

[Content]

.1
.1
.1
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.3
.4
.5
.5
12
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42
43
8 50
59

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 \lceil MV-374/MV-378 \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: MV-374:1 pcs / MV-378: 2 pcs
- 3.5 Rackmount (compatible with 19"Rack) option
- 3.6 User Manual



(3.1) MV-374



(3.2) MV-374



(3.1) MV-378



(3.2) MV-378



(3.3)



(3.5)-option

4.Dimension : 30x28x4 cm



(3.4)

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.

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. <u>http://192.168.0.100</u>) \circ The following page shows up :

Login VolP			
Enter your username and password to login			
	VoIP server		
Username			
Password			
	Login 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.

PORTech Your CTI Partner	Mobile VoIP8 v6.691a		
Route		14/270	
Mobile	Model Name: Model Description:	MV-378 GSM:900/1800/1900MHz	
	Firmware Version:	Thu May 15 14:50:55 2008.	
Network	Codec Version:	Mon Jul 24 10:55:05 2006.	
SIP Settings			
NAT Transform			
Update			
System Authority		© 2007 PORTech Communications Inc.	
Save Change			
Reboot			

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.

PORTech Your CTI Partner	Mobile To LAN Table
	Mobile 1, 2 💌
Route	Page: 1 💌
Mobile	Page: 1
Network	Item CID URL Select
SIP Settings	
NAT Transform	
Update	3
System Authority	4
Save Change	5
Reboot	6
	7
	8
	9
	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 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.

Your CTI Partner	Mobile To LAN Speed Dial				
	Mobile 1, 2	2 🕶			
Route					0.1
Mobile To Lan Settings	ltem O	Name	UI	(L	Select
Mobile To Lan Speed Dial Lan To Mobile Settings	1				
Mobile	2				
	3				
Network	4				
SIP Settings	5				
NAT Transform	6				
Update	7				
System Authority	8				
Save Change	9				
Reboot	Delete	Selected	Delete All Reset		
	Delete				
	Add New I	Phone			
	Position:	0)	~9)		
	Name:				
	URL:				
	Add Re	set			

*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

8.3 LAN to Mobile Settings

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

Your CTI Partner	LAN To Mobile Table		
	Mobile 1, 2 💌		
Route			
Mobile To Lan Settings Mobile To Lan Speed Dial Lan To Mobile Settings	Page: 1 VIL Call Num	Select	
Mobile	1		
Network	2		
SIP Settings	3		
NAT Transform	4		
Update	5		
System Authority	6		
Save Change	7		
Reboot	8		
	Delete Selected Delete All Reset		
	Position: (0~49)		
	URL: Ex:192.168.0.1, 192.168.0.* Call Ex:0911, *:2St, #, #d?, #d?A: 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. 091111111 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-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.

MV-374/MV-378's IP:

The channel 1:192.168.0.100:5060 The channel 2:192.168.0.100:5062 The channel 3:192.168.0.102:5060 The channel 4:192.168.0.102:5062 The channel 5:192.168.0.104:5060 The channel 6:192.168.0.104:5062 The channel 7:192.168.0.106:5060

9.Mobile

9.1 Mobile Status

PORTech Your CTI Partner	Mobile Status		
	2008-05-15 17:13		
Route	Mobile 1 🗸		
Mobile			
Status	Network Registration.:	Chunghwa Telecom LDM	
Settings Fwd Settings	SIM Card ID:	144,0,98889602200752095822	
SMS Agent	Signal Quality.:	27	
Network	GSM S/N:	IMEI: 35815600782656-1	
SIP Settings			
NAT Transform	Incoming IP:		
	Incoming IP Name:		
Update System Authority	Outgoing IP:		
System Authority Save Change	Incoming Mob:		
Reboot	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

Your CTI Partner	Mobile S	etting	
	Mobile 1, 2 💌		
Route			
Mobile	(1) VoIP Tx Gain:	9 (0~12) (2) VolP Rx Gain:	11 (0~15)
Status	 (1) VoIP Tx Gain: (3) LAN Dialtone Gain: 	9 (0~12)	
Settings Fwd Settings			
SMS Agent	Mobile 1 💿 (DN OFF	
Network	(4) Routing Range	0 to 24 (0~49)	
SIP Settings	(5) CODEC Tx Gain:	6 (0~7) (6) CODEC Rx Gain:	6 (0~7)
NAT Transform	(7) SIP From:	Tel/User (Standard) 🛛 Answer Delay	0 (0~15) (8)
Update	(9) CLID Presentation	O Suppression 💿 Invocation	
System Authority	$(10)^{Mobile}$ PIN Code:	On 🗖 Code: Confirmed:	
Save Change	(11)LAN Answer Mode	💿 Answered i O Alerted 💿 Income	
Reboot	(12) Mobile 2 • (
	(12) Mobile 2 • (Routing Range		
	CODEC Tx Gain:		6 (0~7)
	SIP From: CLID Presentation	Tel/User (Standard) 🖌 Answer Delay	0 (0~15)
	Mobile PIN Code:	Suppression Invocation	
	LAN Answer Mode	On Code: Confirmed: On Answered Alerted Income	
	E 447 Alowel Mode		
		Submit Reset	
	Mobile 1:		
	(6)Rx		
LAN VoIP	(5) Tx Co	dec GSM	▶
	→ Mobile 2:	DTMF	
(1)VoIP Tx G	ain Rx		
		dec GSM	
(2) VoIP Rx G	4 1x		
		* *	
		DTMF	

