MV-370

VoIP GSM Gateway

User Manual



PORTech Communications Inc.

Content

1. INTRODUCTION1
2. FUNCTIONS1
3. THE CONTENTS IN PACKAGE1
4. DIMENSION AND PANEL DESCRIPTION2
5. ACCESSORY ATTACHMENT
6. SETTING AND MANAGING VIA WEB PAGE 4
7. SYSTEM INFORMATION
8. ROUTE
9. MOBILE
10. NETWORK 17
11. SIP SETTING21
12. NAT TRANS
13.SYSTEM AUTH
14.SAVE CHANGE
15.UPDATE
16.REBOOT
17. SETTING AND CHECKING VIA IVR
18.SPECIFICATION
19. APPLICATIONS
20. SIMPLE STEPS

21. APPENDIX: SETUP MV-370 WITH	H ASTERISK 4	1
22.HOW TO SETUP ASTERISK TO R	ECEIVE CALLER ID FROM 4	17

1. Introduction

MV-370 series products provide you the best connect solution for heterogeneous network [] including [] WLAN [] GSM or PSTN [] You may use a SIP-protocol VoIP phone or software to connect to the MV-370, then reach this call to the mobile network, and vice versa. With multiple sets of MV-370, you may even build an international call network.

2. Functions

- 2.1 VoIP (SIP)-GSM conversion.
- 2.2 VoIP (SIP)-CDMA conversion.
- 2.3 Voice response for setting and status enquiring. (Dial in GSM numbers of MV-370 to get voice information or to operate.)
- 2.4 50 sets of LAN->MOBILE routing, and 50 sets of MOBILE->LAN routing.
- 2.5 Series connections to save bills.
- 2.6 Standard SIP (RFC2543, RFC3261) protocol to communicate with other gateways or PC.
- 2.7 settings and managing via web page

3. The contents in package

- 3.1 MV-370 main body
- 3.2 AC-DC Adaptor (110V AC 12V DC or 220V AC 12V DC)

- 3.3 Network cable
- 3.4 Antenna
- 3.5 User's Manual



(3.3) When you receive MV-370 package and find it is damaged or incorrect, please contact your vendor.

4. Dimension and Panel description





- 4.1 Antenna Antenna connector.
- 4.2 DC 12V Power socket.
- 4.3 LAN: Standard RJ-45 socket, connecting to Hub circuit.
- 4.4 PWR: Power indicator light, red light. Light is on when system's power supply is normal.
- 4.5 MOBILE: GSM indicator light, green light. Light flashes when GSM status is normal; light turns on constantly when GSM is called.
- 4.6 LAN: LAN indicator light, green light. Light flashes when Lan is called; light turns off when GSM answered.
- 4.7 LINK: Link indicator light, green light. Light is on when network is connected correctly.

5. Accessory attachment

- 5.1 Connect the network cable both to your Hub and to LAN socket of MV-370.
- 5.2 Connect the antenna and place it in a good receiving location (not too close to the device).
- 5.3 Insert a SIM card into back of MV-370.
- 5.4 Plug the adapter in DC 12V socket and PWR socket. The PWR light should turn red at the moment.
- 5.5 Click reset button 3 sec. MV-370 will restore default IP. Other setting as usual.



6. Setting and managing via web page

The default IP address of MV-370 is <u>http://192.168.0.100</u>. Before accessing the web page, please confirm this address is available in your network.

Login VolP	
Enter your us	ername and password to login
	VoIP server
Username	
Password	
	Login Clear .

Enter the default username and password to login. Default username: voip Default password: 1234

7. System Information.

7.1 After login, you could see the system information such as: model name, firmware version, codec version name, etc. in this page.

