

[Content]

1.INTRODUCTION1
2.FUNCTION DESCRIPTION1
3.PARTS LIST1
4.DIMENSION2
5.CHART OF THE DEVICE
6.CABLING4
7.WEB PAGE SETTING5
8.SYSTEM INFORMATION6
9. ROUTE
10.MOBILE12
11.NETWORK
12.SIP SETTING
13. NAT TRANS
14.SYSTEM AUTH
15.SAVE CHANGE
16.UPDATE
17.REBOOT
18. IP SETTING
19.SPECIFICATION40
20. APPENDIX: SETUP MV-370 WITH ASTERISK41
21.HOW TO SETUP ASTERISK TO RECEIVE CALLER ID FROM MV-37247
22. SIMPLE STEPS

1.Introduction

MV-372 is a 2 channels VoIP GSM Gateway for call termination (VoIP to GSM) and origination (GSM to VoIP). It is SIP based and compatible with Asterisk. It can enable to make 2 calls simultaneously from IP phones to GSM networks and GSM network to IP phone.

2. Function description

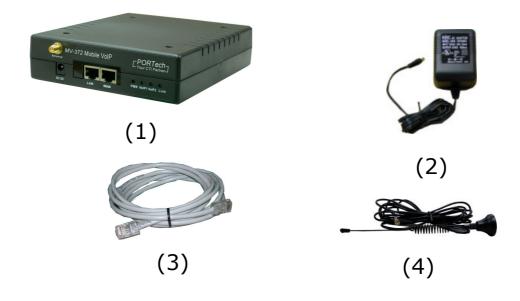
- 2.1 VoIP(SIP)
 GSM(MV-372) conversion.
- 2.2 50 sets of LAN->MOBILE routes setting [,] 50 sets of MOBILE->LAN routes setting.
- 2.3 Voice response for setting and status (dial in from mobile).
- 2.4 Series connections to save bills.
- 2.5 Standard SIP(RFC2543,RFC3261) protocol,

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 \lceil MV-372 \rfloor main body
- 3.2 Power adaptor AC-DC (110V AC 12V DC) or (220V AC 12V DC)
- 3.3 Network cable
- 3.4 Antenna
- 3.5 User Manual



4.Dimension



5.Chart of the device

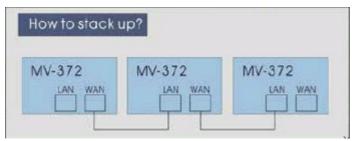


- 5.1 Antenna : Antenna connector.
- 5.2 DC 12V : Power input.
- 5.3 LAN : LAN port. It also can be DHCP Server.
- 5.4 WAN: RJ-45 internet connector [,] standard RJ-45 socket [,] connect to HUB.
- 5.5 PWR (Power LED) : Light up when power is normal.
- 5.6 VoIP1 : an indicator light of VoIP1
- 5.7 VoIP2 : an indicator light of VoIP2
- 5.8 LINK Indicator : Light up when network is connected.

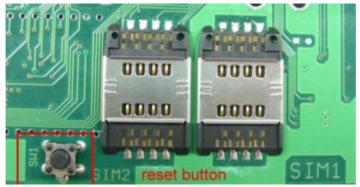
6.CABLING

6.1 Connect the internet cable from HUB to the 'WAN' connector of the MV-372.

*If you need to stack up more MV-372, you can stack up as follows.



- 6.2 Connect the antenna and put it in proper position to get the best signal reception.
- 6.3 Insert the SIM card from back of the main body. (take the slide off first).
- 6.4 <u>Click reset button 3 sec. MV-372 will restore default IP. Other</u> setting as usual.



6.5 Connect the power adaptor. The 'POWER' LED should be light up.

7.Web Page Setting

When the IP setting is done, the operator may setup all the rest parameters via web page. Browse the IP address from Internet Explorer (e.g. <u>http://192.168.0.100</u>) \circ The following page shows up :

Login PORTech VolP			
Enter your username and password to login			
	VoIP server		
Username			
Password			
	Login Clear Remember last login		

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

8.System Information.

- 8.1 When you login the web page, you can see the demo system current system information like firmware version, company... etc in this page.
- 8.2 Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.

PORTech Your CTI Partner
Route
Mobile
Network
SIP Settings
NAT Transform
Update
System Authority
Save Change
Reboot

Mobile VoIP2 v6.691d

Model Name:	MV-372
Model Description:	GSM:900/1800MHz
Firmware Version:	Fri May 16 11:30:35 2008.
Codec Version:	Mon Jul 24 10:55:05 2006.

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9. Route

Important:

The route table -50 sets can share by two channels The setting,please refer 10.2 Mobile setting ex: Mobile 1 use the route table for item 0-24, Mobile 2 use the route table for item 25-49

9.1 Mobile TO LAN Settings

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

For the second s	Mobile	Го LAN	Table	
Route	Page: 1 💌			
Mobile To Lan Settings	ltem	CID	URL	Select
Mobile To Lan Speed Dia Lan To Mobile Settings	0	*	*	
Mobile	1			
	2			
Network	3			
SIP Settings	4 5			
NAT Transform	5			
Update	7			
System Authority	8			
Save Change	9			
Reboot				
	Delete Select	ed 🔹 🛛 Delete Al	ll reset	
	Add New			
	Position:		(0~49)	
	CID:		Ex:0911111111, 0911*	*
	URL:		Ex:192.168.0.1, *:2St	
	Add reset			

The MV-372 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-372 have register proxy server/Asterisk

The proxy server/Asterisk have the route "09"

When the caller's prefix number is 0932,MV-372 will connect 0911123456 automaticlly

(2) Mobile to Lan: *,*

Any caller call the MV-372's sim, MV-372 will prompt dial tone. Caller can enter IP or sip extension or phone number.

