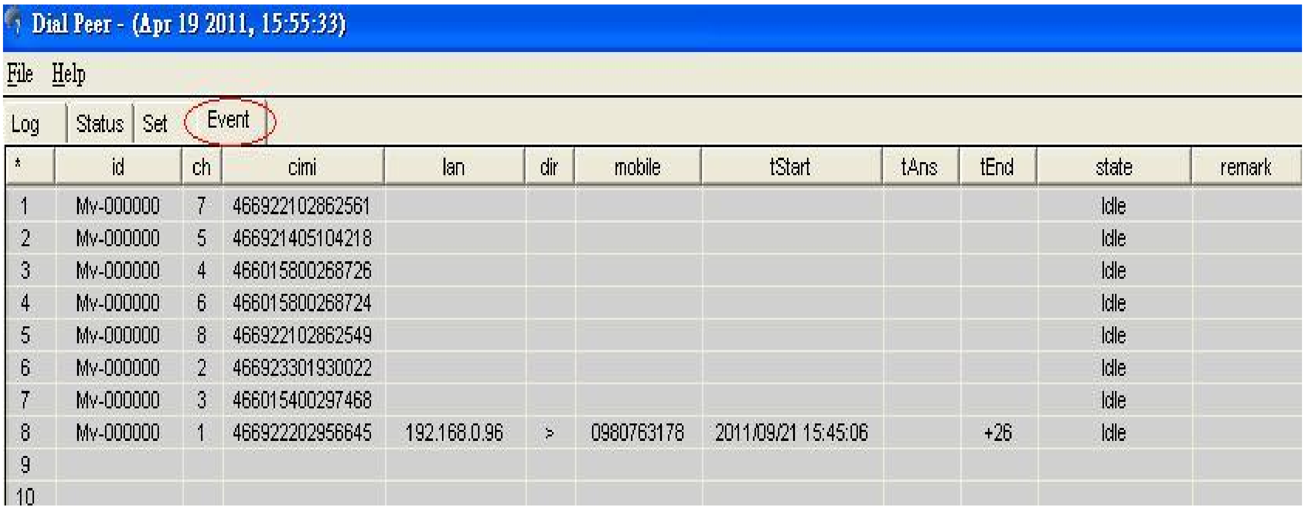
Data ID: MV will create one default Data ID

Data Server: Please fill the PC's IP, which is executed External Dial Peer Software



External Dial Peer

You can check CDR Statue here



Dial Peer - (Apr 19 2011, 15:55:33)

File Help

Log | Status | Set | Event

*	id	ch	cimi	lan	dir	mobile	tStart	tAns	tEnd	state	remark
1	Mv-000000	7	466922102862561							Idle	
2	Mv-000000	5	466921405104218							Idle	
3	Mv-000000	4	466015800268726							Idle	
4	Mv-000000	6	466015800268724							Idle	
5	Mv-000000	8	466922102862549							Idle	
6	Mv-000000	2	466923301930022							Idle	
7	Mv-000000	3	466015400297468							Idle	
8	Mv-000000	1	466922202956645	192.168.0.96	>	0980763178	2011/09/21 15:45:06		+26	Idle	
9											
10											

1. ID: The MV's Data ID
2. CH: The GSM channel of MV-37X
3. Cimi: The SIM Card ID
4. Lan: Show the outgoing Lan IP or Incoming Lan IP
5. Dir: The Arrow shows the route to be Lan to Mobile or Mobile to Lan
6. Mobile: The outgoing mobile number or incoming mobile number
7. tStart: When the call started(date and time)
8. tANS: The second answering the call
9. tEND: The second ending the call(duration)

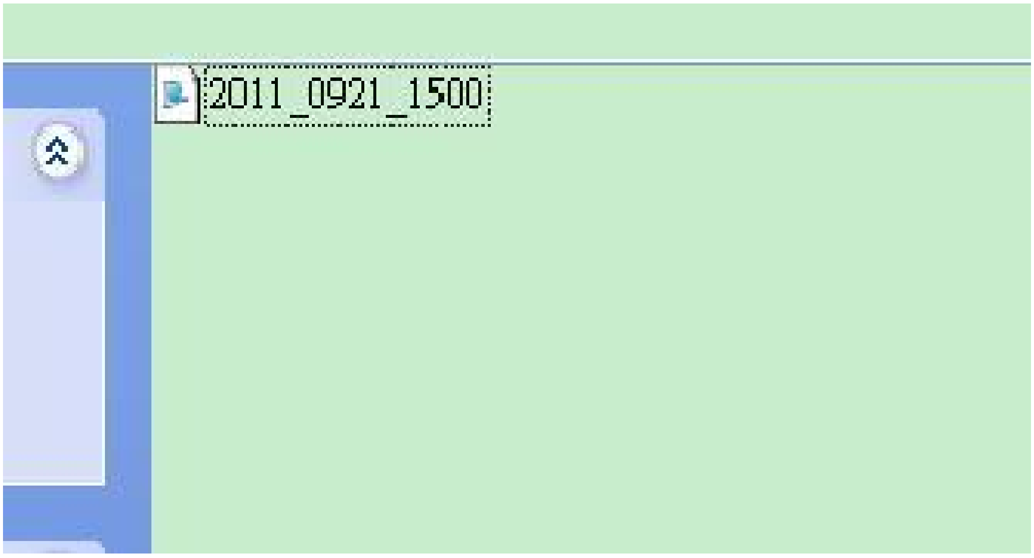
   (tANS, tEND are the exactly talking seconds)
10. State: The GSM Port Sate status

CDR Files store at C:\Program Files\DialPeer

The CDR log is stored in this “cdr” file each hour, which includes all gsm port call detials record.

If there’s no calls in this hour, it won’t creat any log.

CDR File



Example:

```
id=Mv-000000; ch=1; cimi=466922202956645; dir=L2M; iurl=192.168.0.96; omb=0980763178; tStart=4e7a0682(2011/09/21 15:45:06); tEnd=+26; state=LanEnd
```

1. Id=Mv-000000: The MV's Data ID
2. Ch=1: The 1st channel for MV ID
3. Cimi=466922202956645 : The SIM card ID for this GSM port
4. dir=L2M: The route is Lan to Mobile (If it's Mobile to Lan, that shows M2L)
5. iurl=192.168.0.96: The incoming IP
6. omb=0980763178: The outgoing number
7. tStart=4e7a0682(2011/09/21 15:45:06): The duration for the call
8. tEnd=+26: The call end on 26th second
9. state=LanEnd: The call hang up on Lan side.

9. Route

Important:

The route table -50 sets can share by two channels(1,2 ch / 3,4 ch / 5,6 ch / 7,8 ch) . The setting, please refer 9.2 Mobile setting

ex: Mobile 1 use the route table for item 0-24,

Mobile 2 use the route table for item 25-49

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 MV-374/MV-378 will transfer to the URL according to the caller ID of the Mobile.

*CID：

(1) It 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) It 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

MV-374/MV-378 have register proxy server/Asterisk

The proxy server/Asterisk have the route “09”

When the caller’s prefix number is 0932, MV-374/MV-378 will connect 0911123456 automaticlly

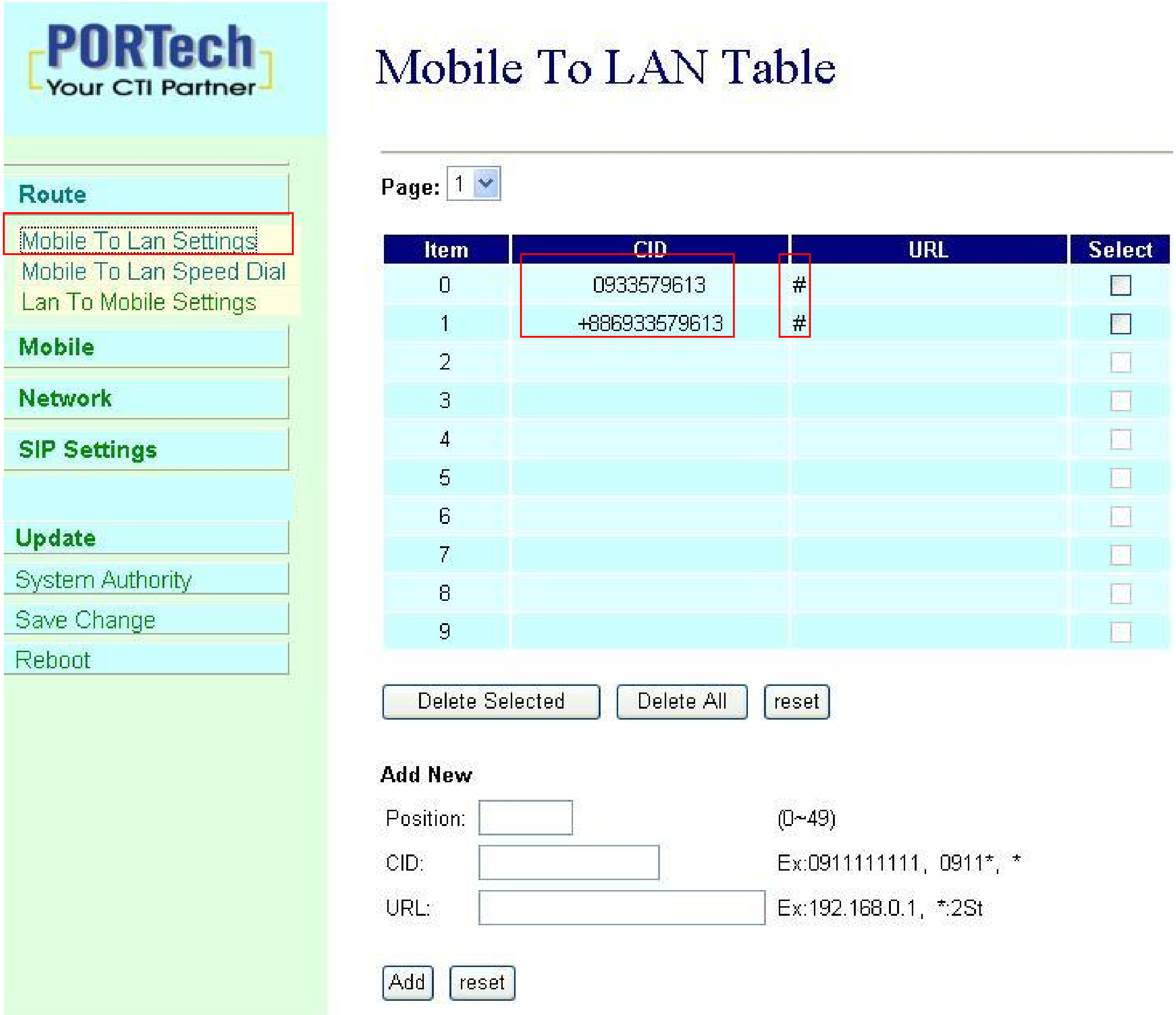
(2) Mobile to Lan: *,*

Any caller call the MV-374/MV-378’s sim,MV-374/MV-378 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 Call Back Service (50 sets)



You can set call back service as the following steps

(1) CID : set the phone number here (up to 50 sets)

(2) URL: # (# is the command of call back)

Application:

a.Call MV-374/MV-378

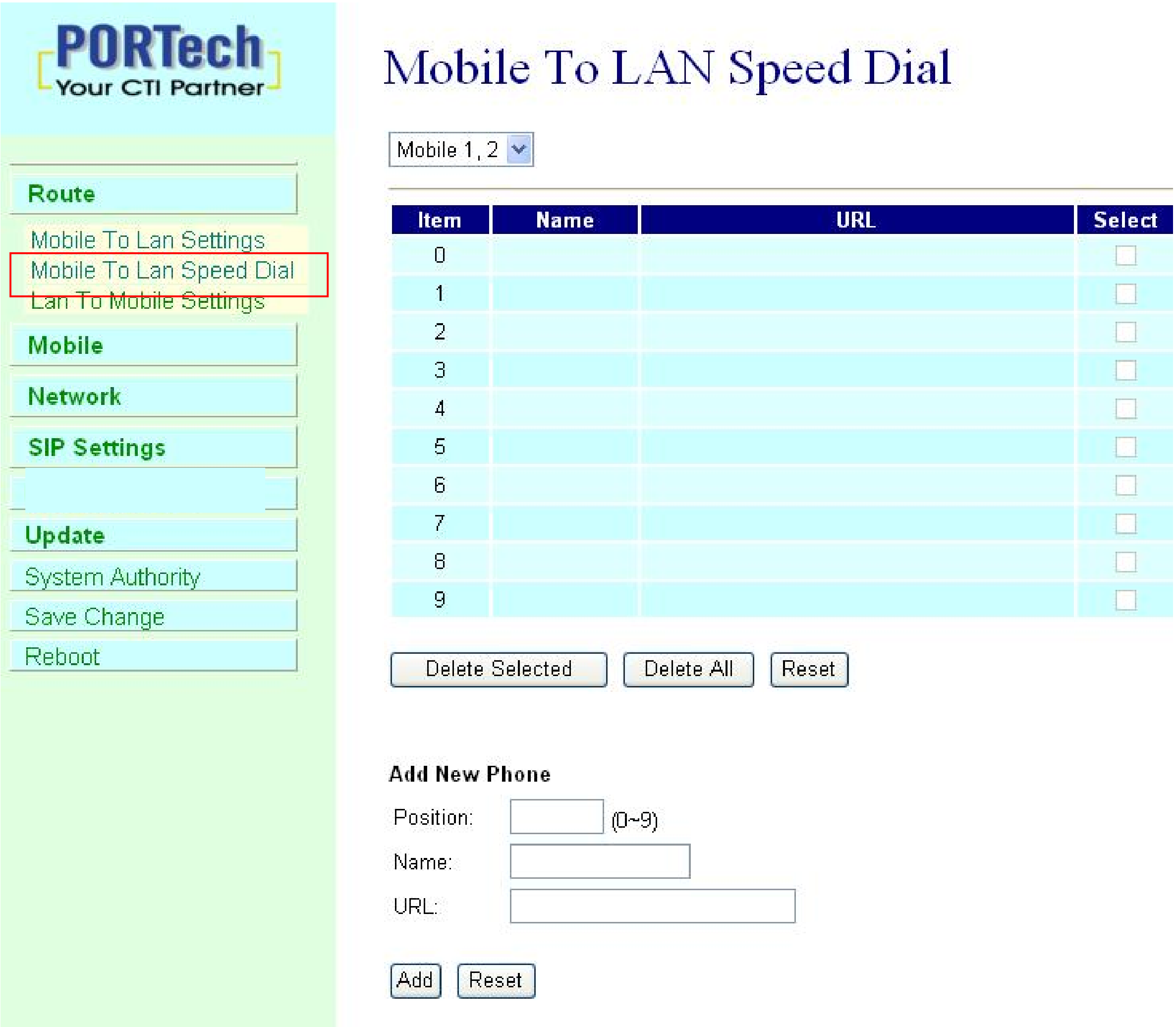
b.MV-374/MV-378 will detect the phone number is in call back list or not

c.If yes, MV-374/MV-378 will reject the call, and call it back

d.You will receive the call from MV-374/MV-378, and prompt a dial tone

9.3 Mobile to LAN Speed Dial Settings

When you set Mobile to LAN Speed Dial Settings and Mobile to LAN at the same time, MV-374/MV-378 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

9.4 LAN to Mobile Settings

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

The MV-374/MV-378 will transfer to the mobile number according to the incoming URL

*URL：It's the IP address of the incoming call.

It 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. # for one-stage dialing
4. # ['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 #d123a456 means one-stage dialing,

   delete the first 123 from your destination number,

   then add 456 in front as the new destination number.

Example:

Lan to Mobile: *, #

(1)MV-374/MV-378 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, MV-374/MV-378 will connect this call auto.

Example of Application:

When you call the ch.1 MV-374/MV-378 GSM number, it will provide dial tone and you enter a destination number.

Then ch.2 MV-374/MV-378 will dial this number and connect.

Ch.1 MV-374/MV-378: mobile to lan set route table *,*

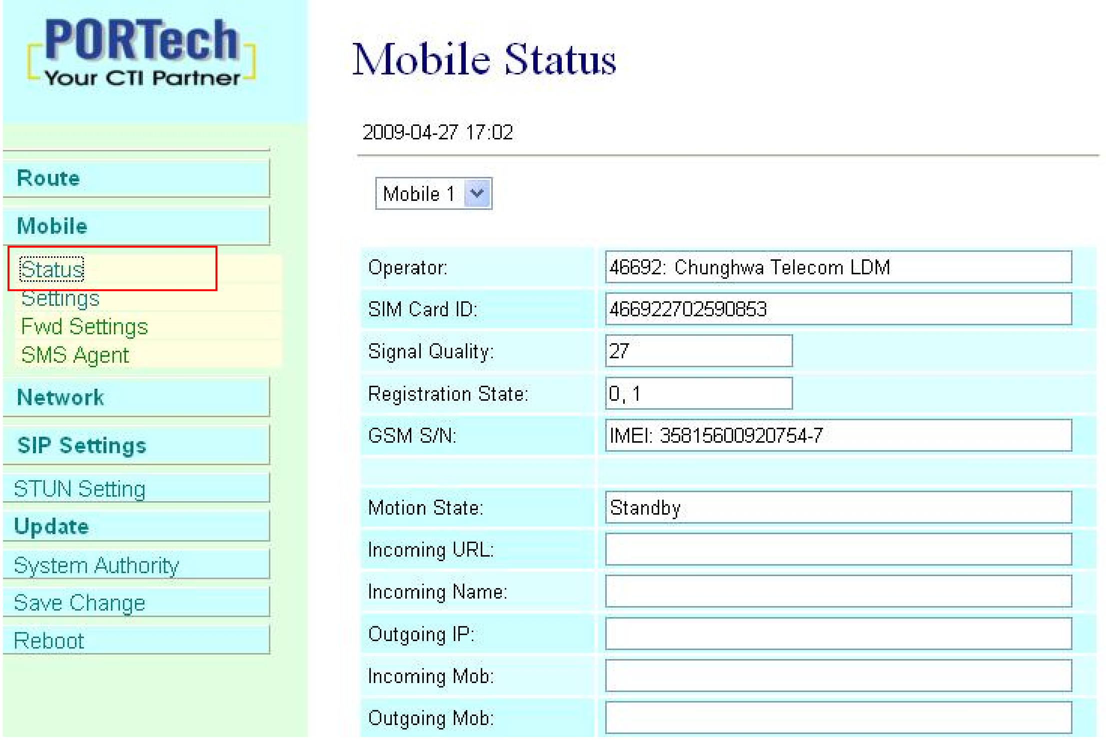
Ch.2 MV-374/MV-378:lan to mobile set route table *,#

Additionally, two channels MV-374/MV-378 both need to register proxy server or Asterisk.

And proxy server/asterisk set the route that the prefix of destination number dials out from ch.2 MV-374/MV-378.

10. Mobile

10.1 Mobile Status



(1)Choose Mobile 1,2,3 or 4 (MV-378: Mobile 1,2,3,4,5,6,7,8)

(2)Network Registration：The telecom carrier, which is the SIM card been registered.

(3)SIM Card ID：SIM card ID. (4)Signal Quality：Signal quality.

(5)GSM S/N: IMEI Number

(6)Motion State: The status of SIM card

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

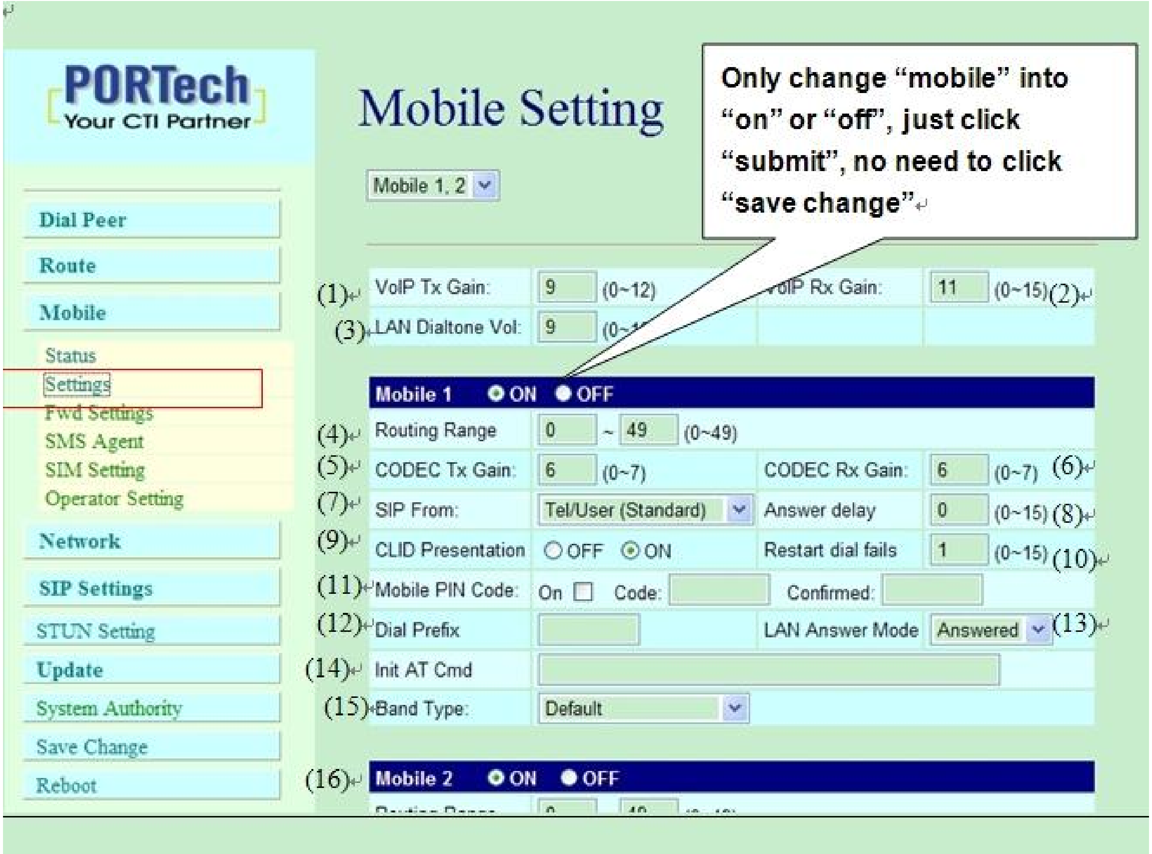
(8)Incoming IP Name: proxy server name

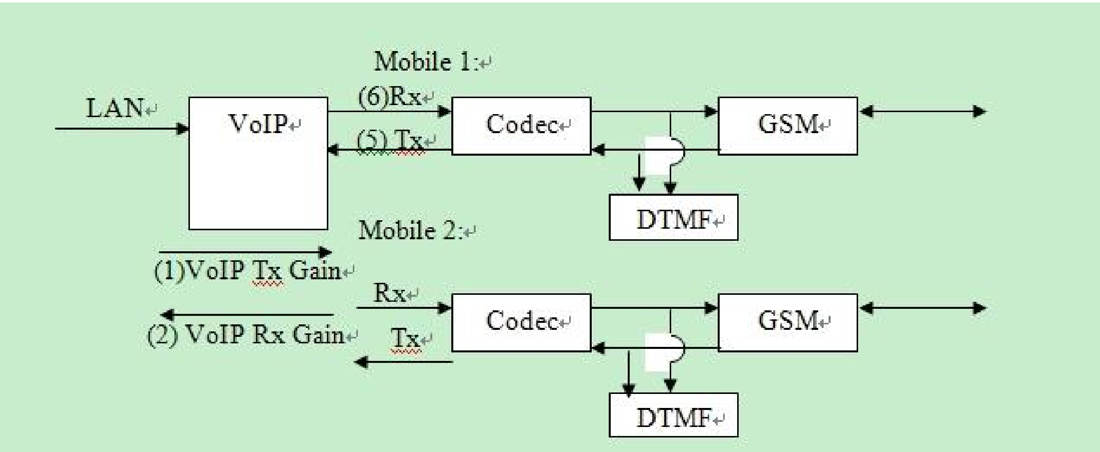
(9)Outgoing IP：The IP address of the last outgoing call to LAN.