PORTech Your CTI Partner	Mobile V	0IP v6.690e		
Route	Madal Nama:	MV/270		
Mobile To Lan Settings	Model Description:	GSM:900/1800/1900MHz		
Mobile To Lan Speed Dial	Firmware Version:	Mon Apr 21 14:24:34 2008.		
Lan To Mobile Settings	Codec Version:	Mon Jul 24 10:55:05 2006.		
Mobile				
Network				
SIP Settings	© 2007 PORTech Communications Inc.			
NAT Transform				
Update				
System Authority				
Save Change				
Reboot				

7.2 You could also see the setting table in the left side. Please click on the option you would like to set. The setting methods are indicated as the following chapters, please input the value or select the item according to your situation.

Note:

Please remember to save change whenever you submit any setting. Click "Save Change" then "Save" button, the system will restart and make the changed function/setting operative.

8. Route

8.1 Route/ Mobile to LAN Settings

In this page: Mobile To Lan Table, you could set the routing rules to transfer the colle incoming from MOBILE to LAN. Maximum 50 sets.

Your CTI Partner		e to LAIN	Table	
Route	Page: 1▼			
Mobile	Item	CID	URL	Select
	• 0	*	*	
Network	1			Γ
	2			
SIP Settings	• 3			
NAT Trans.	4			Г
	• 5			
ystem Auth.	6			
our Change	/			
save change	8			
Update	9			
Reboot	Delete Selec	ted Delete All R	eset	
	Add New			
	Position:		(0~49)	
	CID:		Ex:0911111111, 0	911*, *
	URL:		Ex:192.168.0.1. *:2	?St
	L			
	Add Reset			

When the GSM number of the MV-370 is called, this device transfers the call to URL according to the caller ID of the incoming call.

8.1.1 CID: caller ID, the numbers of incoming call

You could set the CID as the following formats:

- (1) The complete number, e.g. 091111111
- (2) The prefix part plus *, e.g. 0911*. This format means any number starting with 0911 will be accepted to transfer.
- (3) *, this means any incoming call is accepted to transfer.
- (4) N, this means the incoming call without showing its CID is accepted to transfer.

Please note the priority of the routing rules; the CID with more digits

gets the priority.

8.1.2 URL The IP address of destination

You could set the URL as the following formats:

- (1) The complete IP address, e.g. 192.168.0.101
- (2) The proxy extension numbers
- (3) The phone numbers.

Note: If the device has registered at proxy server/Asterisk, you can enter any destination phone number. Also note that in the proxy server/Asterisk, you need to set the route of destination phone number.

- (4) Leave it blank or 'N', this mean to refuse to transfer.
- (5) *, this means to transfer via 2-stage-dialing. The call will be answered with a prompt dial tone for the caller to press the IP address, proxy extension, or **any phone number** as destination. The caller press the IP address on the phone keys: 192*168*0*101# as 192.168.0.101
- 8.1.3 Example of Mobile to Lan setting:
- (1) Mobile to Lan: 0932*, 0911123456

When the GSM numbers of the device is called, if the caller's prefix numbers are 0932, MV-370 transfers the call to 0911123456, then 0911123456 rings (while available). Precondition:

a. MV-370 has registered at proxy server/Asterisk

- b. The proxy server/Asterisk has the route of "09"
- (2) Mobile to Lan: *, *

Any incoming call gets a prompt dial tone; so the caller can enter any IP address, sip extension, or phone number. Precondition:

a. SIP extension or phone number needs to register at SIP Proxy Server or Asterisk.

- b. Phone number, SIP Proxy Server or Asterisk needs to set the route of destination phone number.
- 8.2 Route/ Mobile to LAN Speed Dial Settings When you set both Mobile to LAN Speed Dial Settings and Mobile to LAN settings at the same time, Mobile to LAN Speed Dial Settings gets higher priority. Mobile to Lan setting will be not available.