*sip extension or phone number both need to register SIP Proxy Server or Asterisk.

*Phone number, SIP Proxy Server or Asterisk need to set the route of this phone number.

9.2 Mobile to LAN Speed Dial Settings

When you set Mobile to LAN Speed Dial Settings and Mobile to LAN at the same time, MV-372 will give priority to Mobile to LAN Speed Dial Settings.

Your CTI Partner	Mobile To LAN Speed Dial			
Route	ltem	Name	URL	Select
	0	Test	192.168.0.107	
Mobile To Lan Settings Mobile To Lan Speed Dial	1			
Lan To Mobile Settings	2			
Mobile	3			
	4			
Network	5			
SIP Settings	6			
NAT Transform	7			
Update	8			
System Authority	9			
Save Change	Delete	Selected	Delete All Reset	
Reboot	Delete	Delected	Delete All	

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

E.g Num:0 Name:test URL:192.168.0.107

When the caller hear dial tone and enter 0, system will connect 192.168.0.107

9.3 LAN to Mobile Settings

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

PORTech Your CTI Partner	LAN To Mobile Table			
Route	Page: 1	1		
Mobile To Lan Settings	ltem	URL	Call Num	Select
Mobile To Lan Speed Dial Lan To Mobile Settings	D	*	#	
Mobile	1			
	2			
Network	3			
SIP Settings	4			
NAT Transform	5			
	6			
Update	7			
System Authority	8			
Save Change	9			
Reboot				
	Delete	Selected Delete /	All Reset	

The MV-372 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 091111111#
- 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-372 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-372 will connect this call auto.

Example of Application:

When you call the ch.1 MV-372 gsm number, it will provide dial tone and you enter a destination number.

Then ch.2 MV-372 will dial this number and connect.

ch.1 MV-372: mobile to lan set route table *,*

ch.2 MV-372:lan to mobile set route table *,#

Additionally, two channels MV-372 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-372.

*The channel 2 MV-372's ip: the first ip + :5062 (e.g http://192.168.0.100:5062)

10.Mobile

10.1 Mobile Status

Your CTI Partner	Mobile Status
	2008-05-16 18:10
Route Mobile	Mobile 1 💌
Status	Network Registration.:
Settings Fwd Settings	SIM Card ID: 8
SMS Agent	Signal Quality.:
Network	GSM S/N:
SIP Settings	
NAT Transform	Incoming IP:
	Incoming IP Name:
Update	Outgoing IP:
System Authority	Incoming Mob:
Save Change Reboot	Outgoing Mob:

(1)Network Registration : The telecom carrier which the SIM card been registered.

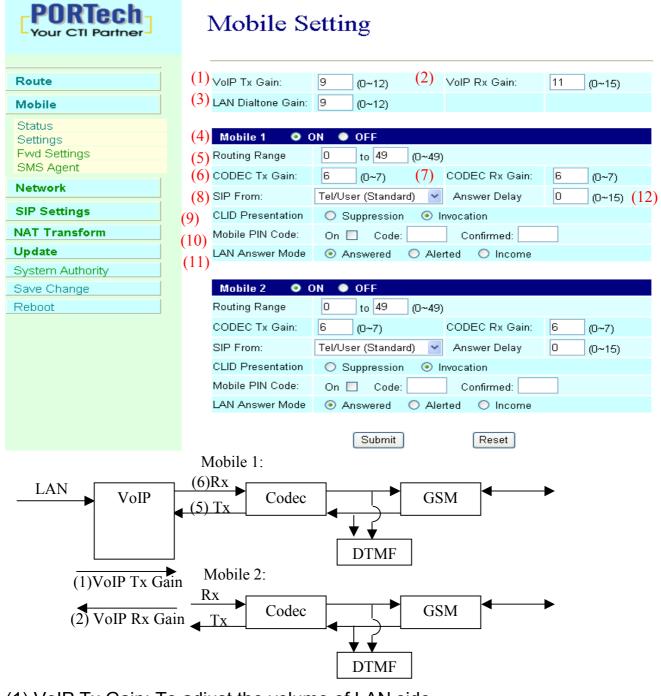
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8988***************

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

10.2 Mobile Setting



- (1) VoIP Tx Gain: To adjust the volume of LAN side.
- (2) VoIP Rx Gain: To adjust the volume of Mobile side.

- (3)LAN Dialtone Gain: DTMF Reciver is not good,you can adjust gain down.
- (4) ON/Off: If you use this channel, please click on. Otherwise, please click off.
- (5)Routing Range: The route table -50 sets can share by two channels

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

Mobile 2 use the route table for item 25-49

(6)CODEC Tx Gain: as above

(7)CODEC Rx Gain: as above

- (8) 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-372 (page 42)

MV-372 will send the message as follows in the Packet.

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

• Tel/Tel :

MV-372 will send the message as follows in the Packet.

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

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

User/Tel

MV-372 will send the message as follows in the Packet.

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

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

- (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-372, 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)Answer Delay: Delay for incoming call when the ring.

