

MV-3716 / MV-3732

VoIP GSM Gateway

User Manual



MV-3716



MV-3732

PORTech Communications Inc.

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1. Introduction

MV-3716/MV-3732 is a 16 / 32 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 16 / 32 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-3716/MV-3732」 main body
- 3.2 Power adaptor
Output 12V/9A, Input 100~240V Auto switching
- 3.3 Network cable
- 3.4 Antenna: MV-3716: 4 pcs / MV-3732: 8 pcs
- 3.5 Rack-mount accessories (compatible with 19"Rack)
- 3.6 User Manual



(3.1) MV-3732

(3.1) MV-3716



(3.2) Power adapter



(3.3)



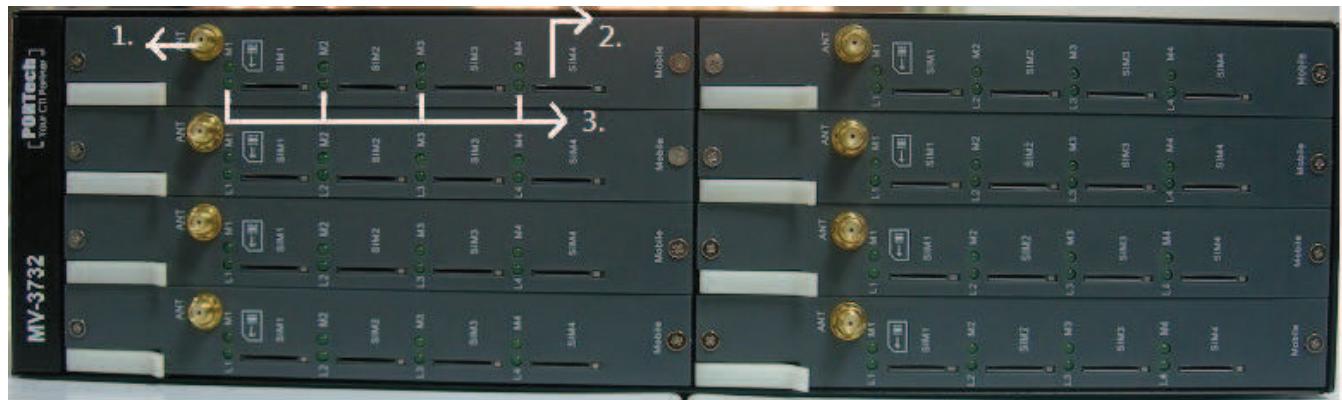
(3.4)



(3.5)

4. Dimension: 37*26*10 cm

5. Chart of the device



- 1 Antenna : Antenna Connector
2. SIM Holder: Insert the SIM card as instruction and hear click sound (the chip side down); Press the SIM to bottom with click sound to remove the SIM card
3. PWR (Power LED) : Light up when power is normal.



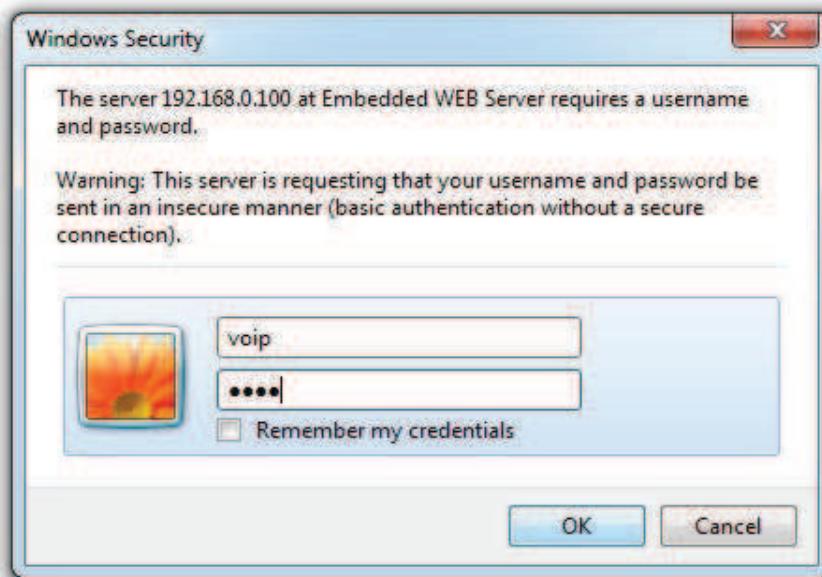
1. DC 12V : Power input.
2. WAN: RJ-45 internet connector
3. LAN: For maintenance use, not for any propose



1. Dial Peer Reset Button
2. IP Reset Button:
Press this button about 10 seconds
IP restore back to 192.168.0.100
3. DHCP mode Button:
Press this button about 10 seconds and switch to DHCP mode

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

User can see the demo system current system information like firmware version, company... etc in this page.

The screenshot shows a web-based configuration interface for a PORTech MV-3732 v10.272 device. The left sidebar contains a navigation menu with the following items:

- Dial Peer
- Status
- Settings
- Prefixs
- CDR
- Route
- Mobile
- Network
- SIP Settings
- STUN Setting
- Update
- System Authority
- Save Change
- Reboot

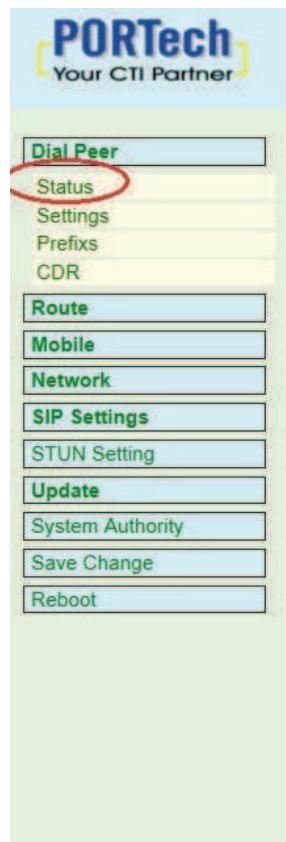
The main content area displays the following system information:

Module Description:	GSM:850/900/1800/1900MHz (M10)
Firmware Version:	Thu May 30 15:45:04 2013.
Codec Version:	Fri Mar 20 17:13:45 2009.
Contact Address:	150, Shiang-Shung N.Road., Taichung, Taiwan, R.O.C.
Tel:	886-4-23058000
Fax:	886-4-23022596
E-Mail:	sales@portech.com.tw
Web Site:	http://www.portech.com.tw

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8. Dial Peer

8.1 Status



Dial Peer Status - 2013-06-06 09:47								
ch	grp	State	MNC	SQ	Mobile	dir	LAN	
1	0	idle/1	46692	21	0963283792	<	123@192.168.0.127:6050	
2	0	idle/1	46692	20	-	-	-	
3	0	idle/1	46692	21	-	-	-	
4	0	idle/1	46692	21	-	-	-	
5	0	idle/1	46692	21	-	-	-	
6	0	idle/1	46692	21	-	-	-	
7	0	idle/1	46692	11	-	-	-	
8	0	idle/1	46692	21	-	-	-	
9	0	idle/1	46692	21	-	-	-	
10	0	idle/1	46692	22	-	-	-	
11	0	idle/1	46692	22	-	-	-	
12	0	idle/1	46692	21	-	-	-	
13	0	idle/1	46692	22	-	-	-	
14	0	idle/1	46692	12	-	-	-	
15	0	idle/1	46601	19	-	-	-	
16	0	idle/1	46692	22	-	-	-	
17	0	idle/1	46692	20	-	-	-	
18	0	idle/1	46692	20	-	-	-	
19	0	idle/1	46692	20	-	-	-	
20	0	idle/1	46692	20	-	-	-	
21	0	idle/1	46692	19	-	-	-	
22	0	idle/1	46692	17	-	-	-	
23	0	idle/1	46692	17	-	-	-	
24	0	idle/1	46697	21	-	-	-	
25	0	idle/1	46692	15	-	-	-	
26	0	idle/1	46601	18	-	-	-	
27	0	idle/1	46692	14	-	-	-	
28	0	idle/1	46692	14	-	-	-	
29	0	idle/1	46692	20	-	-	-	
30	0	idle/1	46692	20	-	-	-	
31	0	init/0	-	-	-	-	-	
32	0	init/0	-	-	-	-	-	

1. ch: The port of GSM channel
2. grp: the group of GSM channel
3. state:
 - INIT/0: GSM module is initialing
 - IDLE/0: GSM module not register
 - IDLE/1: GSM module registered
 - M.ringback/0: Ring Back
 - M.dialed/0: GSM port is dialed
 - M.listen/0: GSM port is engaged

-
-
- 4. MNC: Mobile Network Code
 - 5. SQ: Signal quality
 - 6. Mobile: The caller number of the incoming/outgoing call to Mobile
 - 7. dir: The Arrow shows the route to be LAN to Mobile or Mobile to LAN
 - a. < : LAN to Mobile
 - b. > : Mobile to LAN
 - 8. LAN: the IP address of the last incoming/outgoing call from/to LAN

8.2 Settings

Dial Peer Setting

Transfer SIP Message

Yes No Replace contact to Dial Peer.

SIP Response when all busy

<input checked="" type="radio"/> 600	Busy Everywhere (default)
<input type="radio"/> 408	Request Timeout
<input type="radio"/> 480	Temporarily unavailable
<input type="radio"/> 503	Service unavailable

Dial Peer

Working Mode OFF Internal External

External URL (Dial Peer for XP)

Submit Reset

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

User can select the corresponding response while all ports are busy.

The Default is 600

600 : Busy Everywhere (default)

408 : Request Timeout

480 : Temporarily unavailable

503: Service unavailable

3. Dial Peer

Working Mode→

- a. OFF: To disable Dial Peer, user need to assign the port of GSM channel for the incoming calls from LAN side (E.g. Default ch1 is 5064 port; ch2 should be 5066 port and so on)
- b. Internal: to motivate Dial Peer, all incoming calls from LAN will come to dial peer port. Dial peer will route calls to idle channels(Default: 5060 port)
- c. External: All GSM Channel are 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
[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
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
[SipPort]
1=5060
2=5062
3=5064
4=5066
5=5068
6=5070
7=5072
8=5074
9=5060
10=5062
11=5064
12=5066
13=5068

Total ip / port

The first MV-378

The second MV-378

The first MV-378

The second MV-378

The second MV-378

The first MV-378

The second MV-378

14=5070
15=5072
16=5074

[RtpPort]
1=60000
2=60002
3=60004
4=60006
5=60008
6=60010
7=60012
8=60014
9=60000
10=60002
11=60004
12=60006
13=60008
14=60010
15=60012
16=60014

[PtcPort]
1=40000
2=40000
3=40008
4=40008
5=40016
6=40016
7=40024
8=40024
9=40000
10=40000
11=40008
12=40008
13=40016
14=40016
15=40024
16=40024

External Dial Peer Log

You can check the Statue here

Dial Peer - (Apr 19 2011, 15:55:33)						
Log	Status	Set	Event			
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 Prefix

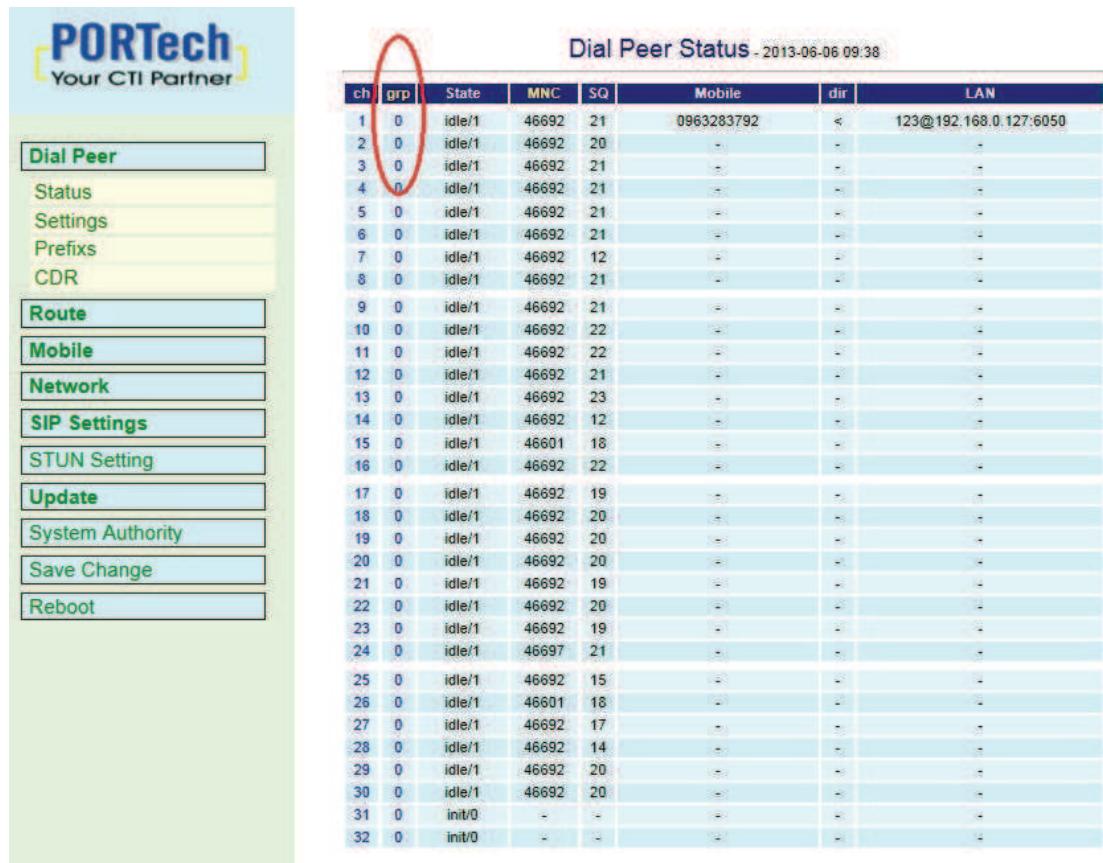
User can setup the prefix number in 15 groups. Dial peer will route the calls based on the prefix settings of each group

Group	Name	Prefix
0	test	09
1		
2		
3		
4		
5		
6		
7		
8		
9		
10		
11		
12		
13		
14		
15		

1. Group Enable

Off: The default is off.