Your CTI Partner	Mobile To LAN Speed Dial				
	You could s	set the speed	dial in this page.		
Mobile To Lan Settings					
Mobile To Lan Speed Dial	Num	Name		URL	Select
Lan To Mobile Settings	0	test	192.168.0.107		
Network	1				
· ·	2				
SIP Settings	3				
	4				
NAT Trans.	5				
System Auth	6				
System Audi.	7				
Save Change	8				
	9				
Update 💦					
Reboot	Delete	Selected	Delete All Re	eset	

The call is answered with a prompt dial tone for the caller to press the "Num", and then the device connects the "URL" as destination.

Example: after you call the GSM number of the device and hear a dial tone, you press 0, then the lan phone of IP address: 192.168.0.107 rings.

8.3 Route/ LAN to Mobile Settings

In this page: Lan To Mobile table, you could set the routing rules to transfer the calls incoming from Lan to Mobile. Maximum 50 sets.



When the Lan of the MV-370 is called, this device transfers the call to Call Num according to the URL of the incoming call.

8.3.1 URL: The IP address or proxy extension numbers of the incoming call.

You could set the URL as the following formats:

- (1) The complete IP address, e.g. 192.168.0.101
- (2) The proxy extension numbers, e.g. 103
- (3) Part of an IP address plus *, e.g. 192.168.0.*. This means the IP address starting with 192.168.0 would be accepted to transfer,

(4) Part of the proxy extension numbers plus, e.g. 10*. This means

the extension numbers starting with 10 would be accepted to transfer.

8.3.2 Call Num: the phone numbers of destination.

You could set the Call Num as the following formats:

(1) The complete number, e.g. 0911111111

(2) *, this means to transfer via 2-stage-dialing. The call will be answered with a prompt dial tone for the caller to press the destination phone numbers, e.g. 091111111

(3) **#**, this allow the caller with lan phone dial directly the destination

numbers.

Precondition:

- (1) MV-370 and incoming Ian Phone are both registered at proxy server or Asterisk.
- (2) Proxy server/asterisk has set the routing rules to assign specific prefix of numbers to be transferred from MV-370.
- (3) Lan to Mobile routing sets: *, #
- **Usage:** You could dial on your lan phone call any destination number with prefix of "09". When your lan phone and MV-370 had registered and "09" prefix is setted the routing rules at proxy server or Asterisk.
- (4) **#**['d'n]['a'ppp], this means to do the above routing, and to modify

the numbers.

Note: 'd'n means to delete the number of prefix,

'a'ppp means to add 'ppp' prefix.

E.g. #d2a09 means to call the registered numbers via one-stagedialing. The numbers are modified to: delete 2 digits of prefix of the original numbers, then add 09 to be new prefix of the destination numbers.

9. Mobile

9.1 Mobile/ Mobile Status

In this page: Mobile Status, you could get the information of your GSM network and the latest operation.



L -	Wioone Status	
D (
Route	Network Registration.:	
Mobile	SIM Card ID:	
· · · · · · · · · · · · · · · · · · ·	Signal Quality.:	19
Network	GSM S/N:	353876012997720
STP Settings		
SIL Settings	Incoming IP:	
NAT Trans.	Incoming IP Name:	
System Auth.	Outgoing IP:	
	Incoming Mob:	
Save Change	Outgoing Mob:	
Undate .		

Mobile Status

Rehoot

Mobile Voiv

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

(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.

9.2 Mobile/ Mobile Setting

In this page: Mobile setting, you could adjust the parameter and click on the option to fit your need. You could leave those default value before you had tried the complete operation of this device.



- (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)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 22.How to setup Asterisk to receive Caller ID from MV-370 (page 43)

MV-370-2 will send the message as follows in the Packet. From: " caller number " <sip:3001@192.168.0.228>;tag=51088abb

• Tel/Tel :

MV-370 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-370 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)Presentation CLIR : If you need to block the Caller Id for call termination, please choose Suppression
- (9)Mobile PIN Code: If you need to unlock pin code via MV-370, you can click "On" and enter pin code.

(10)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

(11)Band Type:When you buy Quad band,you need to choose your GSM frequency

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

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).