(13)When you buy Quad band, you need to choose your GSM frequency

10.3 Mobile / Forward Setting :

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

Your CTI Partner	Forward	Setting	
Route	🔲 Forward Enab	ole	
Mobile		Name	URL:Port
Status	Fwd to Mobile1:		192.168.0.100:5060
Settings Fwd Settings	Fwd to Mobile2:		192.168.0.100:5062
SMS Agent	Fwd to External:		
Network			
SIP Settings		submit cancel	
NAT Transform			
Update			
System Authority			
Save Change			
Reboot			

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

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

General	Initialization	SIP Proxy
Advanced	DTMF) stun
Accept redirection	on replies	
Use short <u>h</u> eader	rs 🖌 a	
✓ Expose software	version	
<u>U</u> se obsolete tra	nsfer mechanism (BYE/	Also)
Dootrigt college id		
different vendor	entity (support varies fo s)	or proxies from
different vendor	s) :tatus messages (otherwi	
different vendor – Use "standard" s	s) tatus messages (otherwi packets)	
different vendor – Use "standard" s taken from SIP j joice mail number	s) tatus messages (otherwi packets)	se messages will be
different vendor – Use "standard" s taken from SIP j <u>J</u> oice mail number	s) status messages (otherwi packets) or address:	se messages will be

	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.

10.4 Mobile / SMS Agent :

PORTech Your CTI Partner	SMS A	Agent Read received SMS
Route	Port	Status Bank
	Mobile 1	Standby. Rx List
Mobile	Mobile 2	Standby. Rx List
Status Settings		SMS Sender
Fwd Settings	Via	Mobile 1 2
SMS Agent		
Network	Dest Num	
SIP Settings		Maximum Number of UCS2 chars for this text box is 70.
NAT Transform	Message	
Update	Ů	
System Authority		You have 70 UCS2 chars remaining for your description
Save Change		
Reboot		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

Read	Status	RemoteID	Date,Time
1	REC READ	886936114545	08/01/01,19:34:22
2	REC READ	886935386862	08/03/12,16:25:27

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

SMS Reader

Index	RemoteID	Date,Time
2	886935386862	08/03/12, 16:25:27
M	/ Serial can send SMS and recei	ve SMS
		*
	Back	Delete

11.Network

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

11.1 Network Status: You can check the current Network setting in this page.

PORTech Your CTI Partner	Network Status		
Route	Ethernet 0	WAN Interface	LAN Interface
Mobile	Туре	Fixed IP Client	Fixed IP Client
	IP	192.168.0.122	192.168.0.102
Network	Mask	255.255.255.0	255.255.255.0
Status	Gateway	192.168.0.254	192.168.0.254
WAN Settings	MAC	00037E009999	00037E008888

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

- (1) The TCP/IP Configuration item is to setup the WAN port's network environment. You may refer to your current network environment to configure the system properly.
- (2) The PPPoE Configuration item is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.
- (3) The Bridge Item is to setuo the system Bridge mode Enable/Disable. If you set the Bridge On, then the two Fast Ethernet ports will be transparent.
- (4) When you finished the setting, please click the Submit button.

Your CTI Partner	WAN Settings		
	You could configu	You could configure the WAN settings in this page.	
Route			
Mobile	Network Mode:	○ Bridge	
Network	WAN Setting		
Status	IP Type	● Fixed IP ○ DHCP Client ○ PPPoE	
WAN Settings	IP	192.168.0.122	
LAN Settings	Mask	255.255.255.0	
SNTP Settings	Gateway	192.168.0.254	
SIP Settings	DNS Server1	168.95.192.1	
NAT Transform	DNS Server2	168.95.1.1	
Update	MAC	00037e009999	
System Authority			
Save Change	PPPoE Setting		
Reboot	User Name		
	Password		
		Submit Reset	

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

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

PORTech Your CTI Partner
Route
Mobile
Network
Status WAN Settings LAN Settings SNTP Settings
SIP Settings
NAT Transform
Update
System Authority
Save Change
Reboot

LAN Settings

LAN Setting	
IP:	192.168.0.102
Mask:	255.255.255.0
MAC:	00037e008888

DHCP Server	
DHCP Server:	On ⊙Off
Start IP:	150
End IP:	200
Lease Time:	1 : 0 (dd:hh)

Submit Reset

11.4 SNTP Settings:

SNTP Setting function: you can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.

PORTech Your CTI Partner
Route
Mobile
Network
Status WAN Settings LAN Settings SNTP Settings
SIP Settings
NAT Transform
Update
System Authority
Save Change
Reboot

SNTP Settings

You could set the SNTP servers in this page.

SNTP:	⊙ On 🔘 Off
Primary Server:	time.windows.com
Secondary Server:	208.184.49.9
Time Zone:	GMT 📴 😶 🛛 🕶 : 👓 💌 (hh:mm)
Sync. Time:	1 : 0 : 0 (dd:hh:mm)
	Submit Reset

12.SIP Setting

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

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

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

- (1)No.,: choose Mobile 1 or Mobile 2
- (2) Display name: you can input the name you want to display.
- (3) User name: you need to input the User Name get from your ISP.
- (4) Register Name: you need to input the Register Name get from your ISP.
- (5) Register Password: you need to input the Register Password get from ISP.
- (6) Domain Server:you need to input the Domain Server get from your ISP.
- (7) Proxy Server: you need to input the Proxy Server get from your ISP.
- (8) Outbound Proxy: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.
- (9) You can see the Register Status in the Status item.
- (10) When you finished the setting, please click the Submit button. Remember to click "Save Charge"

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
NAT Transform

Service Domain Settings

Mobile 1 💌	
Realm 1 (Default)	
Active:	⊙ ON ○ OFF
Display Name:	3001
User Name:	3001
Register Name:	3001
Register Password:	••••
Domain Server:	
Proxy Server:	61.218.151.230
Outbound Proxy:	
Status:	Not Registered

Example: Register VoipBuster

Realm 1 (Default)	
Active:	⊙ On C 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 Port Setting

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

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
NAT Transform
Update
System Authority
Save Change
Reboot

Ports Setting

Port of Mobile 1			
SIP Port:	5060 (1024~65535)		
RTP Port:	60000 (1024~65535)		

Port of Mobile 2			
SIP Port:	5062	(1024~65535)	
RTP Port:	60100	(1024~65535)	

Submit Reset

12.3 Codec Settings:

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



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.711 & G.729:	20 ms 💙			
G.723:	30 ms 💌			
G.723 5.3K				
G.723 5.3K:	🔘 On 💿 Off			
Voice VAD				
Voice VAD:	🔘 On 💿 Off			

12.4 Codec ID Setting

You can setup the Codec ID in this page.

