

# MV-374 / MV-378

## VoIP GSM Gateway

### User Manual



MV-374



MV-378

**PORTech Communications Inc.**

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## **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 ,  
Communicates with other gateway or PC.

## **3.Parts list**

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

3.1 「 MV-374/MV-378 」 main body

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

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 SIM Card

5.5 LINK Indicator : Light up when network is connected.

5.6 CH3 : an indicator light of VoIP3

5.7 CH4 : an indicator light of VoIP4

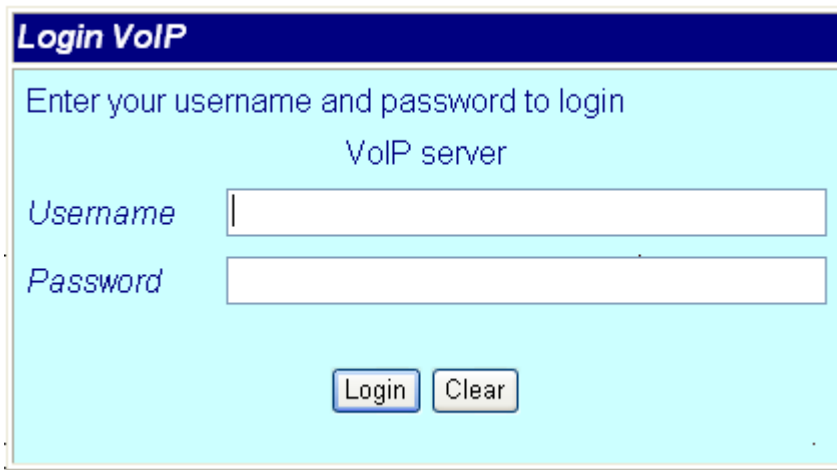
5.8 PWR (Power LED) : Light up when power is normal.

5.9 reboot button: reboot ch1-2 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 :



The screenshot shows a web page titled "Login VoIP". The page has a light blue background. At the top, there is a dark blue header with the text "Login VoIP" in white. Below the header, the text "Enter your username and password to login" is displayed, followed by "VoIP server". There are two input fields: "Username" and "Password". Below the input fields are two buttons: "Login" and "Clear".

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.

Mobile VoIP8 v6.694.u29	
Model Type:	MV-378n
Module Description:	GSM:900/1800/1900MHz (M23)
Firmware Version:	Wed Apr 1 09:08:06 2009.
Codec Version:	Wed Jan 17 15:06:12 2007.
Contact Address:	150, Shiang-Shung N.Road. , Taichung, Taiwan, R.O.C.
Tel:	886-4-23058000
Fax:	886-4-23022596
E-Mail:	<a href="mailto:sales@portech.com.tw">sales@portech.com.tw</a>
Web Site:	<a href="http://www.portech.com.tw">http://www.portech.com.tw</a>

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## 8. 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

## 8.1 Mobile TO LAN Settings

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

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### Mobile To LAN Table

Mobile 1, 2

Page: 1

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

Delete Selected Delete All reset

**Add New**

Position:  (0~49)

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

URL:  Ex:192.168.0.1, \*:2St

Add reset

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

\*CID :

(1) may enter the whole number, e.g. 0911111111

(2) only part of the number (prefix) e.g. 0911\* means any number starting with 0911 will be accepted

---

---

(3) \* means all numbers can be accepted

(4) N means the calls without the CID

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

\*URL : The IP address to transfer this call

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

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

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

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

Example:

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

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.

## 8.2 Call Back Service (50 sets)

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### Mobile To LAN Table

Page: 1

Item	CID	URL	Select
0	0933579613	#	<input type="checkbox"/>
1	+886933579613	#	<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Delete Selected Delete All reset

**Add New**

Position:  (0~49)

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

URL:  Ex:192.168.0.1, \*2St

Add reset

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

### 8.3 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.

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## Mobile To LAN Speed Dial

Mobile 1, 2

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

Delete Selected Delete All Reset

**Add New Phone**

Position:  (0~9)

Name:

URL:

Add Reset

\*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

## 8.4 LAN to Mobile Settings

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

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### LAN To Mobile Table

Mobile 1, 2

Page: 1

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

Delete Selected Delete All Reset

**Add New**

Position:  (0~49)

URL:  Ex:192.168.0.1, 192.168.0.\*

Call Num:  Ex:0911, \*:2St, #, #d?, #d?A?:1St

Add Reset

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

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

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

---

\*Call Num :

1.may enter the whole number, e.g. 0911111111

2.a simple "\*"means 2-stages-dialing. The call will be answered and prompt dial tone again to receive the called number as the destination, e.g. 0911111111 or 0911111111#

3.#['d'n']['a'ppp] for one-stage dialing

[...] is option

'd'n means to delete the beginning n codes,

'a'ppp means to add 'ppp' in front.

for example #d2a09 means one-stage dialing,

delete the first 2 codes from your destination number,  
then add 09 in front as the new destination number.

**Example:**

Lan to Mobile: \*, #

(1)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.

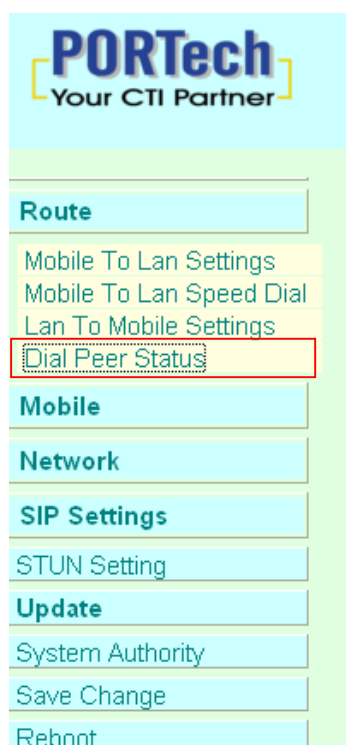
And proxy server/asterisk set the route that the prefix of destination number dial out from ch.2 MV-374/MV-378.

---

## 8.5 Dial Peer Status

You can check Dial Peer Status here

All the information will be shown on this page.



## Dial Peer Status

---

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 resisted

BUSY: Call is online.

Remote Address:

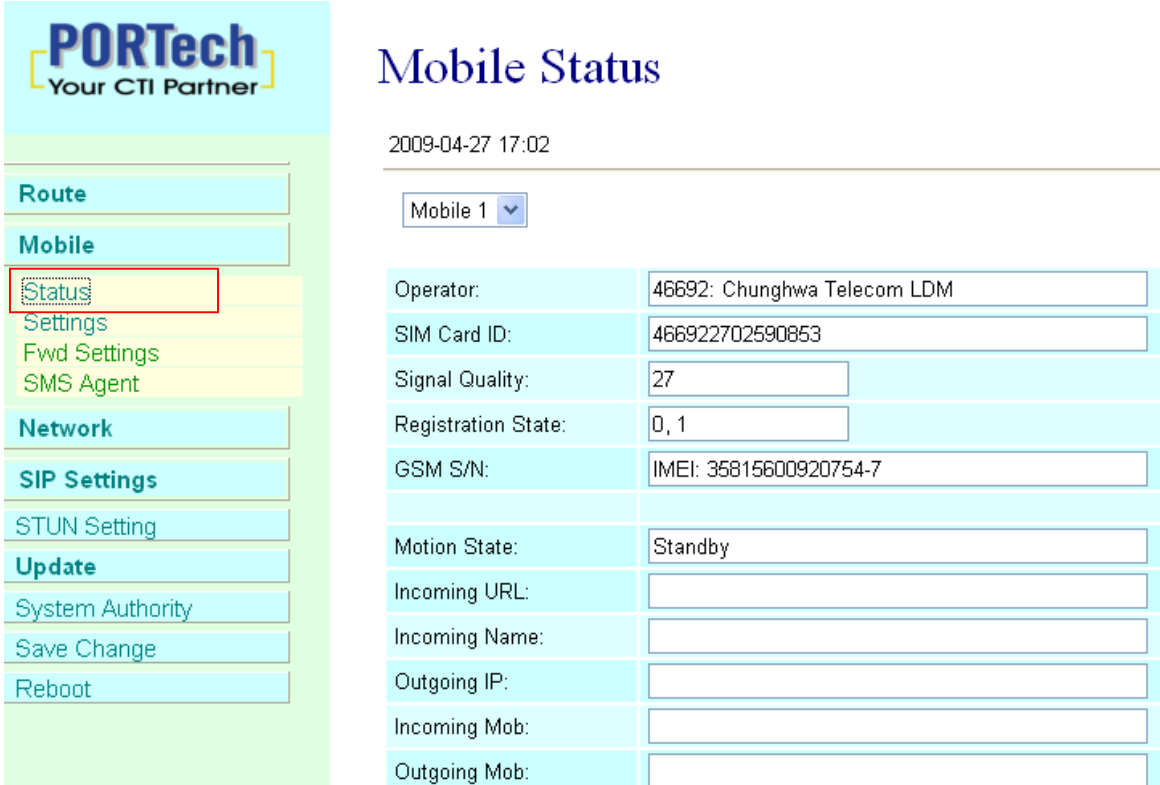
The IP Address which came from Lan side



---

## 9.Mobile

### 9.1 Mobile Status



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### Mobile Status

2009-04-27 17:02

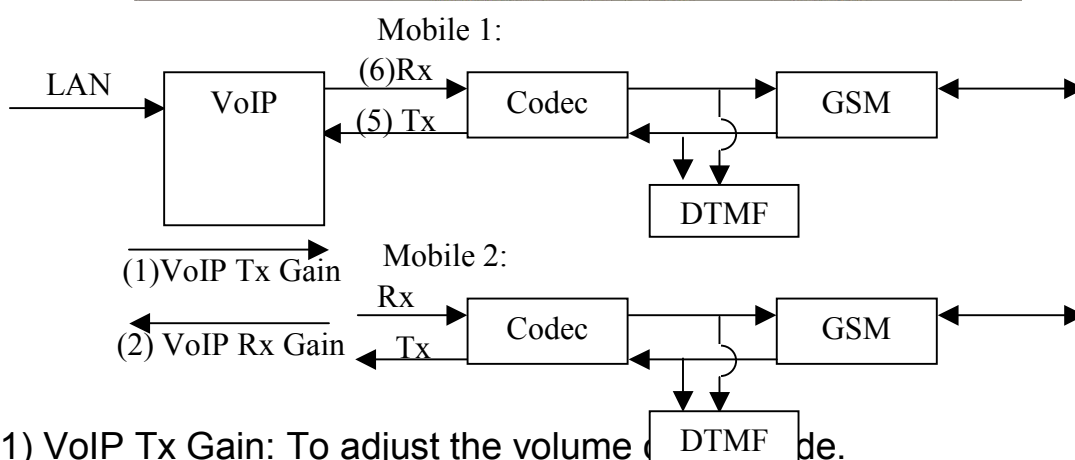
Mobile 1

Operator:	46692: Chunghwa Telecom LDM
SIM Card ID:	466922702590853
Signal Quality:	27
Registration State:	0, 1
GSM S/N:	IMEI: 35815600920754-7
Motion State:	Standby
Incoming URL:	
Incoming Name:	
Outgoing IP:	
Incoming Mob:	
Outgoing Mob:	

- (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 the SIM card been registered.
- (3)SIM Card ID : SIM card ID. (4)Signal Quality : Signal quality.
- (5)GSM S/N : IMEI Number
- (6)Incoming IP : The IP address of the last incoming call from LAN.
- (7)Incoming IP Name: proxy server name
- (8)Outgoing IP : The IP address of the last outgoing call to LAN.
- (9)Incoming Mob : The caller ID of the last incoming call from MOBILE.
- (10)Outgoing Mob : The called number of the last outgoing call to MOBILE.

## 9.2 Mobile Setting

Only change "mobile" into "on" or "off", just click "submit", no need to click "save change"



- (1) VoIP Tx Gain: To adjust the volume of Mobile side.
- (2) VoIP Rx Gain: To adjust the volume of Mobile side.
- (3) LAN Dialtone 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 CLIR : If you need to block the Caller Id for call termination, please choose Suppression
- (10) Mobile PIN Code: If you need to unlock pin code via MV-374/MV-378, you can click "On" and enter pin code.
- (11) LAN Answer Mode:
- Answered : when mobile answer, then connect the call
  - Alerted : when the mobile is ringing back tone, then connect the call
  - Income : when lan dial out, then connect soon
- (12) Band Type: You can manual setting according to your GSM Frequency of carrier.
- (13) ON/Off: If you use this channel, please click on. Otherwise, please click off.

### 9.3 Mobile / Forward Setting :

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

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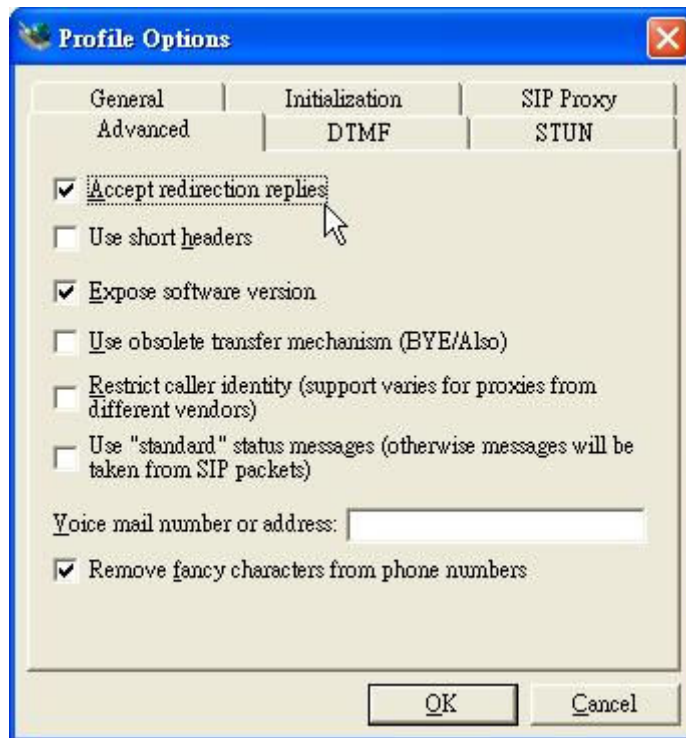
## Forward Setting

Mobile 1, 2 ▼

Forward Enable

	Name	URL:Port
Fwd to Mobile1:	<input type="text"/>	<input type="text"/>
Fwd to Mobile2:	<input type="text"/>	<input type="text"/>
Fwd to External:	<input type="text"/>	<input type="text"/>

- \* "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.)



	Name	URL:Port
Fwd to Mobile1:		192.168.0.100:5060
Fwd to Mobile2:		192.168.0.100:5062
Fwd to External:		

**The Explanation of Picture:**

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

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

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

## 9.4 Mobile / SMS Agent :

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**SMS Agent**

Mobile 1, 2 ▼

Read received SMS

Port	Status	Bank
Mobile 1	Not Ready !!!	Rx List
Mobile 2	Not Ready !!!	Rx List

**SMS Sender**

Encode: ASC7 (ASCII 7bit) ▼

Via: Mobile  1  2

Dest Num:

Message:

Maximum Number of ASC7 chars for this text box is 160.  
You have 160 ASC7 chars remaining for your description..

2 mode:  
ASC7(ASCII 7 bit)  
UCS2(Unicode 16 bit)

Send Now

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

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

## 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	RemotelD	Date, Time
1	886935386862	08/05/15, 15:41:46

MV Serial can send SMS and Receive SMS

[Back](#)

[Delete](#)



---

## 9.5 Send Bulk of SMS via Microsoft Excel

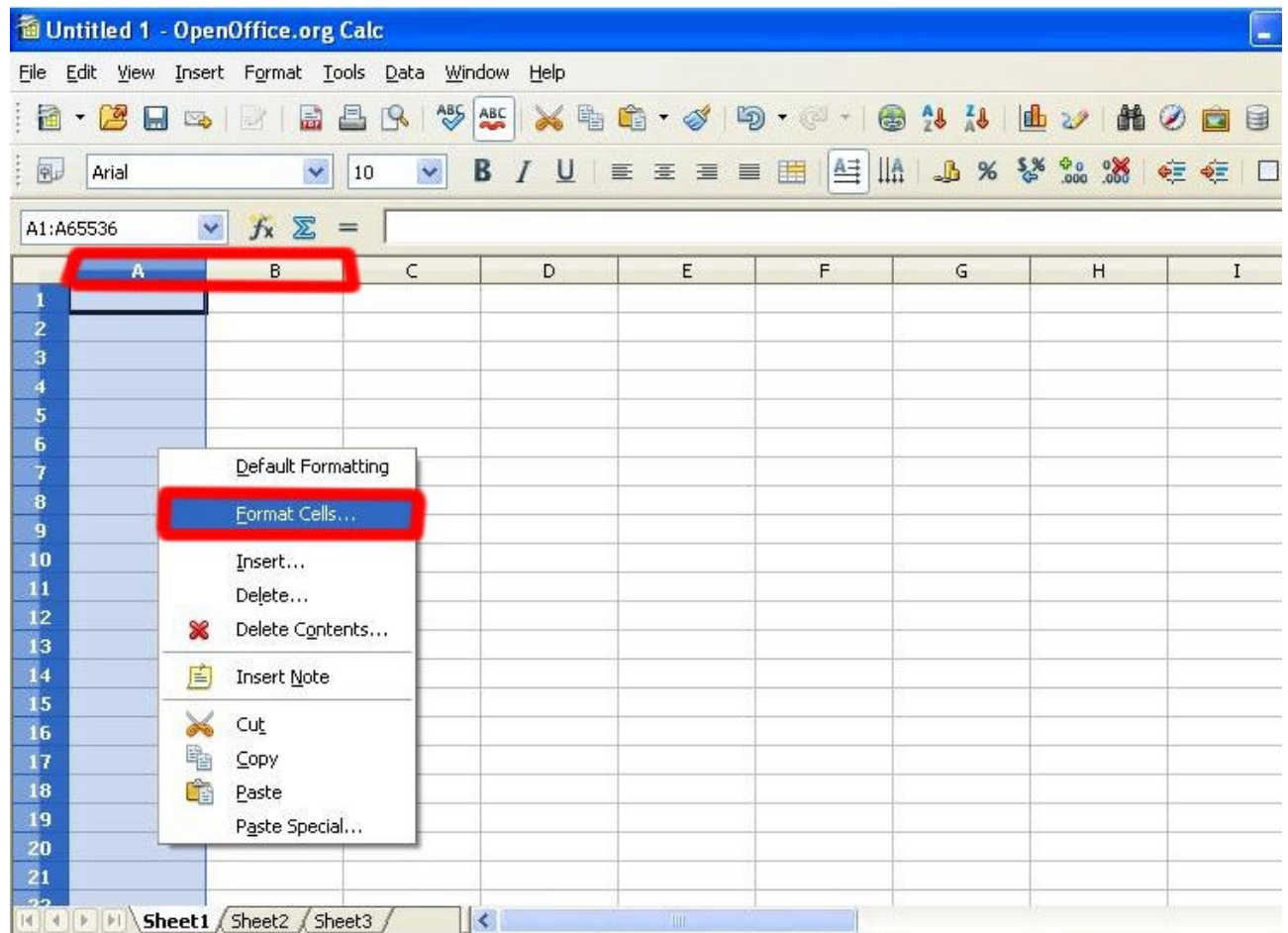
**First of all, please open a new Excel file.**