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

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

10.2 Mobile Setting





(1) VoIP Tx Gain: To adjust the volume of LAN side.

(2) VoIP Rx Gain: To adjust the volume of Mobile side.

(3)LAN Dial tone Gain: DTMF Receiver is not good, you can adjust gain down.

(4)Routing Range: The route table -50 sets can share by two channels(1,2 ch / 3,4 ch / 5,6 ch / 7,8 ch)

ex: Mobile 1 use the route table for item 0-24,

Mobile 2 use the route table for item 25-49

(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 21.How to setup Asterisk to receive Caller ID from MV-374/MV-378 (page 42)

  MV-374/MV-378 will send the message as follows in the Packet.

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

- User/User (Standard): If you need to register to Asterisk and proxy server, please choose this option.

  MV-374/MV-378 will send the message as follows in the Packet.

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

- Tel/Tel :

  MV-374/MV-378 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 domain

- User/Tel

  MV-374/MV-378 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 domain

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

(9)Presentation CLID: If you need to block the Caller Id for call termination, please choose Suppression

(10) Restart Dial Fail: In this feature, user can initialize and register the module while GSM module dials fail in couple times. When GSM module is dysfunctional, it can avoid the device shut down in advance.

(11)Mobile PIN Code: If you need to unlock pin code via MV-374/MV-378, you can click “On” and enter pin code.

(12) Dial Prefix: The prefix number of outgoing calls. When Lan to Mobile, MV-374/MV-378 will automatically add the “Dial prefix” for outgoing mobile.

(13)LAN Answer Mode:

Answered: when mobile answer, and then connect the call

Alerted: when the mobile is ringing back tone, then connect the call

Income: when Lan dial out, then connect soon

(14) Init AT Cmd: User can fill the AT Command for GSM module

(15) Band Type: You can manual setting according to your GSM Frequency of carrier.

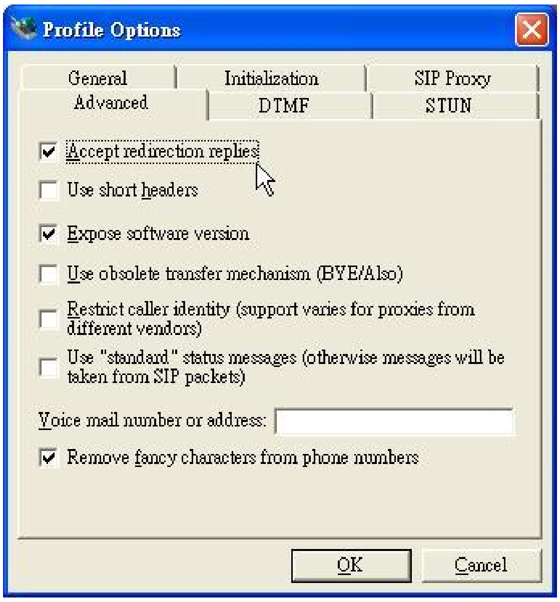
(16) ON/Off: If you use this channel, please click on. Otherwise, please click off.

10.3 Mobile / Forward Setting:

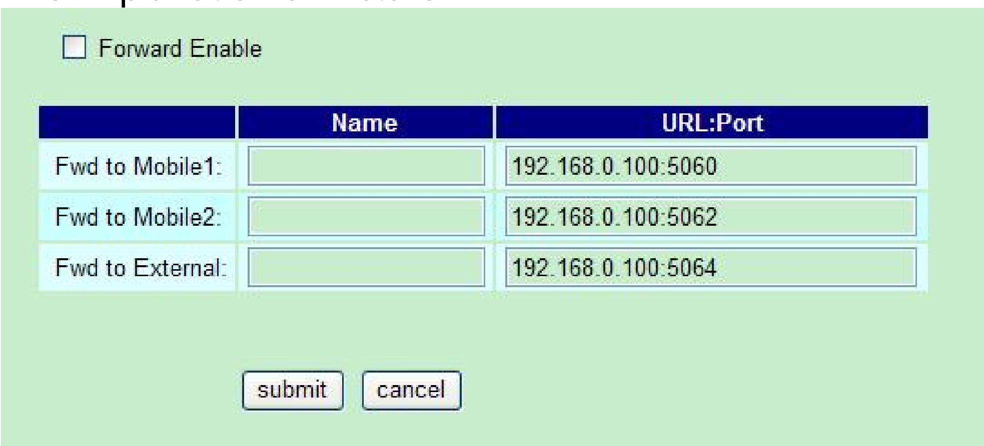
When the first route are busying, SIP can transfer phone call to another free route. When the device are busying, the phone call can be transfer to another device (external equipments).

* "Forward Enable" is not motivate on Default 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.)



The Explanation of Picture:

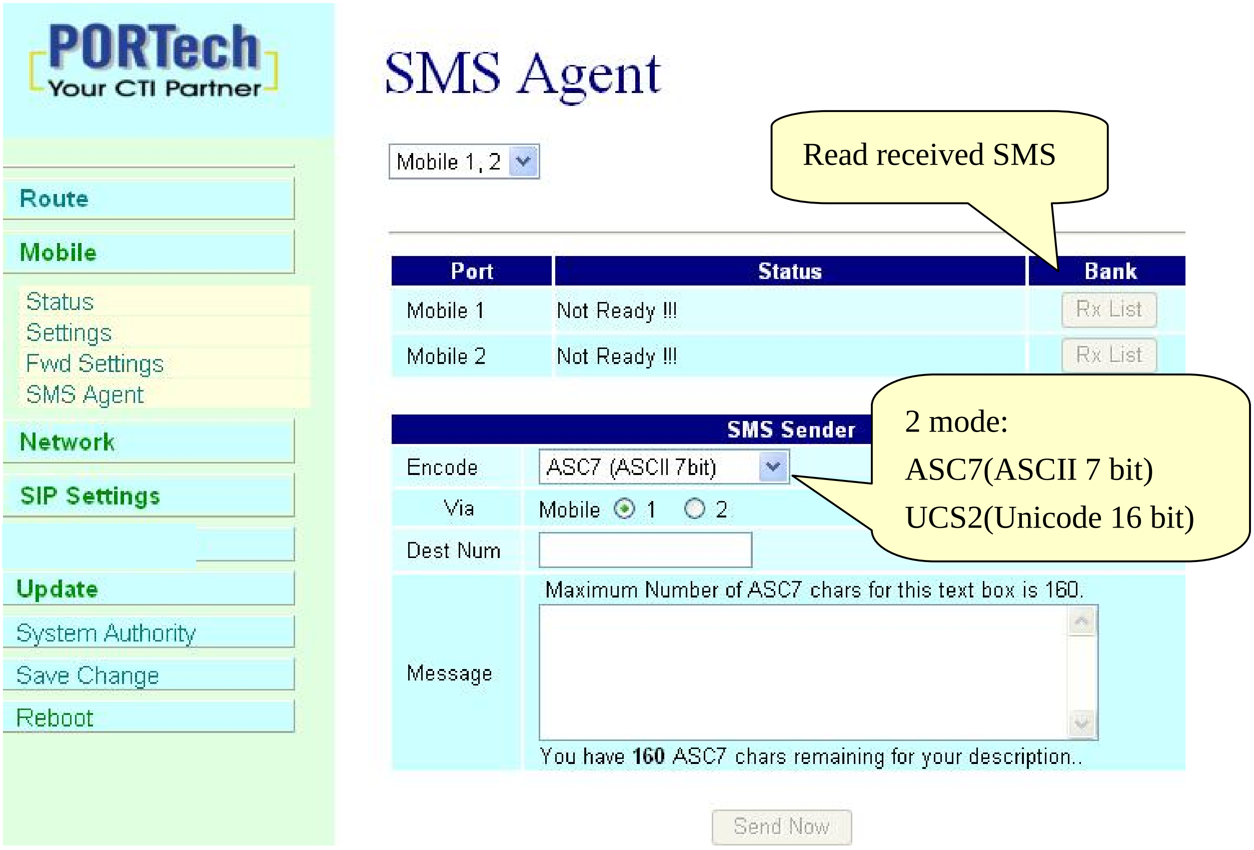


Fwd to Mobile1:192.168.0.100: 5060, it means when 5062 Port are busying, SJ Phone can transfer the call to 5060 Port (192.168.0.100).

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).

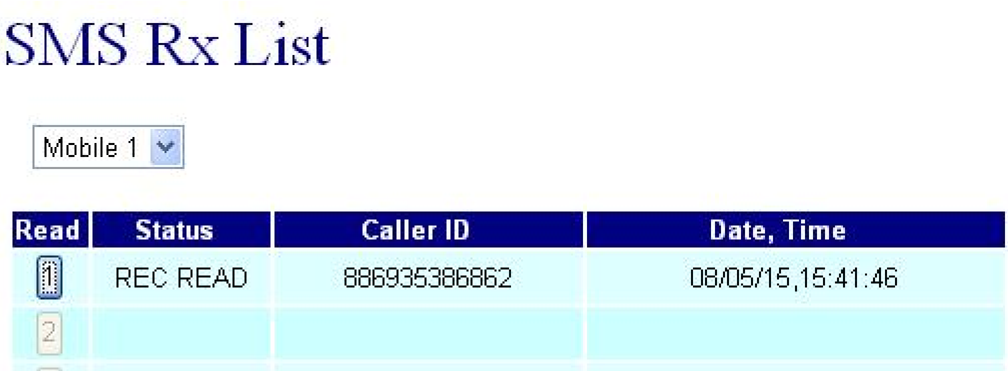
- If both 5060 port and 5062 port are busying at same time, you can set up "Fwd to External", then you can transfer the phone call to another designate device.

10.4 Mobile / SMS Agent:



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

10.5 Send Bulk of SMS via Microsoft Excel

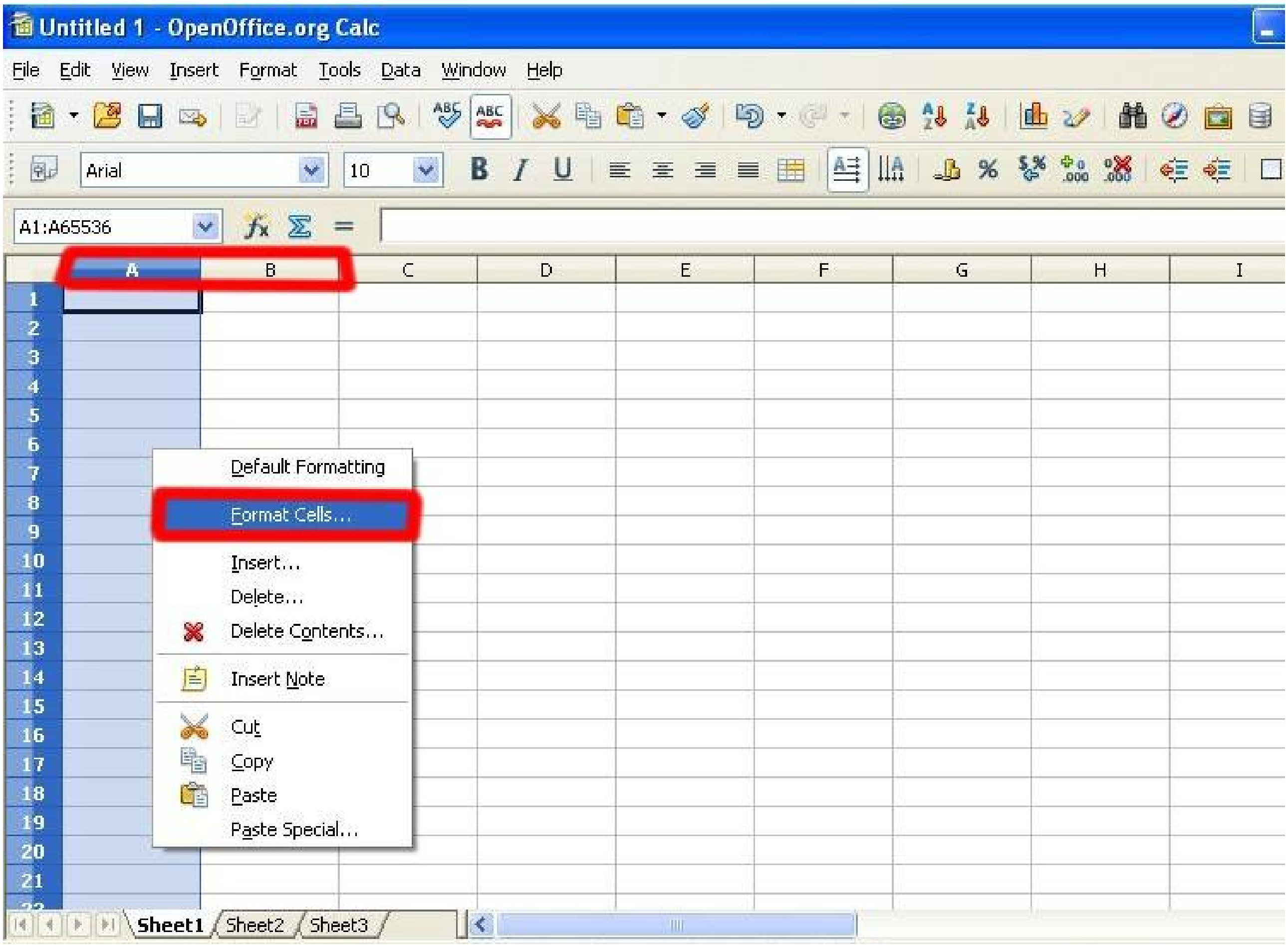
First of all, please open a new Excel file.

Step 1 Format Cells

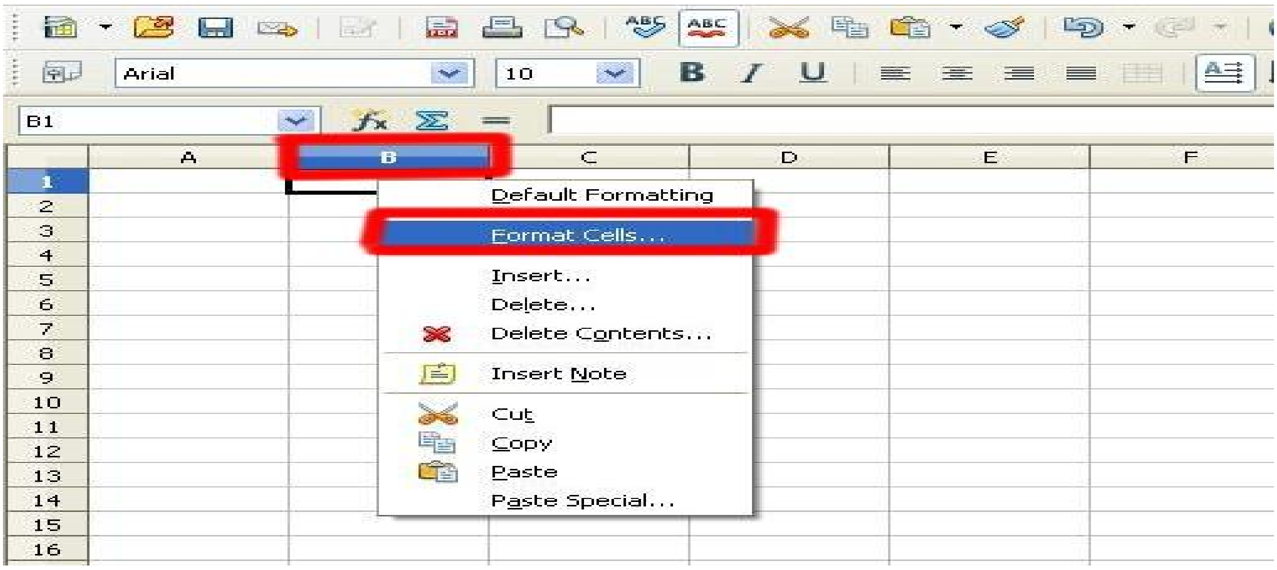
Here, we need you to format cells to "Text" first.

Please click mouse right key, and choose "Format Cells"

Blank A

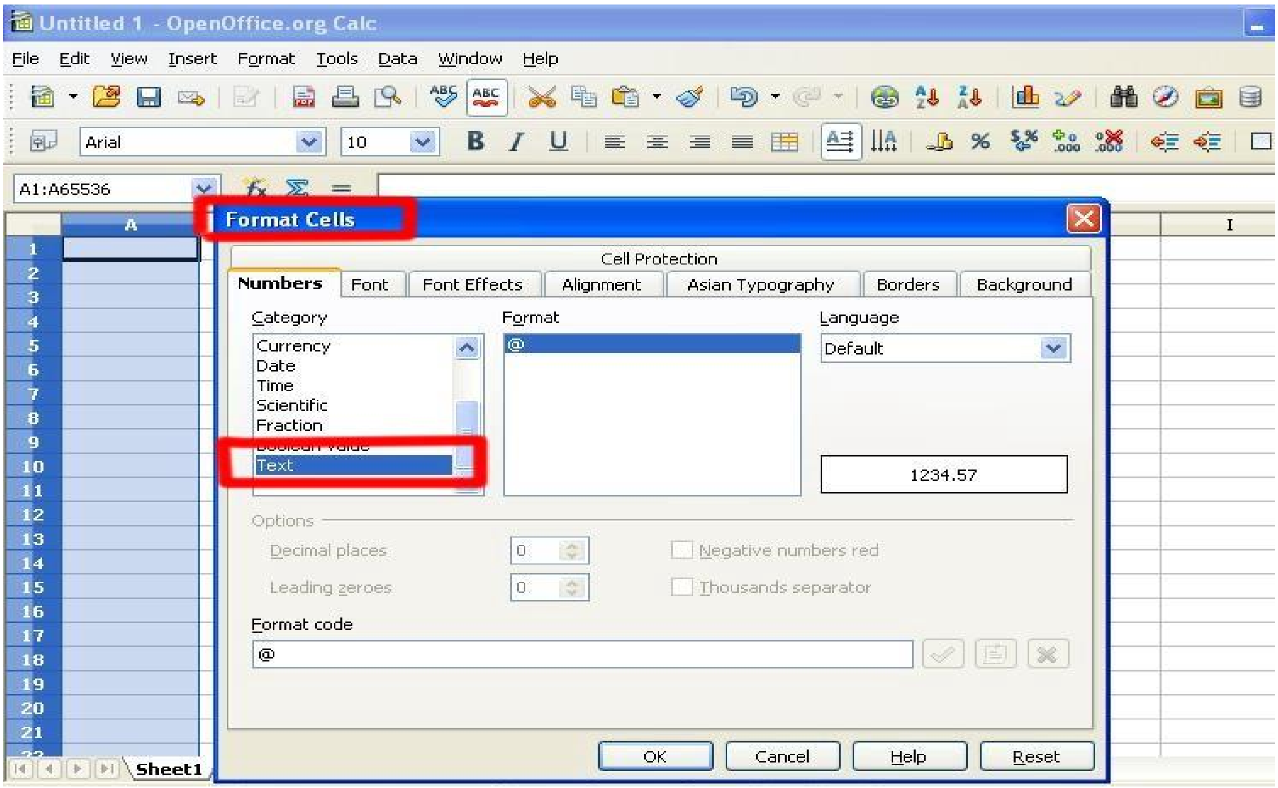


Blank B



Step 2

In the Format Cells, please select “Text”

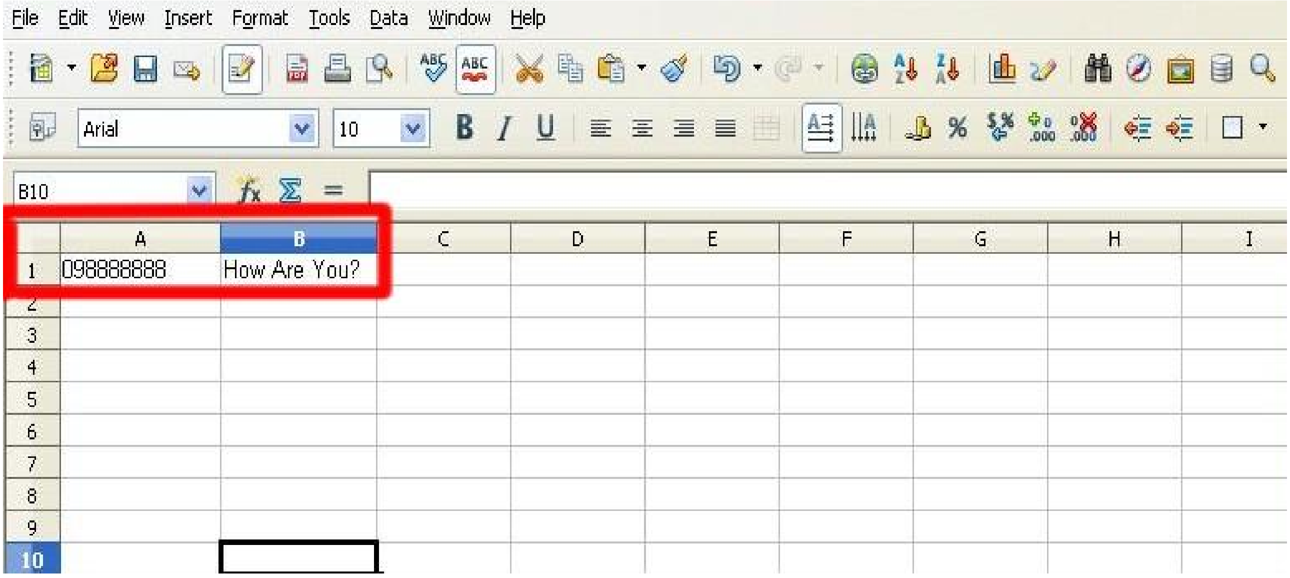


- Please do this action for Blank A and B both.