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
NAT Transform
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:	23 (95~255)	23
G726-24 ID:	22 (95~255)	22
G726-32 ID:	2 (95~255)	2
G726-40 ID:	21 (95~255)	2 1
RFC 2833 ID:	101 (95~255)	101

Submit Reset

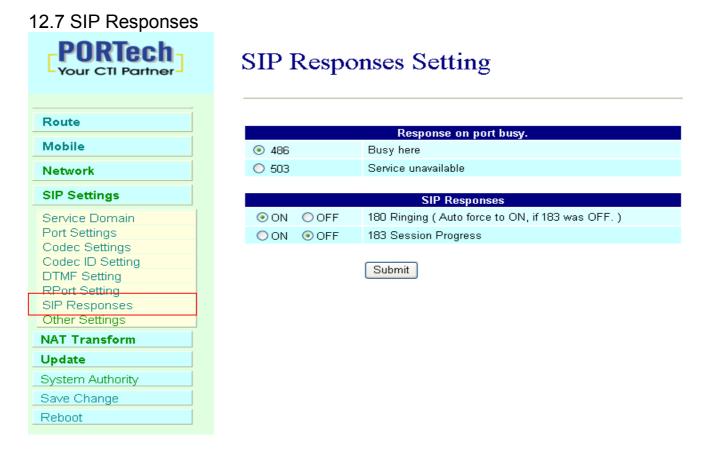
12.5 DTMF Setting You can setup the DTMF Setting in this page.

Your CTI Partner	DTMF Setting		
Route			
Mobile	Mobile DTMF Transfer to Lan • 2833		
Network	Inband DTMF		
	Send DTMF SIP Info		
SIP Settings			
Service Domain	Mobile DTMF debounce: 80 (range:40~200, default:80) step:10ms.		
Port Settings			
Codec Settings	Submit Reset		
Codec ID Setting DTMF Setting			
RPort Setting			
SIP Responses			
Other Settings			
NAT Transform			
Update			
System Authority			
Save Change			
Reboot			

12.6 RPort Function:

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

PORTech Your CTI Partner	RPort Set	ting
Route		
	RPort of Mobile 1:	💿 On 🔘 Off
Mobile	RPort of Mobile 2:	💿 On 🔘 Off
Network		Submit Reset
SIP Settings		Submit Reset
Service Domain		
Port Settings		
Codec Settings		
Codec ID Setting		
DTMF Setting RPort Setting		
SIP Responses		
Other Settings		
NAT Transform		
Update		
System Authority		
Save Change		
Reboot		



12.7.1 486(busy here), 503(Service unavailable): When Device are busying, you can select 486 or 505 to response to SIP.

12.7.2 180 Ring on/off: LAN TO MOBILE two stage dialing can be turn off, therefore there will be no the Ring Back Tone, all the phone call will be transferred to Voice-Mail directly. (For this function, 183 must be turn on)

12.7.3 183(Session Progress)-->[It means"on progressing"] : When you turn 183 on, it means you can hear voicemail while GMS side are busying. We recommend you to turn this on if you use SIP Proxy.

12.8 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.

	Your CTI Partner	Other Settin
	Route	Hold by RFC of Mobile 1
	Mobile	Hold by RFC of Mobile 2
	Network	Voice QoS:
	SIP Settings	SIP QoS:
	Service Domain Port Settings Codec Settings Codec ID Setting DTMF Setting RPort Setting SIP Responses Other Settings	SIP Expire Time:
1	NAT Transform	
	Update	
	System Authority	
	Save Change	
	Reboot	

igs

Hold by RFC of Mobile 1	🔿 On 💿 Off
Hold by RFC of Mobile 2	🔿 On 💿 Off
Voice QoS:	40 (0~63)
SIP QoS:	40 (0~63)
SIP Expire Time:	300 (60~86400 sec)

Submit

Reset

13. NAT Trans

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

13.1 STUN Setting: you can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your VoIP device working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

PORTech Your CTI Partner	
Route	
Mobile	
Network	
SIP Settings	
NAT Transform	
STUN Setting	
Update	
System Authority	
Save Change	
Reboot	

STUN Setting

STUN of Mobile 1	🔿 On 💿 Off	
STUN of Mobile 2	🔿 On 💿 Off	
STUN Server	stun.xten.com	
STUN Port	3478 (1024~65535)	
	Submit Reset	

14.System Auth.

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

PORTech Your CTI Partner	System Aut	hority
	You could change the log	in username/password in this page.
Route		
Mobile	New username:	
Network	New password:	
Network	Confirmed password:	
SIP Settings		Submit Reset
NAT Transform		
Update		
System Authority		
Save Change		
Reboot		

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.