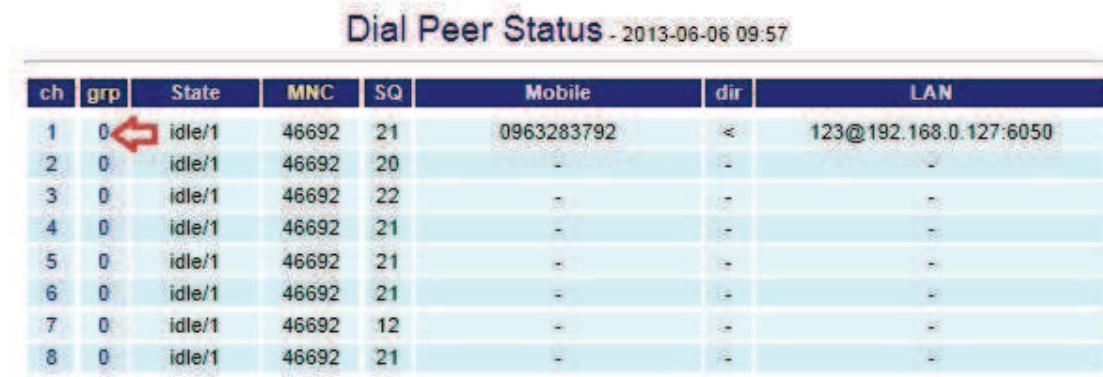
On: Dial peer will route the calls based on the prefix settings of each group. And Dial Peer status will show the grp information as below.



Dial Peer Status - 2013-06-06 09:38

ch	grp	State	MNC	SQ	Mobile	dir	LAN
1	0	idle/1	46692	21	0963283792	<	123@192.168.0.127:6050
2	0	idle/1	46692	20	-	-	-
3	0	idle/1	46692	21	-	-	-
4	0	idle/1	46692	21	-	-	-
5	0	idle/1	46692	21	-	-	-
6	0	idle/1	46692	21	-	-	-
7	0	idle/1	46692	12	-	-	-
8	0	idle/1	46692	21	-	-	-
9	0	idle/1	46692	21	-	-	-
10	0	idle/1	46692	22	-	-	-
11	0	idle/1	46692	22	-	-	-
12	0	idle/1	46692	21	-	-	-
13	0	idle/1	46692	23	-	-	-
14	0	idle/1	46692	12	-	-	-
15	0	idle/1	46601	18	-	-	-
16	0	idle/1	46692	22	-	-	-
17	0	idle/1	46692	19	-	-	-
18	0	idle/1	46692	20	-	-	-
19	0	idle/1	46692	20	-	-	-
20	0	idle/1	46692	20	-	-	-
21	0	idle/1	46692	19	-	-	-
22	0	idle/1	46692	20	-	-	-
23	0	idle/1	46692	19	-	-	-
24	0	idle/1	46697	21	-	-	-
25	0	idle/1	46692	15	-	-	-
26	0	idle/1	46601	18	-	-	-
27	0	idle/1	46692	17	-	-	-
28	0	idle/1	46692	14	-	-	-
29	0	idle/1	46692	20	-	-	-
30	0	idle/1	46692	20	-	-	-
31	0	init/0	-	-	-	-	-
32	0	init/0	-	-	-	-	-

Please click to select the group number of each channel



Dial Peer Status - 2013-06-06 09:57

ch	grp	State	MNC	SQ	Mobile	dir	LAN
1	0	idle/1	46692	21	0963283792	<	123@192.168.0.127:6050
2	0	idle/1	46692	20	-	-	-
3	0	idle/1	46692	22	-	-	-
4	0	idle/1	46692	21	-	-	-
5	0	idle/1	46692	21	-	-	-
6	0	idle/1	46692	21	-	-	-
7	0	idle/1	46692	12	-	-	-
8	0	idle/1	46692	21	-	-	-

After setting, please click submit button

The screenshot shows the PORTech web interface. On the left, there is a vertical sidebar with the following menu items:

- Dial Peer
- Route
- Mobile
- Network
- SIP Settings
- STUN Setting
- Update
- System Authority
- Save Change
- Reboot

The main content area is titled "Group Select". It contains a table with two columns:

MCH	Prefixs Group
1	0: test (09) 1: 2: 3: 4: 5: 6: 7: 8: 9: 10: 11: 12: 13: 14: 15:

At the bottom right of the table, there are two buttons: "submit" and "reset".

Prefix Settings

Group Enable: ON OFF

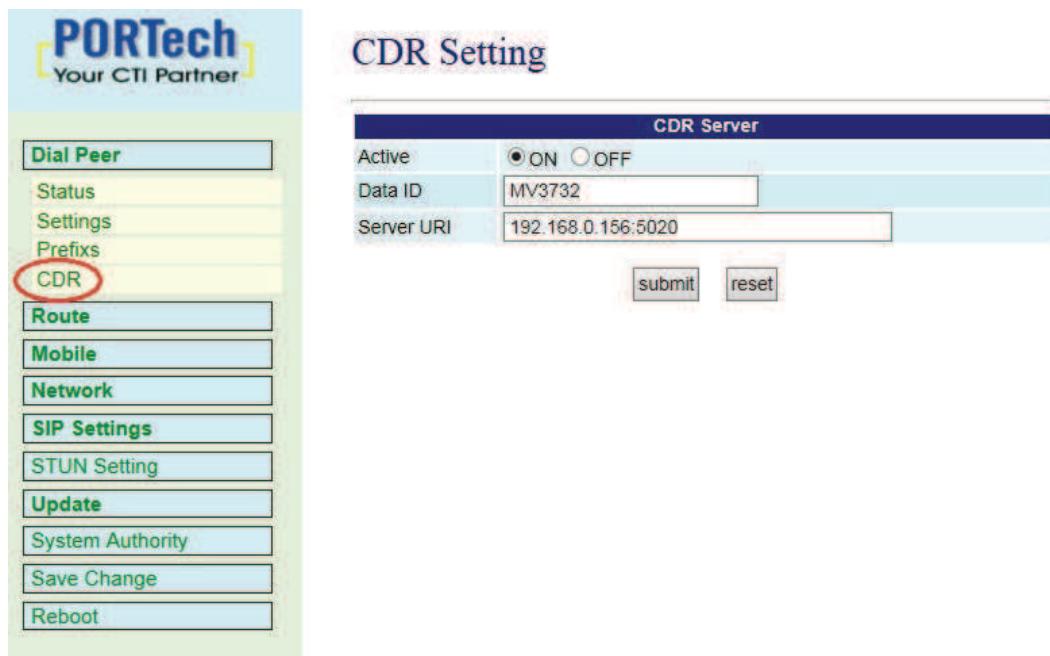
Group	Name	Prefixs
0	test	09
1		
2		
3		
4		
5		
6		
7		
8		
9		
10		
11		
12		
13		
14		
15		

2. Group: The group number, total is 15 sets
3. Name: Fill the name of the group
4. Prefixs: Fill the local area number or prefix numbers of the group

After all settings are done, please click submit button.

8.4 Call Data to Server (CDR)

It 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.



1. Data ID: MV will create one default Data ID
2. Server URL: Fill the IP and port of the CDR server

After the setting, please click Submit and save change button to wait for system reboot

External Dial Peer

You can check CDR Statue here

Dial Peer - (Apr 19 2011, 15:55:33)											
File Help											
Log	Status	Set	Event								
1	Mv-000000	7	466922102862581								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

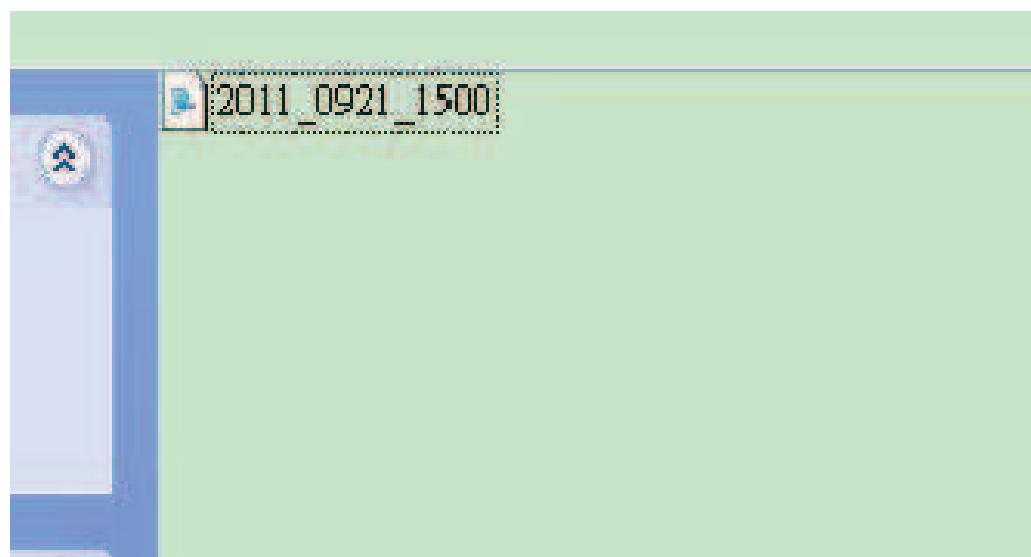
CDR Files store at C:\Program Files\.DialPeer

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

9.1 Mobile TO LAN Settings

User can assign the routing rule to transfer the call incoming on MOBILE to LAN

The screenshot shows the PORTech software interface. On the left, there is a vertical menu bar with various options: Dial Peer, Route (which is circled in red), Mobile To Lan Settings (also circled in red), Mobile To Lan Speed Dial, Lan To Mobile Settings, Mobile, Network, SIP Settings, STUN Setting, Update, System Authority, Save Change, and Reboot. The main window is titled "Mobile to LAN table". It contains a table with columns labeled "MCH", "CID", "URL", and "SEL". A red arrow points to the "CID" column header. The table has 16 rows, each corresponding to a number from 1 to 16. The "CID" column for row 1 contains an asterisk (*). The "URL" column for row 1 also contains an asterisk (*). The "SEL" column for row 1 contains an empty checkbox.

MCH	CID	URL	SEL
1	*	*	<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>
10			<input type="checkbox"/>
11			<input type="checkbox"/>
12			<input type="checkbox"/>
13			<input type="checkbox"/>
14			<input type="checkbox"/>
15			<input type="checkbox"/>
16			<input type="checkbox"/>

Please move the mouse to that red arrow spot and click
It will show the setting bLANK. After the setting, please click Entry.

The screenshot shows a web-based configuration interface for PORTech. On the left, a sidebar lists various menu items: Dial Peer, Route, Mobile, Network, SIP Settings, STUN Setting, Update, System Authority, Save Change, and Reboot. The main area is titled 'Mobile to LAN table'. It contains a table with columns for MCH, CID, URL, and SEL. Row 1 has entries '1' and '*' in the CID field, and '*' in the URL field. Buttons for 'Entry' and 'cancel' are located below row 1. Rows 2 through 16 are empty. A red oval highlights the CID and URL fields for entry 1.

MCH	CID	URL	SEL
1	1 *	*	<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>
10			<input type="checkbox"/>
11			<input type="checkbox"/>
12			<input type="checkbox"/>
13			<input type="checkbox"/>
14			<input type="checkbox"/>
15			<input type="checkbox"/>
16			<input type="checkbox"/>

1. MCH: the code of mobile channel
2. 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

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.
3. 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 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.

4. SEL: Select the one to delete

9.2 Mobile to LAN Speed Dial Settings

NOTE: It's for 2 stage dialing mode

The screenshot shows the PORTech mobile configuration interface. On the left, there is a sidebar with various menu items: Dial Peer, Route, Mobile To Lan Settings (which is circled in red), Mobile To Lan Speed Dial (also circled in red), Lan To Mobile Settings, Mobile, Network, SIP Settings, STUN Setting, Update, System Authority, Save Change, and Reboot. The main area is titled "Mobile To Lan Speed Dial". It features a dropdown menu "Mobile 1, 2" and a table with columns: Item, Name, URL, and Select. The table contains 10 rows, each with a checkbox in the "Select" column. Row 0 has "JACK" in the "Name" column and "192.168.0.156" in the "URL" column. Below the table are buttons for "Delete Selected", "Delete All", and "Reset". At the bottom, there is a section titled "Add New" with fields for Position (containing "(0~9)"), Name, and URL, along with "Add" and "Reset" buttons.

Item	Name	URL	Select
0	JACK	192.168.0.156	<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

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

E.g. item: 0 Name: JACK URL: 192.168.0.156,

When the caller hear dial tone and enter 0, system will connect 192.168.0.156

9.3 LAN to Mobile Settings

User can assign 24 sets of routing rule to transfer the call incoming from LAN to MOBILE. The chart setting is used for all channels.