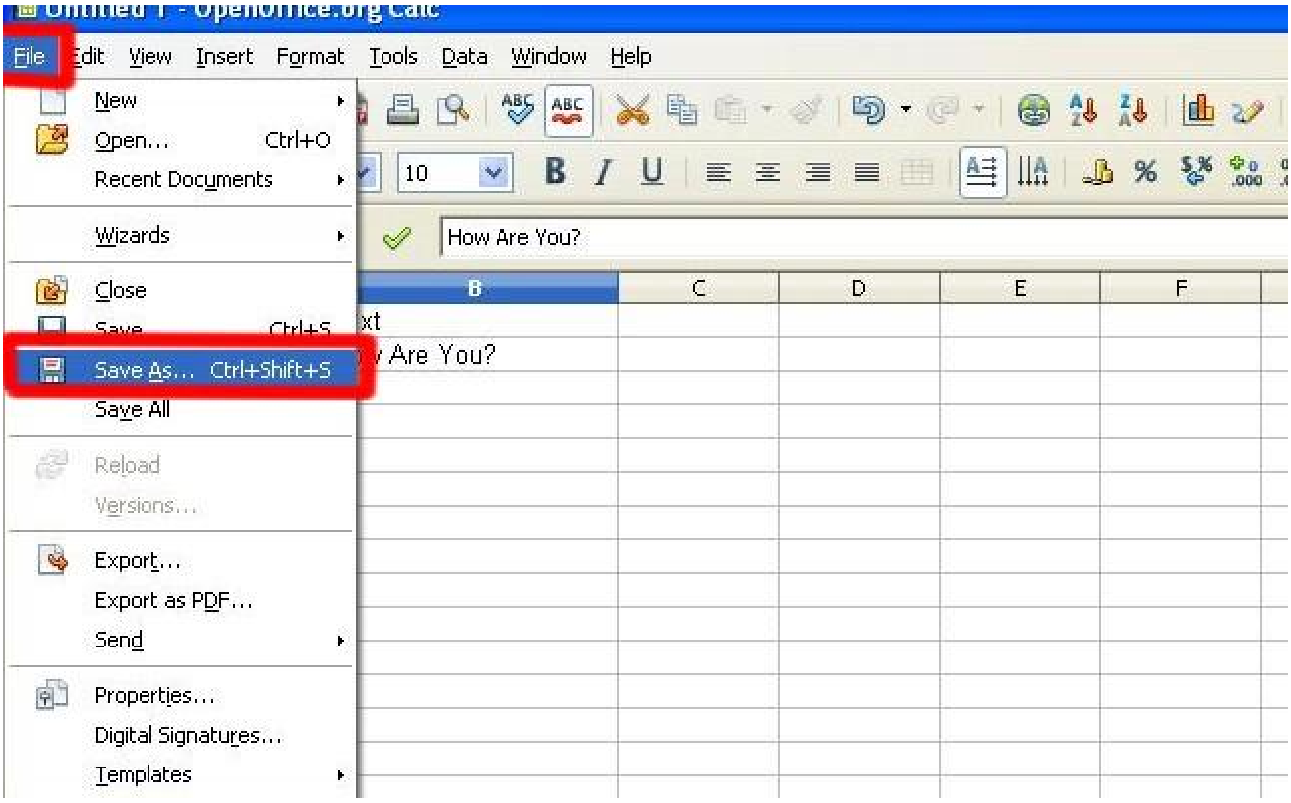
Step 3

Blank A: is for you to key “phone numbers”

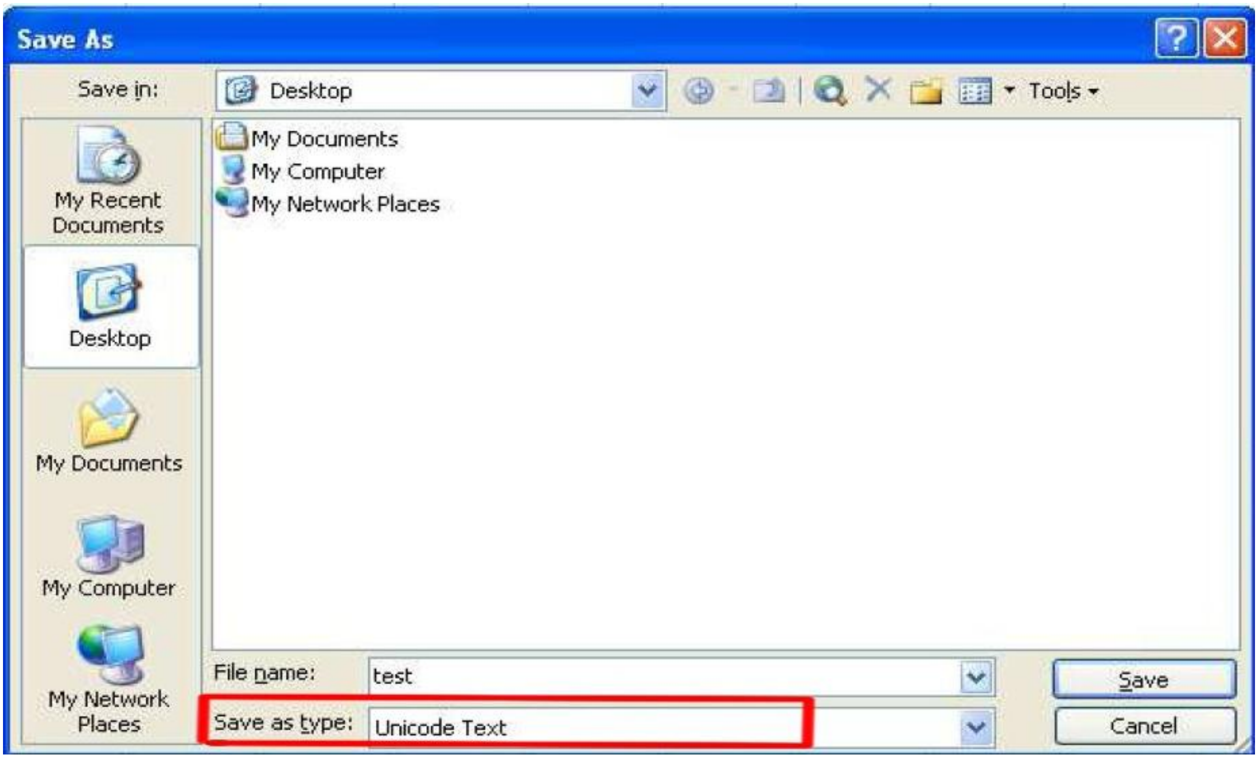
Blank B: is for you to key “text”



Step 4 **save the file**

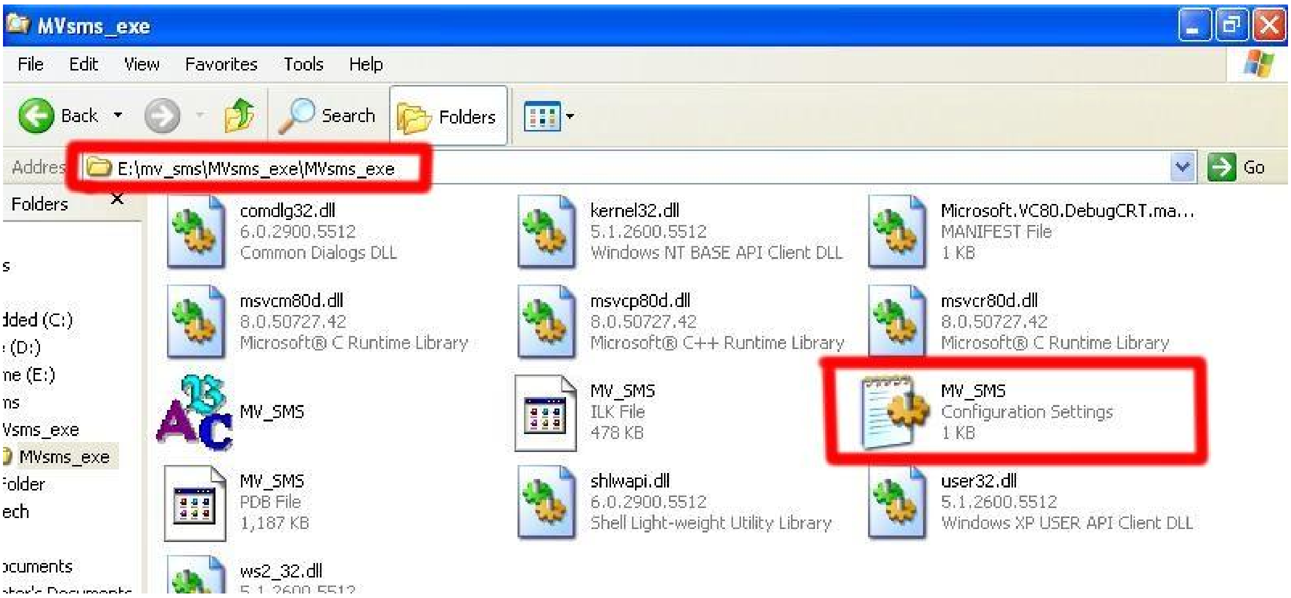


Save the type as “**Unicode Text**”



Step 5

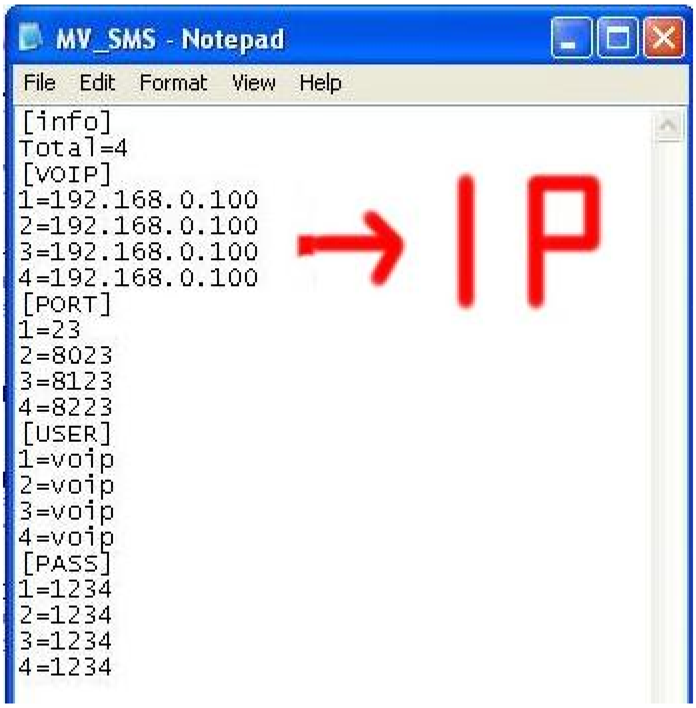
Open MVsms_exe -→ MV-SMS (Configuration Settings)



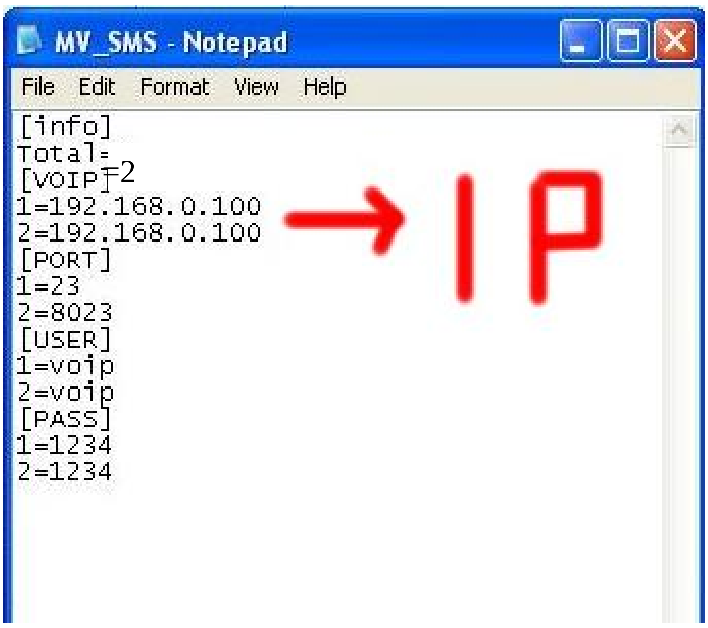
Step 6

Please do the configuration as following:

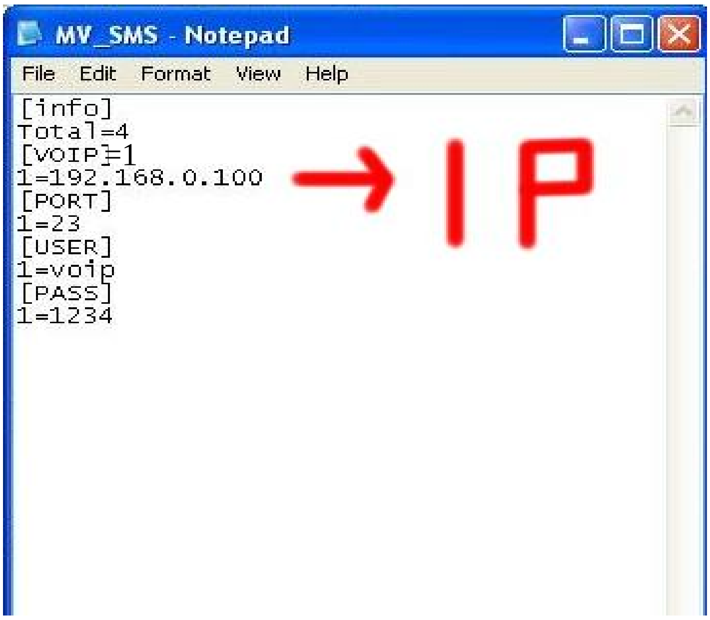
MV-378



MV-374

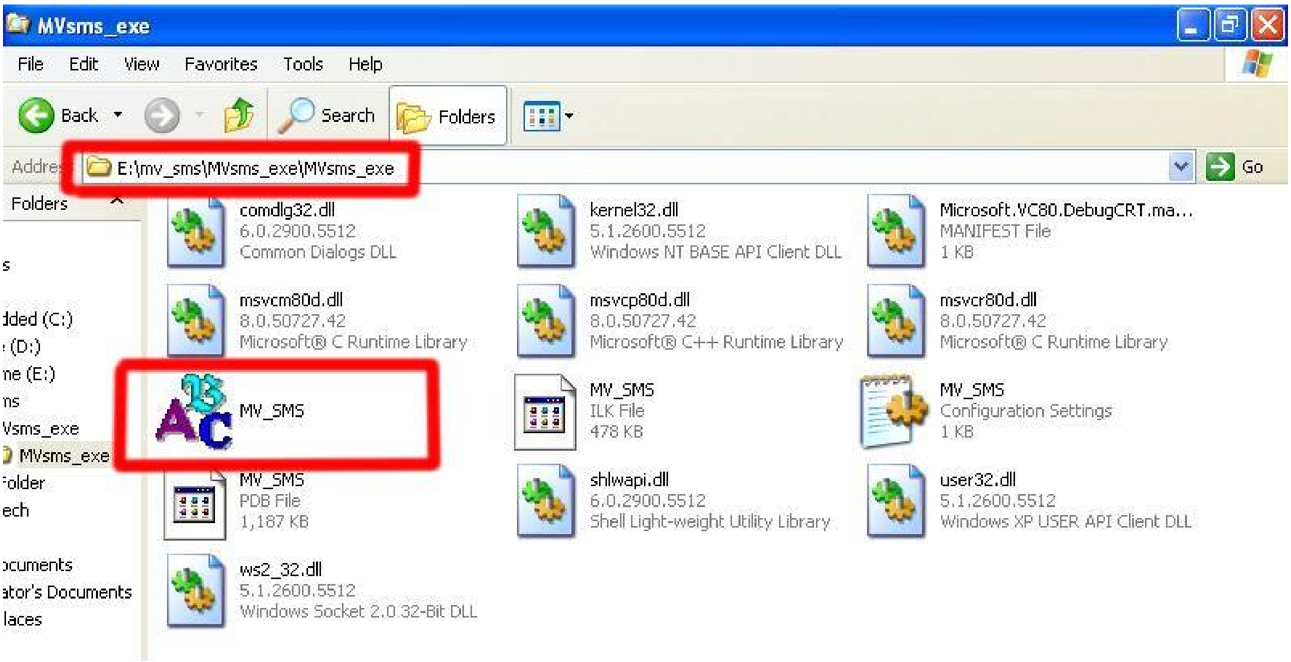


MV-372 & MV-370



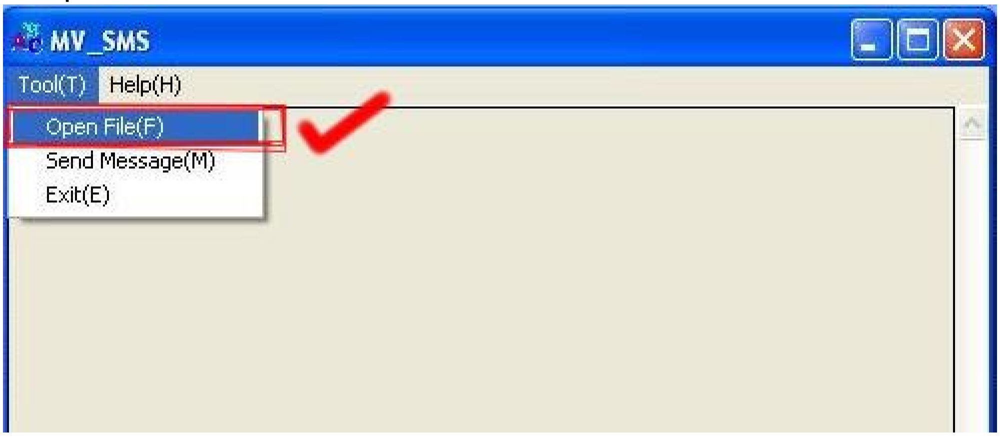
Step 7

Run MV-SMS program



Step 8

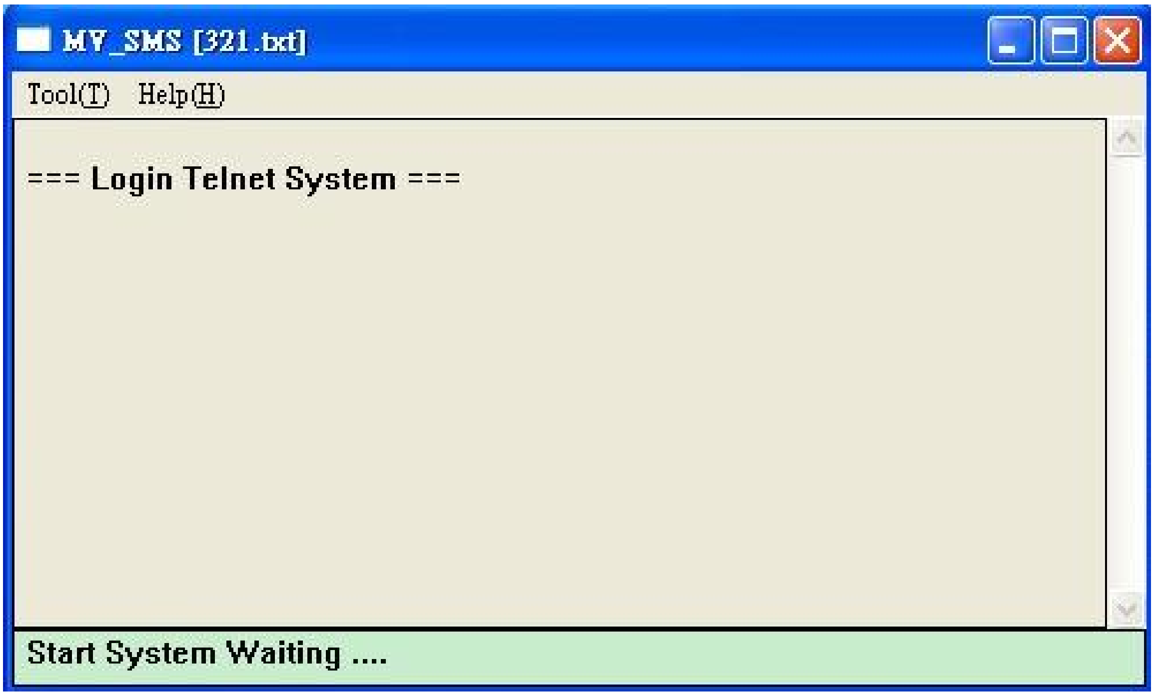
1. Open File



2. Open the “Excel file” that you just saved

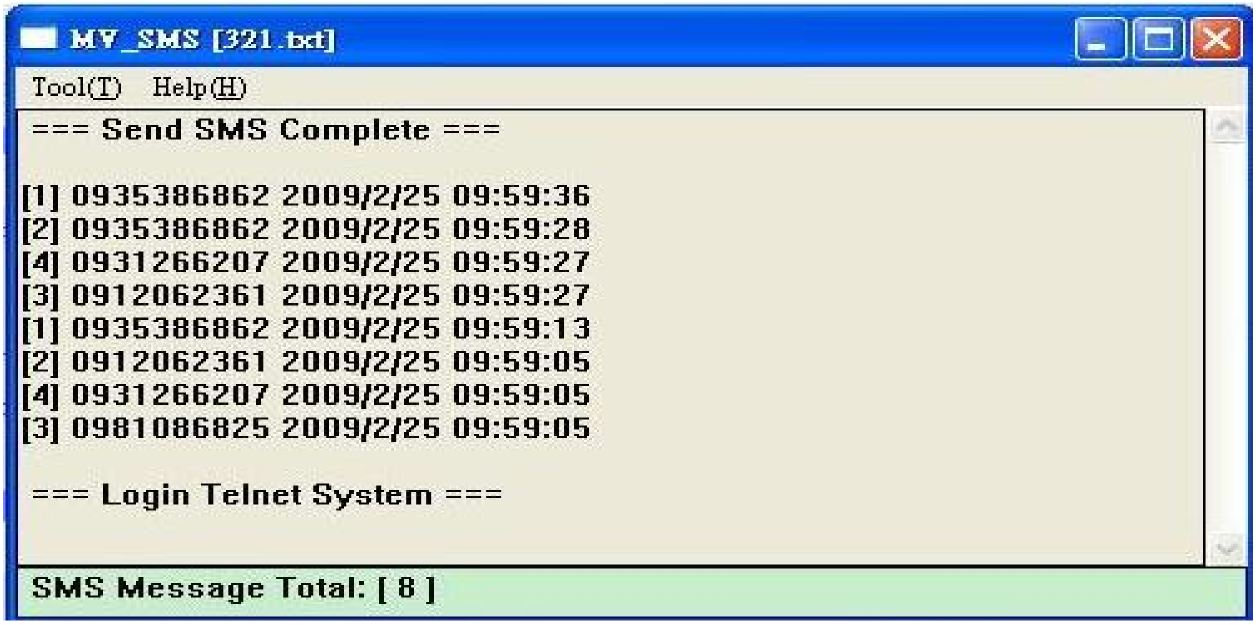
Step 9

Sending



Step 10

Send SMS Complete



10.6 use AT Command via Telnet or your program

Allows your program or Telnet Send/receive SMS with AT Command available in PCB194A (approximately after April , 2008)