For CTI Partner	Save Changes
	You have to save changes to effect them.
Route	
Mobile	Save Changes: Save
Network	
SIP Settings	
NAT Transform	
Update	
New Firmware	
Default Settings	
System Authority	
Save Change	
Reboot	

16.Update

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

- 16.1 Update firmware
- (1) In New Firmware function you can update new firmware via HTTP in this page. You can upgrade the firmware by the following steps:
- (2)Select the firmware code type, Risc code.
- (3)Click the "Browse" button in the right side of the File Location or you can type the correct path and the filename in File Location blank.
- (4)Select the correct file you want to download to the system then click the Update button.
- (5) Please click update/default setting after update firmware

Your CTI Partner	Update	Update Firmware		
	You could updat	You could update the newest firmware. PCB mark: 2K123B		
Route				
Mobile				
Network	Method:	● HTTP ○ TFTP		
SIP Settings	НТТР			
NAT Transform	Code Type:	Risc 💌		
Update	File Location:	瀏覽		
New Firmware	TFTP			
Default Settings	TFTP Server:	192.168.1.250		
System Authority				
Save Change		Update Reset		
Reboot				

16.2 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.

PORTech Your CTI Partner	Restore Default Settings You could click the restore button to restore the factory settings.
Route	
Mobile	Restore default settings: Restore
Network	
SIP Settings	
NAT Transform	
Update	
New Firmware	
Default Settings	
System Authority	
Save Change	
Reboot	

17.Reboot

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

PORTech Your CTI Partner	Reboot System
	You could press the reboot button to restart the system.
Route	
Mobile	Reboot system: Reboot
Network	
SIP Settings	
NAT Transform	
Update	
System Authority	
Save Change	
Reboot	

18. IP Setting

The operator can setup or query the network parameters by dialing in the mobile number which it SIM card has been put in the main body. The status or result is response by voice. In the first 20 seconds after power-on, the VoIP GSM Gateway enters the IP setting mode. The operator may dial in the mobile number during this period to set or query the network parameters.

Item	IVR Action	IVR Menu Choice	Notes		
1	Reboot	#195#	After you hear "Option Successful," hang-up. Unit w reboot automatically.		
2	Factory Reset	#198#	System will automatically Reboot.WARNING: ALL User-Changeable" NONDEFAULT SETTINGS WILL BE LOST! This will include network and service provider data.		
3	Check IP Address	#120#	IVR will announce the current IP address , Default : 192.168.0.100		
4	Check IP Type	#121#	IVR will announce if DHCP in enabled or disabled. default : OFF		
5	Check Network Mask	#123#	IVR will announce the current network mask.Default : 255.255.255.0		
6	Check Gateway IP Address	#124#	IVR will announce the current gateway IP address, Default : 192.168.0.254		
7	Check Primary	#125#	IVR will announce the current		

	DNS Server		setting in the Primary DNS field.		
			Default : 192.168.0.1		
8	Check Firmware	#128#	IVR will announce the version		
	Version		of the firmware running		
9	Set as DHCP	#111#	The system will change to		
	client		DHCP		
			Client type		
10	Set Static IP		DHCP will be disabled and		
	Address	*xxx#	system will change to the		
			Static IP type.		
			Enter IP address using		
			numbers on the telephone key		
			pad. Use the * (star) key when		
11	Cat Natwork Maak	#110,00/*,00/*,00/	entering a decimal point.		
	Set Network Mask				
		*xxx#	Enter value using numbers on		
			the telephone key pad. Use		
			the * (star) key when entering		
40		<i>11444</i> + +	a decimal point.		
12	Set Gateway IP	#114xxx*xxx*xxx			
	Address	*xxx#	Enter IP address using		
			numbers on the telephone key		
			pad. Use the * (star) key		
			when entering a decimal		
			point.		
13	Set Primary DNS	#115xxx*xxx*xxx	Must set Static IP first.		
	Server	*xxx#	Enter IP address using		
			numbers on the telephone key		
			pad. Use the * (star) key		
			when entering a decimal		
			point.		

19.Specification

19.1 Protocols SIP (RFC2543, RFC3261) 19.2 TCP/IP IP/TCP/UDP/RTP/RTCP/ CMP/ARP/RARP/SNTP **DHCP/DNS** Client **IEEE802.1P/Q** ToS/DiffServ NAT Traversal STUN uPnP **IP** Assignment Static IP DHCP **PPPoE** 19.3 Codec G.711 u-Law G.711 a-Law G.723.1 (5.3k) G.723.1 (6.3k) G.729A G.729A/B 19.4 Voice Quality VAD

CNG AEC, LEC Packet loss

19.5 GSM (MV-372)

Dual BAND: 900/1800 MHZ Tri BAND(BenQ M23): 900/1800/1900 MHZ Tri BAND(Siemens MC56): 850/1800/1900 MHZ Quad BAND: 900/1800/1900/850 MHZ

20. Appendix: Setup MV-372 with Asterisk

20.1 Usage

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

Your mobile <----*gsm network*----> MV-372 <--*lan*--> Asterisk <--*internet*--> VOIP provider <--*whatever*--> landline

To do such a call, you just call your MV-372 number (it has its own simcard), then you get an invitation tone, then you dial the number which is handled by Asterisk.

If you have some special deals with your mobile operator, like free special number, you can call your MV-372 for free.

You can then call all around the world from your mobile at voip cost :-)

20.2 MV-372 Configuration

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

To have the MV-372 to work with Asterisk, you need first to configure the

box.

Here are some screen shots showing all the important parameters. You have to note that in all the configuration process, the MV-372 is considered as extension '103' of the IPBX.

In **Bold** are the parameters depending on your installation

WAN Settings

You could configure the WAN settings in this page.