The screenshot shows the PORTech software interface. On the left, there is a sidebar with various menu items: Dial Peer, Route, Mobile To Lan Settings, Mobile To Lan Speed Dial, Lan To Mobile Settings (which is circled in red), Mobile, Network, SIP Settings, STUN Setting, Update, System Authority, Save Change, and Reboot. The main area is titled 'LAN to Mobile table' and contains a table with 16 rows. The table has columns for 'No.', 'URL', 'Call Num', and 'SEL'. The 'URL' column for the first row contains an asterisk (*) with a red arrow pointing to it. The 'Call Num' column for the first row contains a hash (#). The 'SEL' column for the first row has an empty checkbox.

No.	URL	Call Num	SEL
1	*	#	<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>
10			<input type="checkbox"/>
11			<input type="checkbox"/>
12			<input type="checkbox"/>
13			<input type="checkbox"/>
14			<input type="checkbox"/>
15			<input type="checkbox"/>
16			<input type="checkbox"/>

Please move the mouse to that red arrow spot and click
It will show the setting bLANK. After the setting, please click Entry.

No.	URL	Call Num	SEL
1	1 * #		<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>
10			<input type="checkbox"/>
11			<input type="checkbox"/>
12			<input type="checkbox"/>
13			<input type="checkbox"/>
14			<input type="checkbox"/>
15			<input type="checkbox"/>
16			<input type="checkbox"/>
17			<input type="checkbox"/>
18			<input type="checkbox"/>
19			<input type="checkbox"/>
20			<input type="checkbox"/>
21			<input type="checkbox"/>
22			<input type="checkbox"/>
23			<input type="checkbox"/>
24			<input type="checkbox"/>

1. No. : The code number

2. 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.

3. 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-3716/MV-3732 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-3716/MV-3732 will connect this call auto.

4.SEL : Select the one to delete

10. Mobile

10.1 Mobile Status

Operator:	46692: Chunghwa Telecom
SIM Card ID:	466922102862553
Signal Quality:	20
Registration State:	0.1
GSM S/N:	862170016493106
Motion State:	Standby
Incoming URL:	(empty)
Incoming Name:	(empty)
Outgoing IP:	(empty)
Incoming Mob:	(empty)
Outgoing Mob:	(empty)

- (1)Choose Mobile 1,2,3 or 4 (MV-3732: 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 outgoing call to MOBILE.

10.2 Mobile Setting

Mobile Setting

Mobile 1, 2 ▾

VoIP Tx Gain	9 (0~12)	VoIP Rx Gain	11 (0~15)
LAN Dialtone Vol	4 (0~12)		

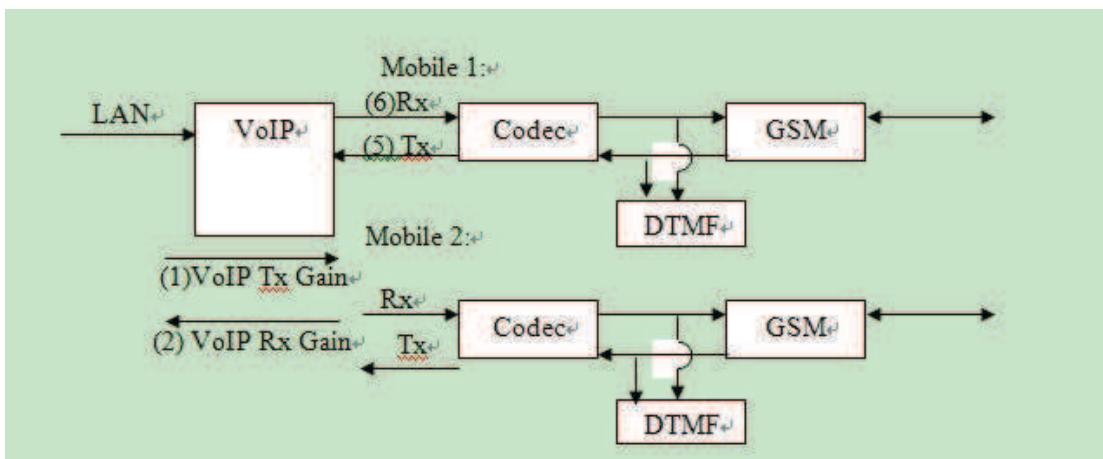
Mobile 1 ON OFF

Routing Range	0 ~ 24	CODEC Rx Gain	6 (0~7)
CODEC TX Gain	6 (0~7)	SIP From:	Tel/User (Standard) ▾
SIP From:	Tel/User (Standard) ▾	Answer delay	0 (0~15)
Hide Caller ID	<input type="radio"/> ON <input checked="" type="radio"/> OFF	Restart dial fails	1 (0~15)
PIN Code	On <input type="checkbox"/> Code: []	Confirm: []	
Dial Prefix	[]	LAN Answer Mode	Answered ▾
Init AT Cmd	[]		

Mobile 2 ON OFF

Routing Range	25 ~ 49	CODEC Rx Gain	6 (0~7)
CODEC TX Gain	6 (0~7)	SIP From:	Tel/User (Standard) ▾
SIP From:	Tel/User (Standard) ▾	Answer delay	0 (0~15)
Hide Caller ID	<input type="radio"/> ON <input checked="" type="radio"/> OFF	Restart dial fails	1 (0~15)
PIN Code	On <input type="checkbox"/> Code: []	Confirm: []	
Dial Prefix	[]	LAN Answer Mode	Answered ▾
Init AT Cmd	[]		

SubmitAll Submit Reset



- (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: To adjust dial tone gain down of LAN.

(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-3716/MV-3732 (page 42)

MV-3716/MV-3732 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-3716/MV-3732 will send the message as follows in the Packet.

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

- Tel/Tel :

MV-3716/MV-3732 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-3716/MV-3732 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-3716/MV-3732, you can click "On" and enter pin code.

(12) Dial Prefix: The prefix number of outgoing calls. When LAN to Mobile, MV-3716/MV-3732 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.

After the setting, please click Submit and save change button to wait for system reboot

You can click Submit All to copy to Mobile setting, and select Yes and save change to wait for the system reboot

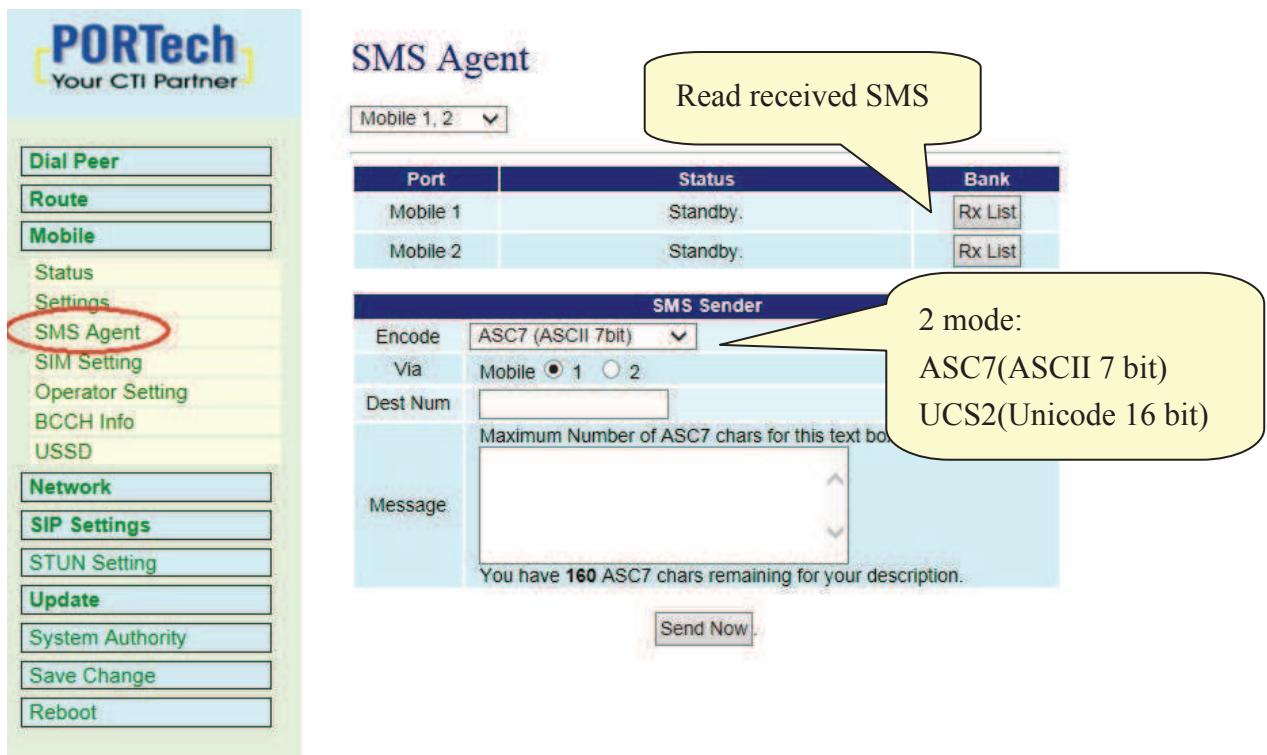
Please check below:

Mobile Setting

Mobile 1, 2 ▼

VoIP Tx Gain	9 (0~12)	VoIP Rx Gain	11 (0~15)
LAN Dialtone Vol	4 (0~12)		
Mobile 1 <input checked="" type="radio"/> ON <input type="radio"/> OFF			
Routing Range	0 ~ 24	CODEC Rx Gain	6 (0~7)
CODEC Tx Gain	6 (0~7)	Answer delay	0 (0~15)
SIP From:	Tel/User (Standard) ▼	Restart dial fails	1 (0~15)
Hide Caller ID	<input type="radio"/> ON <input checked="" type="radio"/> OFF	PIN Code	On <input type="checkbox"/> Code: <input type="text"/> Confirm: <input type="text"/>
Dial Prefix	<input type="text"/>	LAN Answer Mode	Answered ▼
Init AT Cmd	<input type="text"/>		
Mobile 2 <input checked="" type="radio"/> ON <input type="radio"/> OFF			
Routing Range	25 ~ 49	CODEC Rx Gain	6 (0~7)
CODEC Tx Gain	6 (0~7)	Answer delay	0 (0~15)
SIP From:	Tel/User (Standard) ▼	Restart dial fails	1 (0~15)
Hide Caller ID	<input type="radio"/> ON <input checked="" type="radio"/> OFF	PIN Code	On <input type="checkbox"/> Code: <input type="text"/> Confirm: <input type="text"/>
Dial Prefix	<input type="text"/>	LAN Answer Mode	Answered ▼
Init AT Cmd	<input type="text"/>		
<input type="button" value="SubmitAll"/> <input type="button" value="Submit"/> <input type="button" value="Reset"/>			

10.3 Mobile / SMS Agent:



1. Port: The GSM Channel No.
2. Status:
 - a. Standby: The GSM Channel is ready and idle for SMS sending
 - b. Not Ready: The GSM Channel is not registered or engaged, not able to send SMS
3. Encode : ASC7(ASCII 7 bit) or UCS2(Unicode 16 bit)
4. Via : To select the GSM Channel for SMS sending
5. Dest Num: the Receiver's phone number
6. Message: Please fill the message that wants to send to receiver.

After typing the SMS, please click Send Now button

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

SMS Rx List

Mobile 1 ▾

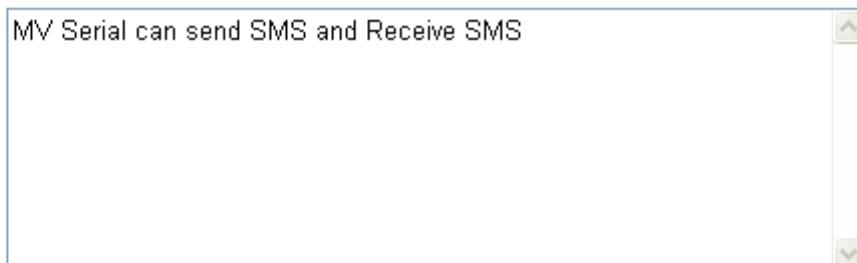
Read	Status	Caller ID	Date, Time
	REC READ	886935386862	08/05/15,15:41:46

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

SMS Reader

Index	RemoteID	Date, Time
1	886935386862	08/05/15, 15:41:46

MV Serial can send SMS and Receive SMS



The scrollable text area displays the message content: "MV Serial can send SMS and Receive SMS". It includes scroll bars on the right side.