- (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)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 demain • 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 demain
- (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 Ian dial out, then connect soon

- (12) 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).

PORTech Your CTI Partner	Forward Settin	ıg	
Route			
Mobile	Forward Enable		
Status Settings Fwd Settings SMS Agent	Na Fwd to Mobile1: Fwd to Mobile2:	me	URL:Port
Network	Fwd to External:		
SIP Settings NAT Transform Update	submit	cancel	
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.)

Advanced DTMF STUN Accept redirection replies Use short headers Expose software version Use obsolete transfer mechanism (BYE/Also) Restrict caller identity (support varies for proxies from different vendors) Use "standard" status messages (otherwise messages will b taken from SIP packets)	General	Initialization	SIP Proxy
Use short <u>h</u> eaders Expose software version Use obsolete transfer mechanism (BYE/Also) <u>Restrict caller identity (support varies for proxies from</u> different vendors) Use "standard" status messages (otherwise messages will b taken from SIP packets)	Advanced	DTMF	STUN
 Expose software version Use obsolete transfer mechanism (BYE/Also) Restrict caller identity (support varies for proxies from different vendors) Use "standard" status messages (otherwise messages will b taken from SIP packets) 	✓ Accept redirecti	on replies	
 Use obsolete transfer mechanism (BYE/Also) Restrict caller identity (support varies for proxies from different vendors) Use "standard" status messages (otherwise messages will b taken from SIP packets) 	Use short <u>h</u> eade	δ_{27}	
 Restrict caller identity (support varies for proxies from different vendors) Use "standard" status messages (otherwise messages will b taken from SIP packets) 	✓ Expose software	e version	
different vendors) – Use "standard" status messages (otherwise messages will b taken from SIP packets)	🗆 <u>U</u> se obsolete tra	nsfer mechanism (BYE)	'Also)
taken from SIP packets)			or proxies from
oice mail number or address:		1000 2022 R	
	— Use "standard"		se mezsefez wm ne
Remove fancy characters from phone numbers	Use "standard" taken from SIP	packets)	ze messages wii be
	Use "standard" taken from SIP Voice mail number	packets) or address:	7.
	 Use "standard" taken from SIP Voice mail number 	packets) or address:	7.

	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 :

PORTech Your CTI Partner	SMS A	Igent
Route	Mobile 1, 2 💌	Read received SMS
Mobile	Port	Status Bank
Status	Mobile 1	Standby. Rx List
Settings Fwd Settings	Mobile 2	Not Ready !!! Rx List
SMS Agent		
Network		SMS Sender
CID Cottinue	Via	Mobile 1 2
SIP Settings	Dest Num	
NAT Transform		Maximum Number of UCS2 chars for this text box is 70.
Update		
System Authority	Message	
Save Change		
Reboot		You have 70 UCS2 chars remaining for your description
		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
1	REC READ	886935386862	08/05/15,15:41:46
2			
-			

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

SMS Reader

ndex	RemoteID	Date, Time	
1	886935386862	08/05/15, 15:41:46	
MV	' Serial can send SMS and Rece	ive SMS	

Back	Delete
------	--------

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.