Forward Enable Fwd to Mobile1: Fwd to Mobile2: Fwd to External:	e Name	URL:Port 192.168.0.100:5060 192.168.0.100:5062
Fwd to Mobile1: Fwd to Mobile2: Fwd to External:	Name	URL:Port 192.168.0.100:5060 192.168.0.100:5062
Fwd to Mobile1: Fwd to Mobile2: Fwd to External:		192.168.0.100:5060 192.168.0.100:5062
Fwd to Mobile2: Fwd to External:		192.168.0.100:5062
Fwd to External:		
	submit cancel]
		submit cancel

* "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
Z Accept redirection	replies	
Use short <u>h</u> eaders	R	
✓ Expose software ∨	ersion	
<u>U</u> se obsolete transf	er mechanism (BYE/	Also)
 <u>R</u>estrict caller iden different vendors) 	tity (support varies fo	or proxies from
 Use "standard" stat taken from SIP particular 	tus messages (otherwi ckets)	se messages will be
<u>/</u> oice mail number or	address:	
	and the second secon	
Remove fancy cha	racters from phone n	umbers
Remove <u>fancy</u> cha	racters from phone n	umbers

	Name	URL:Port
Fwd to Mobile1:		
Fwd to Mobile2:		
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 :

Your CTI Partner	SMS A	Agent	Read received	SMS
Route	Bank		Status	Read
Mobile	Mobile 1	Standby.		Rx List
Status Settings Fwd Settings	Dest Num	Maximum Number	SMS Sender	vic 70
SMS Agent Network				<u> </u>
SIP Settings	Message			•
NAT Transform		You have 70 UCS2	chars remaining for your desc	ription
System Authority Save Change Reboot		1	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
MV	/ Serial can send SMS and recei	ve SMS
	Back	Delete

10. Network

In Network, you could check the Network status; configure the WLAN Settings, LAN Settings and SNTP settings.

10.1 Network/ Status/ Network Status: information of current Network in this page.

P Vo	ORTech		Network Status		
			This page shows current status of network interfaces of the system.		
Rou	te	•			
Mobile			Interface O		
112013		•	Туре:	Fixed IP Client	
Net	St.		IP:	192.168.0.109	
	Status		Mask:	255.255.255.0	
SIP	Network Settings		Gateway:	192.168.0.254	
	SNTP Settings		DNS Server 1:	0.0.0.0	
NAT	Trans.	•	DNS Server 2:	0.0.0.0	
System Auth.					
Save Change					
Upda	ate	•			
Reb	oot				

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

The default IP is 192.168.0.100; you could change it to any available IP address, or select different IP type to suit your environment.

PORTech Your CTI Partner 2		LAN Settings		
		You could configure	e the LAN settings in this page.	
Rou	te 🕨			
Mobile .		LAN Mode:	O Bridge	
Net	Status	WAN Setting		
	Notwork Sottings	- IP Type:		
SIP	SNTD Settings	IP:	192.168.0.109	
NAT	Trans	Mask:	255.255.255.0	
11211	• • • • •	Gateway:	192.168.0.254	
Syst	em Auth.	DNS Server1:	168.95.192.1	
Save Change		DNS Server2:	168.95.1.1	
Save Change		MAC:	00037e000826	
Upd	ate 😱			
		PPPoE Setting		
Reb	oot	User Name:		
		Password:		
			Submit Reset	

- (1) LAN Mode: select NAT
- (2) Fixed IP: 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.
- (3) DHCP client: you could refer to your current network environment to configure the system properly
- (4) PPPoE: If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.
- (5) After you input or modify the value, click the Submit button.

10.3 Network/ SNTP Settings:

You could select "On" to give SNTP function to this device.

Input the primary and secondary IP Address of SNTP Server to get the date/time information. Also you could set the Time Zone according to your location; and set the time to synchronize. After setting, remember to click the Submit button.