10.4 Send Bulk of SMS via Microsoft Excel

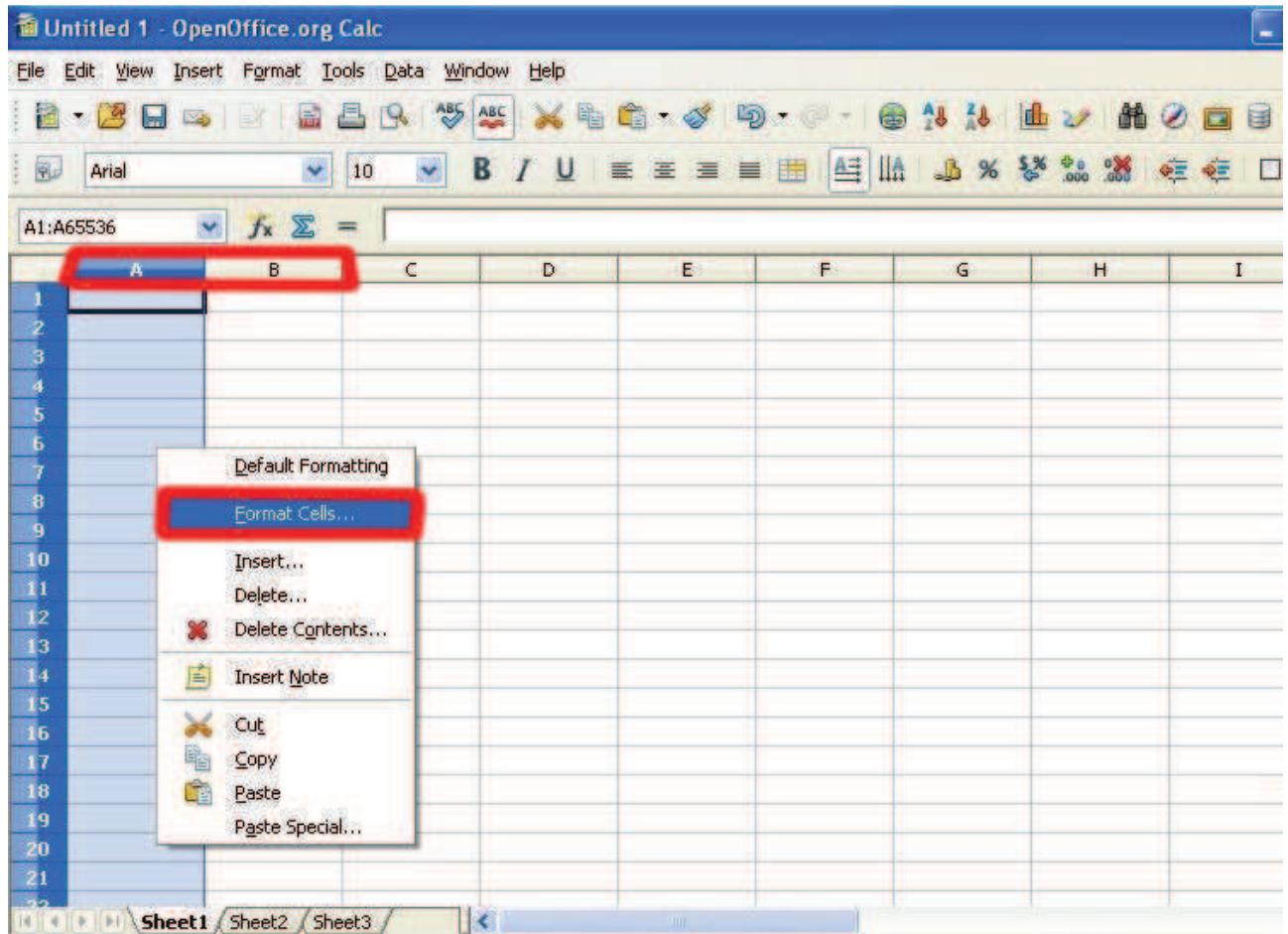
First of all, please open a new Excel file.

Step 1 Format Cells

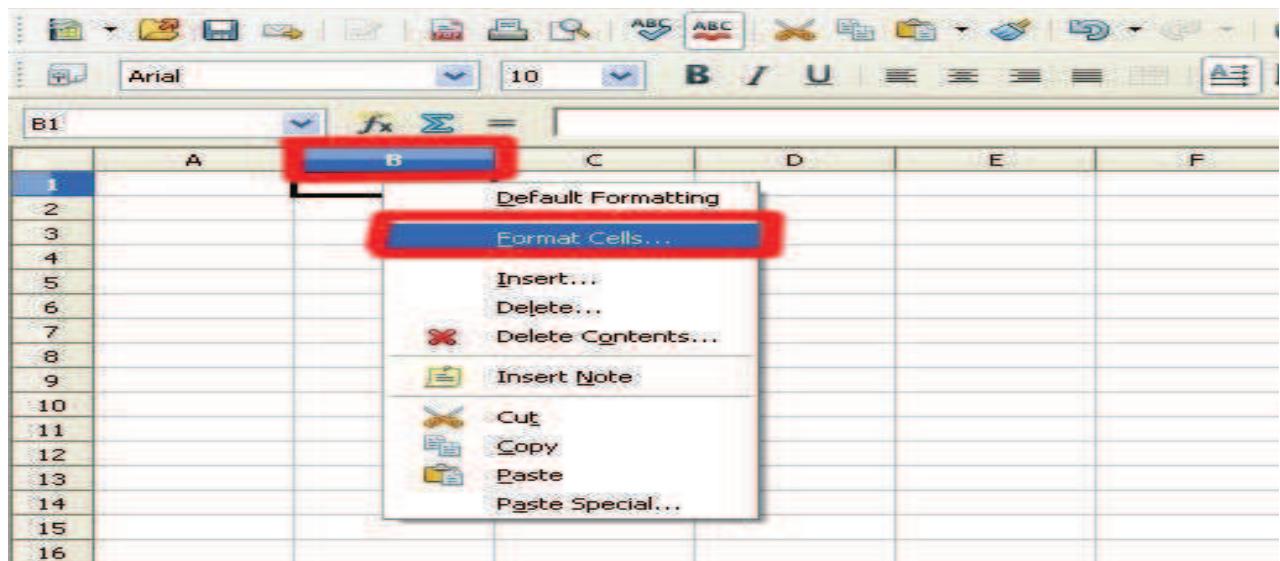
Here, we need you to format cells to “Text” first.

Please click mouse right key, and choose “Format Cells”

BLANK A

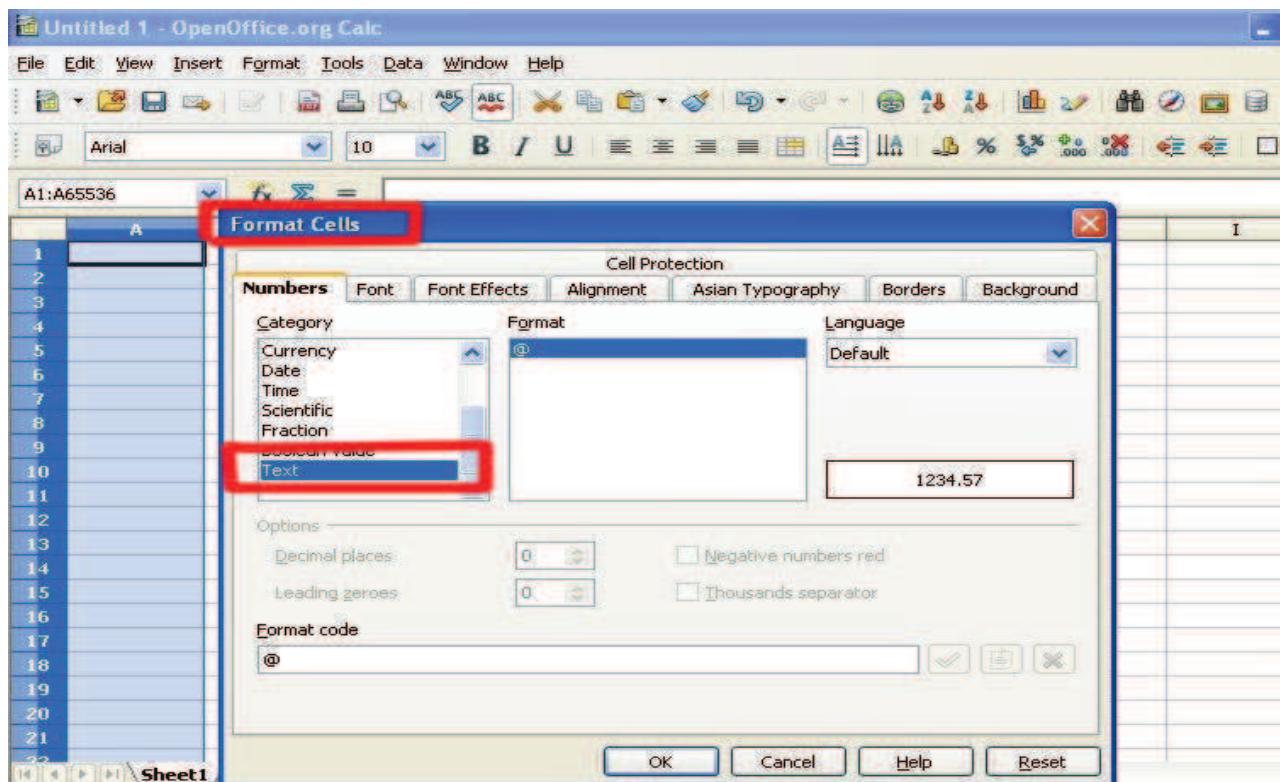


BLANK B



Step 2

In the Format Cells, please select "Text"



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

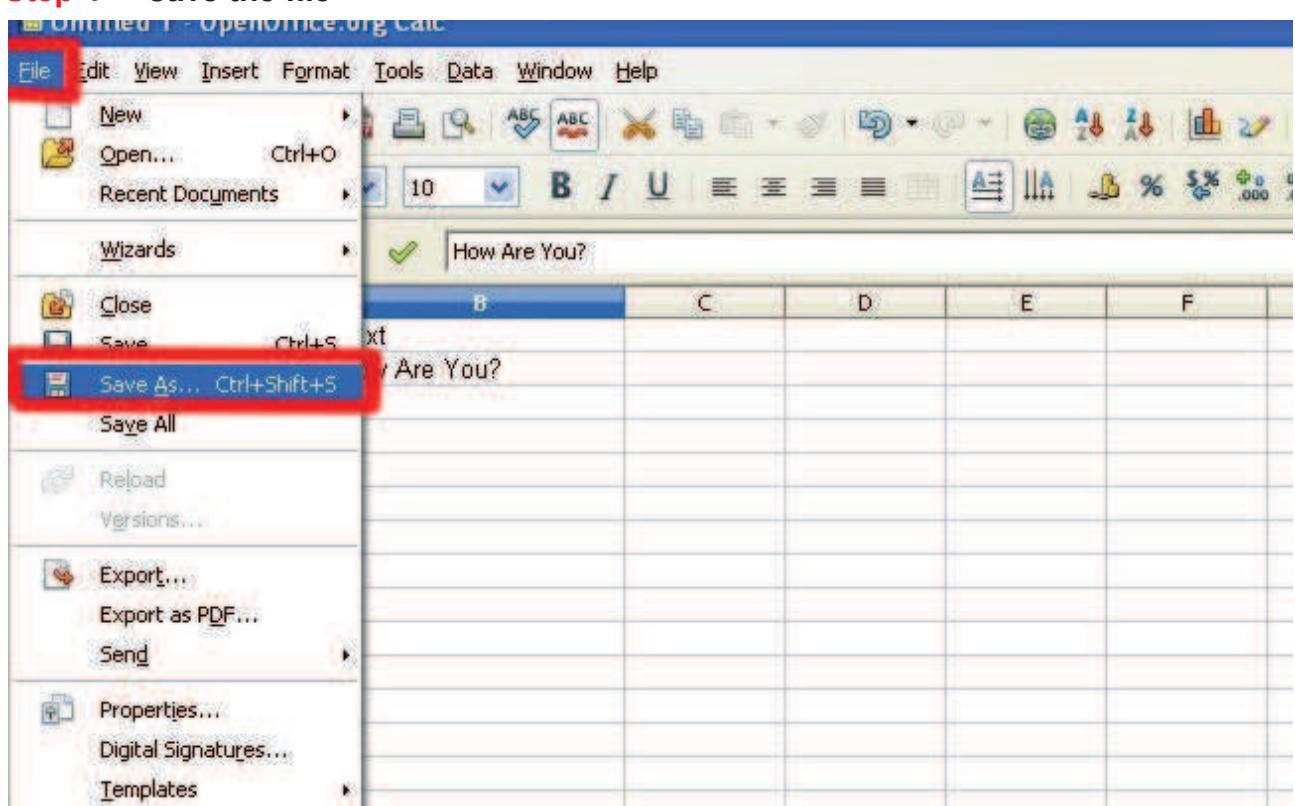
Step 3

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

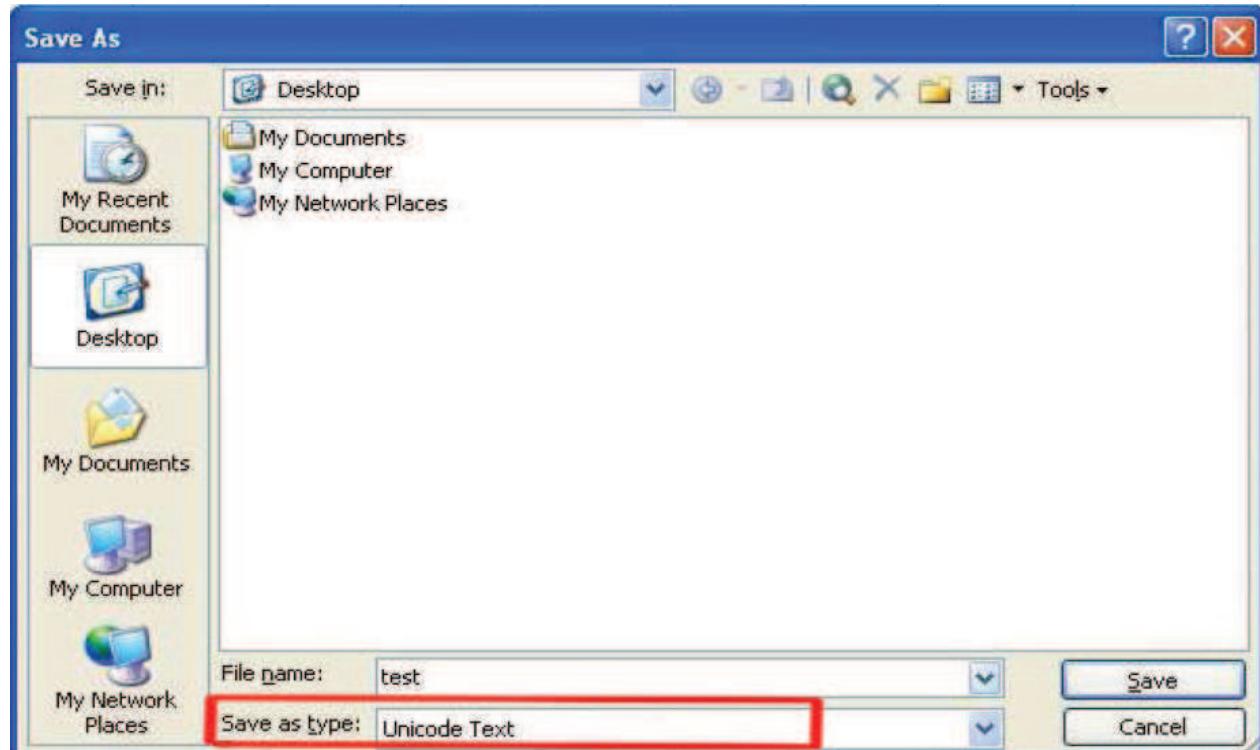
BLANK B: is for you to key “text”