### Step 1 Format Cells

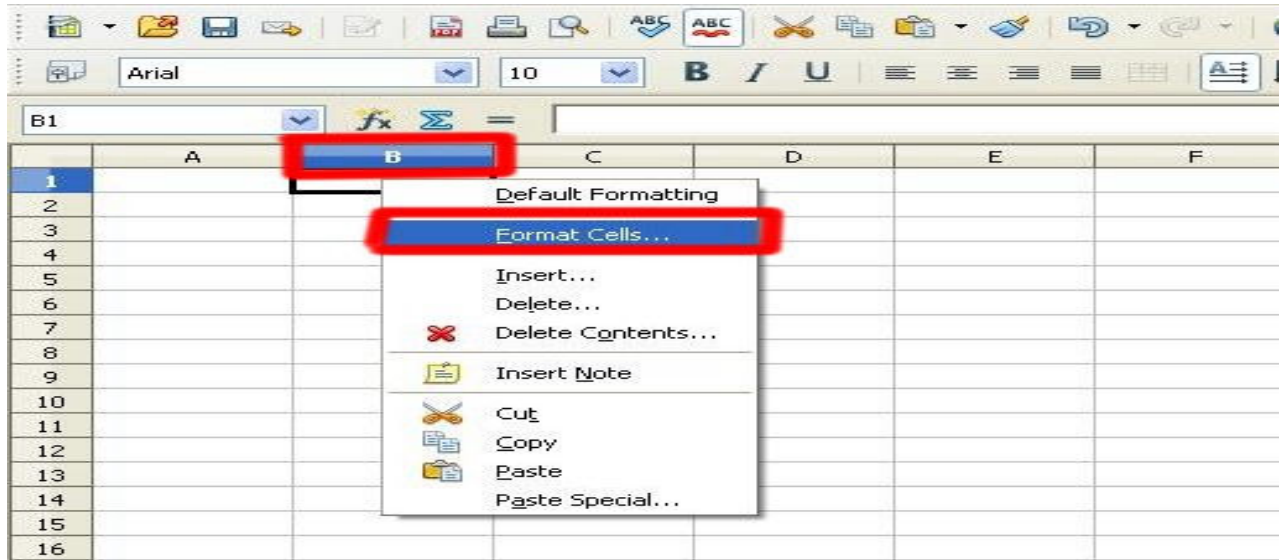
Here, we need you to format cells to “Text” first.

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

#### Blank A

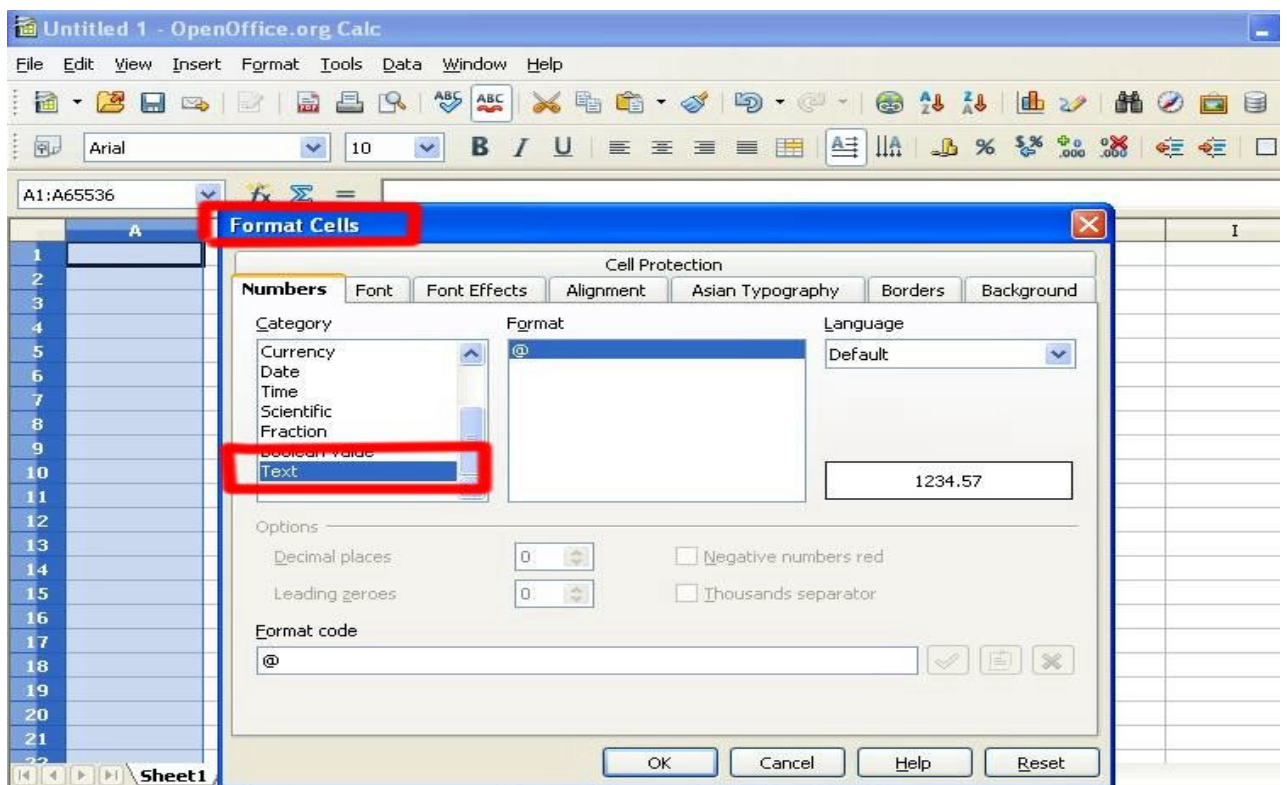


## Blank B



## Step 2

In the Format Cells, please select “Text”



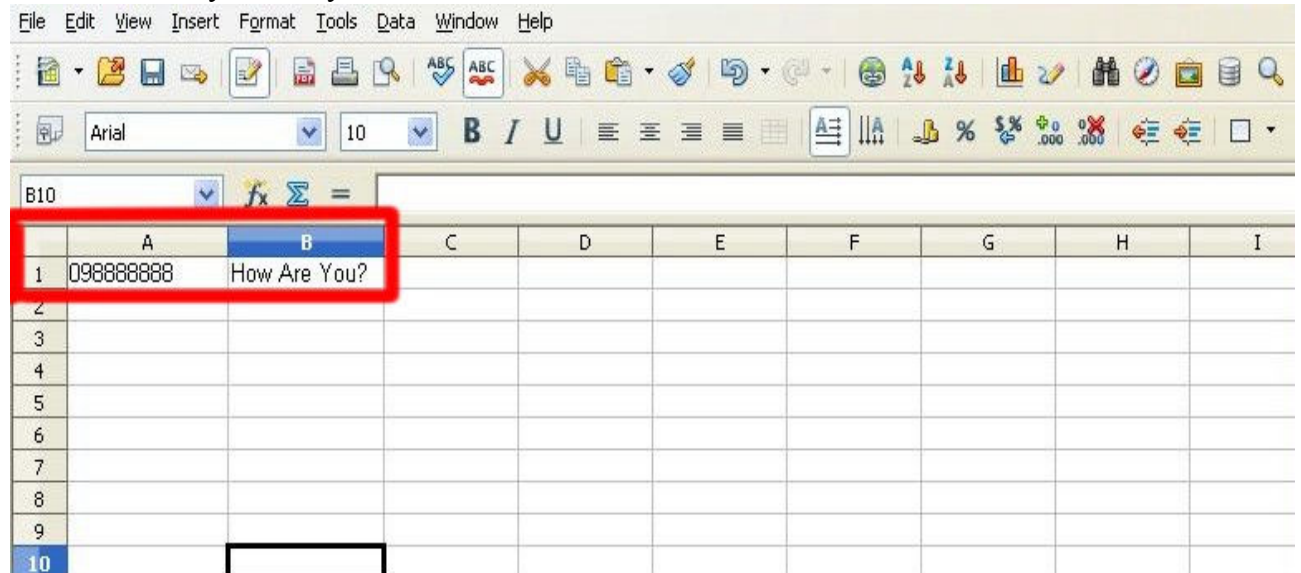
- Please do this action for Blank A and B both.