F	Route
Ν	lobile
Ν	letwork
4	Status
	/VAN Settings
	LAN Settings
	SNTP Settings
4	Slave Setting
S	SIP Settings
Ν	IAT Transform
U	lpdate
S	ystem Authority
S	ave Change
Б	?eboot

Network Status

Ethernet 0	WAN Interface	LAN Interface
Туре	Fixed IP Client	-
IP	192.168.0.110	-
Mask	255.255.255.0	-
Gateway	192.168.0.254	-
MAC	00037E005555	-

Ethernet 1	WAN Interface	LAN Interface
Туре	Fixed IP Client	-
IP	192.168.0.112	-
Mask	255.255.255.0	-
Gateway	192.168.0.254	-
MAC	00037E000077	-

Ethernet 2	WAN Interface	LAN Interface
Туре	Fixed IP Client	-
IP	192.168.0.114	-
Mask	255.255.255.0	-
Gateway	192.168.0.254	-
MAC	00037E000432	-

Ethernet 3	WAN Interface	LAN Interface
Туре	Fixed IP Client	Fixed IP Client
IP	192.168.0.116	192.168.0.108
Mask	255.255.255.0	255.255.255.0
Gateway	192.168.0.254	192.168.0.254
MAC	00037E000002	00037E000003

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

PORTech Your CTI Partner	WAN Se	ettings
	You could configu	re the WAN settings in this page.
Route	Ethernet 0 🗸	
Mobile		
Network	Network Mode:	Intersection ● State ■ Sta
Status WAN Settings	WAN Setting IP Type	● Fixed IP ○ DHCP Client ○ PPPoE
LAN Settings SNTP Settings Slave Setting	IP Mask	192.168.0.110
SIP Settings	Gateway	192.168.0.254
NAT Transform	DNS Server1	168.95.192.1
Update	DNS Server2	168.95.1.1
System Authority	MAC	00037e005555
Save Change		
Reboot	PPPoE Setting	
	User Name	
	Password	
		Submit Reset

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

PORTech Your CTI Partner	LAN Settings	
Route	Ethernet 0 🗸	
Mobile	Linemeto	
Network	LAN Setting	
	IP:	192.168.0.101
Status WAN Settings	Mask:	255.255.255.0
LAN Settings	MAC:	00037e006666
SNTP Settings		
Slave Setting	DHCP Server	
SIP Settings	DHCP Server:	◯ On ⊙ Off
NAT Transform	Start IP:	0
Update	End IP:	0
System Authority	Lease Time:	0 : 0 (dd:hh)
Save Change		
Reboot		
		Submit Reset

- (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

10.4 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 Slave Setting	
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 📴 🔽 08 💌 : 00 💌 (hh:mm)
Sync. Time:	1 : 0 : 0 (dd:hh:mm)

Reset

Submit

10.5 Slave Settings: Record Slave IP for Master

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

Interlink Setting

	IP Address : Port	
Master:	192.168.0.110 : 40000 (Local)	
Slave 1:	192.168.0.112 : 40000	
Slave 2:	192.168.0.114 : 40000	
Slave 3:	192.168.0.116 : 40000	

submit cancel

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 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.

Your CTI Partner	Service Domain Settings		
Route	Mobile 1 💌		
Mobile	Realm 1 (Default)		
Network	Active:	⊙ ON ○ OFF	
SIP Settings	Display Name:	803	
Service Domain	User Name:	803	
Port Settings	Register Name:	803	
Codec Settings	Register Password:	•••	
Codec ID Setting	-		
DTMF Setting	Domain Server:		
RPort Setting	Proxy Server:	192.168.0.1	
SIP Responses Other Settings	Outbound Proxy:		
NAT Transform	Status:	Registered	
Update	Realm 2		
System Authority	Active:	○ ON ⊙ OFF	
Save Change	Display Name:		
Reboot	User Name:		
	Register Name:		
	Negister Marile.		

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

-25-

- (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"

Realm 1 (Default)	
Active:	⊙On COff
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

Example:

11.2 Port Setting

Reboot

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	Ports Setting		
	You could set the	port number in this page.	
Route			
Mobile		Port of Mobile 1	
Network	SIP Port:	5060 (1024~65535)	
SIP Settings	RTP Port:	60000 (1024~65535)	
Service Domain	Port of Mobile N		
Port Settings	SIP Port:	5062 (1024~65535)	
Codec Settings Codec ID Setting	RTP Port:		
DTMF Setting	RTE FOIL	60100 (1024~65535)	
RPort Setting		Submit Reset	
SIP Responses			
Other Settings			
NAT Transform			
Update			
System Authority			
Save Change			

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.