B10	A	B	C	D	E	F	G	H	I
1	0988888888	How Are You?							
2									
3									
4									
5									
6									
7									
8									
9									
10									

Step 4 save the file

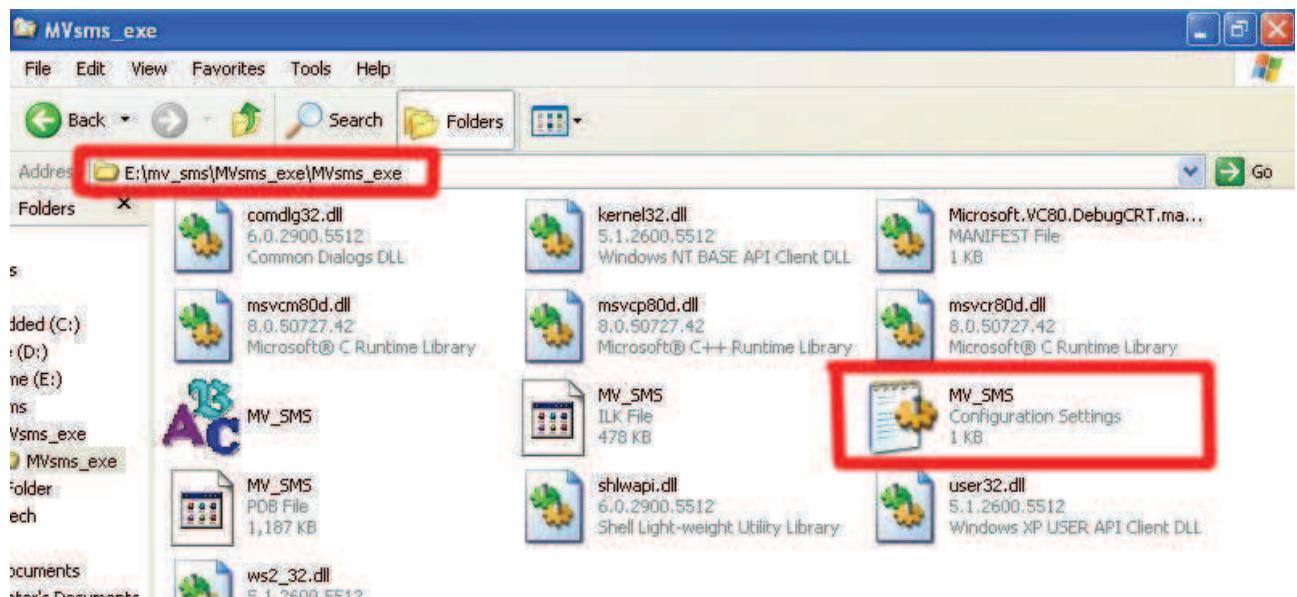


Save the type as “**Unicode Text**”



Step 5

Open MVsms_exe → MV-SMS (Configuration Settings)





Step 6

Please do the configuration as following:

MV-3732



```
MV_SMS - Notepad
File Edit Format View Help
[info]
Total=4
[VOIP]
1=192.168.0.100
2=192.168.0.100
3=192.168.0.100
4=192.168.0.100
[PORT]
1=23
2=8023
3=8123
4=8223
[USER]
1=voip
2=voip
3=voip
4=voip
[PASS]
1=1234
2=1234
3=1234
4=1234
```

MV-3716



```
MV_SMS - Notepad
File Edit Format View Help
[info]
Total=2
[VOIP]
1=192.168.0.100
2=192.168.0.100
[PORT]
1=23
2=8023
[USER]
1=voip
2=voip
[PASS]
1=1234
2=1234
```

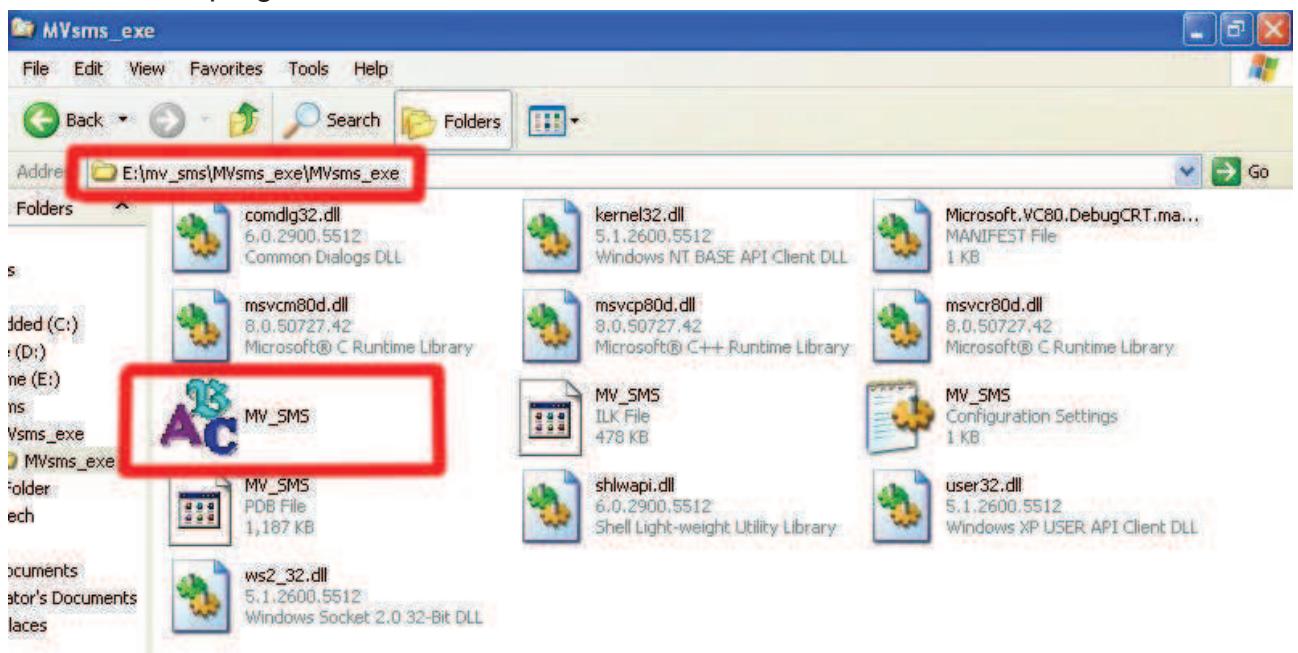
MV-372 & MV-370



```
[info]
Total=4
[VOIP]
1=192.168.0.100
[PORT]
1=23
[USER]
1=voip
[PASS]
1=1234
```