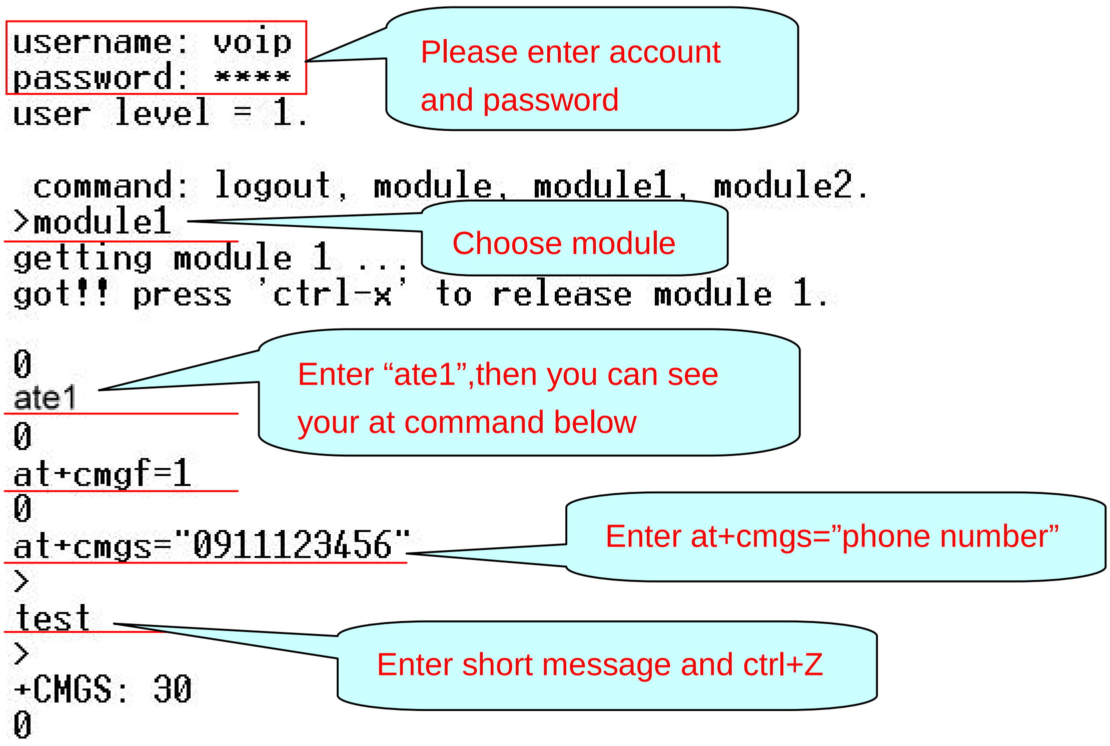
Telnet PORT Corresponding port as follows:

SLAVE 1:8023

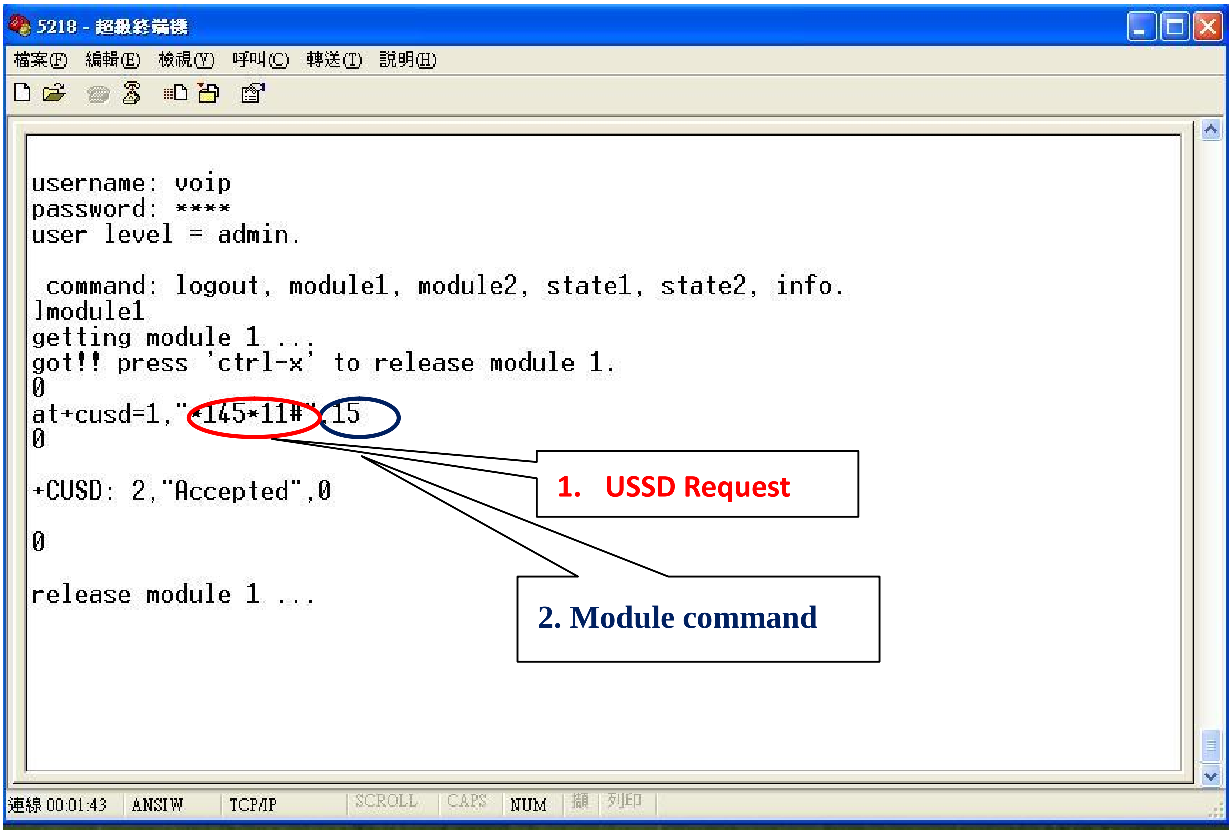
SLAVE 2:8123

SLAVE 3:8223

SLAVE 4:8323



10.7 USSD SIM Balance Check via Telnet



1. USSD Request: Please enter USSD code for your operator to check balance

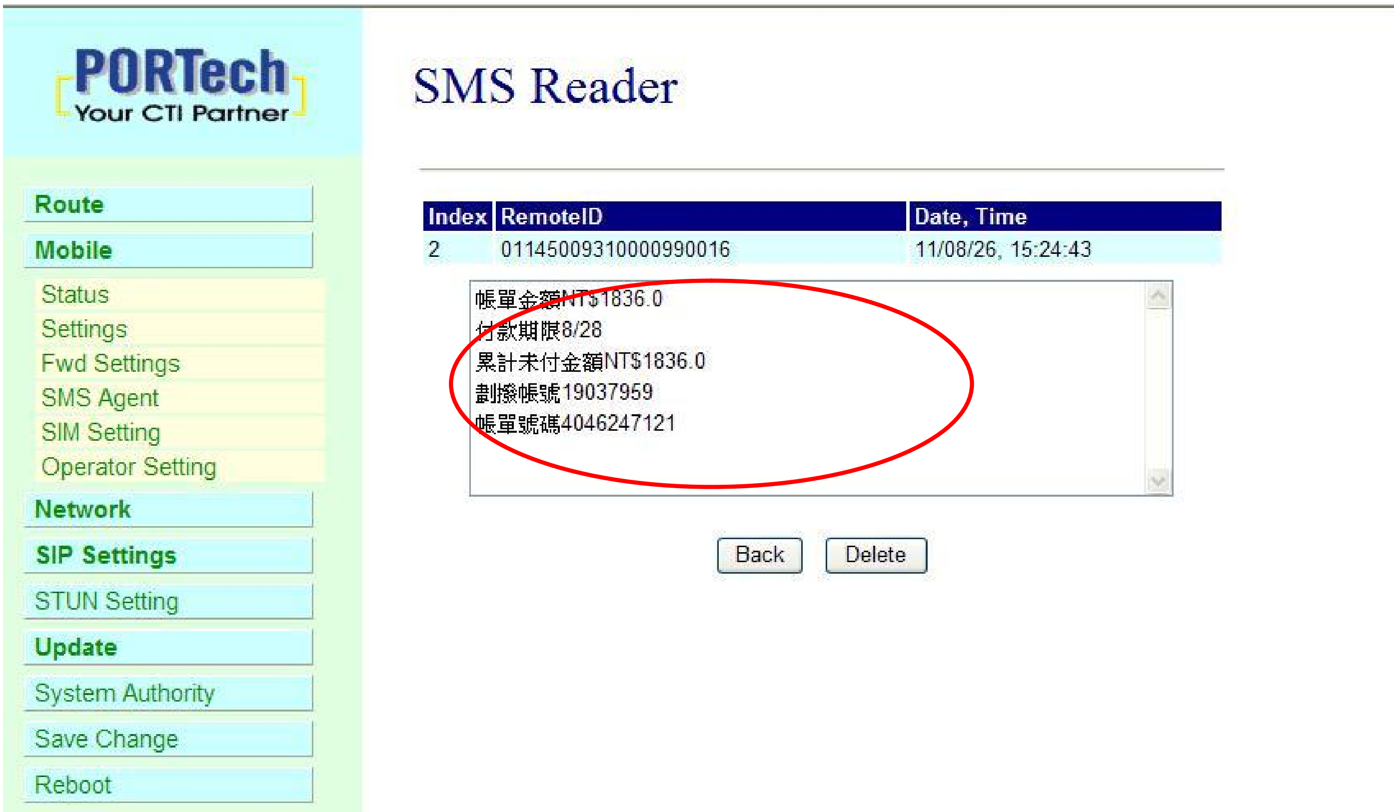
2. Module command:

Please enter “15” for Siemens BG2W module

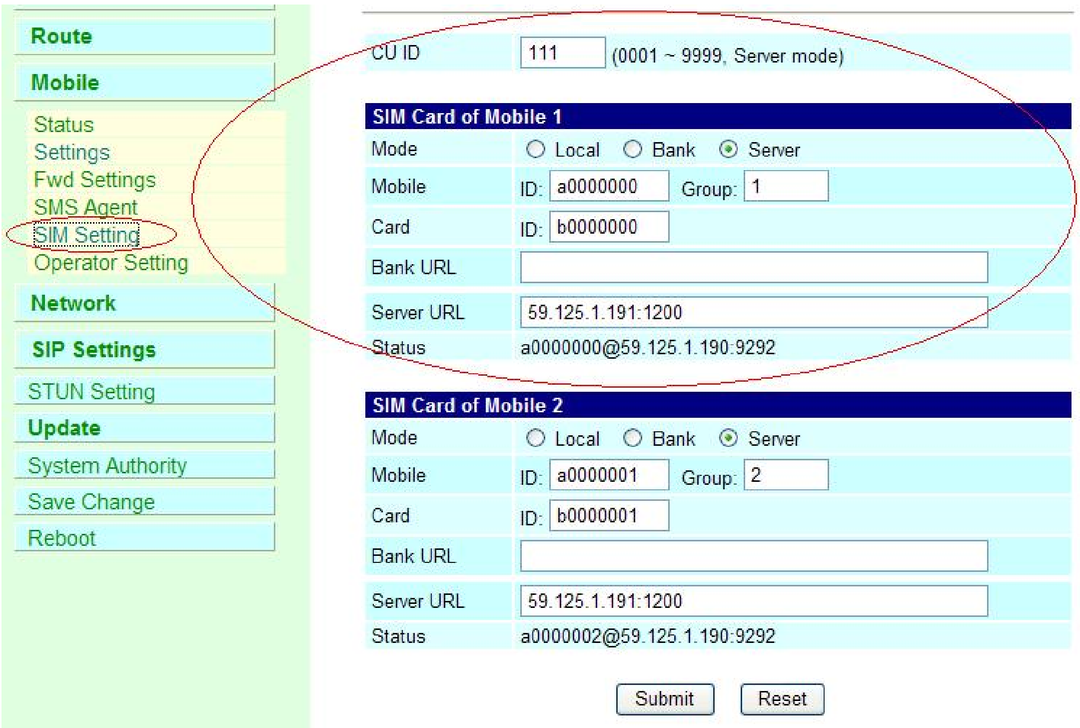
Please enter “0” for Simcom module

- You can check this information on main page in **Module Description**

After you send the USSD request, MV will receive the SMS from operator
Please check the incoming SMS on SMS Agent



10.8 SIM Setting

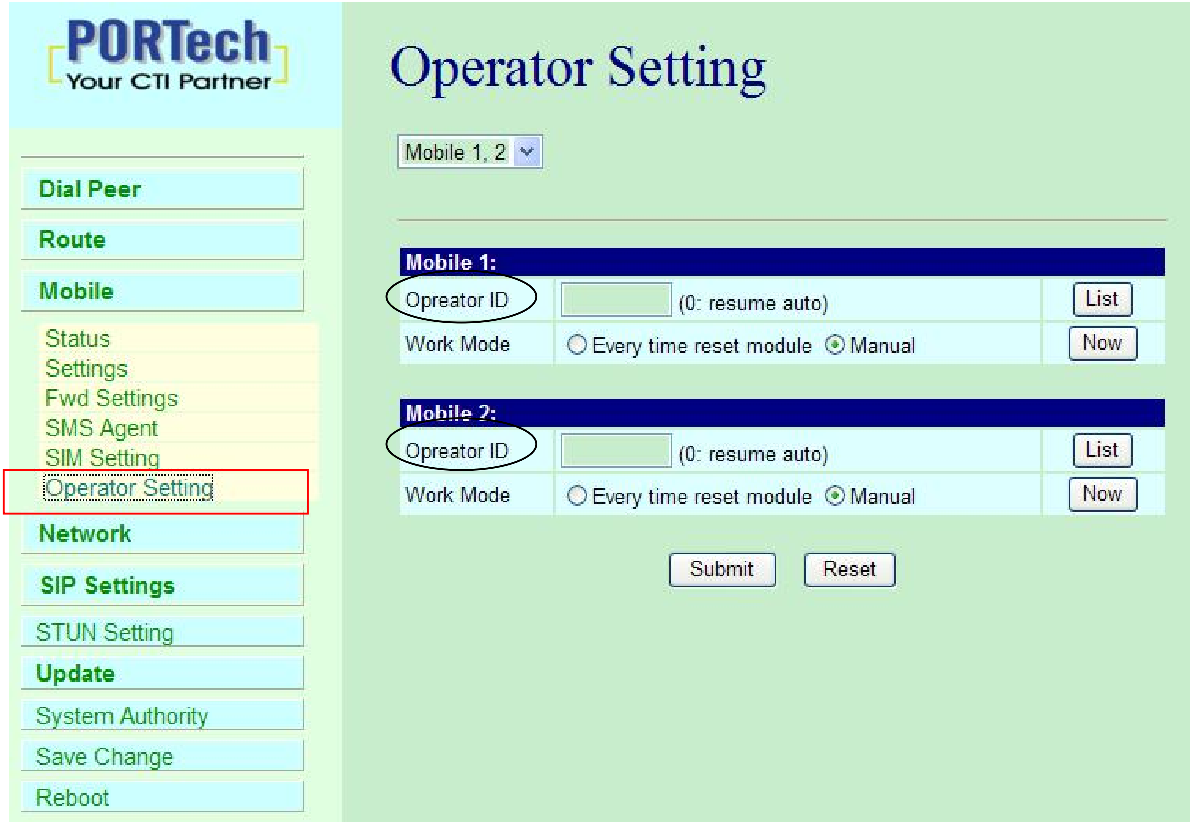


1. CU ID: It's the ID for MV and SIM Server Transfer Protocol, within 1~9999. Each MV under same SIM Sever should setup different CU ID, and no reusing parameter. E.g. If you put "888" on 1st MV-378 that you can't use "888" on 2nd MV-378, and so on.
2. Mode
   a. Local: Disable Remote SIM feature
   b. Bank: Enable Remote SIM Bank feature, and manage SIM card on SBK-32 SIM Bank.
   c. Server: Enable Remote SIM Server feature, and allocate SIM cards on SBK-32 SIM Bank.
3. Mobile
   a. ID: Put in 8 digits (hexadecimal, also base 16), which used for GSM Module ID identification to Remote SIM protocol. User can define the ID. IF it's Server Mode, just leave it default. If it's Bank Mode, No reusing GSM Module ID for same SIM Bank.

b. Group: Fill in SIM Group number for Remote GSM module. Server follow SIM Group Number to allocate SIM card to correspond GSM module

4. Card ID: Put in 8 digits (hexadecimal, also base 16), which used for SIM Card ID identification to Remote SIM protocol. User can define the ID. If it's in Server Mode, Card ID can be blank or default. As for Bank Mode, Card ID must be corresponding to SIM Card ID of SIM Bank.
5. Bank URL: If it's Bank Mode, please fill SIM Bank IP and Port Number. On other hand, please leave blank for Server Mode.
6. Server URL: If it's Server Mode, please fill SIM Server IP and Port Number. On other hand, please leave blank for Bank Mode.
7. Status: User can check the SIM Card ID of GSM module and IP, Port Number of SIM bank.

10.9 Operator Setting



1. Operator ID: When GSM module is registered, user can click the List to show all available operators in that area. You will see like follows diagram.

No	Status	Name	ID	Use
00	Current	Chunghwa Telecom (CHT)	46692	
01	Forbidden	Far EasTone (FET)	46601	
02	Forbidden	Pacific GSM 1800 (TCC)	46697	
03				
04				
05				
06				
07				

2. Work Mode:

a.Every time reset module:

Fill the assigned Operator ID, then press **Submit** bottom and save change. After reboot, GSM module will research the operator ID and registered the base station.

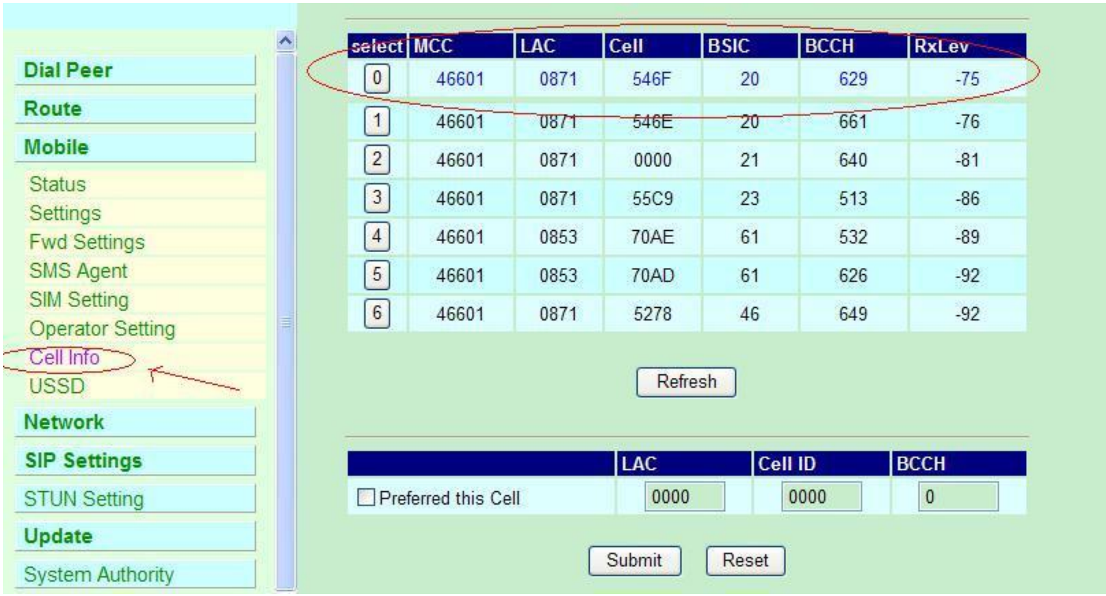
b.Manual:

Fill the assigned Operator ID, then press **Now** bottom. GSM module will search that Operator ID and registered after reboot.

10.10 Cell Info

It shows BTS (BCCH) cells of the cellular network and register to new BCCH selection. Support Quad band-BG2W, Quad band-M10 and firmware V10.185 above only.

Please work with this feature when the mobile status is “Stand by/Active”. It detects the surrounding active cell, up to 7 cells and shows Cell ID, signal and best signal (RXlev). The No.0 shows the data of current registered cell. Follow by No.1 to No.6 cell is based on cell signal (best to low).



MCC : Mobile Country Code

LAC : Location Area Code

Cell : Cell Identifier

BSIC: Base Station Identity Code

BCCH: Broadcast Control Channel

RxLev: Received Signal level in dbm

How to Configure

1. You can choose a BCCH channel by clicking on the cell. The module will automatically register in the new BCCH.

E.g. If you would like to register BCCH channel on No.4 cell, please click no4 select like below.

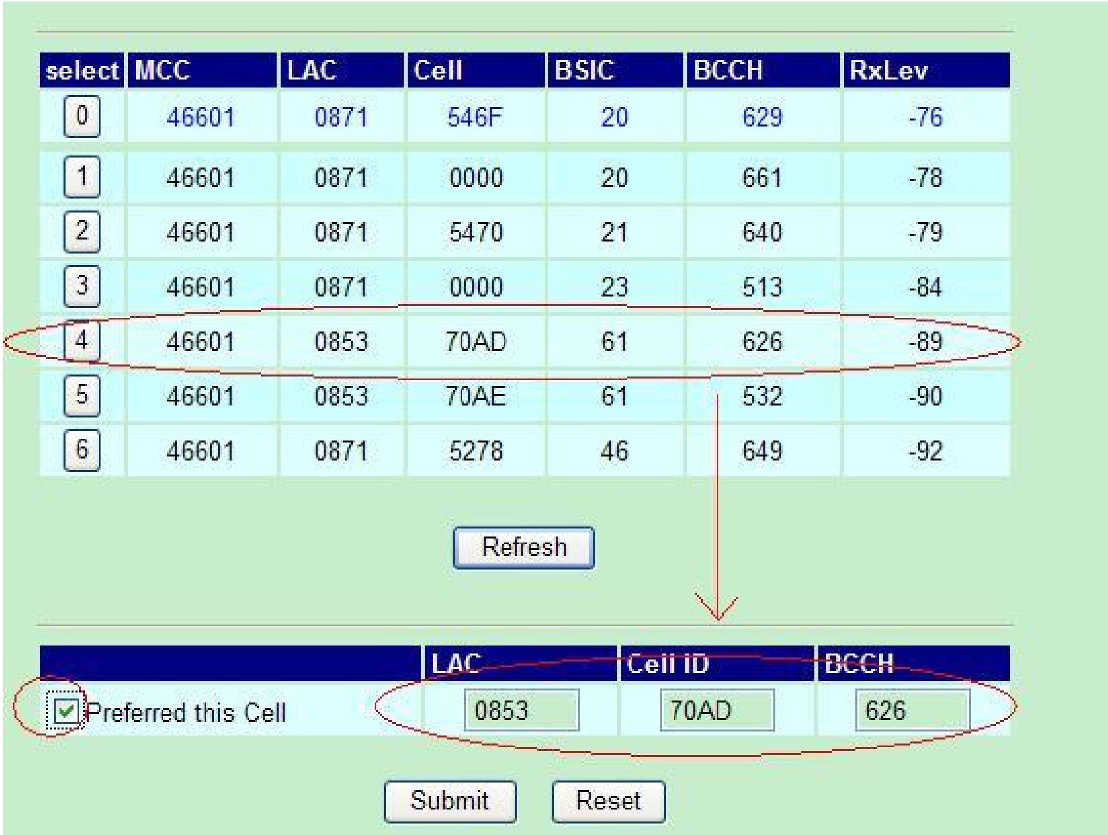

Cell Info

Mobile 1

select	MCC	LAC	Cell	BSIC	BCCH	RxLev
0	46601	0871	546F	20	629	-76
1	46601	0871	0000	20	661	-78
2	46601	0871	5470	21	640	-79
3	46601	0871	0000	23	513	-84
4	46601	0853	70AD	61	626	-89
5	46601	0853	70AE	61	532	-90
6	46601	0871	5278	46	649	-92

Refresh

2. System will show the cell number information once you select on Preferred this Cell form. Please click the submit button and Save Change on left to restart the module.