Your CTI Partner		SNTP Settings		
		You could set the SNTP servers in this page.		
Route				
Mobile •		SNTP:	⊙On COff .	
Net	Status		Primary Server:	time.windows.com
SIP	Network Settings		Secondary Server:	208.184.49.9
	SNTP Settings			
NAT	Trans.	•	Time Zone:	GMT + 💌 08 💌 : 00 💌 (hh:mm)
Syst	em Auth.		Sync. Time:	1 : 0 : 0 (dd:hh:mm)
Save	e Change			Submit Reset
Upd	ate	•		
Reboot				

11. SIP Setting

If you need, you could setup the Service Domain, Port Settings, Codec Settings, RTP setting, RPort Setting and Other Settings. If ISP provides the VoIP service, you need to input the related information correctly to register at SIP Proxy Server.

11.1 SIP Setting/ Service Domain:

In this page, you should input the data refer to your ISP. Maximum is 3 accounts (Realm 1 to 3). You could dial out via first SIP account, and receive via the three SIP accounts.

	CTI Partner	Service Domain Settings		
		You could set information	on of service domains in this page.	
Route	•			
Mobile .		Realm 1 (Default)		
	•	Active:	⊙ On C Off	
Netwo	ork 😱	Display Name:	david	
orn or		User Name:	5007	
215.20	Service Domain	Register Name:	5007	
NAT	Port Settings	Register Password:	****	
	Codec Settings	Domain Sower	192 168 0 228	
Syste	Codec ID Setting	Domain Server.	400.400.0.220	
Same	DTMF Setting	Proxy Server:	192.168.0.228	
Save	RPort Setting	Outbound Proxy:		
Updat	Other Settings	Status:	Registered	
Rebor	at	Realm 2		
		n anti-		

(1) Active: click "On" to enable the function in Service Domain, then input the following items.

(2) Display name: input the name you would like to display.

- (3) User name: input your user name in ISP.
- (4) Register Name: input your register name in ISP.
- (5) Register Password: input your password in ISP.

- (6) Domain Server: input the Domain Server IP address.
- (7) Proxy Server: input the Proxy Server IP address.
- (8) Outbound Proxy: input the Outbound Proxy IP address. If your ISP does not provide the information, you could skip this item.
- (9) After setting, click the Submit button. Remember to click "Save Charge"
- (10) You can see the Register Status in the Status item.

Example: Register VoipBuster

Realm 1 (Default)	
Active:	⊙On COff
Display Name:	jenny0922
User Name:	jenny0922
Register Name:	jenny0922 Vour Voinbuster password
Register Password:	****
Domain Server:	
Proxy Server:	194.221.62.207
Outbound Proxy:	
Status:	Registered

11.2 Port Setting

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

Your CTI Partner	Port Settings
	You could set the port number in this page
Route	
Mobile	SIP Port: 5060 (10~65533)
Notwork	RTP Port: 60000 (10~65533)
THE WOLK	
SIP S Service Domain	Submit Reset
NAT Port Settings	
Codec Settings	
Syster Codec ID Setting	
Save DTMF Setting	
RPort Setting	
Updat Other Settings	
Reboot	

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. After setting, remember to click the Submit button.

PORTech Your CTI Partner	Codec Settings You could set the codec settings in this page.		
	Codec Priority		
	Codec Priority 1:	G.711 u-law 💌	
Route	Codec Priority 2:	G.711 a-law 💌	
Mohile	Codec Priority 3:	G.729 💌	
MIDDIE .	Codec Priority 4:	G.723 💌	
Network	Codec Priority 5:	G.726 - 16 💌	
STD G'	Codec Priority 6:	G.726 - 24 💌	
SLP Settings	Codec Priority 7:	G.726 - 32 💌	
NAT Trans.	Codec Priority 8:	G.726 - 40 💌	
,			
System Auth.	RTP Packet Length		
S (1	G.711 & G.729:	20 ms 💌	
Save Change	G.723:	30 ms 💌	
Update			
	G.723 5.3K		
Reboot	G.723 5.3K:	○ On	
	Voice VAD		
	Voice VAD:	C On © Off	
	S	ubmit Reset	

11.4 Codec ID Setting

You can setup the Codec ID in this page.