Your CTI Partner	Codec Settings	
Route		
Mobile	Codec Priority 1:	Codec Priority G.711 u-law 🗸
letwork	Codec Priority 2:	G.711 a-law 🗸
SIP Settings	Codec Priority 3:	G.723 👻
	Codec Priority 4:	G.729 V
Service Domain	Codec Priority 5:	G.726 - 16 💙
Codec Settings	Codec Priority 6:	G.726 - 24 💙
Codec ID Setting	Codec Priority 7:	G.726 - 32 V
TMF Setting	-	
Port Setting	Codec Priority 8:	G.726 - 40 💌
Other Settings		RTP Packet Length
AT Transform	G.711 & G.729:	20 ms V
Jpdate	G.723:	30 ms 🗸
System Authority		
Save Change		G.723 5.3K
Reboot	G.723 5.3K:	🔘 On 💿 Off
CEDOOL		
		Voice VAD
	Voice VAD:	🔘 On 💿 Off
		Submit Reset

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 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	5) 🗹 23
G726-24 ID:	22 (95~255	5) 🗹 22
G726-32 ID:	2 (95~255	5) 🗹 2
G726-40 ID:	21 (95~255	5) 🗹 21
RFC 2833 ID:	101 (95~255	5) 🔽 101

Submit Reset

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

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

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.

PORTech Your CTI Partner	RPort Setting	
e	Mobile 1, 2 💌	
bile		
etwork	RPort of Mobile 1: (🖲 On 🔘 Off
	RPort of Mobile 2: (🖲 On 🔘 Off
? Settings		
vice Domain		Submit Reset
Settings		
c Settings		
ec ID Setting		
/IF Setting		
ort Setting		
Responses er Settings		
Transform		
date		
em Authority		
e Change		
ot		

11.7 SIP Responses

Your CTI Partner	SIP Kesponses Selling		
Route			
Mobile	• 486		Response on port busy. Busy here
Network	0 503		Service unavailable
SIP Settings			SIP Responses
Service Domain	⊙ ON	○ OFF	180 Ringing (Auto force to ON, if 183 was OFF.)
Port Settings	OON	⊙ OFF	183 Session Progress
Codec Settings		0	
Codec ID Setting			Submit
DTMF Setting			Sabinit
RPort Setting			
SIP Responses			
Other Settings			
NAT Transform			
Update			
System Authority			
Save Change			
Reboot			

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

11.7.2 180 Ring on/off: LAN TO MOBILE two stage dialing can be turn off, therefore there will be no the Ring Back Tone, all the phone call will be transferred to Voice-Mail 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 voicemail while GMS side are busying. We recommend you to turn this on if you use SIP Proxy.

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.

Your CTI Partner	Other Settings			
Route	Mobile 1, 2 💌			
Mobile	Hold by RFC of Mobile 1	◯ On ⊙ Off		
Network	Hold by RFC of Mobile 2	On ⊙ Off		
SIP Settings				
	Voice QoS:	40 (0~63)		
Service Domain Port Settings	SIP QoS:	40 (0~63)		
Codec Settings	SIP Expire Time:			
Codec ID Setting	SIF Expire fille.	300 (60~86400 sec)		
DTMF Setting				
RPort Setting		Submit Reset		
SIP Responses				
Other Settings				
NAT Transform				
Update				
System Authority				
Save Change				
Reboot				

12. NAT Transform

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.

PORTech Your CTI Partner	STUN Setting		
	Mobile 1, 2 💙		
Route			
Mobile	STUN of Mobile 1	◯ On ⊙ Off	
Network	STUN of Mobile 2	On ⊙ Off	
SIP Settings			
NAT Transform	STUN Server	stun.xten.com	
STUN Setting	STUN Port	3478 (1024~65535)	
Update		Submit Reset	
System Authority			
Save Change			
Reboot			

13.System Authority

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

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

System Authority

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

New username:	
New password:	
Confirmed password:	
	Submit 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

PORTech Your CTI Partner	Update Firmware
	You could update the newest firmware. PCB mark: 2N149A
Route	
Mobile	
Network	Method: 💿 HTTP 🔘 TFTP
SIP Settings	НТТР
NAT Transform	Code Type: Risc 💌
Update	File Location: 瀏覽
New Firmware Default Settings	TFTP
System Authority	TFTP Server: 192.168.1.250
Save Change	
Reboot	Update

- (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.

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	

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.

PORTech Your CTI Partner	Save Changes You have to save changes to effect them.
Route	
Mobile	
MODILE	Save Changes: Save
Network	
SIP Settings	
NAT Transform	
Update	
System Authority	
Save Change	
Reboot	

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.

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	

17. 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 will 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	#112xxx*xxx*xxx	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
			entering a decimal point.
11	Set Network Mask		
		*xxx#	Enter value using numbers on
			the telephone key pad. Use
			the * (star) key when entering
			a decimal point.
12	Set Gateway IP	#114xxx*xxx*xxx	Must set Static IP first.
	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.