---

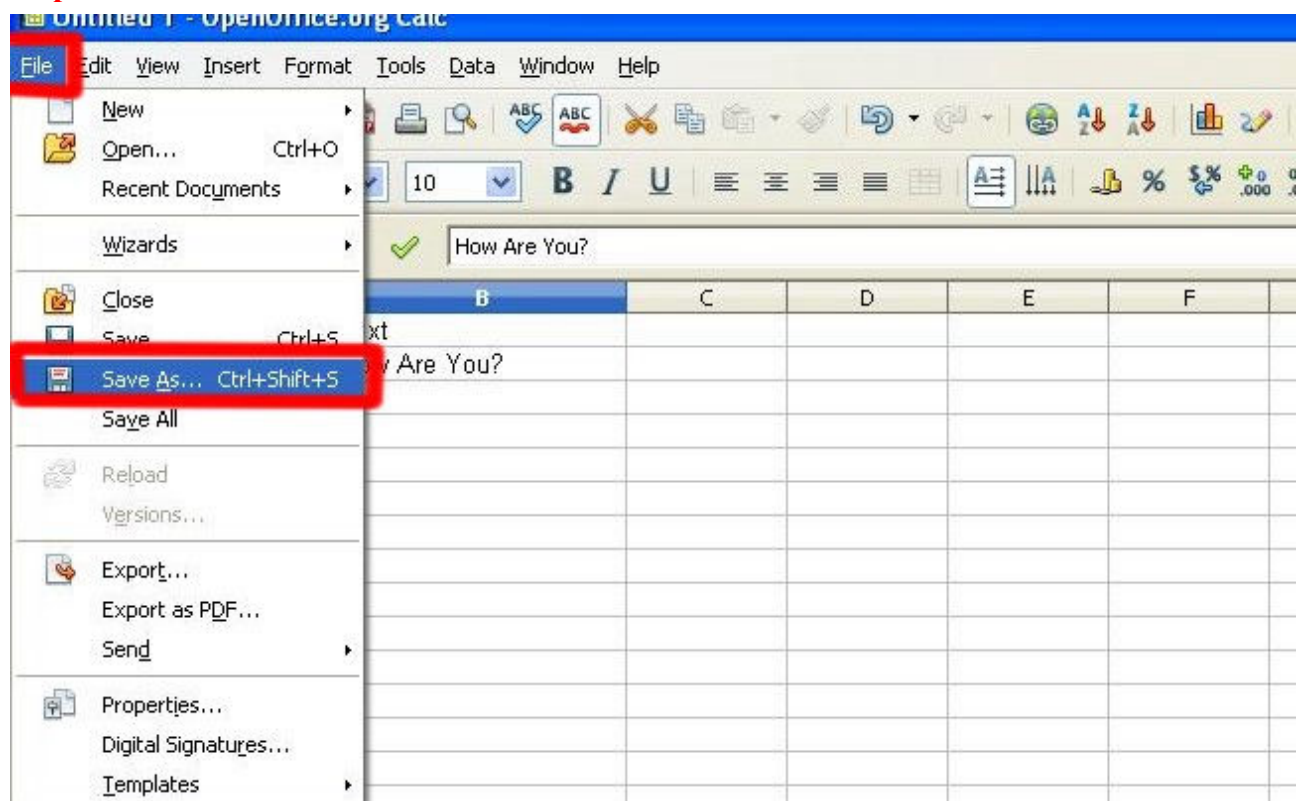
### Step 3

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

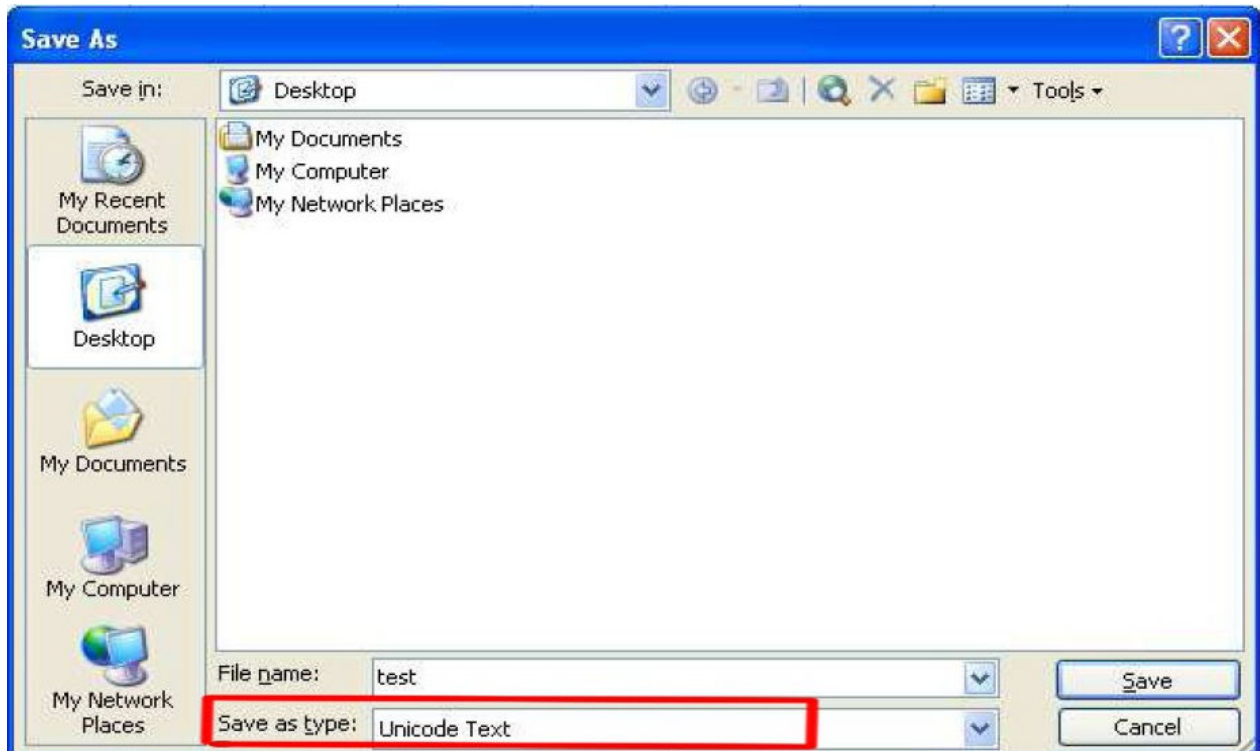
Blank B: is for you to key “text”



### Step 4 Save the file

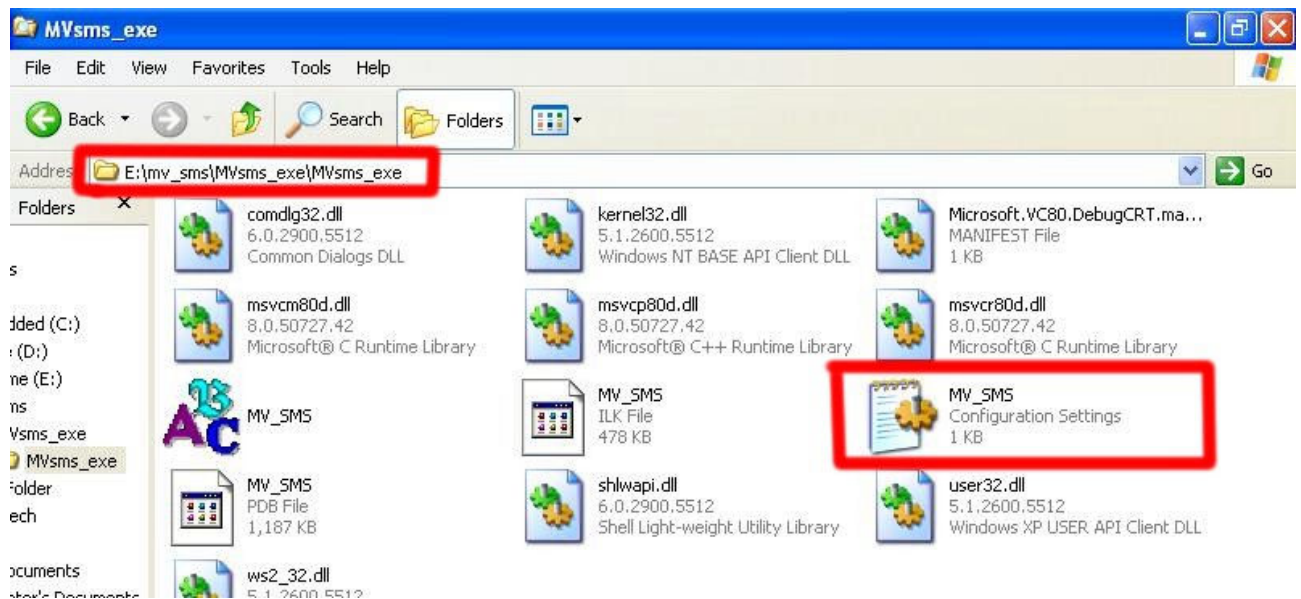


Save the type as “Unicode Text”



### Step 5

Open MVsms\_exe -> MV-SMS (Configuration Settings)





### Step 6

Please do the configuration as following:

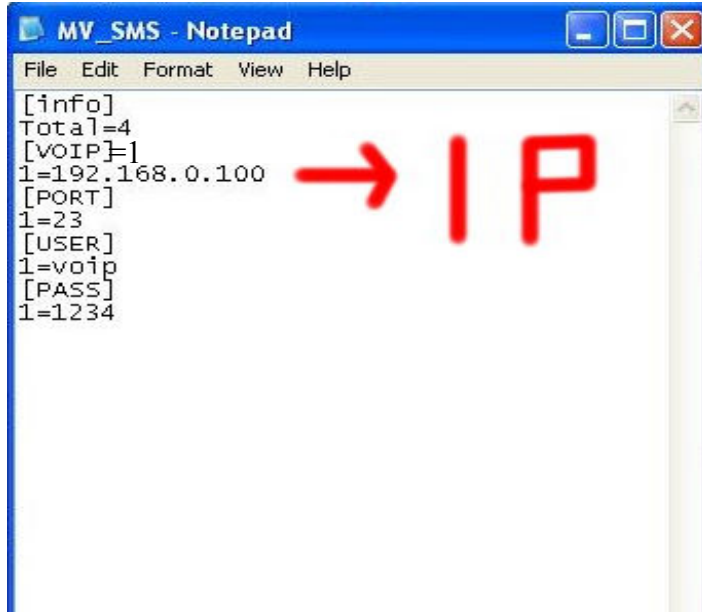
#### MV-378

```
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-374

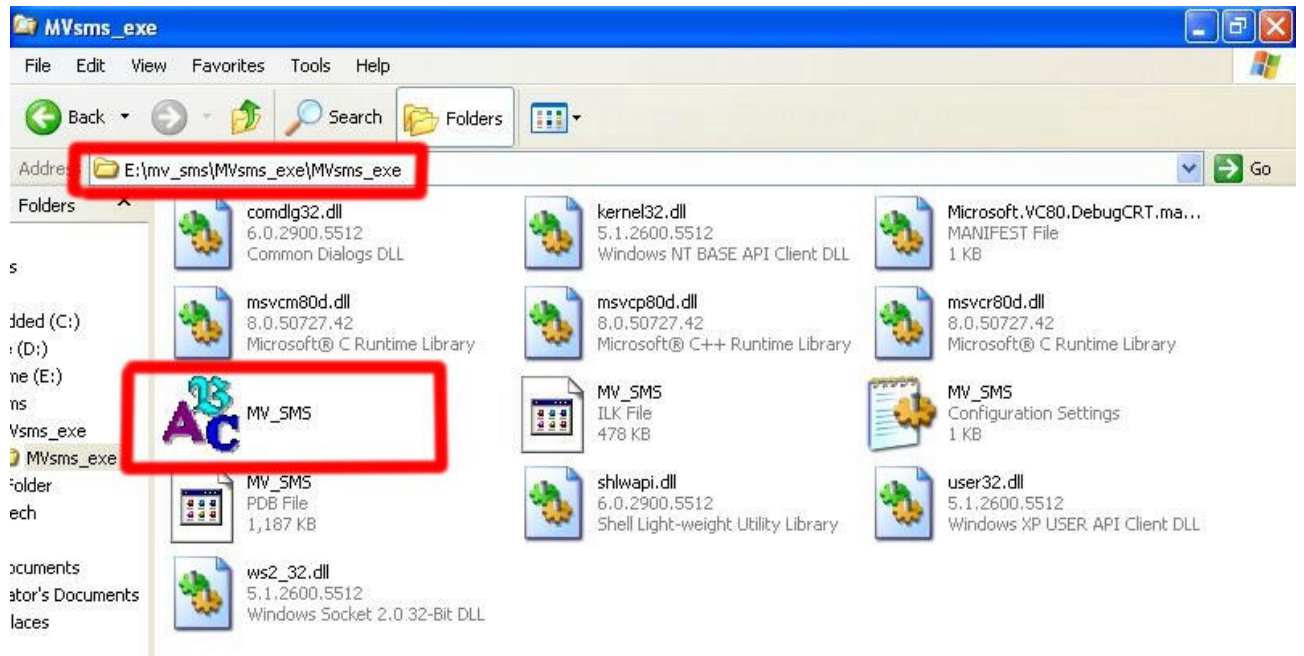
```
File Edit Format View Help
[info]
Total=2
[VOIP]
1=192.168.0.100
2=192.168.0.100
[PORT]
1=23
2=8023
[USER]
1=voip
2=voip
[PASS]
1=1234
2=1234
```

## MV-372 & MV-370



### Step 7

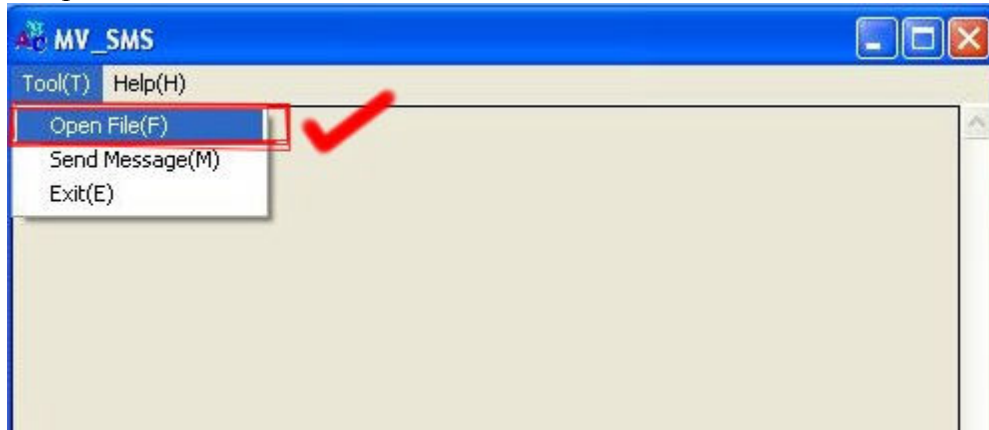
Run MV-SMS program



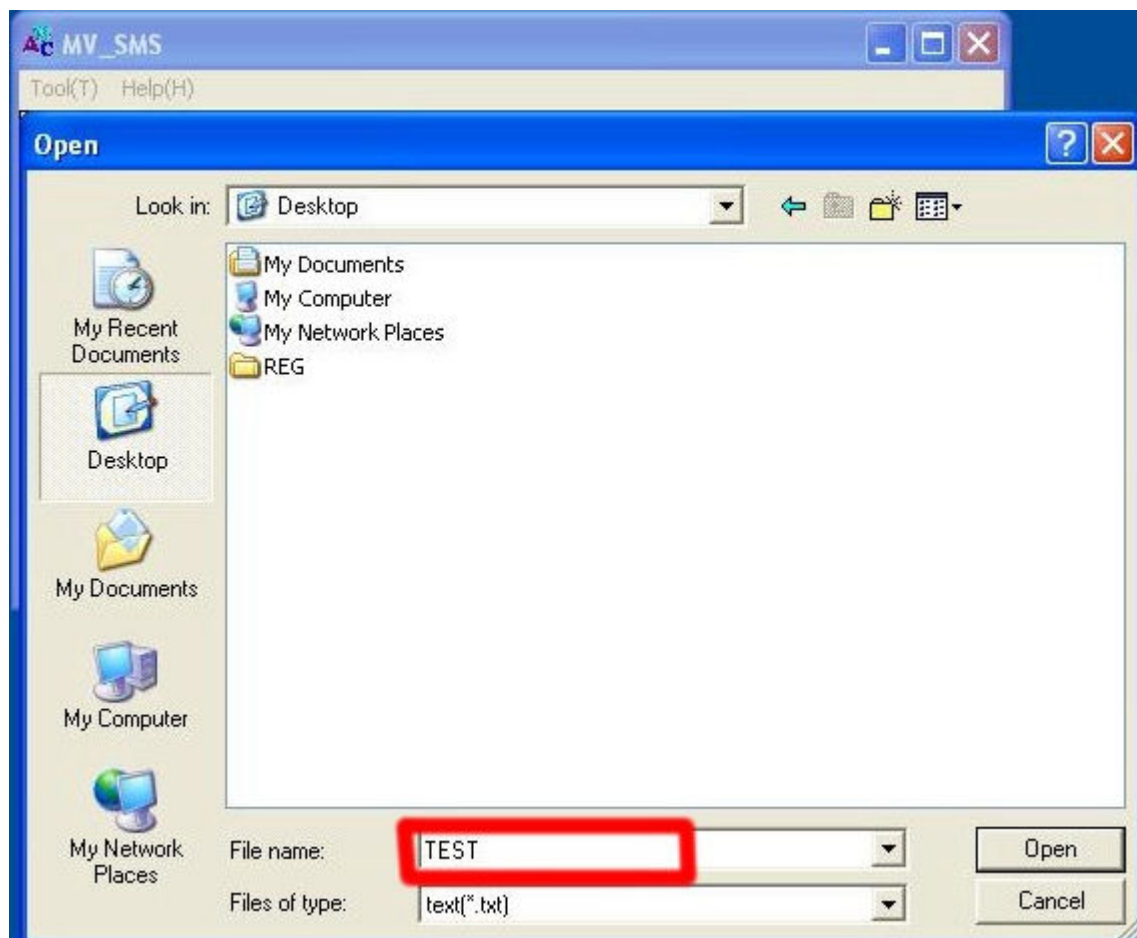
---

## Step 8

### 1. Open File



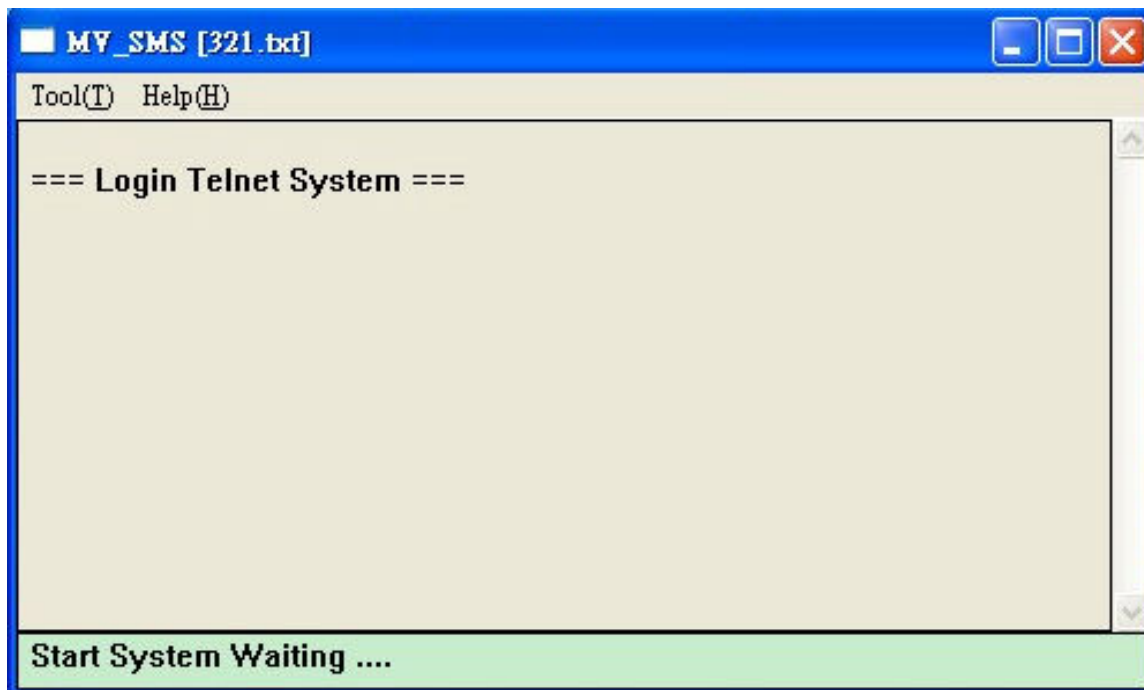
### 2. Open the “Excel file” that you just saved



---

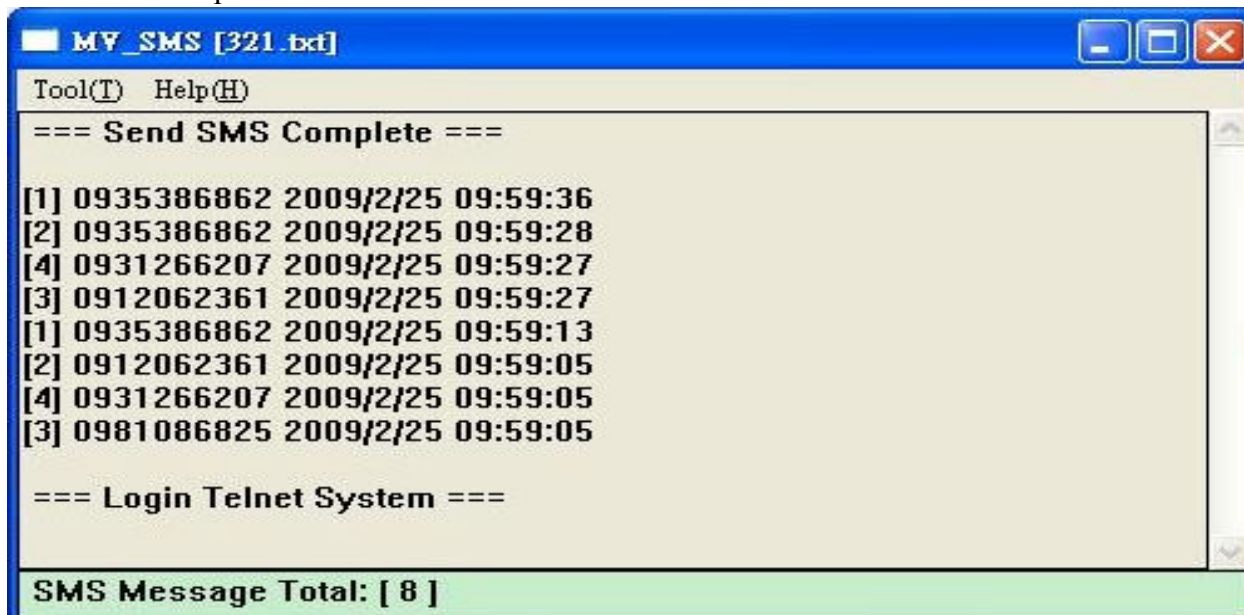
### Step 9

Sending



### Step 10

Send SMS Complete





---

## 9.6 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