WAN Setting	
IP Type	● Fixed IP ○ DHCP Client ○ PPPoE
IP	MV370 IP
Mask	255.255.255.0
Gateway	Router IP
DNS Server1	168.95.192.1
DNS Server2	168.95.1.1
MAC	
PPPoE Setting	
User Name	
Password	
	Submit Reset

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

LAN To Mobile Table

Page: 1 💌	•		
ltem	URL	Call Num	Select
0	Your Asterisk IP	#	
1			
2			
3			
4			
5			
6			
7			
8			
9			

Mobile To LAN Table

Page: 1 🔽

ltem	CID	URL	Select
0	Authorised Mobile	103	
1	Another Authorised Mobile	103	
2			
3			
4			
5			
6			
7			
8			
9			

The mobile number you give in that page are the authorised mobile which can call GSM to Asterisk.

These mobile number must be defined as your GSM provider displays the number.

If you don't know how it is displayed, just give a call to the box and check the number given in the 'Incoming Mob' field of the 'Mobile Status' page. Any number which is not in that list won't have acces to the LAN side, so to Asterisk.

If you want to allow any number, just set '*' in that field ... but beware of the bill ;-)

Service Domain Settings

Realm 1 (Default)	
Active:	💿 ON 🔘 OFF
Display Name:	103
User Name:	103
Register Name:	103
Register Password:	
Domain Server:	Asterisk IP
Proxy Server:	
Outbound Proxy:	
Status:	Not Registered

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

Codec Settings

	Codec
Codec Priority 1:	G.711 u-law 💙
Codec Priority 2:	G.711 a-law 🗙
Codec Priority 3:	Not Used 🛛 🛩
Codec Priority 4:	Not Used 🛛 🛩
Codec Priority 5:	Not Used 🛛 🛩
Codec Priority 6:	Not Used 🛛 🛩
Codec Priority 7:	Not Used 🛛 🛩
Codec Priority 8:	Not Used 🛛 👻
	RTP Pack
G.711 & G.729:	20 ms 🛩
G.723:	30 ms 🛩
	G.72
G.723 5.3K:	🔘 On 💿 Off
	Voice
Voice VAD:	🔘 On 💿 Off

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

Mobile Setting

VoIP Tx Gain:	10 (0~12) VoIP Rx Gain: 3 (0~15)
LAN Dialtone Gain:	10 (0~12)
Mobile 💿 ON	I OFF
Routing Range	0 to 49 (0~49)
Routing Range CODEC Tx Gain:	0 to 49 (0~49) 6 (0~7) CODEC Rx Gain: 6 (0~7)

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

20.3 Antenna position

Another important thing is to properly place the provided antenna. If your gsm reception is good, you should get around 18 or 19 as Signal Quality in the "Mobile Status" page.

With that level of signal quality, your audio quality will be very good. On the other end,the signal quality down to 11, audio becomes very jerky. So, maximum signal quality = maximum audio quality.

20.4 Asterisk configuration

Once the MV-372 is set, you have to configure Asterisk. On that side, you have to setup files as follow :

20.5 sip.conf

; GSM VOIP Gateway MV-372 [103] type=friend

```
username=103
fromuser=103
regexten=103; When they register, create extension 401
secret=xxxxxxx ; Asterisk extension password
context=gateway; Incoming calls context
dtmfmode=inband; Very important for DISA to work
call-limit=1; Limit to 1 call max
callerid=GSM Gateway <103>
host=dynamic
nat=no; Gateway is not behind a NAT router
canreinvite=no; Typically set to NO if behind NAT
insecure=very
qualify=yes
disallow=all
allow=ulaw; prefered codec for DTMF detection
allow=alaw
```

20.6 extensions.conf

; ******* GSM Gateway incoming calls *********

[gateway]

exten => _103,1,Answer()

exten => _103,2,DigitTimeout(3) ; give enough time to do second stage dialing

exten => _103,3,ResponseTimeout(5)

exten => _103,4,DISA(no-password|outgoing) ; here 'outgoing' is the normal context to deal with the dial plan

[outgoing]

... ; example of LAN to GSM call ; call the MV-372 sim card mail box thru GSM exten => _888,1,SetCallerID("xxxxxxxxx") exten => _888,2,Dial(SIP/\${EXTEN}@103,60,r) exten => _888,3,Hangup()

21. How to setup Asterisk to receive Caller ID from MV-372

Test version trixbox-2.2

SIP Softphone

- SJPhone 1.60.289a
- X-Lite 1105x

Modify file

Add the following setting to/etc/asterisk/sip.conf
 [1000]
 type=friend
 secret=1000
 qualify=yes
 nat=yes
 host=dynamic
 canreinvite=no
 context=internal
 [1001]
 type=friend
 secret=1001
 qualify=yes

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

[1002] type=friend secret=1002 qualify=yes

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

Add the following setting to /etc/asterisk/extensions.conf
 [internal]
 exten => 1000,1,Dial(SIP/1000)
 exten => 1001,1,Dial(SIP/1001)
 exten => 1002,1,Dial(SIP/1002)

configure:

trixbox-2.2: address=192.168.66.202:5060

SJPhone: address=192.168.66.145:5060; username=1000,

displayname=user_1000

X-Lite: address=192.168.66.145:7331; username=1001, displayname=user_1001 MV-372: address=192.168.66.203:5060; username=1002, displayname=user_1002