After system restart and turn to Standby, please check on No.0 cell and confirm the current registered cell you selected. At the point, the GSM module won’t provide the data of surrounding cell signal, but shows -110dbm on No.1 to No.6 RxLev, which means GSM signal 0.

select	MCC	LAC	Cell	BSIC	BCCH	RxLev
0	46601	0853	70AD	61	626	-88
1	46601	0871	546F	20	629	-110
2	46601	0871	546E	20	661	-110
3	46601	0871	0000	23	513	-110
4	46601	0853	0000	61	532	-110
5	46601	0853	0000	23	656	-110
6	46601	0871	0000	27	667	-110

Refresh

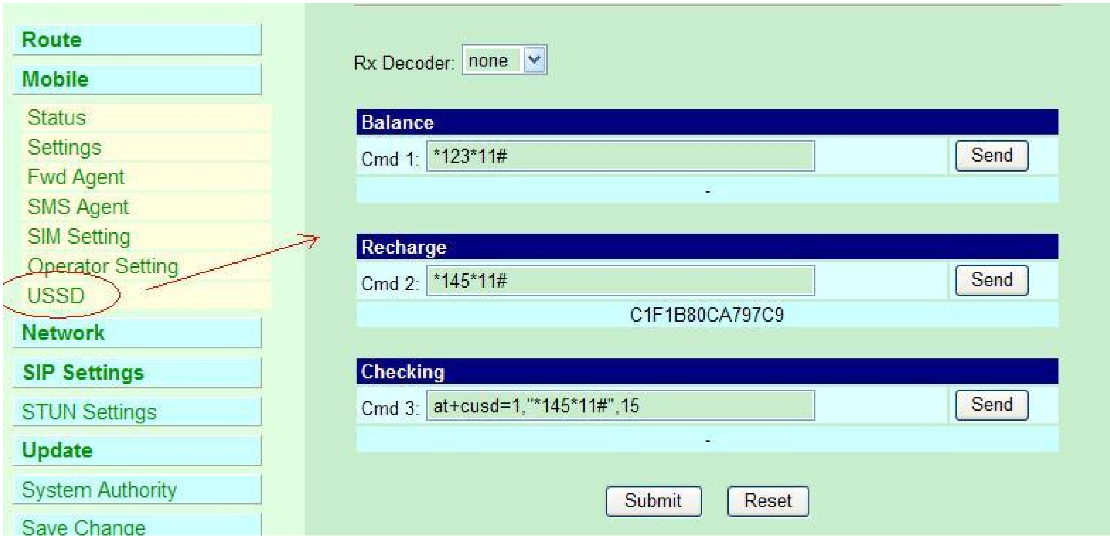
3. If you would like to research all the surrounding BCCH cells again, please cancel Preferred this Cell selection first and send Submit, Save Change to restart the gateway. That, System can detect the surrounding active cell, up to 6 cells and display Cell ID, signal and best signal (RXlev).

select	MCC	LAC	Cell	BSIC	BCCH	RxLev
0	46601	0871	546E	20	661	-76
1	46601	0871	546F	20	629	-77
2	46601	0871	5470	21	640	-79
3	46601	0871	0000	23	513	-83
4	46601	0853	70AE	61	532	-90
5	46601	0853	70AD	61	626	-89
6	46601	0871	5278	46	649	-92

Refresh

10.11 USSD (Unstructured Supplementary Service Data)

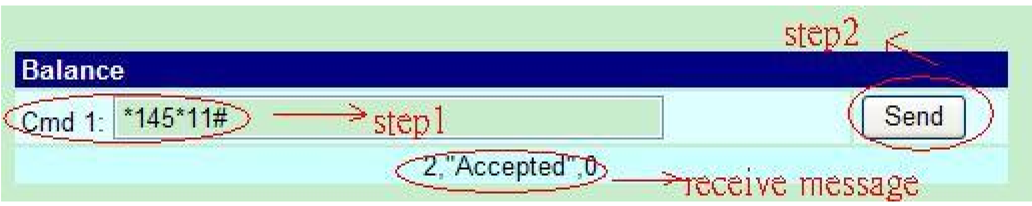
You can check USSD screen for SIM balance remaining and SIM recharge (add value) automatically. Please work with this feature when the mobile status is "Stand by/Active". And ensure your Service provider has given you a USSD string(Command) for checking SIM Balance and Recharge the SIM Card.



1. Balance (SIM balance remaining)

Step1: Enter Balance checking USSD command in column
Step 2: Click Send button

When selected, system will check the balance of SIM and display the reply of receive message as below

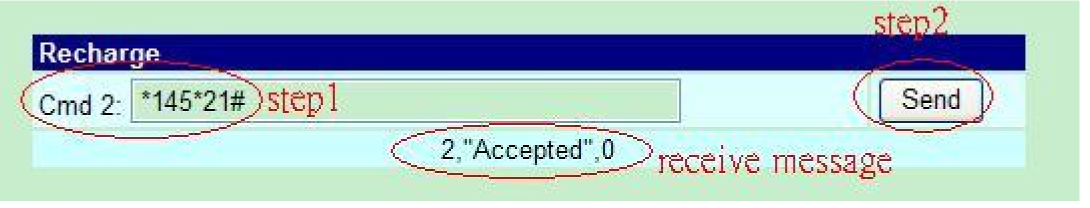


2. Recharge (add value)

Step1: Enter the Recharging USSD command in column

Step 2: Click Send button

When selected, system will display the reply of receive message as below



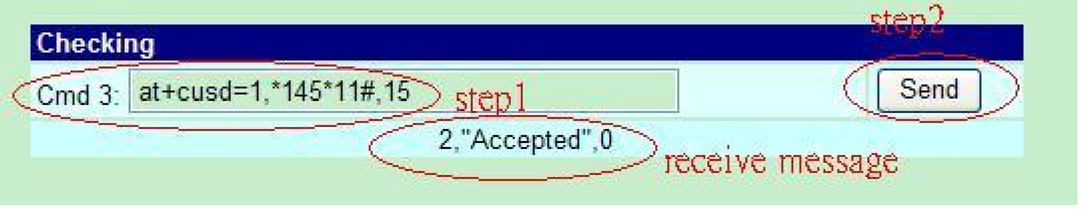
3. Checking (If above ways are failed, please select this)

Step 1: Enter the complete AT command in Cm3 column

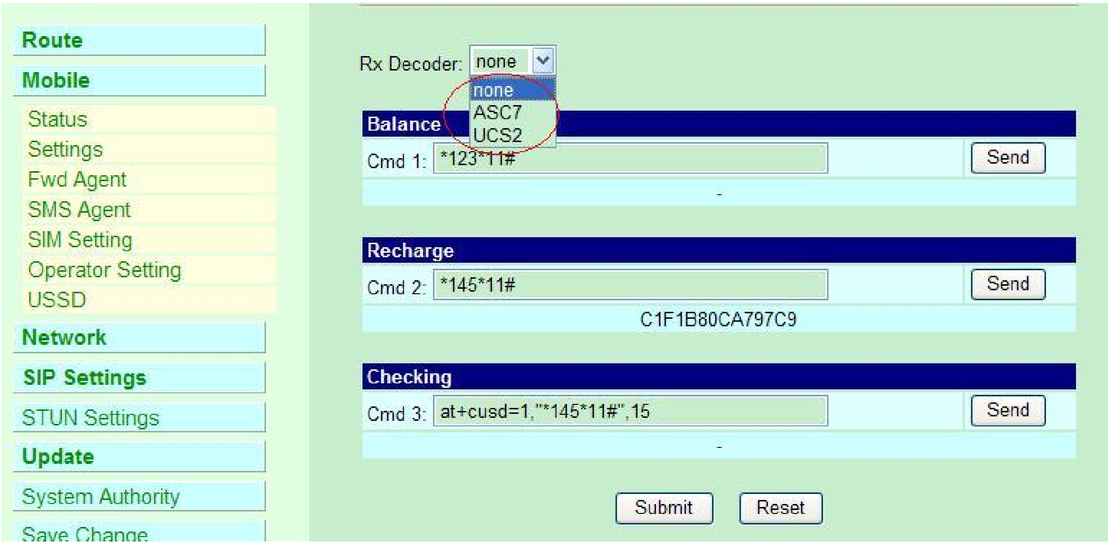
Ex. AT+CUSD=1,*145*11#,15

Step 2: Click Send button

When selected, system will display the reply of receive message as below



4. Rx Decoder



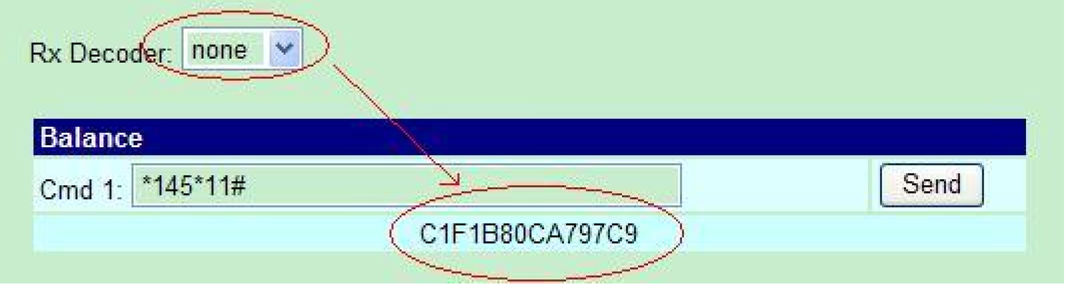
a. None: GSM Format (Default)
b. ASC7: ASCII 7bit
c.UCS2: Unicode 16bit

When user select default GSM Format(None), it may not receive correct GSM code due to the different operator or GSM module/chipset. Please check below example,

In this case, user need to select other RX Decoder (ASCII or UCS2) to receive correct message.

For Example,

None format: When user send command, “*145*11#”, the return message show on system, “C1F1B80CA797C9”

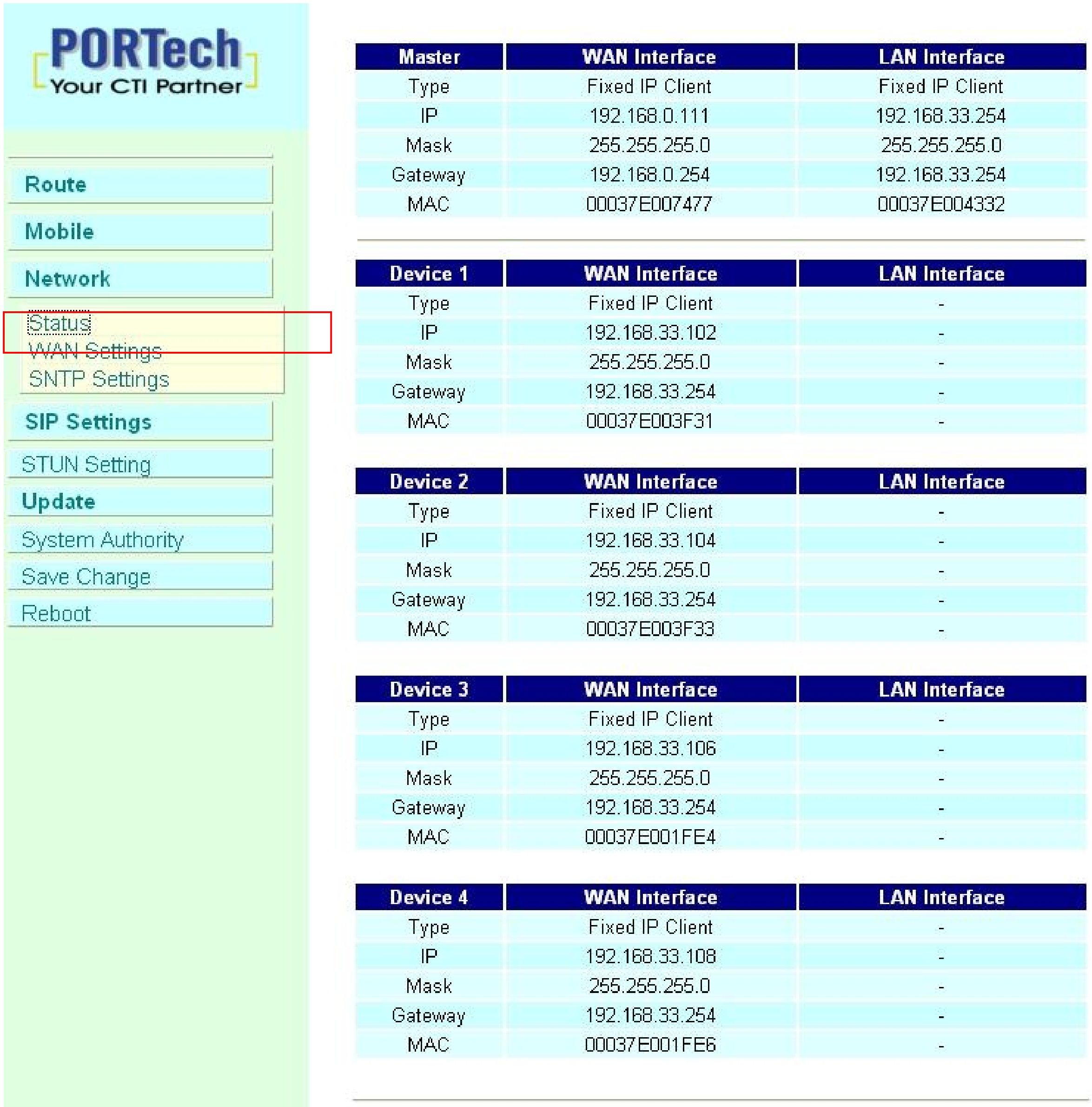


ASC7 Format: In this format, the return message is “Accepted”

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.



Master	WAN Interface	LAN Interface
Type	Fixed IP Client	Fixed IP Client
IP	192.168.0.111	192.168.33.254
Mask	255.255.255.0	255.255.255.0
Gateway	192.168.0.254	192.168.33.254
MAC	00037E007477	00037E004332

Device 1	WAN Interface	LAN Interface
Type	Fixed IP Client	-
IP	192.168.33.102	-
Mask	255.255.255.0	-
Gateway	192.168.33.254	-
MAC	00037E003F31	-

Device 2	WAN Interface	LAN Interface
Type	Fixed IP Client	-
IP	192.168.33.104	-
Mask	255.255.255.0	-
Gateway	192.168.33.254	-
MAC	00037E003F33	-

Device 3	WAN Interface	LAN Interface
Type	Fixed IP Client	-
IP	192.168.33.106	-
Mask	255.255.255.0	-
Gateway	192.168.33.254	-
MAC	00037E001FE4	-

Device 4	WAN Interface	LAN Interface
Type	Fixed IP Client	-
IP	192.168.33.108	-
Mask	255.255.255.0	-
Gateway	192.168.33.254	-
MAC	00037E001FE6	-

11.2 WAN Settings:

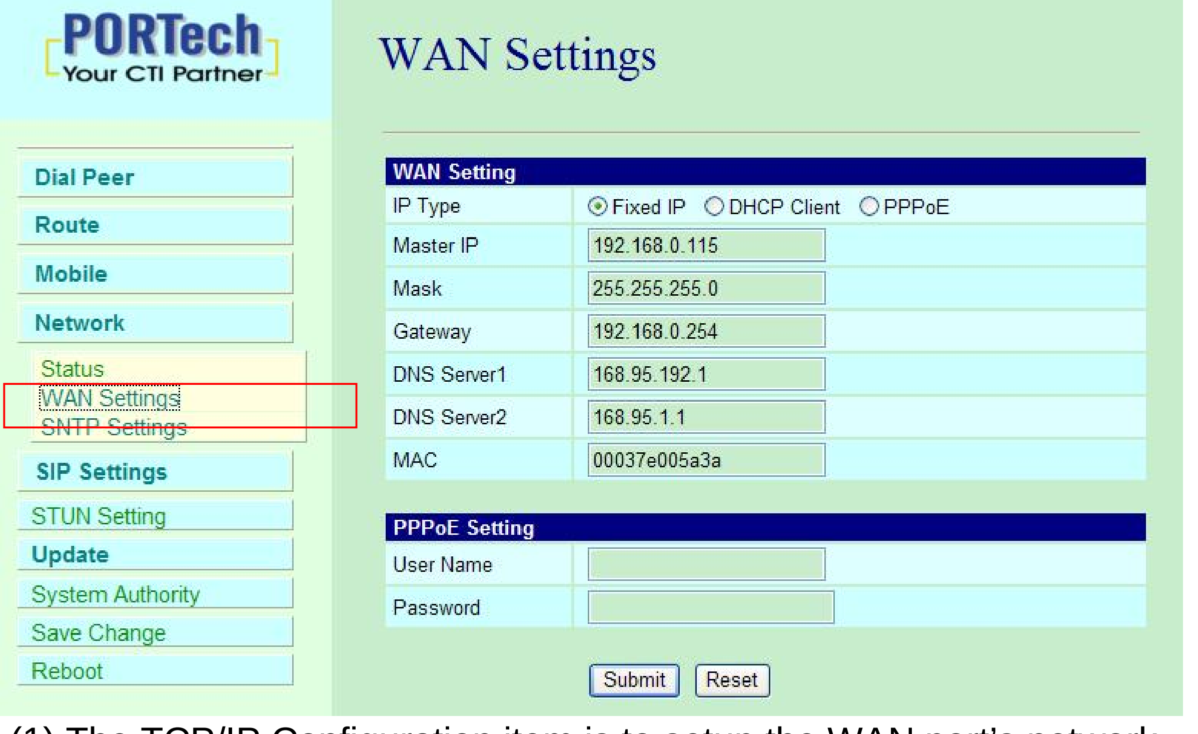
WAN IP (Master)   Default: 192.168.0.100

Slaver1: Master IP: 8080

Slaver2: Master IP: 8180

Slaver3: Master IP: 8280   Slaver4: Master IP: 8380

WAN IP Corresponding port 5064 5066 5068 5070 5072 5074 5076 5078



(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.

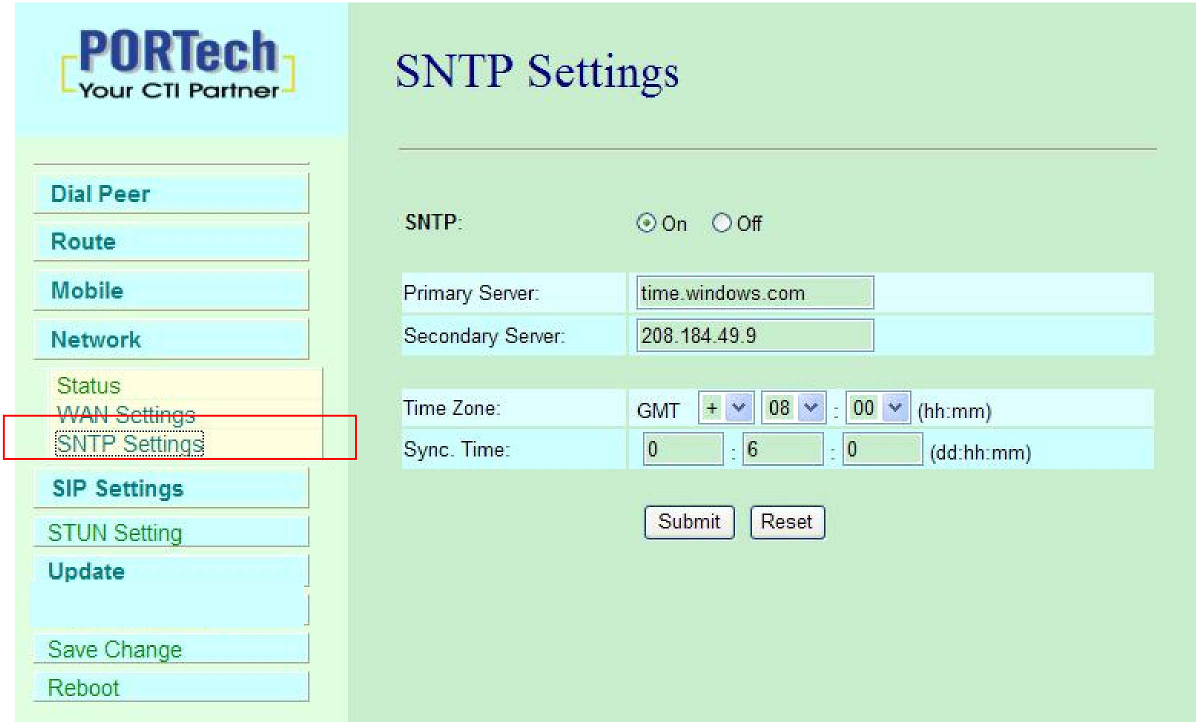
(2) The PPPoE Configuration item is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.

(3) The Bridge Item is to setup 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 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.