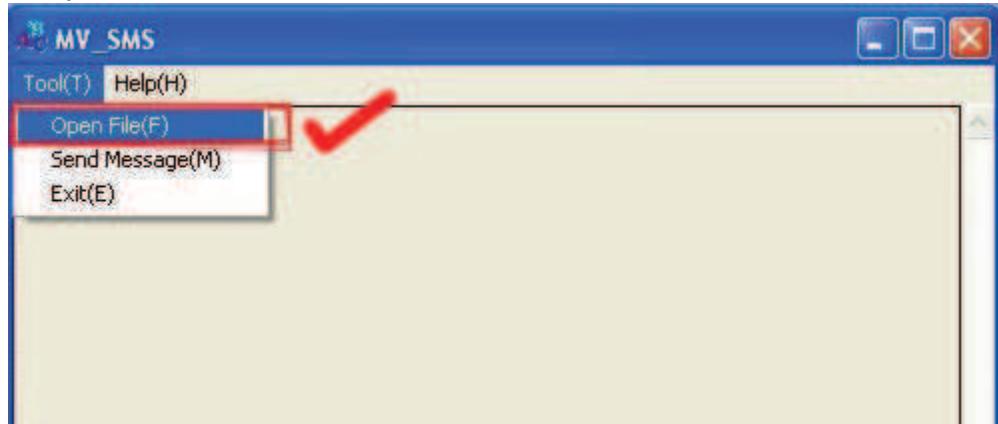
Step 7

Run MV-SMS program

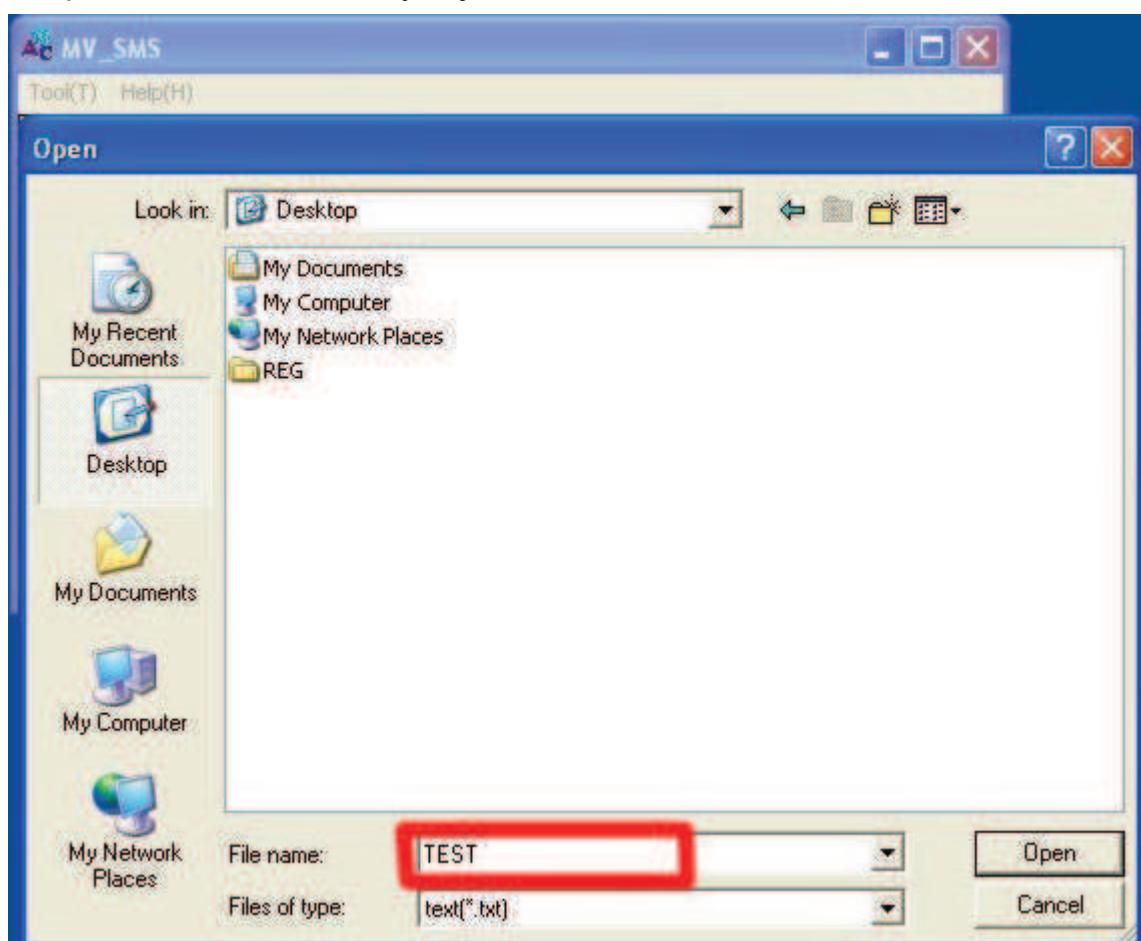


Step 8

1. Open File

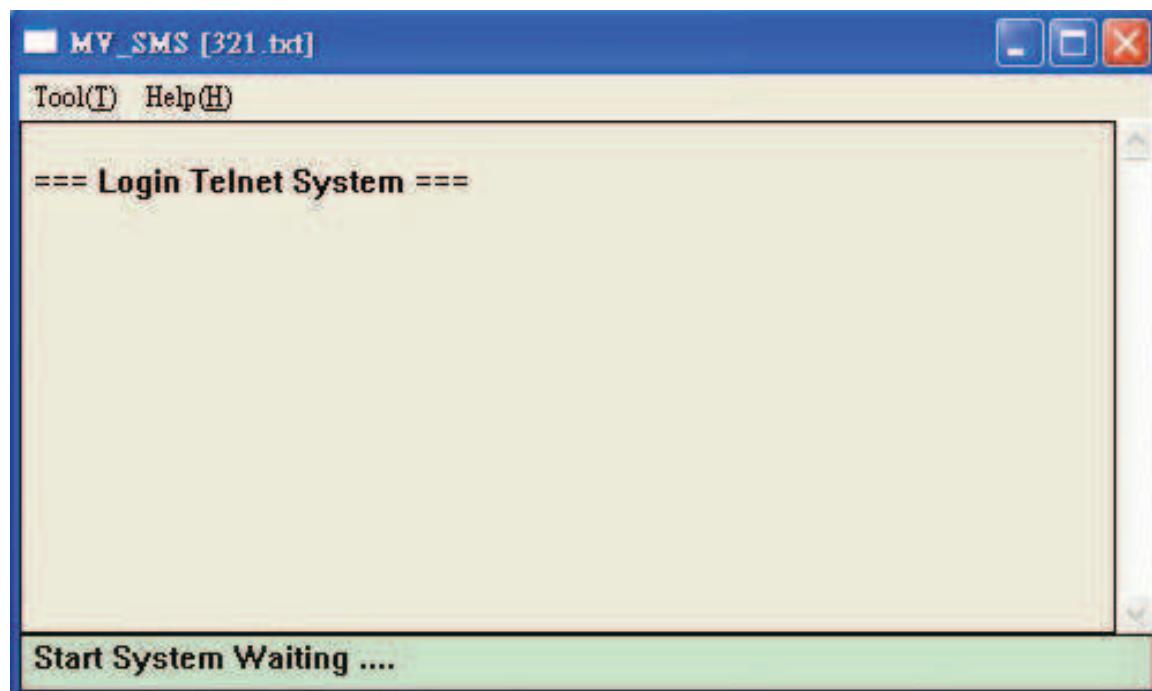


2. Open the "Excel file" that you just saved



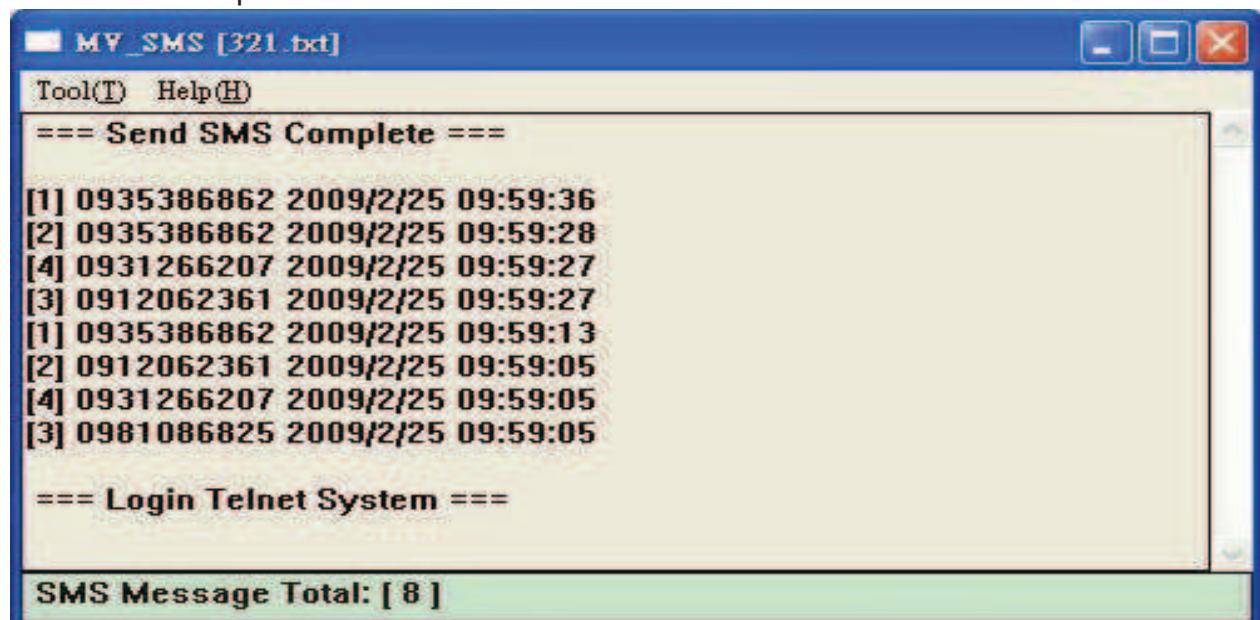
Step 9

Sending



Step 10

Send SMS Complete



10.5 Use AT Command via Telnet or your program

Allows your program or Telnet Send/receive SMS with AT Command

Telnet PORT Corresponding port as follows:
(2 modules in one SLAVE)

SLAVE 1:1301

SLAVE 2:1302

SLAVE 3:1303

SLAVE 4:1304

SLAVE 5:1305

SLAVE 6:1306

SLAVE 7:1307

SLAVE 8:1308..... *MV-3716

SLAVE 9:1309

SLAVE 10:1310

SLAVE 11:1311

SLAVE 12:1312

SLAVE 13:1313

SLAVE 14:1314

SLAVE 15:1315

SLAVE 16:1316..... *MV-3732

```
username: voip  
password: ****  
user level = 1.  
  
command: logout, module, module1, module2.  
>module1  
getting module 1 ... Choose module  
got!! press 'ctrl-x' to release module 1.  
  
0  
ate1 Enter “ate1”,then you can see  
0 your at command below  
at+cmgf=1  
0  
at+cmgs="0911123456" Enter at+cmgs="phone number"  
>  
test  
> Enter short message and ctrl+Z  
+CMGS: 30  
0
```

10.6 USSD SIM BaLANce Check via Telnet

```
username: voip
password: ****
user level = admin.

command: logout, module1, module2, state1, state2, info.
lmodule1
getting module 1 ...
got!! press 'ctrl-x' to release module 1.
0
at+cusd=1, "*145*11#",15
0
+CUSD: 2, "Accepted",0
0
release module 1 ...
```

The screenshot shows a Telnet session window titled "5218 - 超級終端機". The session content is as follows:

```
username: voip
password: ****
user level = admin.

command: logout, module1, module2, state1, state2, info.
lmodule1
getting module 1 ...
got!! press 'ctrl-x' to release module 1.
0
at+cusd=1, "*145*11#",15
0
+CUSD: 2, "Accepted",0
0
release module 1 ...
```

Annotations with arrows point to specific lines:

- An arrow points from the text "1. USSD Request" to the line "at+cusd=1, "*145*11#",15". This line is circled in red.
- An arrow points from the text "2. Module command" to the line "0".

At the bottom of the window, there is a status bar with the text: 連線 00:01:43 ANSIW TCP/IP SCROLL CAPS NUM 滾 | 列印 |

1. USSD Request: Please enter USSD code for your operator to check baLANce
 - 2.
 3. Module command:
Please enter “15” for Siemens BG2W module
Please enter “0” for Simcom module
- ! User can check this information on main page on **Module Description****

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

The screenshot shows the PORTech SMS Reader interface. On the left, there is a navigation menu with the following items:

- Route
- Mobile
 - Status
 - Settings
 - Fwd Settings
 - SMS Agent
 - SIM Setting
 - Operator Setting
- Network
- SIP Settings
- STUN Setting
- Update
- System Authority
- Save Change
- Reboot

The main area is titled "SMS Reader". It displays a table with the following data:

Index	RemoteID	Date, Time
2	01145009310000990016	11/08/26, 15:24:43

The message content is highlighted with a red oval and contains the following text:

帳單金額NT\$1836.0
付款期限8/28
累計未付金額NT\$1836.0
劃撥帳號19037959
帳單號碼4046247121

At the bottom right of the message area, there are "Back" and "Delete" buttons.

10.7 SIM Setting

The screenshot shows the PORTech SIM Card Setting interface. The left sidebar contains a navigation menu with various options like Dial Peer, Route, Mobile, SMS Agent, Operator Setting, BCCH Info, USSD, Network, SIP Settings, STUN Setting, Update, System Authority, Save Change, and Reboot. The 'Mobile' option is expanded, and 'SIM Setting' is highlighted with a red circle. The main right panel is titled 'SIM Card Setting' and displays two sections: 'SIM Card of Mobile 1' and 'SIM Card of Mobile 2'. Each section includes fields for Mode (Local, Bank, Server), Mobile ID, Group, Card ID, Bank URL, Server URL, and Status. At the bottom of the right panel are three buttons: 'SubmitAll', 'Submit', and 'Reset'.

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-3732 that you can't use "888" on 2nd MV-3732, 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.

After the setting, please click submit and save change button and wait for system reboot

10.8 Operator Setting

PORTech
Your CTI Partner

Dial Peer
Route
Mobile
Status
Settings
SMS Agent
SIM Setting
Operator Setting (highlighted with a red circle)
BCCH Info
USSD
Network
SIP Settings
STUN Setting
Update
System Authority
Save Change
Reboot

Operator Setting

Mobile 1, 2 ▾

Mobile 1 :
Opreator ID: [] (0: resume auto)
Work Mode: Every time reset module Manual

Mobile 2 :
Opreator ID: [] (0: resume auto)
Work Mode: Every time reset module Manual

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.

Operator List				
Mobile 1				
No	Status	Name	ID	Use
00	Current	Chunghwa Telecom (CHT)	46692	<input type="radio"/>
01	Forbidden	Far EasTone (FET)	46601	<input type="radio"/>
02	Forbidden	Pacific GSM 1800 (TCC)	46697	<input type="radio"/>
03				<input type="radio"/>
04				<input type="radio"/>
05				<input type="radio"/>
06				<input type="radio"/>
07				<input type="radio"/>

Submit **Reset**

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.

After the setting, please click submit and save change button and wait for system reboot

10.9 BCCH Info

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).

NOTE: Support Quad band-BG2W, Quad band-M10 and firmware V10.185 above only.

select	MCC	LAC	Cell	BSIC	BCCH	RxLev
0	46692	0FAB	D3D2	14	31	-70
1	46692	0FAB	AC9D	10	30	-84
2	46692	0FAB	ACC2	11	49	-92
3	46692	0FAB	AC4E	14	28	-92
4	46692	0FAB	D3AD	14	34	-93
5	46692	0FAB	3790	8	572	-94
6	46692	0FAB	1140	10	43	-97

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.

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

2. System will show the cell number information once you select on Preferred this Cell form. Please click the submit button and Save Change, and wait for system reboot

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

	LAC	Cell ID	BCCH
<input checked="" type="checkbox"/> Preferred this Cell	0853	70AD	626

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.

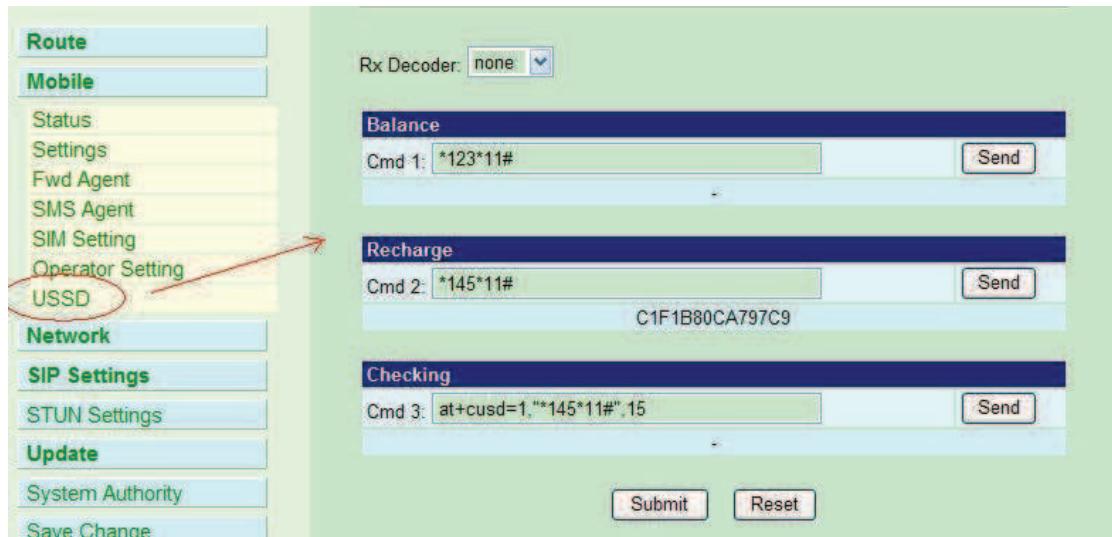
select	MCC	LAC	Cell	BSIC	BCCH	RxLev
0	46601	0853	70AD	61	626	-88
1	46601	0871	546F	20	629	-110
2	46601	0871	546E	20	661	-110
3	46601	0871	0000	23	513	-110
4	46601	0853	0000	61	532	-110
5	46601	0853	0000	23	656	-110
6	46601	0871	0000	27	667	-110

-
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

10.10 USSD (Unstructured Supplementary Service Data)

User 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.