```
username: voip  
password: ****  
user level = 1.
```

Please enter account and password

```
command: logout, module, module1, module2.
```

```
>module1
```

```
getting module 1 ...
```

```
got!! press 'ctrl-x' to release module 1.
```

Choose module

```
0  
ate1
```

Enter "ate1", then you can see your at command below

```
0
```

```
at+cmgf=1
```

```
0
```

```
at+cmgs="0911123456"
```

Enter at+cmgs="phone number"

```
>
```

```
test
```

```
>
```

```
+CMGS: 30
```

```
0
```

Enter short message

## 10.Network

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

10.1 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	-

## 10.2 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

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### WAN Settings

You could configure the WAN settings in this page.

WAN Setting	
IP Type	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP Client <input type="radio"/> PPPoE
Master IP	<input type="text" value="192.168.0.110"/>
Mask	<input type="text" value="255.255.255.0"/>
Gateway	<input type="text" value="192.168.0.254"/>
DNS Server1	<input type="text" value="168.95.192.1"/>
DNS Server2	<input type="text" value="168.95.1.1"/>
MAC	<input type="text" value="00037e001f7d"/>

PPPoE Setting	
User Name	<input type="text"/>
Password	<input type="text"/>


 

- (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.

---

### 10.3 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.



## SNTP Settings

You could set the SNTP servers in this page.

**SNTP:**  On  Off

Primary Server:

Secondary Server:

Time Zone: GMT -  :  (hh:mm)

Sync. Time:  :  :  (dd:hh:mm)

**Route**

**Mobile**

**Network**

Status

WAN Settings

LAN Settings

**SNTP Settings**

Slave Setting

**SIP Settings**

**Update**

System Authority

Save Change

Reboot

---

## 11.SIP Setting

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

### 11.1 Service Domain Setting

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

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Your CTI Partner

Route  
Mobile  
Network  
SIP Settings  
Service Domain  
Port Settings  
Codec Settings  
Codec ID Setting  
DTMF Setting  
RPort Setting  
SIP Responses  
Other Settings

Update  
System Authority  
Save Change  
Reboot

### Service Domain Settings

Mobile 1 ▾

Realm 1 (Default)	
Active:	<input checked="" type="radio"/> ON <input type="radio"/> OFF
Display Name:	<input type="text" value="803"/>
User Name:	<input type="text" value="803"/>
Register Name:	<input type="text" value="803"/>
Register Password:	<input type="password" value="..."/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text" value="192.168.0.1"/>
Outbound Proxy:	<input type="text"/>
Status:	Registered

Realm 2	
Active:	<input type="radio"/> ON <input checked="" type="radio"/> OFF
Display Name:	<input type="text"/>
User Name:	<input type="text"/>
Register Name:	<input type="text"/>

First you need to click Active to enable the Service Domain, 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

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

## 11.2 Ports Setting

**PORTech**  
Your CTI Partner

**Ports Setting**

Internal Dial Peer Port:  (1024~19900)

	SIP Port (1024~19900)	RTP Port (20000~59900)
Mobile 1	<input type="text" value="5064"/>	<input type="text" value="20004"/>
Mobile 2	<input type="text" value="5066"/>	<input type="text" value="20006"/>
Mobile 3	<input type="text" value="5068"/>	<input type="text" value="20008"/>
Mobile 4	<input type="text" value="5070"/>	<input type="text" value="20010"/>
Mobile 5	<input type="text" value="5072"/>	<input type="text" value="20012"/>
Mobile 6	<input type="text" value="5074"/>	<input type="text" value="20014"/>
Mobile 7	<input type="text" value="5076"/>	<input type="text" value="20016"/>
Mobile 8	<input type="text" value="5078"/>	<input type="text" value="20018"/>

In Ports Setting, you can change dialpeer 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 dialpeer 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 dialpeer port or SIP port)

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

### 11.3 Codec Settings:

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



## Codec Settings

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



## 11.4 Codec ID Setting

You can setup the Codec ID in this page.



**Route**

**Mobile**

**Network**

**SIP Settings**

Service Domain

Port Settings

Codec Settings

**Codec ID Setting**

DTMF Setting

RPort Setting

SIP Responses

Other Settings

**Update**

System Authority

Save Change

Reboot

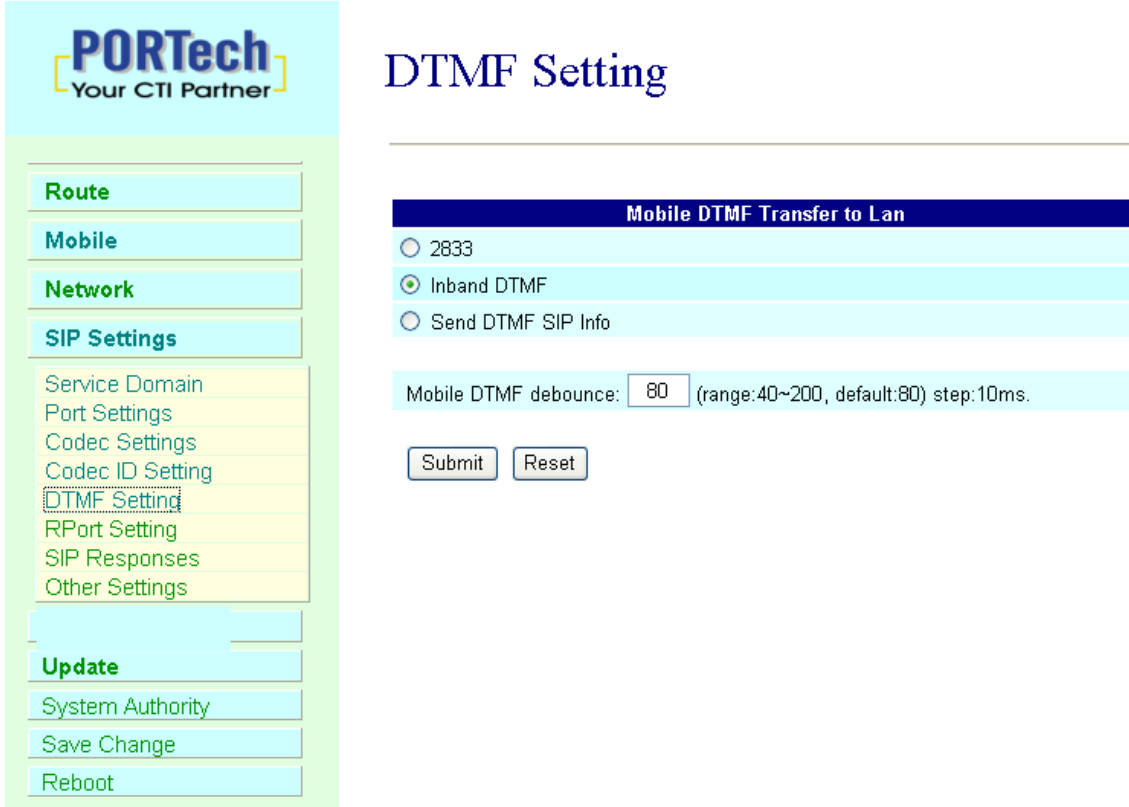
### Codec ID Setting

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

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

## 11.5 DTMF Setting

You can setup the DTMF Setting in this page.



The screenshot shows the PORTech web interface for DTMF settings. On the left is a navigation menu with the following items: Route, Mobile, Network, SIP Settings, Service Domain, Port Settings, Codec Settings, Codec ID Setting, DTMF Setting (highlighted), RPort Setting, SIP Responses, Other Settings, Update, System Authority, Save Change, and Reboot. The main content area is titled "DTMF Setting" and contains a section for "Mobile DTMF Transfer to Lan" with three radio button options: 2833, Inband DTMF (selected), and Send DTMF SIP Info. Below this is a text input field for "Mobile DTMF debounce" with a value of 80 and a range of 40~200. At the bottom of the main area are "Submit" and "Reset" buttons.

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### DTMF Setting

**Mobile DTMF Transfer to Lan**

2833

Inband DTMF

Send DTMF SIP Info

Mobile DTMF debounce:  (range:40~200, default:80) step:10ms.

## 11.6 RPort Function:

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

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**Route**

**Mobile**

**Network**

**SIP Settings**

Service Domain

Port Settings

Codec Settings

Codec ID Setting

DTMF Setting

RPort Setting

SIP Responses

Other Settings

.....

**Update**

System Authority

Save Change

Reboot

## RPort Setting

Mobile 1, 2 ▼

---

RPort of Mobile 1:  On  Off

RPort of Mobile 2:  On  Off

## 11.7 SIP Responses

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### SIP Responses Setting

**Response on port busy.**

<input checked="" type="radio"/> 486	Busy here
<input type="radio"/> 503	Service unavailable

**SIP Responses**

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