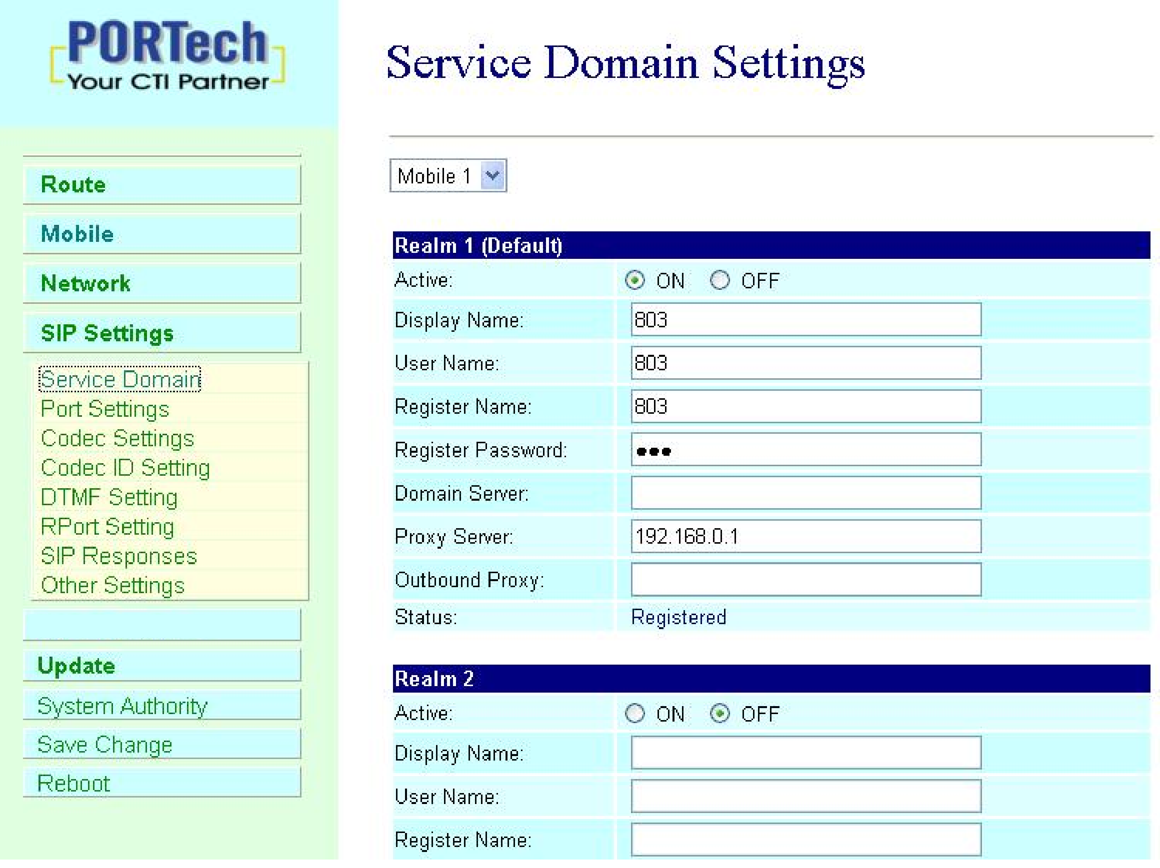


12. SIP Setting

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 information correctly then you can register to SIP Proxy Server correctly.

12.1 Service Domain Setting

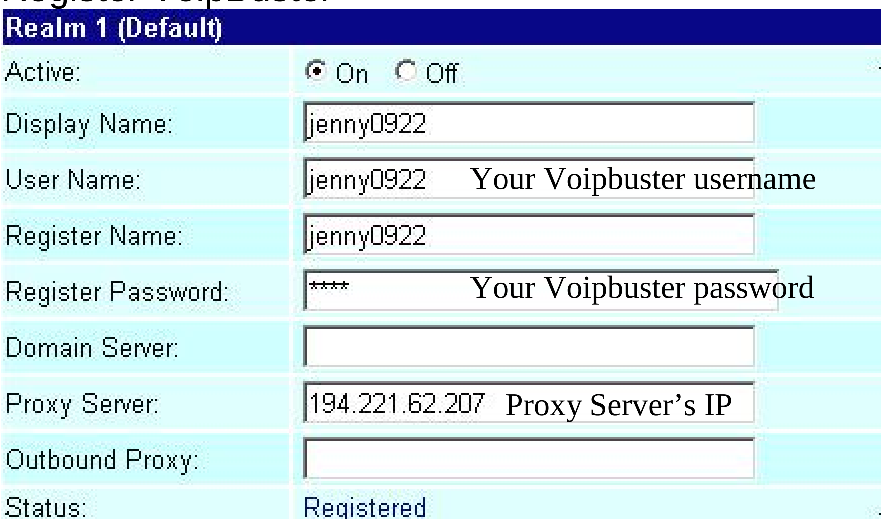
In Service Domain Function you need to input the account and the related information in this page please refer to your ISP Provider. You can register three SIP accounts. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from the tree SIP account.



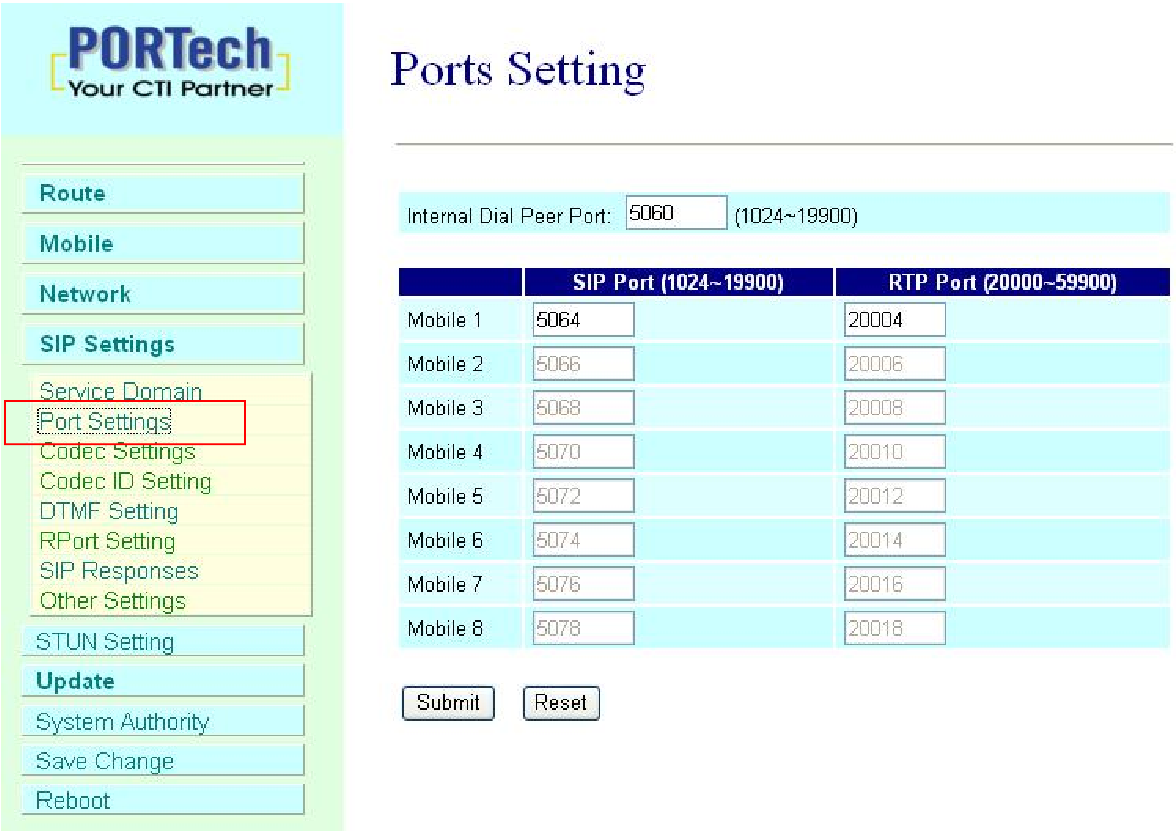
First you need to click Active to enable the Service Domain, and then you can input the following items.

(1) Choose Mobile 1, 2, 3 or 4
(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



12.3 Ports Setting



In Ports Setting, you can change dial peer port, SIP port, and RTP port.

Internal Dial Peer Port: default = **5060** (*important* this port number can't coincide with SIP port or RTP port)

SIP port: default = ch1:5064 ch2:5066 ch3:5068…etc (*important* this port number can't coincide with dial peer port or RTP port)

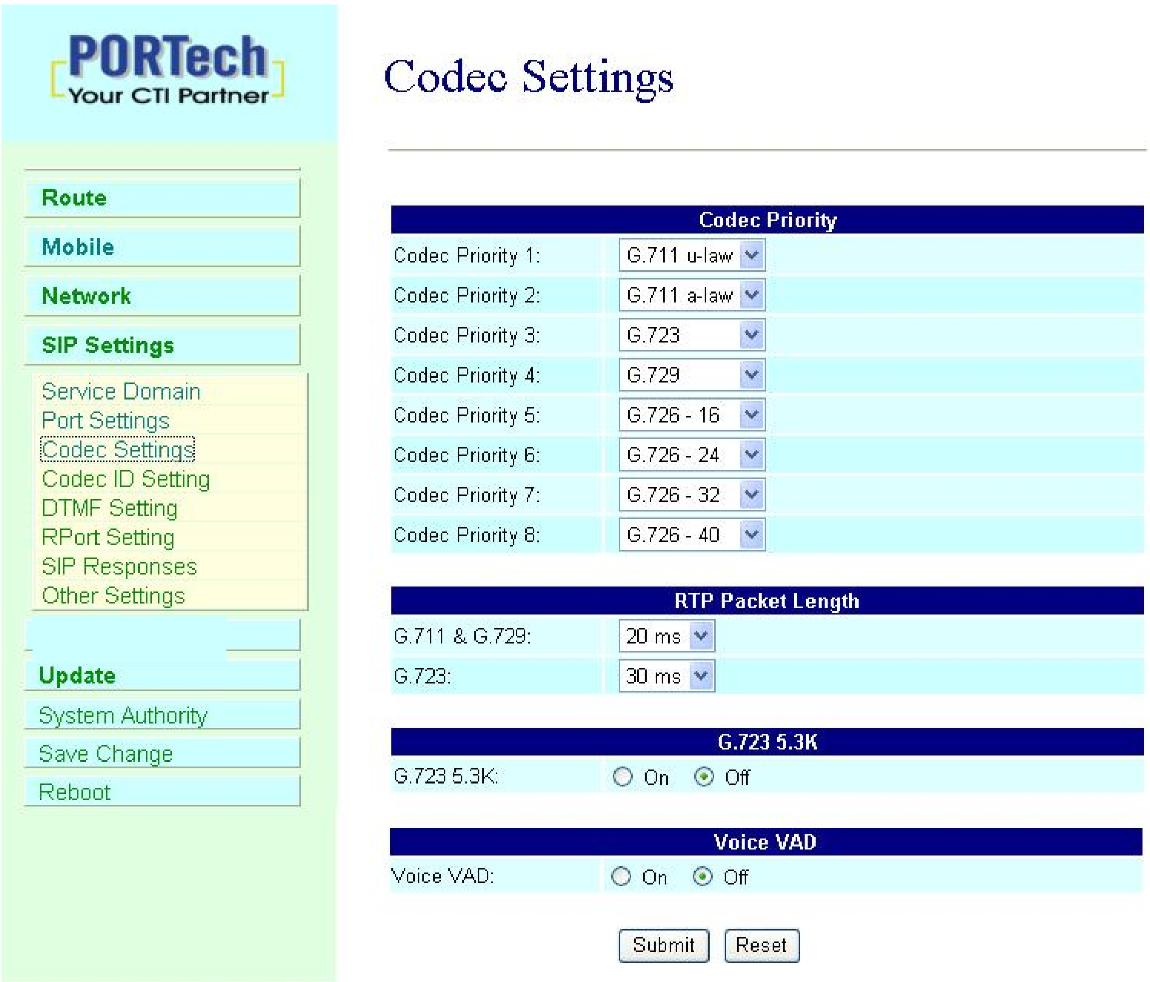
You can only change the port number on Ch1; other Channels will be changed automatically

RTP port: default = ch1:20004 ch2:20006 ch3:20008…etc (*important* this port number can't coincide with dial peer port or SIP port)

You can only change the port number on Ch1; other Channels will be changed automatically

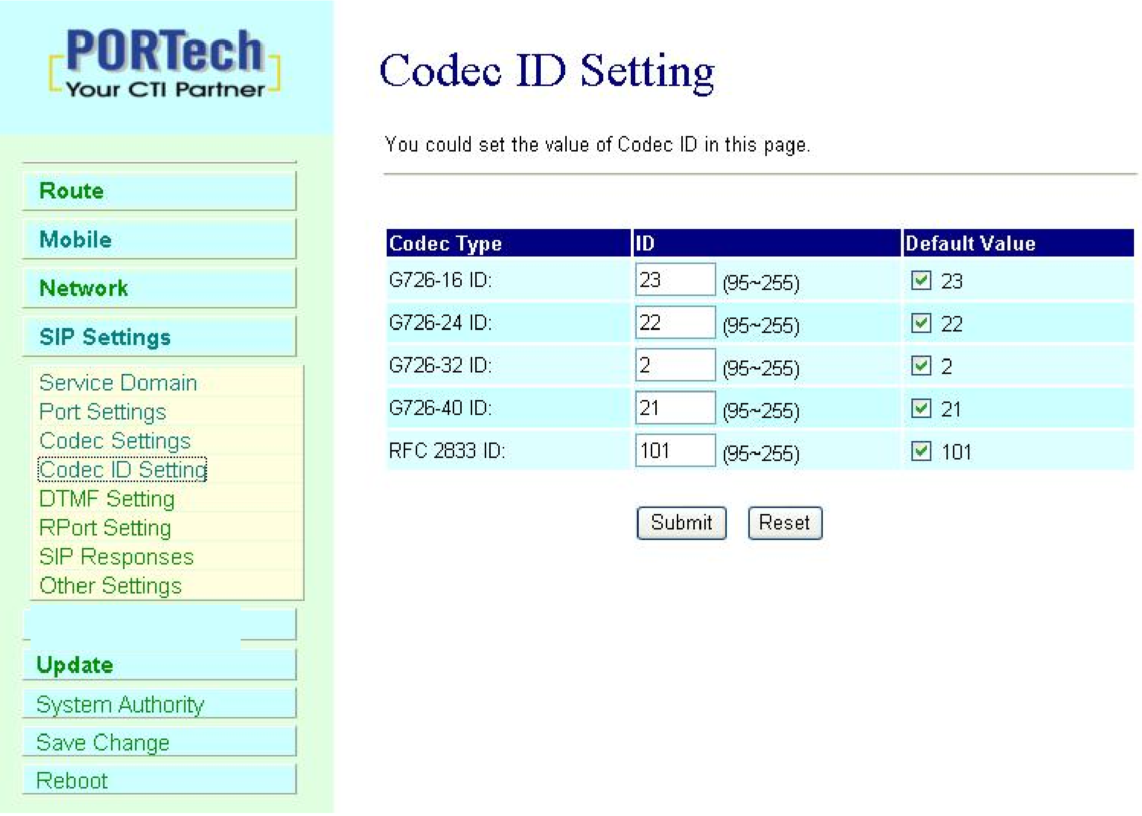
12.3 Codec Settings:

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.

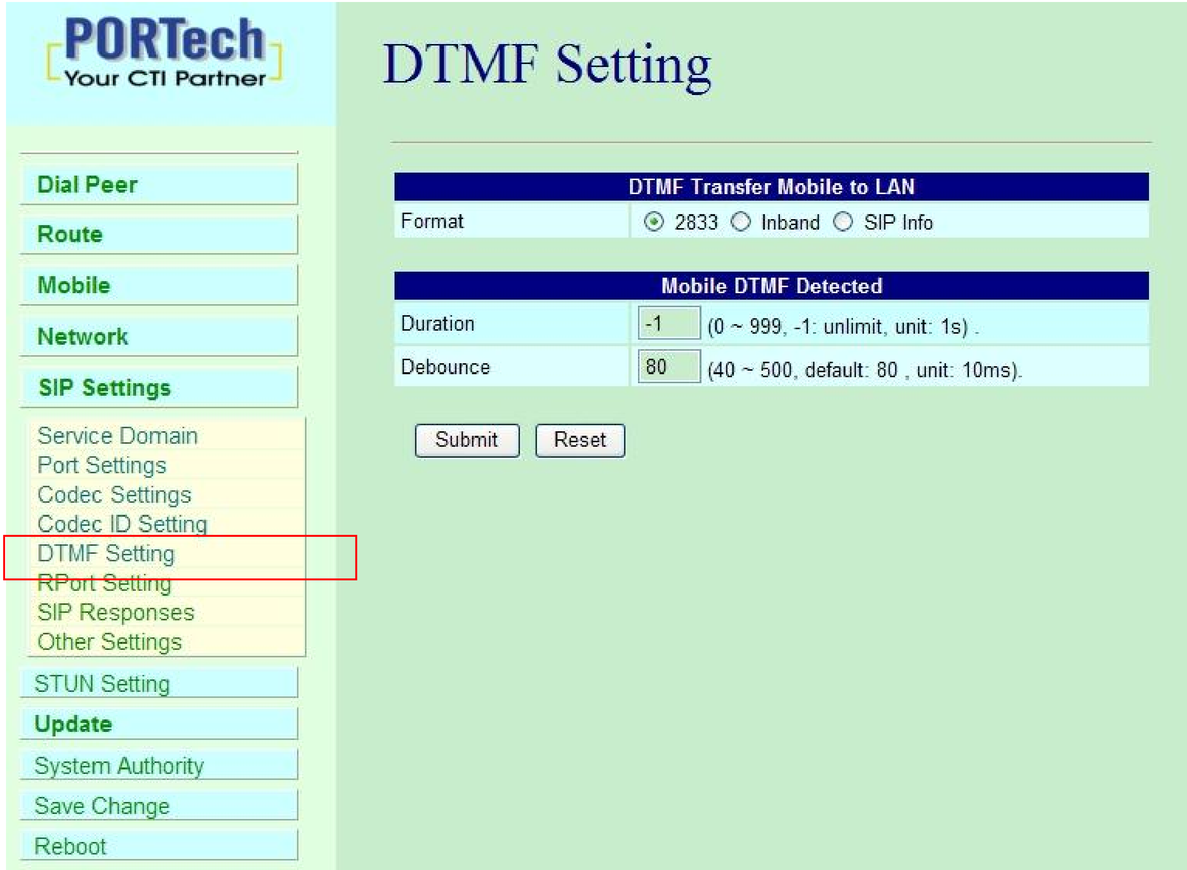


12.4 Codec ID Setting

You can setup the Codec ID in this page.



12.5 DTMF Setting



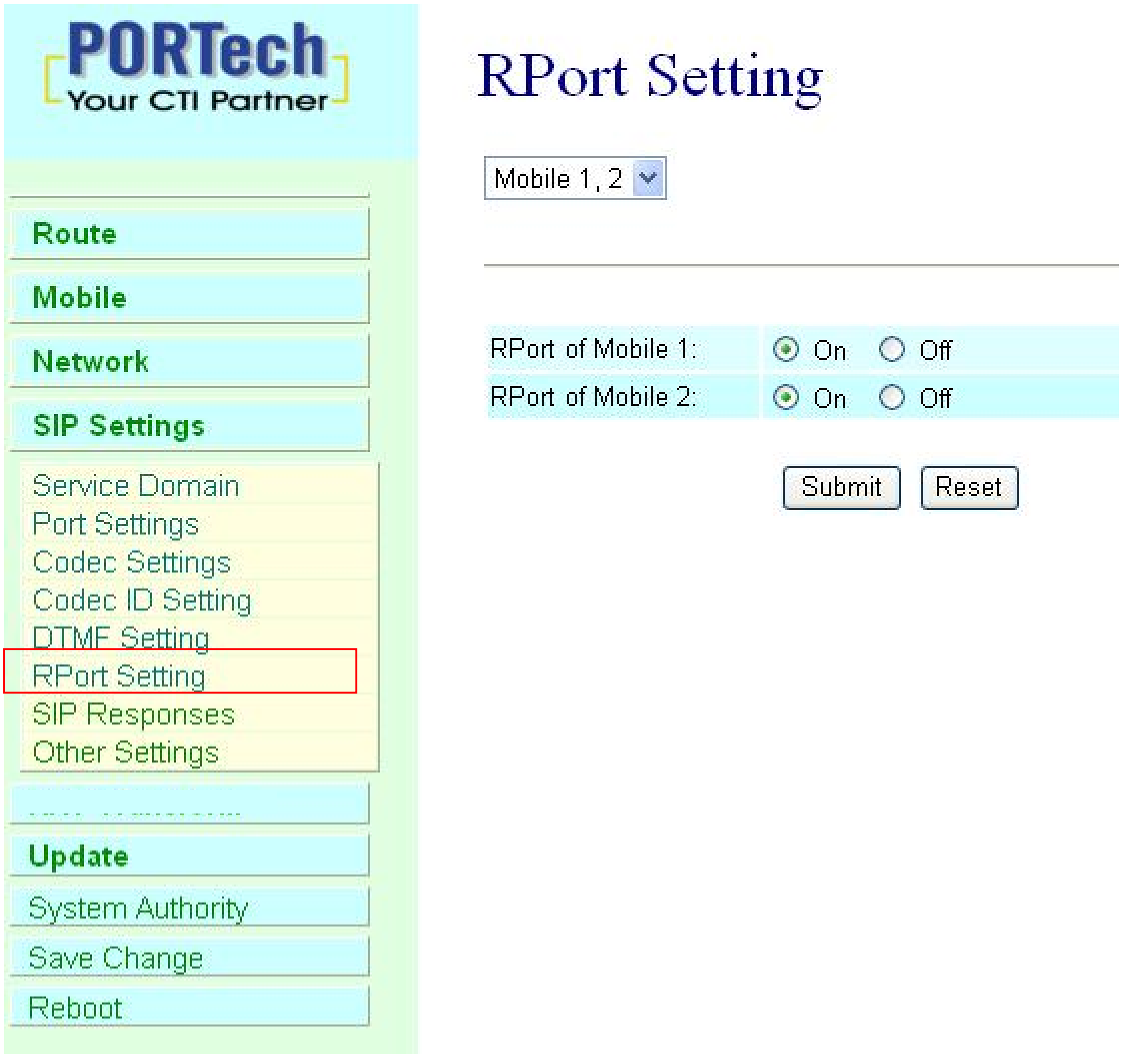
1. Format:
   a. 2833: Default RFC2833, the type of DTMF Data Transfer Format
   b. Inband: The Type of Inband DMTF Data Transfer Format
   c. SIP Info: The Type of SIP-Info DMTF Data Transfer Format;

2. Duration: Default is -1. It's the duration for MV-374/MV-378 to defect sender's DTMF. If the parameter is 0, MV-374/MV-378 won't detect sender's DTMF. Parameter is 0~999 seconds. After that duration, MV-374/MV-378 won't detect DTMF.