Your CTI Partner	Codec ID Setting		
	You could set the val	ue of Codec ID in this page.	
Route			
Mobile	Codec Type	ID	Default Value
	G726-16 ID:	23 (95~255)	23
Network	G726-24 ID:	22 (95~255)	22
SIP Settings	G726-32 ID:	2 (95~255)	2
	G726-40 ID:	21 (95~255)	2 1
NAT Trans.	RFC 2833 ID:	101 (95~255)	☑ 101
System Auth. Save Change		Submit Reset	
Update			
Reboot			

11.5 DTMF Setting

You can setup the DTMF Setting in this page.

Your CTI Partner	DTMF Setting
Route	
Trouce	Mobile DTMF Transfer to Lan
Mobile	O 2833
Network	Inband DTMF
SIP Settings	Send DTMF SIP Info
NAT Transform	Mobile DTMF debounce: 80 (range:40~200, default:80) step:10ms.
Update	
New Firmware	Submit Reset
Default Settings	
System Authority	
Save Change	
Reboot	

Note:

If this device has registered at SIP Proxy Server/Asterisk, please select "2833". If not, please select "Inband DTMF".

11.6 RPort Setting:

You can setup the RPort Enable/Disable according to your ISP information. After setting, remember to click the Submit button.

	RTech	RPort S	etting
		You could enable	e/disable the RPort setting in this page.
Route			
Mobil	le 🖡	RPort:	⊙On COff
Network			Submit Reset
SIP S	Service Domain		
NAT Port Settings Codec Settings			
Syster Codec ID Setting			
Save	DTMF Setting		
	RPort Setting		
Updat	Other Settings		
Rebo	ot		

11.7 SIP Setting: SIP Responses

		You could	I set the La	n to Mobile sip message in this page.
Route	,			
Iobile				Response on port busy.
	•			Busy here
Network		○ 503		Service unavailable
SIP Sterning Dee				SIP Responses
Service Don	nain	ON	OFF	180 Ringing (Auto force to ON, if 183 was OFF.)
NAT Port Setting	S	OON	⊙ OFF	183 Session Progress
Codec Setti	ngs			
Systel Codec ID Se	etting			Submit
DTMF Sett	ing			
RPort Settin	ng			
Updat SIP Respon	ses			
Other Settin	igs			
Reboon				

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-Attempt 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 Voice-Attempt while GMS side are busying. We recommend you to turn this on if you use SIP Proxy.

11.8 Other Settings

You could setup the RFC and QoS according to your ISP information. After setting, remember to 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 get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.

PO	CTI Partner	
Route	I.	•
Mobil	e	•
Netwo	ırk	×
SIP S	Service Domain	
NAT	Port Settings	
	Codec Settings	
Syster	Codec ID Setting	
Save	DTMF Setting	
Save	RPort Setting	
Updat	Other Settings	
Rebo	nt	

Other Settings

You could set other settings in this page.

Hold by RFC:	🔿 On 💿 Off
Voice QoS:	40 (0~63)
SIP QoS:	40 (0~63)
SIP Expire Time:	300 (60~86400 sec)
Virtual Ring:	On ⊙Off
	Submit Reset

12. NAT Trans

In this page: NAT Trans./ STUN, you could setup the STUN Enable/Disable and STUN Server IP address. This function helps your VoIP device work properly behind NAT. Change these settings according to your ISP information. After setting, remember to click the Submit button.