**Dial Peer**

Working Mode:  OFF  Internal  External

External URL:  ( Dial Peer for XP )

Dial Peer Configuration Table corresponding IP (please read next page)

**\*\*\* If you have dial peer server, Sip server/Asterisk set GSM route, please set Dial Peer server's IP\*\***

11.7.1 486(busy here), 503(Service unavailable):

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

11.7.2 180 Ring on/off:

LAN TO MOBILE two stage dialing can be turn off, therefore there will be no the Ring Back Tone, all the phone call will be transferred to prompt voice directly. (For this function, 183 must be turn on)

11.7.3 183(Session Progress)

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

---

## 11.7.4 Dial Peer

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

Dial Peer	
Working Mode	<input type="radio"/> OFF <input checked="" type="radio"/> Internal <input type="radio"/> External
External URL	<input type="text"/> ( <a href="#">Dial Peer</a> for XP )

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

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

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

**E.g SIP Server send 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 dialpeer program.

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

### Edit DialPeer.ini (External Dial Peer)

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

Total ip / port

[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

The first  
MV-378

The second  
MV-378

[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

The first  
MV-378

The second  
MV-378

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

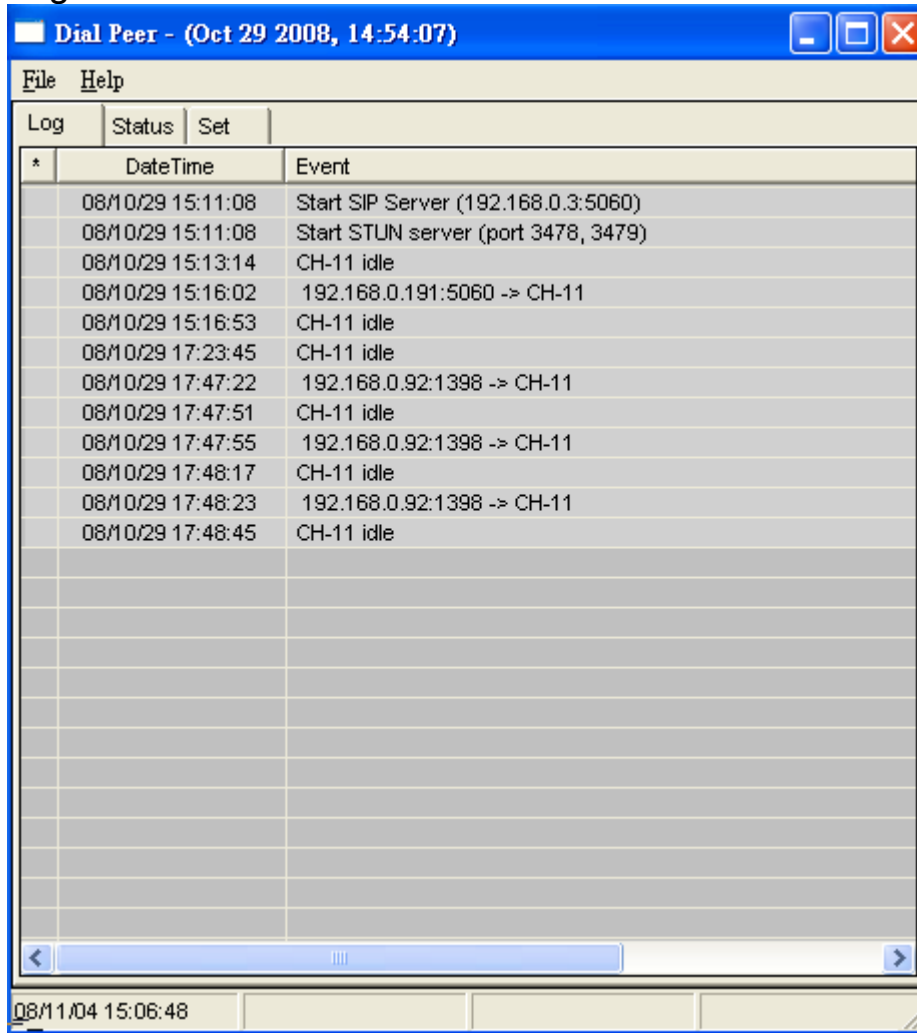
[PtcPort]  
1=40000  
2=40000  
3=40008  
4=40008  
5=40016  
6=40016  
7=40024  
8=40024

The first  
MV-378

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

The second  
MV-378

## Log



The screenshot shows a window titled "Dial Peer - (Oct 29 2008, 14:54:07)". The window has a menu bar with "File" and "Help". Below the menu bar is a table with columns "Log", "Status", and "Set". The main content area is a table with columns "DateTime" and "Event". The table contains the following data:

* DateTime	Event
08/10/29 15:11:08	Start SIP Server (192.168.0.3:5060)
08/10/29 15:11:08	Start STUN server (port 3478, 3479)
08/10/29 15:13:14	CH-11 idle
08/10/29 15:16:02	192.168.0.191:5060 -> CH-11
08/10/29 15:16:53	CH-11 idle
08/10/29 17:23:45	CH-11 idle
08/10/29 17:47:22	192.168.0.92:1398 -> CH-11
08/10/29 17:47:51	CH-11 idle
08/10/29 17:47:55	192.168.0.92:1398 -> CH-11
08/10/29 17:48:17	CH-11 idle
08/10/29 17:48:23	192.168.0.92:1398 -> CH-11
08/10/29 17:48:45	CH-11 idle

The window also features a scrollbar at the bottom and a status bar at the very bottom showing the date and time "08/11/04 15:06:48".

## Status

**Dial Peer - (Oct 29 2008, 14:54:07)**

File Help

Log	Status	Set			
CH	Mv IP	port	state	remote	
1	192.168.0.100	5060	-s-	-r-	
2	192.168.0.100	5062	-s-	-r-	
3	192.168.0.100	5064	-s-	-r-	
4	192.168.0.100	5066	-s-	-r-	
5	192.168.0.100	5068	-s-	-r-	
6	192.168.0.100	5070	-s-	-r-	
7	192.168.0.100	5072	-s-	-r-	
8	192.168.0.100	5074	-s-	-r-	
9	192.168.0.110	5060	OFF/0	-r-	
10	192.168.0.110	5062	IDLE/0	-r-	
11	192.168.0.110	5064	IDLE/1	-r-	
12	192.168.0.110	5066	OFF/0	-r-	
13	192.168.0.110	5068	OFF/0	-r-	
14	192.168.0.110	5070	OFF/0	-r-	
15	192.168.0.110	5072	OFF/0	-r-	
16	192.168.0.110	5074	OFF/0	-r-	

08/11/04 15:07:14

**Callout 1:** The first MV-378 doesn't register dial peer software (points to rows 1-8)


**Callout 2:** The 2,3ch of Second MV-378 idle (points to rows 10-11)

**Callout 3:** The 1,4-8ch of Second MV-378 turn off (points to rows 12-16)



## 11.8 Other Settings

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



### Other Settings

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:	<input type="text" value="40"/>	(0~63)
SIP QoS:	<input type="text" value="40"/>	(0~63)
SIP Expire Time:	<input type="text" value="300"/>	(60~86400 sec)

---

**Route**

**Mobile**

**Network**

**SIP Settings**

- Service Domain
- Port Settings
- Codec Settings
- Codec ID Setting
- DTMF Setting
- RPort Setting
- SIP Responses
- Other Settings**

System Authority

Save Change

Reboot

---

## 12. STUN Setting

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

12.1 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.

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Your CTI Partner

### Public STUN Setting

Public STUN  On  Off

STUN Server

STUN Port  (1024~65534)

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

---

## 13. System Authority

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



The sidebar menu is located on the left side of the page. It features the PORTech logo at the top, which includes the text 'PORTech' in a large, bold, blue font and 'Your CTI Partner' in a smaller, black font below it. Below the logo, there is a vertical list of menu items, each enclosed in a light blue rectangular box with a thin border. The items are: 'Route', 'Mobile', 'Network', 'SIP Settings', 'Update', 'System Authority', 'Save Change', and 'Reboot'. The 'System Authority' item is highlighted with a darker blue background.

### System Authority

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

---

New username:	<input type="text"/>
New password:	<input type="password"/>
Confirmed password:	<input type="password"/>
	<input type="button" value="Submit"/> <input type="button" value="Reset"/>

---

## 14.Update

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

### 14.1 Update firmware

You can down load new firmware from here:

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

Please update firmware by to following steps.

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

Step 1: MASTER: 192.168.0.100 please update this firmware → nat\_portech\_XXX

Step 2: Slave 1 : 192.168.0.100:8080 please update this firmware → Mv\_4-8p\_portech\_XXX

Step 3: Slave 2 : 192.168.0.100:8180 please update this firmware → Mv\_4-8p\_portech\_XXX

#### 14.1.2 MV-378 Need to update firmware for 5 times (Master, Slave1, Slave2, Slave3, and Slave4)

Step 1: MASTER: 192.168.0.100 please update this firmware → nat\_portech\_XXX

Step 2: Slave 1 : 192.168.0.100:8080 please update this firmware → Mv\_4-8p\_portech\_XXX


Step 3: Slave 2 : 192.168.0.100:8180 please update this firmware → Mv\_4-8p\_portech\_XXX

Step 4: Slave 3 : 192.168.0.100:8280 please update this firmware → Mv\_4-8p\_portech\_XXX