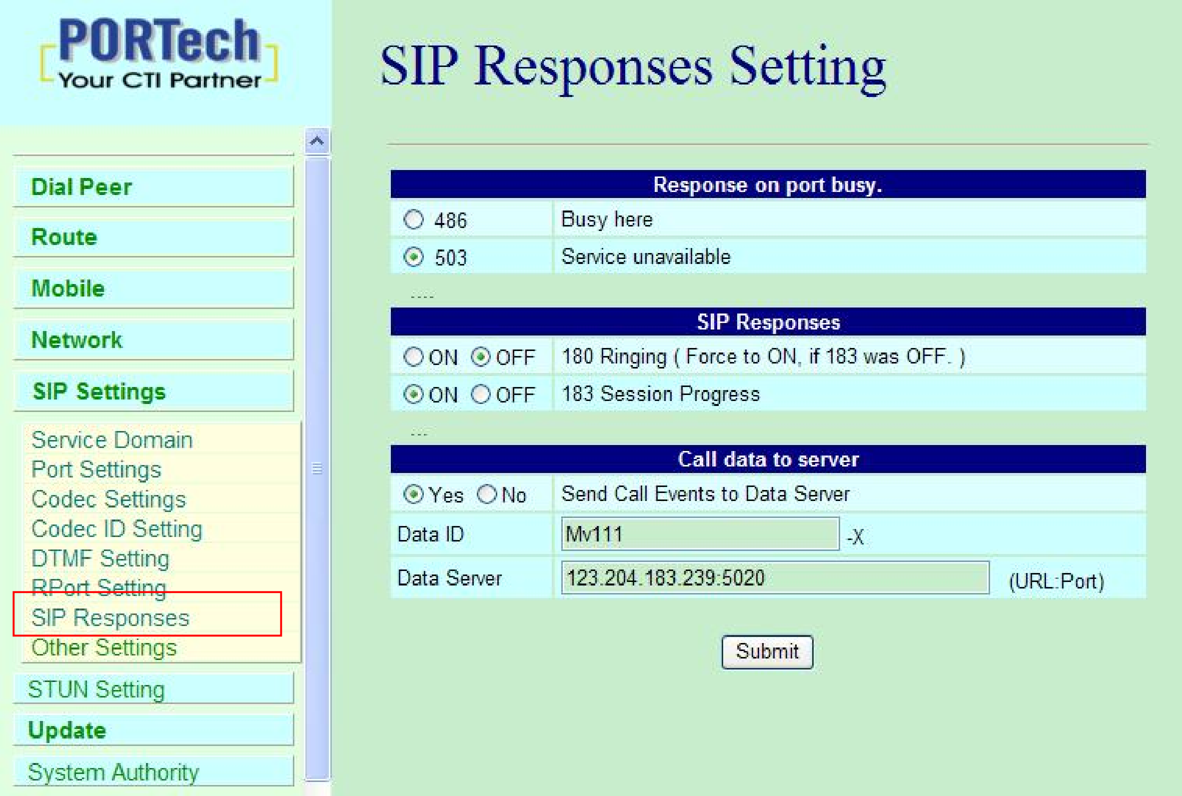
3. Debounce: Default is 80ms.User can adjust for own. If DTMF is adding more digits, please increase parameter over 80. If DMTF is lost digit, please decrease parameter less than 80.

12.6 RPort Function:

You can setup the RPort Enable/Disable in this page. Please follow your ISP information to make the settings. When you finished the setting, please click the Submit button.



12.7 SIP Responses



12.7.1 486(busy here), 503(Service unavailable):

When Device is busy, you can select 486 or 505 to response to SIP.

12.7.2 180 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 prompt voice directly. (For this function, 183 must be turn on)

12.7.3 183(Session Progress)

[It means "on progressing"]: When you turn 183 on, it means you can hear the prompt voice while GSM side is busy we recommend you to turn this on if you use SIP Proxy.

12.7.4 Call data to server(CDR)

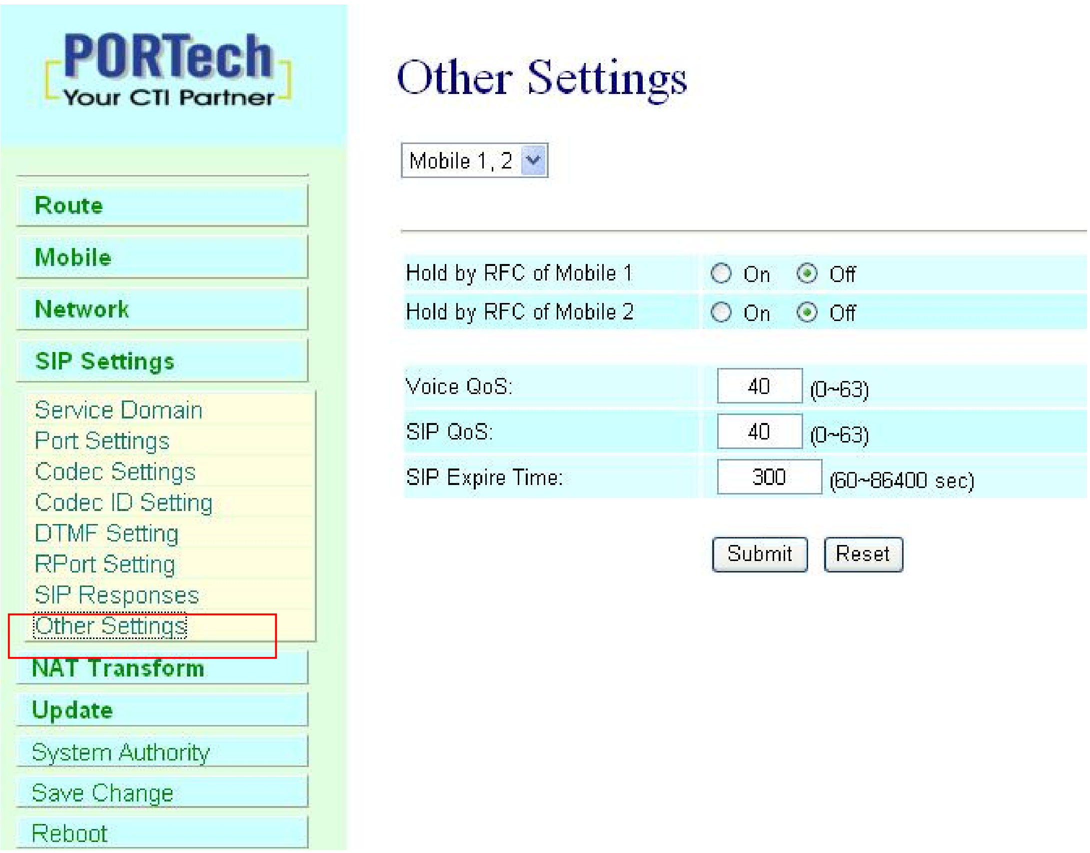
MV can provide Call Detail Record (CDR) for traffic and accounting management. User need to download external Dial Peer software on PC and can monitor traffic.

Data ID: MV will create one default Data ID

Data Server: Please fill the PC's IP, which is executed External Dial Peer Software

12.8 Other Settings

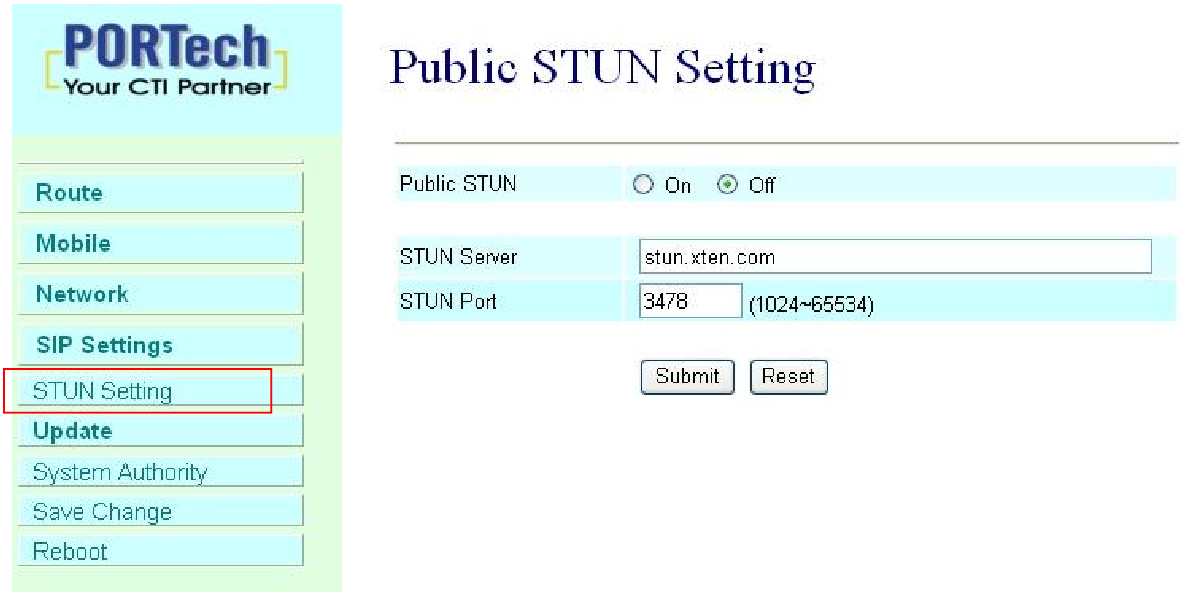
Other Settings: you can setup the Hold by RFC and QoS in this page. To change these settings, please follow your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets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



13. STUN Setting

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.



Public STUN OFF → Default is OFF; While MV-374/MV-378's WAN Setting is in Static IP or Private IP please selects Public STUN OFF.

Public STUN ON → While MV-374/MV-378 is working under Firewall or behind NAT, It will cause SIP can't register, or one side communicate, please select Public STUN ON.

14. Update

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

14.1 Update firmware

You can download new firmware from here, and follow those steps

https://www.portech.com.tw/p3-HowtoupdateMV-374.asp

NOTE: Please open the webpage from Internet Explorer, not compatible with FF or Google Chrome

14.1.1 **MV-374** Need to update firmware for 3 times (Slave1, Slave2, and Master)

Step 1: Slave 1: 192.168.0.100.8080, please update the firmware

Step 2: Slave 2: 192.168.0.100:8180, please update the firmware

Step 3: MASTER: 192.168.0.100 please updates the firmware

14.1.2 **MV-378**

Need to update firmware for 5 times (Slave1, Slave2, Slave3, Slave4, and Master)

Step 1: Slave 1: 192.168.0.100:8080 please update the firmware

Step 2: Slave 2: 192.168.0.100:8180 please update the firmware

Step 3: Slave 3: 192.168.0.100:8280 please update the firmware

Step 4: Slave 4: 192.168.0.100:8380 please update the firmware

Step 5: MASTER: 192.168.0.100 please update the firmware

Important

1. After you upgrade all the firmware, please click Default Setting on 192.168.0.100

2. Please do not change firmware's files name.

(1) Select the firmware code type, Risc code only.

(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.

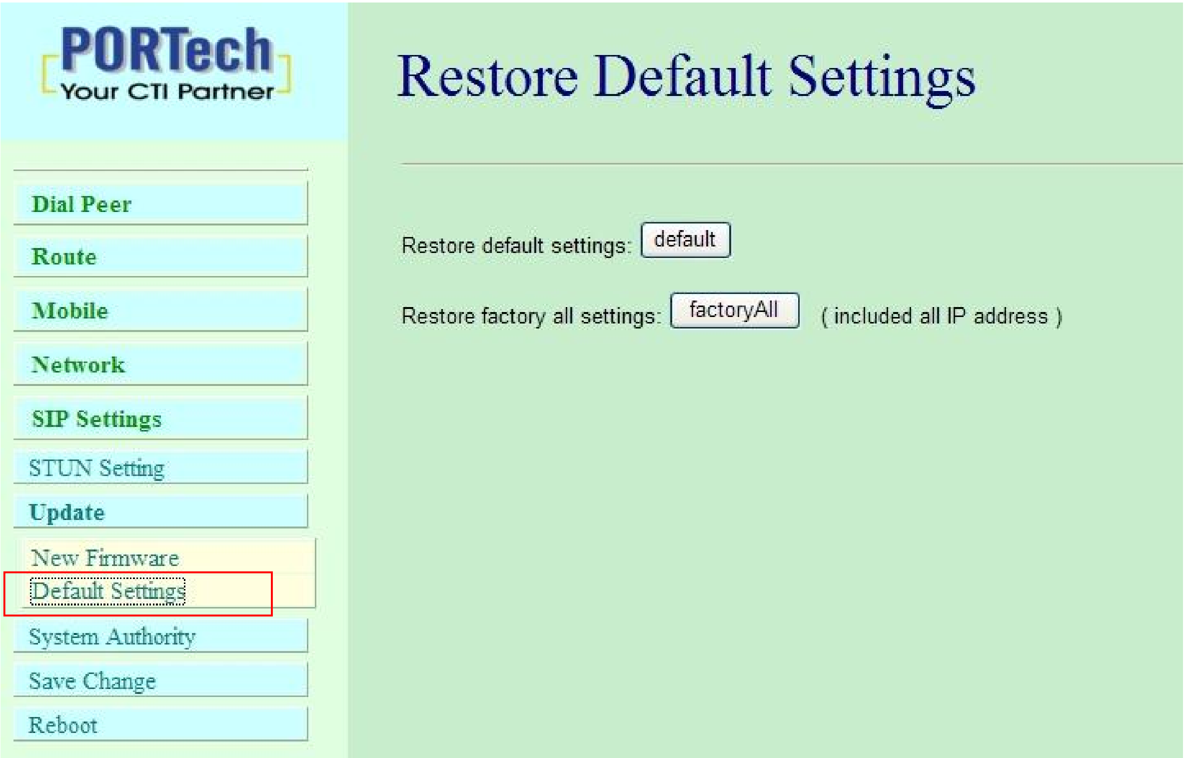
(4)Select the correct file you want to download to the system then click the Update button.

(5) Please click update/default setting after update firmware

14.2 Restore Default Settings

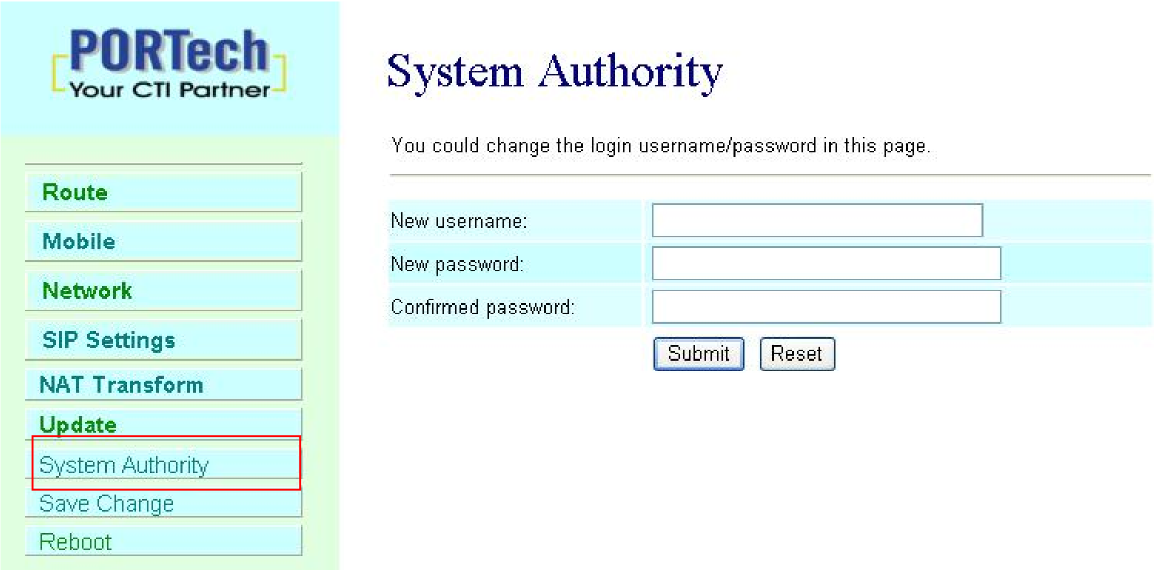
In this page: Update/ Default Settings, you could restore the factory default settings to the system. All setting will restore default setting. <u>The device IP will back to user original IP, but not the default IP.</u>

Factory all: all setting includes IP will restore default setting.



15. System Authority

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

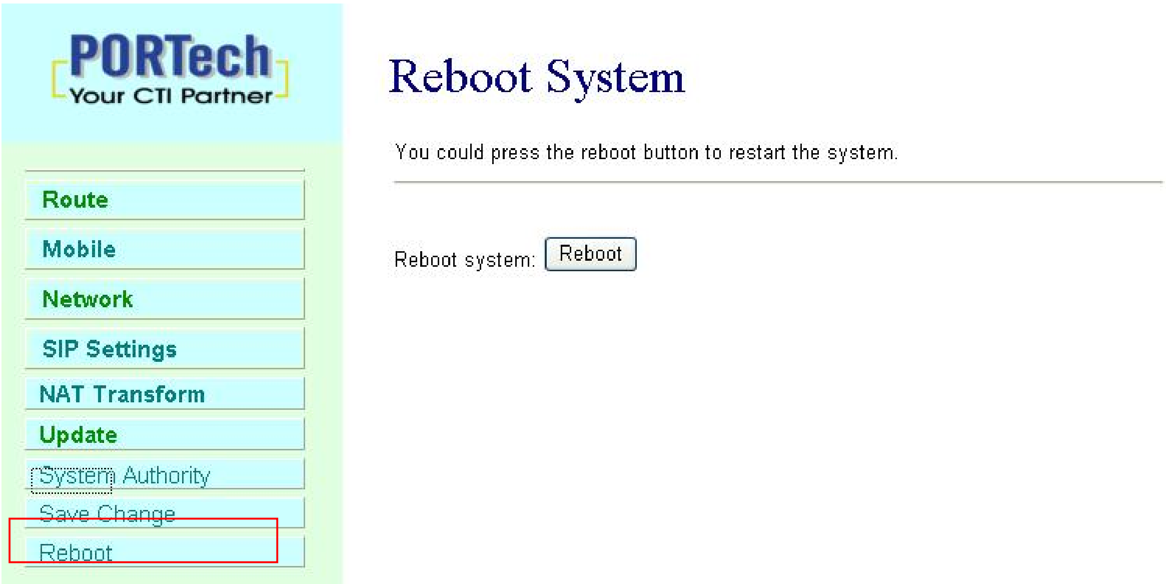


16. Save Change

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.

17. Reboot

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



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.729A

G.729A/B

18.4 Voice Quality

VAD

CNG

AEC, LEC

Packet loss

18.5 GSM (MV-374/MV-378)

Quad Band: 900/1800/1900/850MHZ

3G/UMTS: for all world and Japan (SoftBank and Docomo)

3G: EDGE/GPRS 850, 900, 1800, 1900 MHz / HSDPA/UMTS 850, 1900, 2100 MHz

CDMA 2000(800MHZ/1900MHZ)

Please note

1. Most CDMA -2000 operators don't offer Answer signal.
   So VoIP to Mobile, MV-378 will connect soon.
   CDMA -2000 operators will start billing soon. It doesn't wait mobile side answer
2. CDMA Version doesn't support SMS Feature and 180/183 unavailable
3. CDMA version doesn't have Remote SIM feature

19. Simple Steps

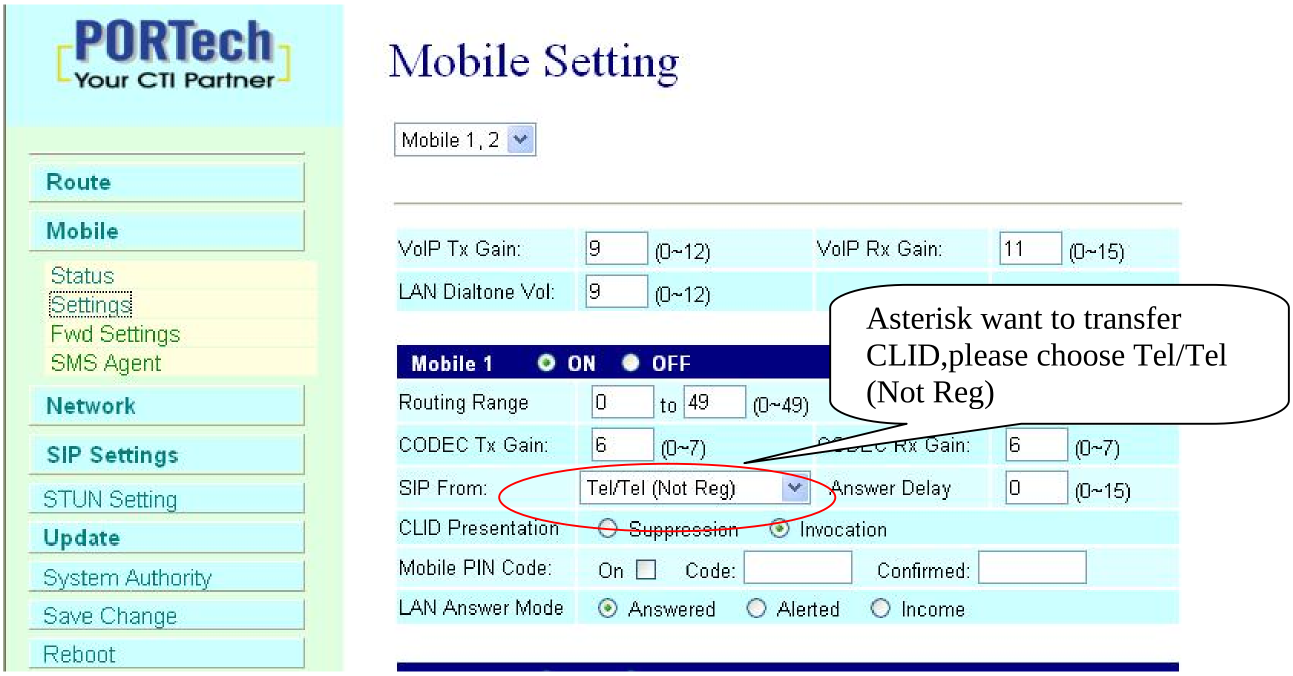
Step 1. Change the Network setting as you need (Network/network setting)
Step 2. Register SIP proxy Server or Asterisk or VoipBuster as you need (sip setting/service domain)
Step 3. Set Mobile setting –adjust your gain as you need
Step 4. Set Route (**request**)