PORTech Your CTI Partner	STUN Setting		
	You could set th	e IP of STUN server in this page.	
Route			
Mobile .	STUN:	COn ତOff	
Network	STUN Server:	stun.xten.com	
SIP Settings	STUN Port:	3478 (1024~65535)	
NAT TI STUN Setting		Submit Reset	
System Auth.			
Save Change			
Update .			
Reboot			

13.System Auth.

In this page: System Authority, you could change your login name and password.

For CTI Partner		System Auth	ority
		You could change the login	username/password in this page.
Route	•		
Mobile		New username:	
NT . 1		New password:	
Network		Confirmed password:	
SIP Settings	•		Submit Reset
NAT Trans.	•		
System Auth.			
Save Change			
Update	•		
Reboot			

14.Save Change

Please remember this step whenever you submit any setting. Click "Save Change" then "Save" button, the system will restart and make the changed function/setting operative.

Your CTI Partner	Save Changes
	You have to save changes to effect them.
Route	
Mobile •	Save Changes: Save
Network	
SIP Settings	
NAT Trans.	
System Auth.	
Save Change	
Update •	
Reboot	

15.Update

Here you could update the latest firmware and restore the default settings.

15.1 Update/ New Firmware/ Update Firmware

Download the latest firmware, then

- (1) Method: select "HTTP"
- (2) Code Type: select "Risc".
- (3) File Location: Click the "Browse" button in the right side of the File Location for the file.

Please note: no need to unzip the firmware file.

(4) Click "Update", it takes few minutes to generate new firmware.

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

Update Firmware

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

Method:	● HTTP ○ TFTP
НТТР	
Code Type:	Risc 💌
File Location:	瀏覽
TFTP	
TFTP Server:	192.168.1.250
	Update Reset

15.2 Restore Default Settings

In this page: Update/ Default Settings, you could restore the factory default settings to the system. Click the Restore button, then the system returns to default IP <u>http://192.168.0.100</u> (the other settings e.g SIP setting, mac address remains), and automatically restart.

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 Trans.	
System Auth.	
Save Change	
Updat New Firmware	
Rebo Default Settings	

16.Reboot

In this page, you could click the Reboot button to restart the system.

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 Trans.	
System Auth.	
Save Change	
Update 🔸	
Reboot	

17. Setting and checking via IVR

User could get or set some parameters of the system by dialing in the mobile numbers of the device. The status or result is reported via voice response system. In the first 20 seconds after power-on (when only Mobile light flash), you could dial its mobile numbers. When you hear the dial tone, press the following codes to set or check the device.

Item	Function	Code	Action
1	Reboot	#195#	Reboot the device
2	Factory Reset	#198#	Return to default settings
3	Check IP Address	#120#	IVR announces the current IP
			address. Default: 192.168.0.100
4	Check IP Type	#121#	IVR announces DHCP is on or off.
			Default: off
5	Check Network	#123#	IVR announces the current network
			mask. Default: 255.255.255.0
	Mask Chask Cataway	#404#	
6		#124#	IVR announces the current
	IP Address		gateway IP address. Default:
			192.168.0.254
(Check Primary	#125#	IVR announces the current setting
	DNS Server		in the Primary DNS field.
			Default: 192.168.0.1
8	Check Firmware	#128#	IVR announces the version of the
	Version		firmware.
9	Set as DHCP	#111#	The system is changed to DHCP
	client		Client type
10	Set Static IP	#112xxx*xx	DHCP is disable and system is
	Address	x*xxx*xxx#	changed to static IP type.
			Enter IP address using numbers on
			the telephone keypad. Use the *
			(star) key when entering a decimal

			point.
11	Set Network Mask	#113xxx*xx	Must set Static IP first.
		x*xxx*xxx#	Enter value using numbers on the
			telephone keypad. Use the * (star)
			key when entering a decimal point.
12	Set Gateway IP	#114xxx*xx	Must set Static IP first.
	Address	x*xxx*xxx#	Enter IP address using numbers on
			the telephone keypad. Use the *
			(star) key when entering a decimal
			point.
13	Set Primary DNS	#115xxx*xx	Must set Static IP first.
	Server	x*xxx*xxx#	Enter IP address using numbers on
			the telephone keypad. 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-370) Dual BAND: 900/1800 MHZ Tri BAND: 900/1800/1900 MHZ Quad BAND: 900/1800/1900/850 MHZ