🔿 🗸 🥫 http://192.1		Explorer n.cgi			• *+	X Live Search	- 5
		*(A) 工具(D) 説明(H)			₩ FlashGet 🔐 選項 🗸		a 🖄 🔍
🕸 🔏 VolP Web Mau		1 THO 11410	1				œ <u>ш</u>) ™ ¹ / ₁ • ⊡•• (
Voir web Mau	ragement] 🖬 . 🗠 . (
Mobile V	in	с · р	1. 0				
Moone V	<u>np</u>	Service Do	main Settings				
		You could set informati	on of service domains in this page.				
Route			on or control domains in this page.	ž			
Koute	•		-				
Mobile		No.: Mobile 1					
Network		Realm 1 (Default)					
THEINOIR	•	Active:	⊙ On O Off				
SIP Settings		Display Name:	user_1002				
NAT Trans.		User Name:	1002				
	- * I	Register Name:	1002				
System Auth.		Register Password:	••••				
Save Change		Domain Server:	192.168.66.202				
_		Proxy Server:	192.168.66.202				
Update	•	Outbound Proxy:	192.168.66.202				
Reboot		Status:	Registered				
		Realm 2					
		Active:	OOn Ooff				
		Display Name:					
		User Name:					
		Register Name:					
		Register Password:					

test1

pstn → call 0928492911(mobile number) → MV-372 → hear the second dial tone,call SoftPhone's number → SoftPhone → show pstn caller id

This Is X-Lite receiving packet, red word is pstn number. Test ok.

INVITE sip:1001@192.168.66.145:7331 SIP/2.0 Via: SIP/2.0/UDP 192.168.66.202:5060;branch=z9hG4bK3d0bbaf7;rport From: "035678238" <sip:1002@192.168.66.202>;tag=as580472a7 To: <sip:1001@192.168.66.145:7331> Contact: <sip:1002@192.168.66.202> Call-ID: 20fa417265e6a26d0b0aae4f551f06f3@192.168.66.202 CSeq: 102 INVITE User-Agent: Asterisk PBX Max-Forwards: 70 Date: Tue, 22 May 2007 02:50:37 GMT Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY Content-Type: application/sdp Content-Length: 242

v=0 o=root 2737 2737 IN IP4 192.168.66.202 s=session c=IN IP4 192.168.66.202 t=0 0 m=audio 15852 RTP/AVP 0 8 101 a=rtpmap:0 PCMU/8000 a=rtpmap:8 PCMA/8000 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-16 a=silenceSupp:off - - - - SIP/2.0 200 Ok Via: SIP/2.0/UDP 192.168.66.202:5060;branch=z9hG4bK3d0bbaf7;rport From: "035678238" <sip:1002@192.168.66.202>;tag=as580472a7 To: <sip:1001@192.168.66.145:7331>;tag=677373503 Contact: <sip:1001@192.168.66.145:7331> Call-ID: 20fa417265e6a26d0b0aae4f551f06f3@192.168.66.202 CSeq: 102 INVITE Content-Type: application/sdp Server: X-Lite release 1105x Content-Length: 254

v=0 o=1001 4804366 4807851 IN IP4 192.168.66.145 s=X-Lite c=IN IP4 192.168.66.145 t=0 0 m=audio 8000 RTP/AVP 0 8 3 101 a=rtpmap:0 pcmu/8000 a=rtpmap:8 pcma/8000 a=rtpmap:3 gsm/8000 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 a=sendrecv

test 2

SoftPhone \rightarrow call 1002 \rightarrow MV-372 \rightarrow hear second dial tone and call pstn \rightarrow pstn answer \rightarrow show caller id-mobile number 0928492911

This Is X-Lite receiving packet. Test ok.

INVITE sip:1002@192.168.66.202 SIP/2.0

Via: SIP/2.0/UDP 192.168.66.145:7331;rport;branch=z9hG4bK4C4315351FC84CA582D14FB8C25F C3BF From: user 1001 <sip:1001@192.168.66.202:7331>;tag=1121869743 To: <sip:1002@192.168.66.202> Contact: <sip:1001@192.168.66.145:7331> Call-ID: F4B32CA6-1835-4E68-941A-C685B39C43FF@192.168.66.145 CSeq: 63148 INVITE Proxy-Authorization: Digest username="1001",realm="asterisk",nonce="0d3b2879",response="8aaaaa5b5ad53 654bf0a2ab0fa9bb118",uri="sip:1002@192.168.66.202",algorithm=MD5 Max-Forwards: 70 Content-Type: application/sdp User-Agent: X-Lite release 1105x Content-Length: 254 v=0 o=1001 5111461 5111501 IN IP4 192.168.66.145 s=X-Lite c=IN IP4 192.168.66.145 t=0 0 m=audio 8000 RTP/AVP 0 8 3 101 a=rtpmap:0 pcmu/8000

a=rtpmap:8 pcma/8000 a=rtpmap:3 gsm/8000 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 a=sendrecv

SIP/2.0 200 OK Via: SIP/2.0/UDP 192.168.66.145:7331;branch=z9hG4bK4C4315351FC84CA582D14FB8C25FC3BF ;received=192.168.66.145;rport=7331 From: user_1001 <sip:1001@192.168.66.202:7331>;tag=1121869743 To: <sip:1002@192.168.66.202>;tag=as2a2fbf98 Call-ID: F4B32CA6-1835-4E68-941A-C685B39C43FF@192.168.66.145 CSeq: 63148 INVITE User-Agent: Asterisk PBX Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY Contact: <sip:1002@192.168.66.202> Content-Type: application/sdp Content-Length: 242

```
v=0
o=root 2737 2737 IN IP4 192.168.66.202
s=session
c=IN IP4 192.168.66.202
t=0 0
m=audio 13798 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
```

```
register issue
The packet date from Asterisk as follows.
Please note, user_1002's display name don't appear
So the website's Display Name is not available
```