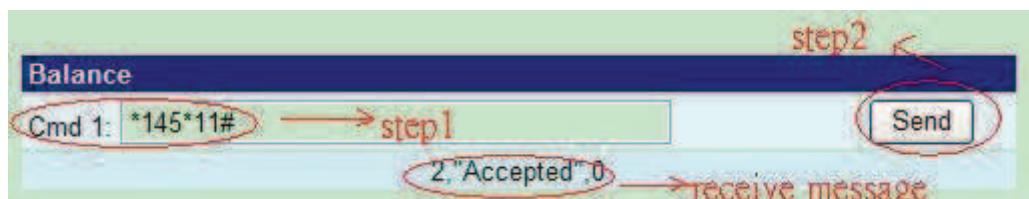


1. BaLANce (SIM baLANce remaining)

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

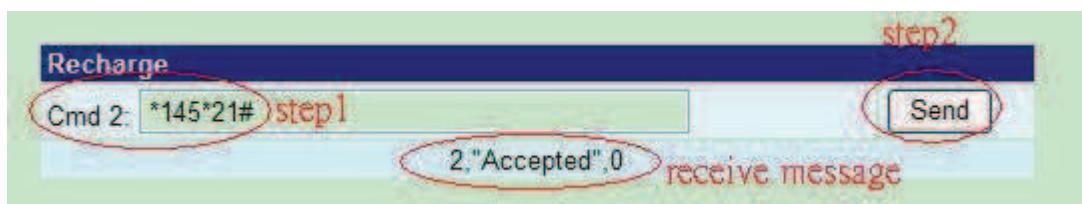


2. Recharge (add value)

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



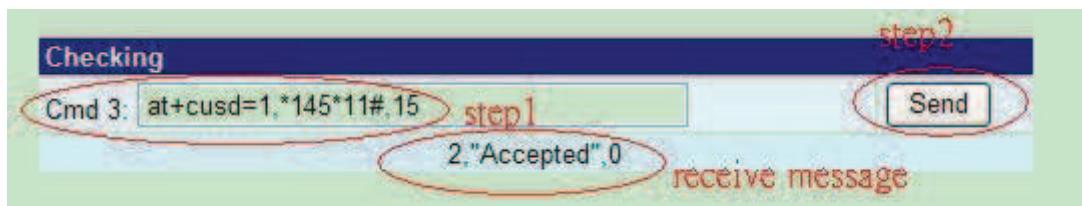
3. Checking (If above ways are failed, please select this)

Step 1: Enter the complete AT command in Cm3 column

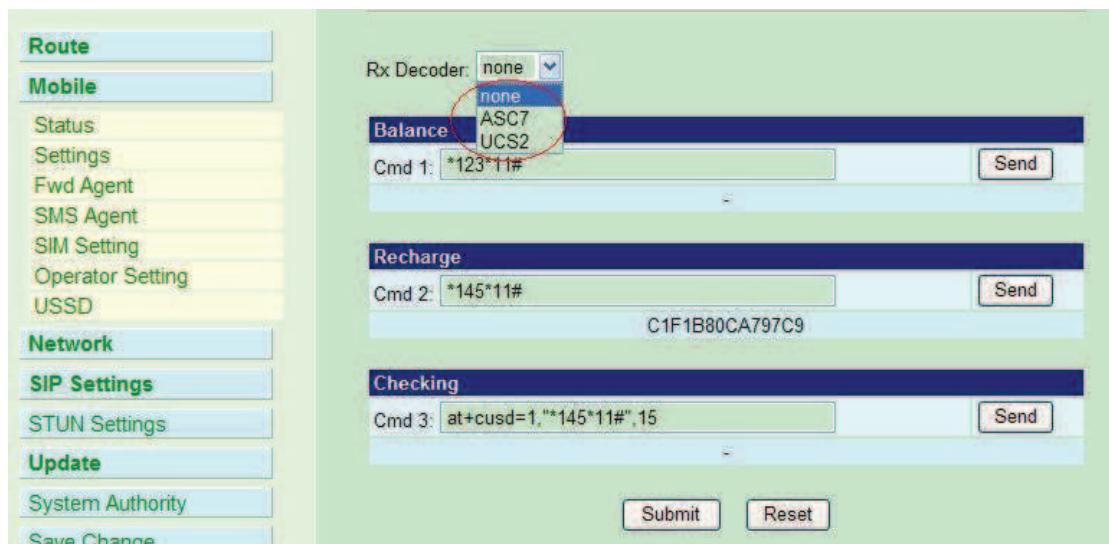
Ex. **AT+CUSD=1,*145*11#,15**

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

User can check the Network status and configure the WLAN Settings and SNTP settings.

11.1 WAN Setting

The screenshot shows the PORTech network configuration interface. On the left, there is a sidebar with the following menu items: Dial Peer, Route, Mobile, Network, WAN Settings (which is highlighted with a red oval), SNTP Settings, SIP Settings, STUN Setting, Update, System Authority, Save Change, and Reboot. The main area is titled "WAN Setting (RT)". It contains two sections: "WAN Settings" and "PPPoE Settings". The "WAN Settings" section includes fields for IP Type (radio buttons for Fixed IP, DHCP Client, and PPPoE, with Fixed IP selected), Main IP (192.168.0.98), Mask (255.255.255.0), Gateway (192.168.0.254), DNS 1 (168.95.192.1), DNS 2 (168.95.1.1), and MAC (00037E011BF2). The "PPPoE Settings" section includes fields for Username and Password. At the bottom right are "Submit" and "Reset" buttons.

1. IP Type
 - a. Fixed IP (Default IP: 192.168.0.100)
 - b. DHCP Client
 - c. PPPoE
2. Main IP: The current IP address. The IP chaning need to under the Fixed IP mode.
3. PPPoE Setting

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

After the setting, please click submit and save change button and wait for system reboot

11.2 SNTP Settings

User 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.

The screenshot shows the 'SNTP Settings' page of the PORTech software. On the left, there's a vertical menu with several tabs: Dial Peer, Route, Mobile, Network, WAN Settings, **SNTP Settings** (which is circled in red), SIP Settings, STUN Setting, Update, System Authority, Save Change, and Reboot. The main area is titled 'SNTP Settings'. It has two radio buttons for 'SNTP': 'On' (selected) and 'Off'. Below that are two input fields: 'Primary Server' containing 'time.windows.com' and 'Secondary Server' containing '208.184.49.9'. Further down are two sets of dropdowns for 'Time Zone': 'GMT' and '+08' (selected), followed by ':00' (selected). To the right of these are '(hh:mm)' and '(dd:hh:mm)' placeholder text. At the bottom are 'Submit' and 'Reset' buttons.

SNTP settings (Default: On)

After the setting, please click submit and save change button and wait for system reboot

12. SIP Setting

User 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.

Realm 1 (Default)	
Active:	<input type="radio"/> ON <input checked="" type="radio"/> OFF
Display Name:	
User Name:	
Register Name:	
Register Password:	
Domain Server:	
Proxy Server:	
Outbound Proxy:	
Status:	Not Registered

Realm 2	
Active:	<input type="radio"/> ON <input checked="" type="radio"/> OFF
Display Name:	
User Name:	
Register Name:	
Register Password:	
Domain Server:	
Proxy Server:	
Outbound Proxy:	
Status:	Not Registered

-
- (1) Active: On /OFF
 - (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) Status: Register or Not register

After the setting, please click submit and save change button and wait for system reboot

Example:

Register VoipBuster

Realm 1 (Default)	
Active:	<input checked="" type="radio"/> On <input type="radio"/> Off
Display Name:	jenny0922
User Name:	jenny0922 Your Voipbuster username
Register Name:	jenny0922
Register Password:	**** Your Voipbuster password
Domain Server:	
Proxy Server:	194.221.62.207 Proxy Server's IP
Outbound Proxy:	
Status:	Registered

12.2 Ports Setting

MCH	SIP Port (2000~59000)	RTP Port (2000~59000)
1	5064	20004
2	5066	20006
3	5068	20008
4	5070	20010
5	5072	20012
6	5074	20014
7	5076	20016
8	5078	20018
9	5080	20020
10	5082	20022
11	5084	20024
12	5086	20026
13	5088	20028
14	5090	20030
15	5092	20032
16	5094	20034
17	5096	20036
18	5098	20038

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

After the setting, please click Submit and save change button to wait for system reboot

12.3 Codec Settings:

User can setup the Codec priority, RTP packet length in this page.
Please follow the ISP suggestion to setup these items.

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

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

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

Voice VAD	
Voice VAD:	<input type="radio"/> On <input checked="" type="radio"/> Off

RTP Packet Length

1. G.711& G.729: Default is 20ms.
Range: 10ms,20ms,30ms,40ms,50ms,60ms,70ms,80ms,90ms
2. G.723: Default:
Range: 30ms ,60ms, 90ms

After the setting, please click Submit and save change button to wait for system reboot

12.4 Codec ID Setting

User can setup the Codec ID in this page.

After the setting, please click Submit and save change button to wait for system reboot

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Codec ID Setting

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

Service Domain
Port Settings
Codec Settings
Codec ID Settings
DTMF Settings
SIP Responses
Other Settings
STUN Setting
Update
System Authority
Save Change
Reboot

12.5 DTMF Setting

The screenshot shows the PORTech configuration interface. On the left, a sidebar lists various settings. The 'DTMF Settings' option is highlighted with a red circle. The main panel is titled 'DTMF Setting' and contains two configuration sections: 'DTMF Transfer Mobile to LAN' and 'Mobile DTMF Detection'. In the 'DTMF Transfer Mobile to LAN' section, the 'Format' dropdown is set to '2833'. In the 'Mobile DTMF Detection' section, the 'Duration' field is set to '-1' and the 'Debounce' field is set to '80'. At the bottom of the panel are 'Submit' and 'Reset' buttons.

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-3716/MV-3732 to detect sender's DTMF. If the parameter is 0, MV-3716/MV-3732 won't detect sender's DTMF. Parameter is 0~999 seconds. After that duration, MV-3716/MV-3732 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.

After the setting, please click Submit and save change button to wait for system reboot

12.6 SIP Responses

SIP Responses

2013-06-05 16:37

Mobile Busy Response	
Unavailable	486 Busy here
Ring Timeout	486 Busy here

SIP Ring Responses	
<input checked="" type="radio"/> ON	180 Ringing (Force to ON, if 183 was OFF.)
<input type="radio"/> OFF	183 Session Progress

submit reset

Mobile Busy Response

1. Unavailable: User can setup the SIP response code of LAN side while the call dial failed or in busy line
 - a. 486 Busy Here (Default)
 - b. 503 Service unavailable
 - c. 480 Temporarily unavailable
2. Ring Timeout: User can setup the response SIP code of LAN side while operators hang up the no answered calls
 - a. 486 Busy Here (Default)
 - b. 503 Service unavailable
 - c. 480 Temporarily unavailable

SIP Ring Response

1. 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)

2. 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.

After the setting, please click Submit and save change button to wait for system reboot

12.7 Other Settings

User can setup the Hold by RFC and QoS in this page. To change these settings, please follow your ISP information. 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.

The screenshot shows the PORTech web interface with a sidebar on the left containing various configuration options. The 'Other Settings' option is highlighted with a red circle. The main area is titled 'Other Setting' and contains the following configuration fields:

Mobile 1, 2	
Hold by RFC of Mobile 1	<input type="radio"/> ON <input checked="" type="radio"/> OFF
Hold by RFC of Mobile 2	<input type="radio"/> ON <input checked="" type="radio"/> OFF
Voice QoS:	40 (0~63)
SIP QoS:	40 (0~63)
SIP Expire Time:	60 (30~86400 sec)

At the bottom right are three buttons: 'SubmitAll', 'Submit', and 'Reset'.

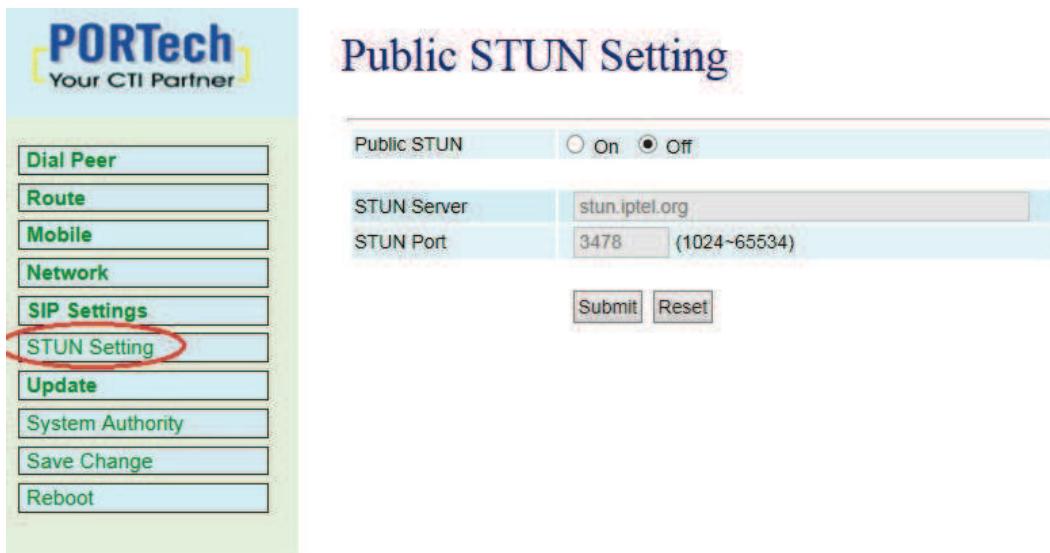
1. Hold RFC of Mobile:
 - a. On: To activate Hold RFC of Mobile
 - b. OFF (Default)
2. Voice QoS : The setting of Voice QoS, Default is 40
3. SIP QoS : The setting of SIP QoS, Default is 40
4. SIP Expire Time : The setting of SIP Expire Time, Default is 40

After the setting, please click Submit and save change button to wait for system reboot

You can click Submit All to copy to Mobile setting, and select Yes and save change to wait for the system reboot

13. STUN Setting

User 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. Please following your ISP information to change the settings



Public STUN OFF → Default is OFF; While the WAN setting of MV-3716/MV-3732 is in Static IP or Private IP please selects Public STUN OFF.

Public STUN ON → While MV-3716/MV-3732 is working under Firewall or behind NAT, It will cause SIP can't register, or one side communicate, please select Public STUN ON.