18.Specification

18.1 Protocols SIP (RFC2543, RFC3261) 18.2 TCP/IP IP/TCP/UDP/RTP/RTCP/ CMP/ARP/RARP/SNTP **DHCP/DNS** Client **IEEE802.1P/Q** ToS/DiffServ NAT Traversal STUN uPnP **IP** Assignment Static IP DHCP **PPPoE** 18.3 Codec G.711 u-Law G.711 a-Law G.723.1 (5.3k) G.723.1 (6.3k) G.729A G.729A/B 18.4 Voice Quality VAD

CNG AEC, LEC Packet loss 18.5 GSM (MV-374/MV-378) 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

19. Appendix: Setup MV-374/MV-378 with Asterisk

19.1 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-374/MV-378 <--*lan*--> Asterisk <--*internet*--> VOIP provider <--*whatever*--> landline

To do such a call, you just call your MV-374/MV-378 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-374/MV-378 for free. You can then call all around the world from your mobile at voip cost :-)

19.2 MV-374/MV-378 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-374/MV-378 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-374/MV-378 is considered as extension '103' of the IPBX. In **Bold** are the parameters depending on your installation

LAN Settings

You could configure the LAN settings in this page.

LAN Mode:	C Bridge
WAN Setting	
IP Type:	Fixed IP C DHCP Client C PPPoE
IP:	mv-370 IP
Mask:	255.255.255.0
Gateway:	Router IP
DNS Server1:	168.95.192.1
DNS Server2:	168.95.1.1
MAC:	60

LAN To Mobile Table

item	URL		Call Num	Selec
0	your asterisk IP	#		
1				Г
2				E
З				Γ
4				Г
5				F
6				E
7				Г
8				F
9				E

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

Mobile To LAN Table

tem	CID		URL	Selec
0	authorised mobile n°	103		Г
1	another authorised n°	103		
2				E
3				F
4				E
5				F
6				F
7				F
8				Г
9				Г

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

You could set information of service domains in this page.

Realm 1 (Default)	
Active:	COn COff
Display Name:	103
User Name:	103
Register Name:	103
Register Password:	Asterisk extension password
Domain Server:	
Proxy Server.	Asterisk IP
Outbound Proxy:	
Status:	Registered

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

Codec Settings

You could set the codec settings in this page.

Codec Priority	
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:	NotUsed 💌
RTP Packet Length	
G.711 & G.729:	20 ms 💌
G.723:	30 ms 💌
G.723 5.3K	
G.723 5.3K	COn COM
Voice VAD	
Voice VAD:	C On C 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

You could set the v	olume	of your phone	in this page.		
@					
VolP Volume:	10	(0~12)	VoIP Gain:	12	(0~15)
i@					
LAN DTMF Gain:	10	(0~12)	Mobile In Gain	3	(0~4)
1@					
Caller ID	C Clid		← Fix(SIP User)		
1@					
Mobile PIN Code:	On 🗖		Code:	Confirmed:	

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

19.3 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.

19.4 Asterisk configuration

Once the MV-374/MV-378 is set, you have to configure Asterisk. On that side, you have to setup files as follow :

19.5 sip.conf

```
; GSM VOIP Gateway MV-374/MV-378
[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
```

19.6 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-374/MV-378 sim card mail box thru GSM exten => _888,1,SetCallerID("xxxxxxxxx") exten => _888,2,Dial(SIP/\${EXTEN}@103,60,r) exten => _888,3,Hangup()

20.How to setup Asterisk to receive Caller ID from

MV-374/MV-378

Test version trixbox-2.2

SIP Softphone

- SJPhone 1.60.289a
- X-Lite 1105x

Modify file

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

[1000]

type=friend secret=1000 qualify=yes nat=yes host=dynamic canreinvite=no context=internal

[1001] type=friend secret=1001 qualify=yes nat=yes host=dynamic canreinvite=no context=internal

[1002] type=friend secret=1002 qualify=yes nat=yes host=dynamic canreinvite=no context=internal

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

configure: trixbox-2.2: address=192.168.66.202:5060 SJPhone: address=192.168.66.145:5060; username=1000, displayname=user_1000 X-Lite: address=192.168.66.145:7331; username=1001, displayname=user_1001 MV-374/MV-378: address=192.168.66.203:5060; username=1002, displayname=user_1002

test1

pstn → call 0928492911(mobile number) → MV-374/MV-378 → 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-374/MV-378 \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:3 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

21. 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-374/MV-378 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-374/MV-378 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.				
Lar	n to Mobile:				
(1)	*,*>it is two stage dialing.				
	when lan phone call in,MV-374/MV-378 will provide dial tone and you can enter mobile number.				
(2)	*, specific mobile number				
	when Ian phone call in,MV-374/MV-378 will connect with the specific mobile number auto.				
(3)	*,#>It is 1 stage dialing				
	When lan phone and MV-374/MV-378 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-374/MV-378				

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