<-- SIP read from 192.168.66.203:5060: REGISTER sip:192.168.66.202 SIP/2.0 Via: SIP/2.0/UDP 192.168.66.203:5060;rport;branch=z9hG4bK590e92b551233a10a0ae71944c19b5 aa From: <sip:1002@192.168.66.202>;tag=4e36d8f1 To: <sip:1002@192.168.66.202> Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203 Contact: <sip:1002@192.168.66.203:5060> CSeq: 10 REGISTER Expires: 300 Authorization: Digest username="1002",realm="asterisk",nonce="3ca93a1e",response="4d39ccb0dae64 bb2f1341e9896ac1ea7",uri="sip:192.168.66.202",algorithm=MD5 User-Agent: CMI CM5K Content-Length: 0

--- (11 headers 0 lines) ---Using latest REGISTER request as basis request Sending to 192.168.66.203 : 5060 (NAT) Transmitting (NAT) to 192.168.66.203:5060: SIP/2.0 100 Trying Via: SIP/2.0/UDP 192.168.66.203:5060;branch=z9hG4bK590e92b551233a10a0ae71944c19b5aa;rec eived=192.168.66.203;rport=5060 From: <sip:1002@192.168.66.202>;tag=4e36d8f1 To: <sip:1002@192.168.66.202> Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203 CSeq: 10 REGISTER User-Agent: Asterisk PBX Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY Contact: <sip:1002@192.168.66.202> Content-Length: 0

Transmitting (NAT) to 192.168.66.203:5060: SIP/2.0 401 Unauthorized

Via: SIP/2.0/UDP 192.168.66.203:5060;branch=z9hG4bK590e92b551233a10a0ae71944c19b5aa;rec eived=192.168.66.203;rport=5060 From: <sip:1002@192.168.66.202>;tag=4e36d8f1 To: <sip:1002@192.168.66.202>;tag=as13a32ae8 Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203 CSeq: 10 REGISTER User-Agent: Asterisk PBX Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY WWW-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="5def9231" Content-Length: 0

Scheduling destruction of call

'7e45b773130f1fc945efcee502f84042@192.168.66.203' in 15000 ms

asterisk1*CLI>

<-- SIP read from 192.168.66.203:5060:

REGISTER sip:192.168.66.202 SIP/2.0

Via: SIP/2.0/UDP

192.168.66.203:5060; rport; branch=z9hG4bK672fa67f59c2223275f5ee286d27597a

From: <sip:1002@192.168.66.202>;tag=4e36d8f1

To: <sip:1002@192.168.66.202>

Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203

Contact: <sip:1002@192.168.66.203:5060>

CSeq: 11 REGISTER

Expires: 300

Authorization: Digest

username="1002",realm="asterisk",nonce="5def9231",response="046a412f4e7ed4

e98fd507416994a80a",uri="sip:192.168.66.202",algorithm=MD5

User-Agent: CMI CM5K

Content-Length: 0

---- (11 headers 0 lines) ----

Using latest REGISTER request as basis request

Sending to 192.168.66.203 : 5060 (NAT)

Transmitting (NAT) to 192.168.66.203:5060:

SIP/2.0 100 Trying

Via: SIP/2.0/UDP

192.168.66.203:5060;branch=z9hG4bK672fa67f59c2223275f5ee286d27597a;recei

ved=192.168.66.203;rport=5060

From: <sip:1002@192.168.66.202>;tag=4e36d8f1

To: <sip:1002@192.168.66.202>

Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203

CSeq: 11 REGISTER

User-Agent: Asterisk PBX

Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY

Contact: <sip:1002@192.168.66.202>

Content-Length: 0

12 headers, 0 lines

Reliably Transmitting (NAT) to 192.168.66.203:5060:

OPTIONS sip:1002@192.168.66.203:5060 SIP/2.0

Via: SIP/2.0/UDP 192.168.66.202:5060;branch=z9hG4bK7b92dd8a;rport

From: "Unknown" <sip:Unknown@192.168.66.202>;tag=as5dee3942

To: <sip:1002@192.168.66.203:5060>

Contact: <sip:Unknown@192.168.66.202>

Call-ID: 5ebc2211278e2cb7699911ad39454d4e@192.168.66.202

CSeq: 102 OPTIONS

User-Agent: Asterisk PBX

Max-Forwards: 70

Date: Tue, 22 May 2007 03:11:54 GMT

Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY Content-Length: 0 ---

Transmitting (NAT) to 192.168.66.203:5060: SIP/2.0 200 OK Via: SIP/2.0/UDP 192.168.66.203:5060;branch=z9hG4bK672fa67f59c2223275f5ee286d27597a;recei ved=192.168.66.203;rport=5060 From: <sip:1002@192.168.66.202>;tag=4e36d8f1 To: <sip:1002@192.168.66.202>;tag=as13a32ae8 Call-ID: 7e45b773130f1fc945efcee502f84042@192.168.66.203 CSeq: 11 REGISTER User-Agent: Asterisk PBX Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY Expires: 300 Contact: <sip:1002@192.168.66.203:5060>;expires=300 Date: Tue, 22 May 2007 03:11:54 GMT Content-Length: 0

22. Simple Steps

Step 1. Change the Network setting if you need (Network/network setting)

Step 2. Register SIP proxy Server or Asterisk or VoipBuster if you need (sip setting/service domain)

Step 3. Set Route (request)

mobile to lan:

(1)	*,*>it is two stage dialing.
	when mobile call in,MV-372 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-372 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.
	n to Mobile:
(1)	*,*>it is two stage dialing.
	when lan phone call in,MV-372 will provide dial tone and you can enter mobile number.
(2)	*, specific mobile number
	when lan phone call in,MV-372 will connect with the specific mobile number auto.
(3)	*,#>It is 1 stage dialing
	When lan phone and MV-372 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-372
* Al	I changes both need to click "save and change"

All changes both need to click "save and change"