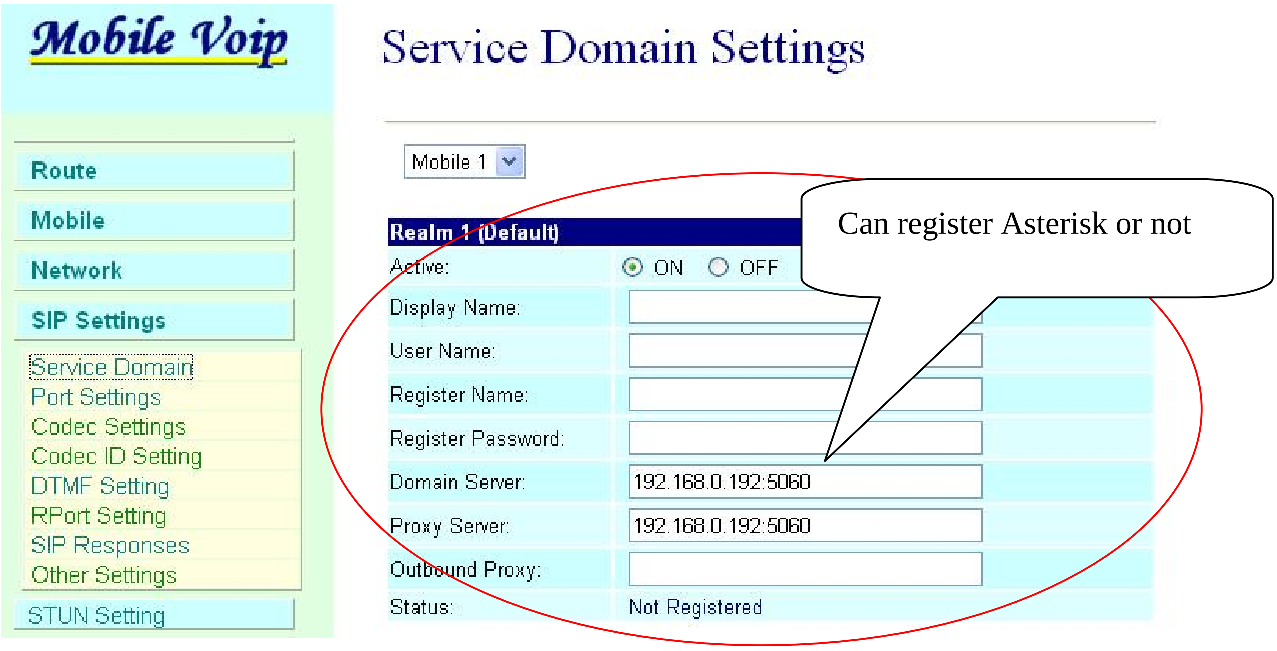
mobile to Lan:		
(1)	*,* --->it is two stage dialing.	
	when mobile call in,MV-37x will provide dial tone and you can enter ip or asterisk extension or phone number.	
	*	If you want to enter phone number, please note your asterisk need to have route of destination number.
(2)	*, specific extension or IP or phone number	
	when mobile call in,MV-37x 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-37x will provide dial tone and you can enter mobile number.	
(2)	*, specific mobile number	
	When lan phone call in, MV-37x will connect with the specific mobile number auto.	
(3)	*,#--->It is 1 stage dialing	
	When Lan phone and MV-37x 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-37x

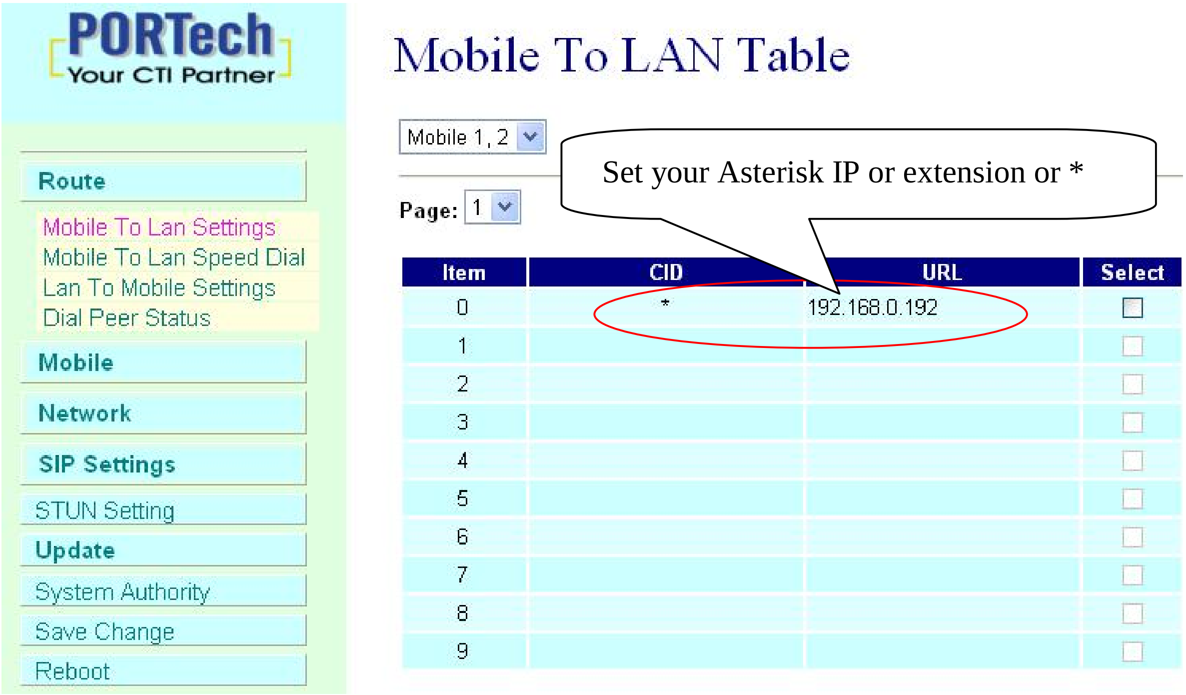
* All changes both need to click "save and change"

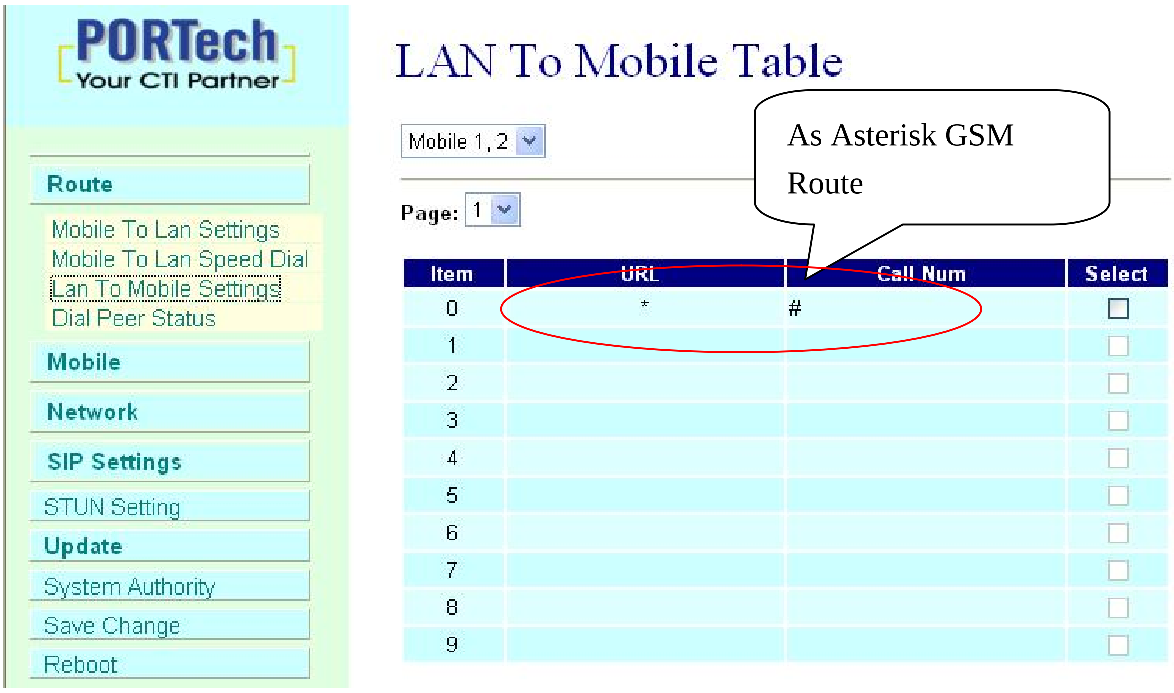
20. Appendix: Setup MV-37x with Asterisk

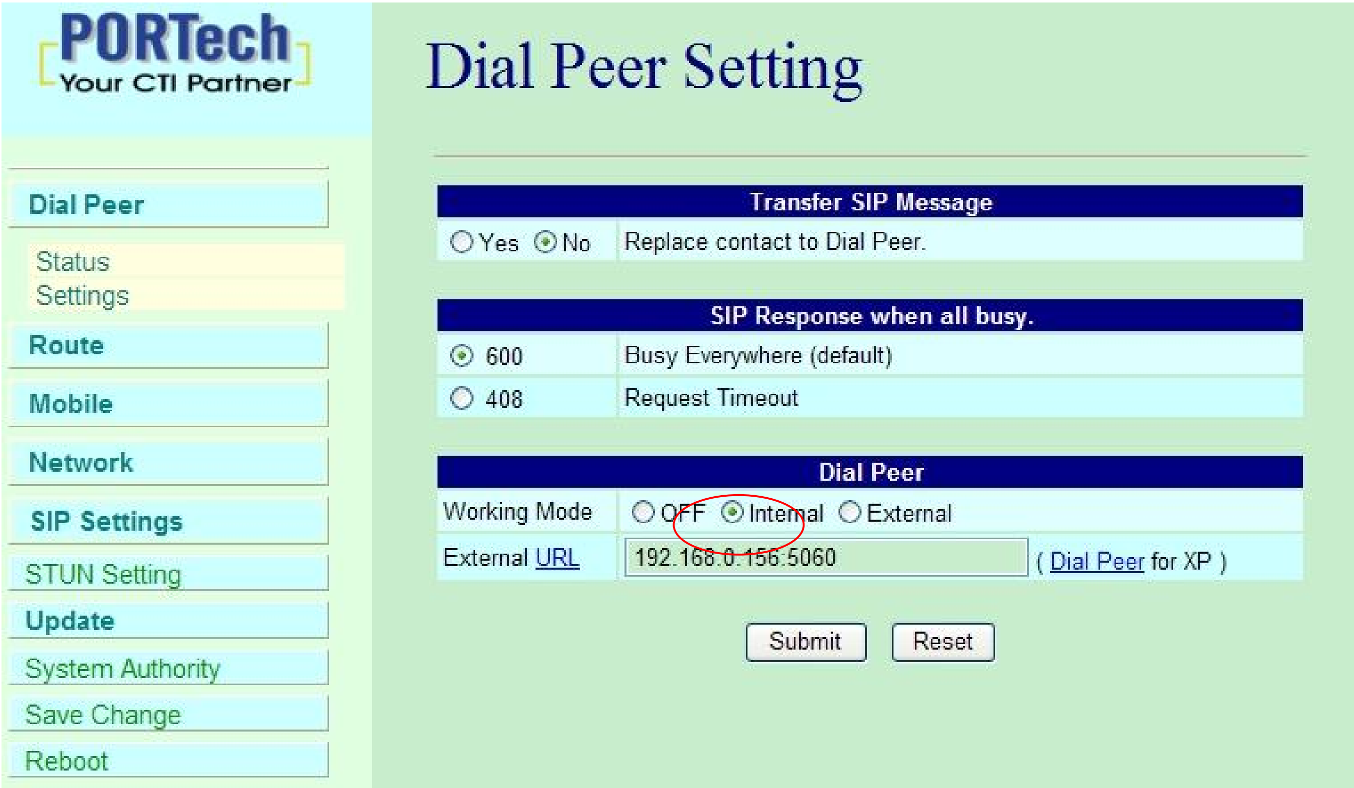
MV-37x Settings

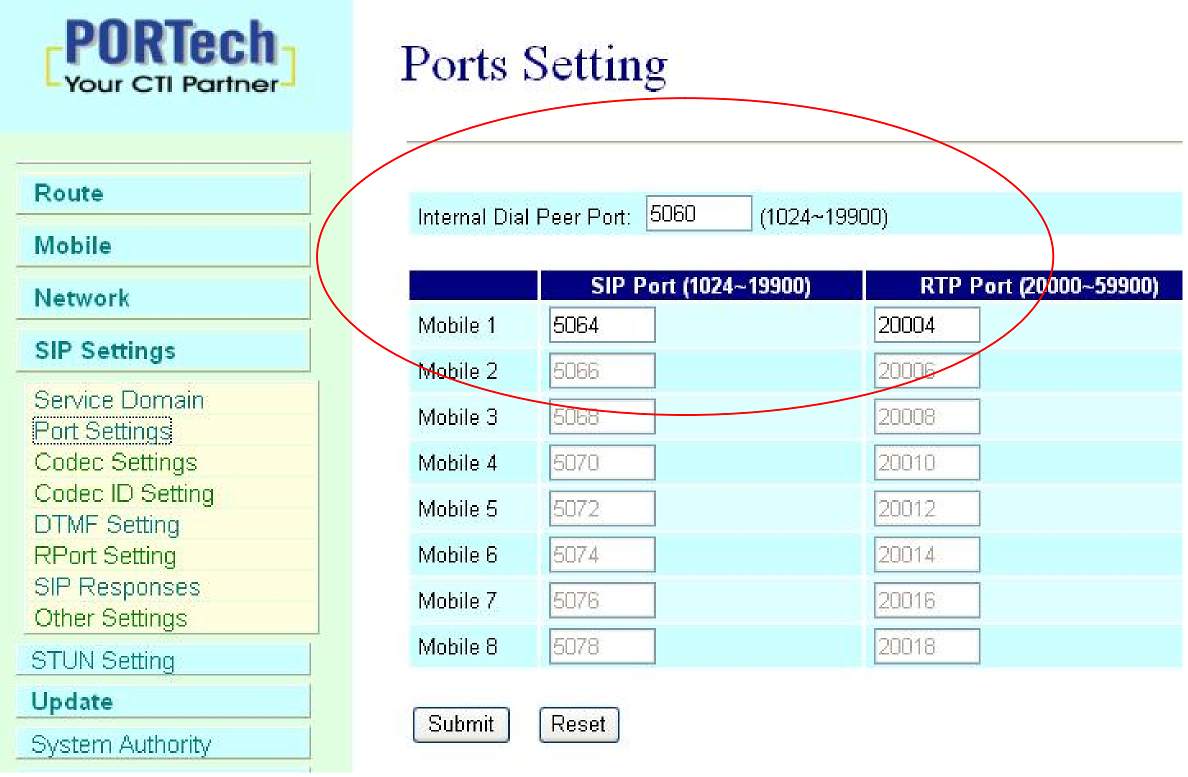

PORTech
Your CTI Partner
Route
Mobile
Status
Settings
Fwd Settings
SMS Agent
Network
SIP Settings
STUN Setting
Update
System Authority
Save Change
Reboot
Mobile Setting
Mobile 1, 2
VoIP Tx Gain: 9 (0~12)
VoIP Rx Gain: 11 (0~15)
LAN Dialtone Vol: 9 (0~12)
Mobile 1 ON OFF
Routing Range 0 to 49 (0~49)
CODEC Tx Gain: 6 (0~7)
Rx Gain: 6 (0~7)
SIP From: Tel/Tel (Not Reg)
Answer Delay 0 (0~15)
CLID Presentation Suppression Invocation
Mobile PIN Code: On Code: Confirmed:
LAN Answer Mode Answered Alerted Income
Asterisk want to transfer CLID,please choose Tel/Tel (Not Reg)


Mobile Voip
Route
Mobile
Network
SIP Settings
Service Domain
Port Settings
Codec Settings
Codec ID Setting
DTMF Setting
RPort Setting
SIP Responses
Other Settings
STUN Setting
Service Domain Settings
Mobile 1
Realm 1 (Default)
Active: ON OFF
Display Name:
User Name:
Register Name:
Register Password:
Domain Server: 192.168.0.192:5060
Proxy Server: 192.168.0.192:5060
Outbound Proxy:
Status: Not Registered
Can register Asterisk or not


PORTech
Your CTI Partner
Route
Mobile To Lan Settings
Mobile To Lan Speed Dial
Lan To Mobile Settings
Dial Peer Status
Mobile
Network
SIP Settings
STUN Setting
Update
System Authority
Save Change
Reboot
Mobile To LAN Table
Mobile 1, 2
Page: 1
Set your Asterisk IP or extension or *
Item
CID
URL
Select
0
*
192.168.0.192
1
2
3
4
5
6
7
8
9

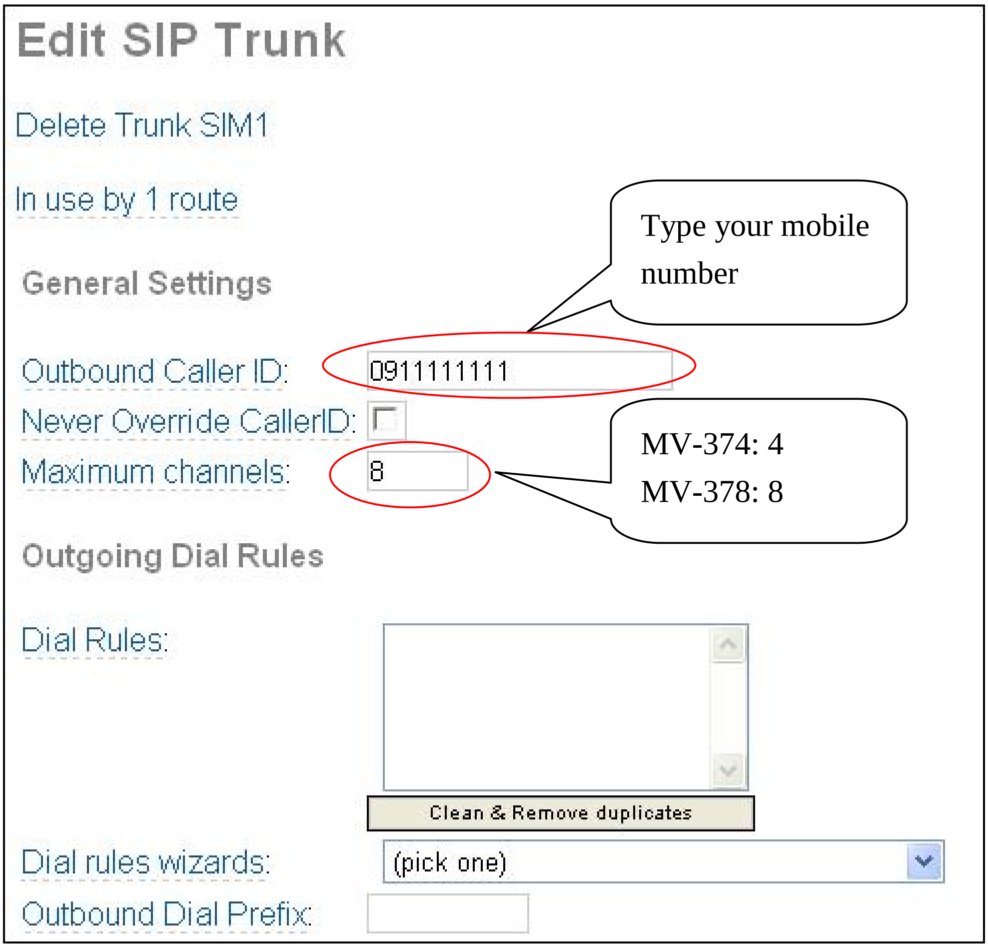

PORTech
Your CTI Partner
Route
Mobile To Lan Settings
Mobile To Lan Speed Dial
Lan To Mobile Settings
Dial Peer Status
Mobile
Network
SIP Settings
STUN Setting
Update
System Authority
Save Change
Reboot
LAN To Mobile Table
Mobile 1, 2
Page: 1
As Asterisk GSM Route
Item
URL
Call Num
Select
0
*
#
1
2
3
4
5
6
7
8
9

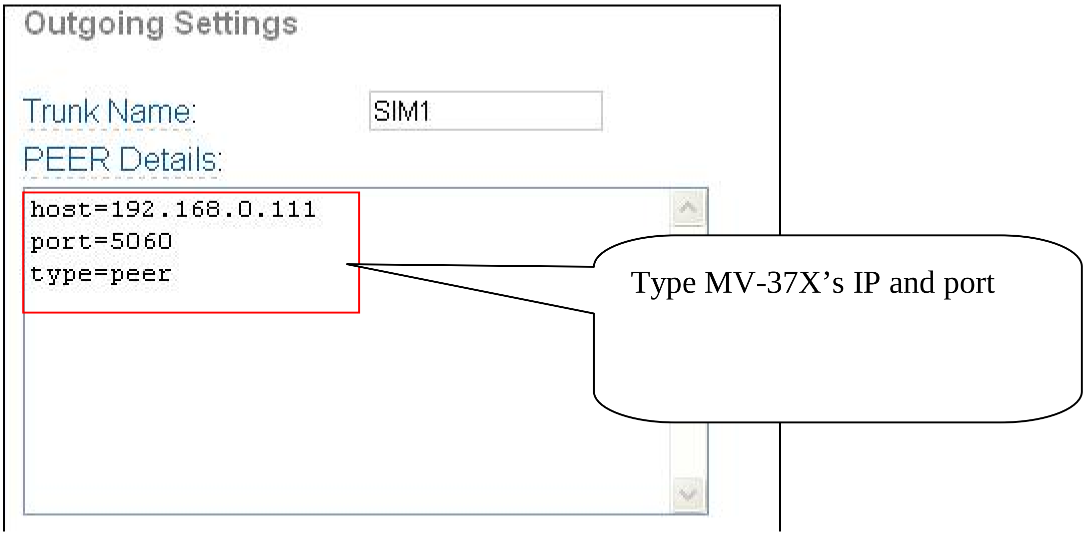

PORTech
Your CTI Partner
Dial Peer Setting
Dial Peer
Status
Settings
Route
Mobile
Network
SIP Settings
STUN Setting
Update
System Authority
Save Change
Reboot
Transfer SIP Message
Yes No
Replace contact to Dial Peer.
SIP Response when all busy.
600
Busy Everywhere (default)
408
Request Timeout
Dial Peer
Working Mode
OFF Internal External
External URL
192.168.0.156:5060
(Dial Peer for XP)
Submit
Reset



Don't forget to Save changes and then reboot

Asterisk / Trixbox setting
Add SIP Trunk:




Outgoing Settings
Trunk Name:
SIM1
PEER Details:
host=192.168.0.111
port=5060
type=peer
Type MV-37X's IP and port

Set GSM Route that dial out via MV-37X

Frequency: Quad Band:900/1800/1900/850MHZ
GSM Module use Simcom sim340
Compliant to GSM phase 2/2+
-Class 4 (2W@850/900 MHz)
-Class 1 (1W@1800/1900 MHz)

15.21

Federal Communications Commission (FCC) Statement

You are cautioned that changes or modifications not expressly approved by the part responsible for compliance could void the user's authority to operate the equipment.

15.105(b)

Federal Communications Commission (FCC) Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

-Reorient or relocate the receiving antenna.
-Increase the separation between the equipment and receiver.
-Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
-Consult the dealer or an experienced radio/TV technician for help.

Operation is subject to the following two conditions:
1) this device may not cause interference and
2) this device must accept any interference, including interference that may cause undesired operation of the device.

FCC RF Radiation Exposure Statement:

1. This Transmitter must not be co-located or operating in conjunction with any other antenna or transmitter.

2. This equipment complies with FCC RF radiation exposure limits set forth for an uncontrolled environment. This equipment should be installed and operated with a minimum distance of 20 centimeters between the radiator and your body.

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