Step 5: Slave 4 : 192.168.0.100:8380 please update this firmware → Mv\_4-8p\_portech\_XXX

**\*Important\* After you upgrade all the firmwares, please click Default Setting on 192.168.0.100**

**\*Important\* Please do not change firmware 's files name.**



Route

Mobile

Network

SIP Settings

Update

**New Firmware**

Default Settings

System Authority

Save Change

Reboot

## Update Firmware

You could update the newest firmware. PCB mark: 2N149A

Method:  HTTP  TFTP

### HTTP

Code Type: Risc

File Location:  瀏覽...

### TFTP

TFTP Server: 192.168.1.250

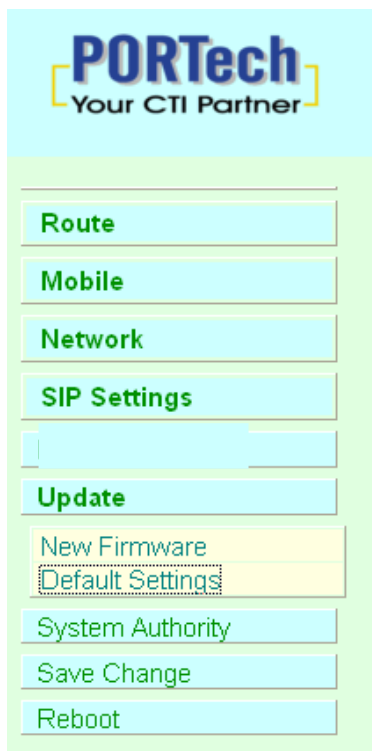
- (1) In New Firmware function you can update new firmware via HTTP in this page. You can upgrade the firmware by the following steps:
- (2) Select the firmware code type, Risc code.
- (3) Click the "Browse" button in the right side of the File Location or you can type the correct path and the filename in File Location blank.
- (4) Select the correct file you want to download to the system then click the Update button.
- (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. IP will retain original IP as usual not default IP.

Factory all: all setting include ip will restore default setting.



## Restore Default Settings

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

Restore default settings:

Restore factory all settings:  ( included all IP address )

---

## 15. Save Change

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



---

## 16.Reboot

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



### Reboot System

You could press the reboot button to restart the system.

---

Reboot system:



---

## **17.Specification**

### 17.1 Protocols

SIP (RFC2543,RFC3261)

### 17.2 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

### 17.3 Codec

G.711 u-Law

G.711 a-Law

G.723.1 (5.3k)

G.723.1 (6.3k)

G.729A

G.729A/B

### 17.4 Voice Quality

VAD

---

CNG

AEC, LEC

Packet loss

17.5 GSM (MV-374/MV-378)

Quad Band:900/1800/1900/850MHZ

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

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

CDMA 2000(800MHZ/1900MHZ)

\*\*Please note\*\*

Most CDMA operators don't offer Polarity reversing . So VoIP to  
Mobile, MV-370 will connect soon. CDMA operators will start billing  
soon. It doesn't wait mobile side answer.

CDMA Version doesn't support SMS Feature and 180/183  
unavailable

## 18. 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"

## 19. 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)		
<b>Mobile 1</b> <input checked="" type="radio"/> ON <input type="radio"/> OFF			
Routing Range	0 to 49 (0~49)		
CODEC Tx Gain:	6 (0~7)	CODEC Rx Gain:	6 (0~7)
SIP From:	Tel/Tel (Not Reg)	Answer Delay	0 (0~15)
CLID Presentation	<input type="radio"/> Suppression <input checked="" type="radio"/> Invocation		
Mobile PIN Code:	On <input type="checkbox"/> Code: <input type="text"/>	Confirmed:	<input type="text"/>
LAN Answer Mode	<input checked="" type="radio"/> Answered <input type="radio"/> Alerted <input type="radio"/> Income		

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

<b>Realm 1 (Default)</b>	
Active:	<input checked="" type="radio"/> ON <input type="radio"/> OFF
Display Name:	<input type="text"/>
User Name:	<input type="text"/>
Register Name:	<input type="text"/>
Register Password:	<input type="text"/>
Domain Server:	192.168.0.192:5060
Proxy Server:	192.168.0.192:5060
Outbound Proxy:	<input type="text"/>
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	<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

**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	*	#	<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"/>

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

## SIP Responses Setting

### Response on port busy.

- 486 Busy here  
 503 Service unavailable

### SIP Responses

- ON  OFF 180 Ringing ( Force to ON, if 183 was OFF. )  
 ON  OFF 183 Session Progress

### Dial Peer

- Working Mode  OFF  Internal  External  
External URL  ([Dial Peer](#) for XP )

Submit

**PORTech**  
Your CTI Partner

- 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

## Ports Setting

Internal Dial Peer Port:  (1024~19900)

	SIP Port (1024~19900)	RTP Port (20000~59900)
Mobile 1	<input type="text" value="5064"/>	<input type="text" value="20004"/>
Mobile 2	<input type="text" value="5066"/>	<input type="text" value="20006"/>
Mobile 3	<input type="text" value="5068"/>	<input type="text" value="20008"/>
Mobile 4	<input type="text" value="5070"/>	<input type="text" value="20010"/>
Mobile 5	<input type="text" value="5072"/>	<input type="text" value="20012"/>
Mobile 6	<input type="text" value="5074"/>	<input type="text" value="20014"/>
Mobile 7	<input type="text" value="5076"/>	<input type="text" value="20016"/>
Mobile 8	<input type="text" value="5078"/>	<input type="text" value="20018"/>

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:](#)  Type your mobile number

[Never Override CallerID:](#)

[Maximum channels:](#)  MV-374: 4  
MV-378: 8

#### Outgoing Dial Rules

[Dial Rules:](#)

[Dial rules wizards:](#)

[Outbound Dial Prefix:](#)



---

**Outgoing Settings**

Trunk Name:

PEER Details:

```
host=192.168.0.111
port=5060
type=peer
```

Type MV-37X's ip and port

The image shows a screenshot of a configuration interface titled "Outgoing Settings". It contains a "Trunk Name" field with the value "SIM1" and a "PEER Details" section. The PEER Details section contains a text area with the following text: "host=192.168.0.111", "port=5060", and "type=peer". A red rectangular box highlights the first line of text. A callout box with a pointer indicates that the user should enter the IP address and port for an MV-37X device.

Set GSM Route that dial out via MV-37X

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

The screenshot shows the 'Edit Route' page in freePBX. The page title is 'Delete Route SIM1'. The 'Route Name' is 'SIM1'. The 'Dial Patterns' section contains a list with '09XXXXXXXX' selected. The 'Trunk Sequence' section shows 'SIP/SIM1' selected. The 'Submit Changes' button is at the bottom left. A red bar at the top of the page contains the text 'Apply Configuration Changes'. Three callout boxes provide instructions: one pointing to the 'Apply Configuration Changes' bar, one pointing to the '09XXXXXXXX' dial pattern, and one pointing to the 'SIP/SIM1' trunk selection.

The prefix of route :  
09XXXXXXXX

Choose SIP Trunk

Submit 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.