STUN Server → The STUN Server IP (Default: stun.uptel.org)

STUN Port → The STUN Port (Default: 3478)

After the setting, please click Submit and save change button to wait for system reboot

14. Update

14.1 Update Firmware

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

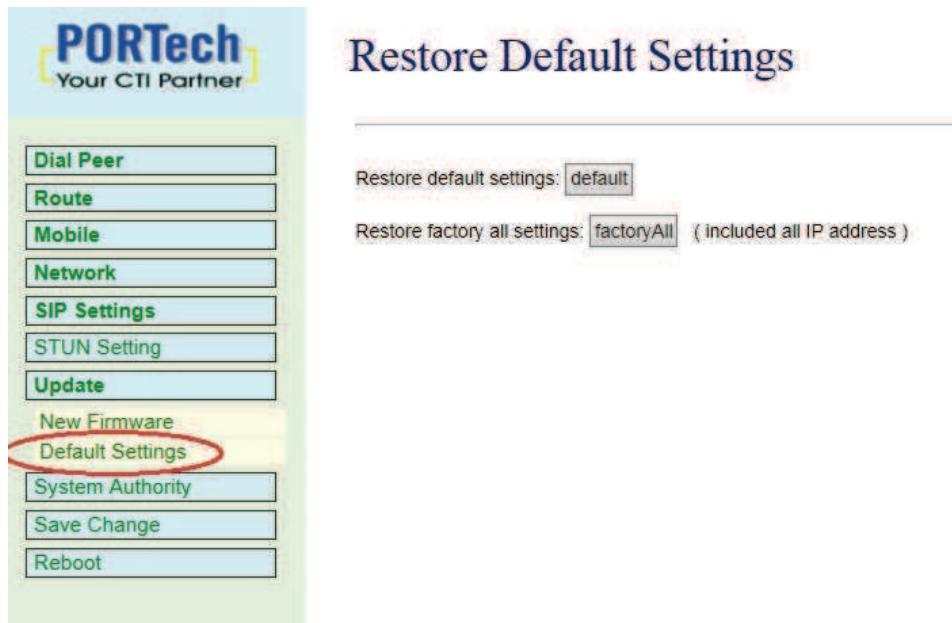
NOTE: Please open the webpage from Internet Explorer, not compatible with FF or Google Chrome

The screenshot shows the PORTech web interface. The top navigation bar includes links for Dial Peer, Route, Mobile, Network, SIP Settings, STUN Setting, Update, New Firmware (which is circled in red), Default Settings, System Authority, Save Change, and Reboot. The main content area is titled "Update Firmware" and displays the message "Ver = v10.272 , GZ = r4nat , PCB = 3748NAT .". Below this, there is an "HTTP" section with fields for "Code Type" (set to RISC) and "File Location" (with a "Browse..." button). At the bottom of the form are "Submit" and "Reset" buttons.

Step:

- (1) Select the firmware code type, Risc code only.
- (2) 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.
- (3) Select the correct file you want to download to the system then click the Update button.
- (4) Please click update/default setting after update firmware

14.2 Default Settings



1. Restore default settings: User can restore the factory default settings to the system. All setting will restore default setting.
The device IP still is the user original IP.
2. Restore factory all settings: All setting will be restored to default setting. The device IP will be back to 192.168.0.100

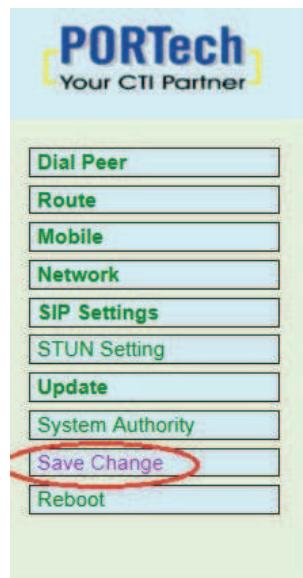
15. System Authority

User can change the login name and password

The screenshot shows a web-based configuration interface for PORTech. At the top left is the PORTech logo with the tagline "Your CTI Partner". On the left side, there is a vertical menu bar with several options: Dial Peer, Route, Mobile, Network, SIP Settings, STUN Setting, Update, System Authority, Save Change, and Reboot. The "System Authority" option is circled in red. The main content area is titled "System Authority". It contains three input fields: "New username:" with an empty input field, "New password:" with an empty input field, and "Confirmed password:" with an empty input field. Below these fields are two buttons: "Submit" and "Reset".

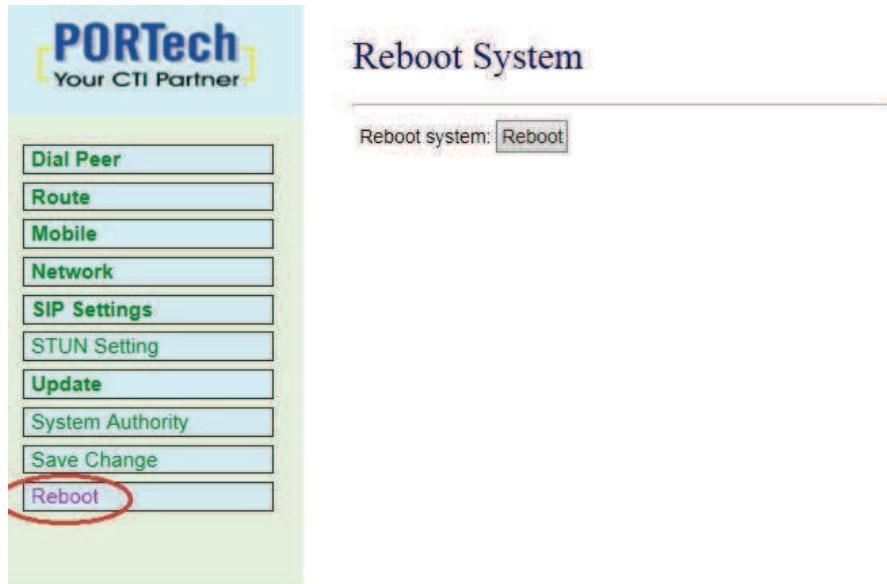
16. Save Change

User can save the changes after the setting is 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



17. Reboot

User 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-3716/MV-3732)

Quad Band: 900/1800/1900/850MHZ

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

The screenshot shows the PORTech web-based configuration interface. On the left, a sidebar menu includes: Route, Mobile, Status, Settings, Fwd Settings, SMS Agent, Network, SIP Settings, STUN Setting, Update, System Authority, Save Change, and Reboot. The main content area is titled "Mobile Setting". It features a dropdown menu "Mobile 1, 2" and several configuration fields. A callout bubble highlights the "SIP From" dropdown, which is currently set to "Tel/Tel (Not Reg)". The bubble contains the text: "Asterisk want to transfer CLID, please choose Tel/Tel (Not Reg)". Other visible fields include: VolP Tx Gain (set to 9), VolP Rx Gain (set to 11), LAN Dialtone Vol (set to 9), Routing Range (set to 0 to 49), CODEC Tx Gain (set to 6), CODEC Rx Gain (set to 6), SIP From (highlighted), Answer Delay (set to 0), CLID Presentation (set to Invocation), Mobile PIN Code (checkboxes for On, Code, Confirmed), and LAN Answer Mode (radio buttons for Answered, Alerted, Income).

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:	<input checked="" type="radio"/> ON <input type="radio"/> OFF
Display Name:	[Text Box]
User Name:	[Text Box]
Register Name:	[Text Box]
Register Password:	[Text Box]
Domain Server:	192.168.0.192:5060
Proxy Server:	192.168.0.192:5060
Outbound Proxy:	[Text Box]
Status:	Not Registered

Can register Asterisk or not

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- 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	<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>



LAN To Mobile Table

Mobile 1, 2

Page: 1

As Asterisk GSM Route

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

PORTech
Your CTI Partner

Dial Peer

Status
Settings

Route

Mobile

Network

SIP Settings

STUN Setting

Update

System Authority

Save Change

Reboot

Dial Peer Setting

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 (Dial Peer for XP)

Submit

Reset

PORTech
Your CTI Partner

Ports Setting

Internal Dial Peer Port: (1024~19900)

	SIP Port (1024~19900)	RTP Port (20000~59900)
Mobile 1	5064	20004
Mobile 2	5066	20006
Mobile 3	5068	20008
Mobile 4	5070	20010
Mobile 5	5072	20012
Mobile 6	5074	20014
Mobile 7	5076	20016
Mobile 8	5078	20018

Route
Mobile
Network
SIP Settings
Service Domain
Port Settings
Codec Settings
Codec ID Setting
DTMF Setting
RPort Setting
SIP Responses
Other Settings
STUN Setting
Update
System Authority

Don't forget to Save changes and then reboot

Asterisk / Trixbox setting

Add SIP Trunk:

Edit SIP Trunk

Delete Trunk SIM1

In use by 1 route

General Settings

Outbound Caller ID:

0911111111

Type your mobile
number

Never Override CallerID:

Maximum channels:

8

MV-374: 4
MV-378: 8

Outgoing Dial Rules

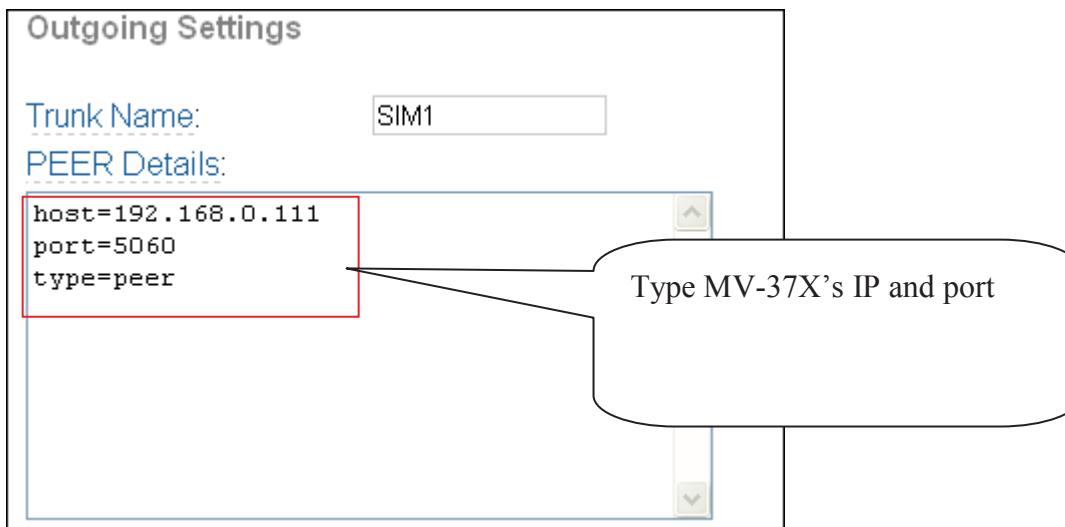
Dial Rules:

	<input type="button" value="^"/>
	<input type="button" value="▼"/>
Clean & Remove duplicates	

Dial rules wizards:

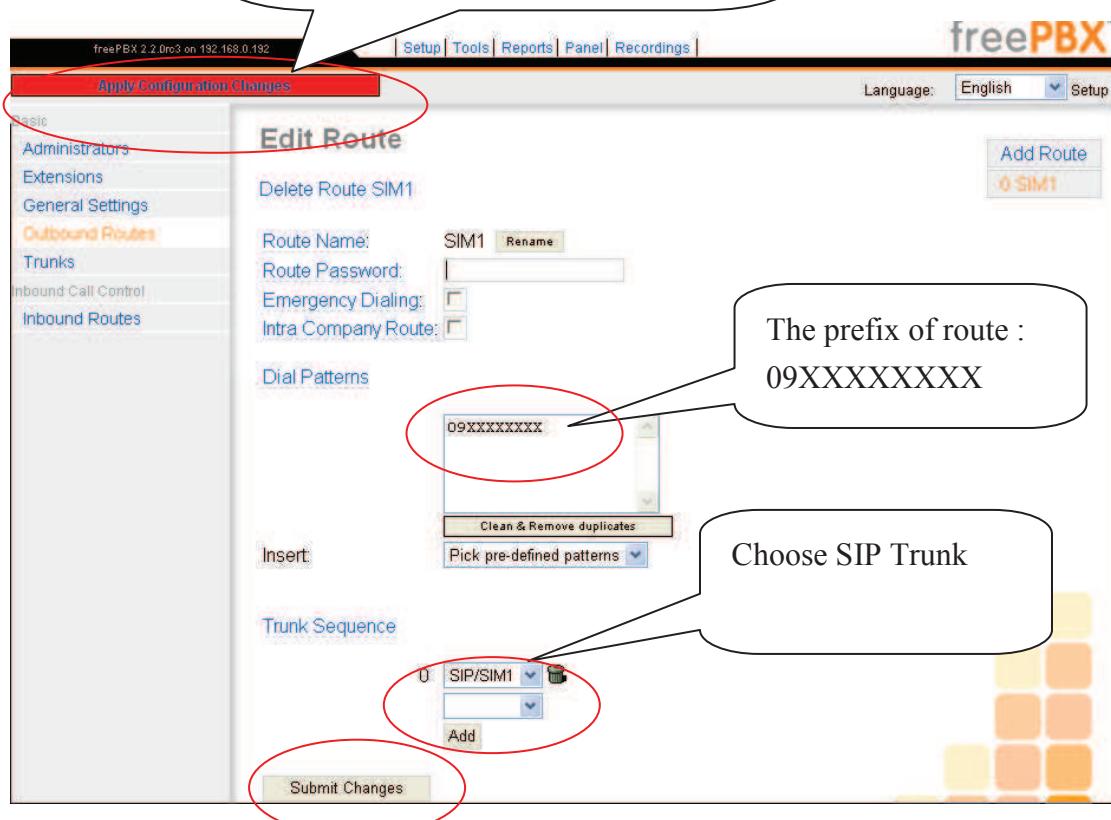
(pick one)

Outbound Dial Prefix:



Set GSM Route that dial out via MV-37X

After change, please press “**Submit changes**” and “**apply configuration changes**”



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.
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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.