19. Applications

- 1.Connect to VoipBuster
 - a). Register VoipBuster account at Service Domain.
 - b). Route setting: Mobile to Lan set: *,*

When you call in GSM number of MV-370, you can enter destination number that will dial out from VoipBuster. (Landline is free, GSM rate is cheap)

- 2.How to apply 2 sets of MV-370?
 - (1) When you call the no.1 MV-370 gsm number, it will provide dial tone and you enter a destination number. Then no.2 MV-370 will dial this number and connect.
 - Step 1:no.1 MV-370: mobile to lan set route table *,*
 - Step 2:no.2 MV-370:lan to mobile set route table *,#
 - Step 3:Additionally, two pcs MV-370 both need to register proxy server.
 - Step 4:And proxy server set the route that the prefix of destination number to dial out from no.2 MV-370.
 - (2) When you call the no.1 MV-370 gsm number,no.2 MV-370 will dial this specific number and connect
 - Step 1:no.1 MV-370: mobile to lan set route table *, specific destination number
 - Step 2:no.2 MV-370:lan to mobile set route table *,#
 - Step 3:Additionally, two pcs MV-370 both need to register proxy server.
 - Step 4:And proxy server set the route that the prefix of destination number to dial out from no.2 MV-370.

20. 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)

Мо	bile to Lan:			
(1)) *,*>it is two stage dialing.			
	when mobile call in, MV-370 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-370 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-370 will provide dial tone and you can enter mobile number.			
(2)	*, specific mobile number			
	when lan phone call in, MV-370 will connect with the specific mobile number auto.			
(3)	*,#>It is 1 stage dialing			
	When lan phone and MV-370 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-370 			

All changes both need to click "save and change"

21. Appendix: Setup MV-370 with Asterisk

21.1 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-370 <--lan--> Asterisk <--

internet--> VOIP provider <--*whatever-->* landline

To do such a call, you just call your MV-370 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-370 for free.

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

21.2 MV-370 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-370 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-370 is considered as extension '103' of the IPBX.

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

LAN Settings

You could configu	ure the LAN settings in this page.
LAN Mode:	C Bridge I NAT
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:	CONT

LAN To Mobile Table

@ Page: 1 💌	I			
ltem	URL		Call Num	Select
0	your asterisk IP	#		
1				Г
2				E
З				E
4				Г
5				Г
6				П
7				Γ
8				F
9				Г

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		
1	another authorised n°	103		Γ
2				П
3				Г
4				E
5				F
6				F
7				Г
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.

Active:	€ On € Off
Display Name:	103
User Name:	103
Register Name:	103
Register Password:	Asterisk extension password
Domain Server:	
Proxy Server	Asterisk IP
Outbound Proxy:	

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

Codec Settings

You could set the codec settings in this page.

20200 2020 D2	
Codec Priority	
Codec Priority 1:	G.711 u-lew 💌
Codec Priority 2:	G.711 a-law 💌
Codec Priority 3:	NotUsed 💌
Codec Priority 4:	Not Used 🔹
Codec Priority 5:	Not Used
Codec Priority 6:	NotUsed 💌
Codec Priority 7:	Not Used 💌
Codec Priority 8:	Not Used
-	
RTP Packet Length	1
G.711 & G.729:	20 ms 💌
G.723:	30 ms 💌
G.723 5.3K	
G.723 5.3K:	C On @ Off
Voice VAD	
Vaice 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.		
1@					
VoIP 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		irmed:

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

21.3 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.

21.4 Asterisk configuration

Once the MV-370 is set, you have to configure Asterisk. On that side, you have to setup files as follow :

21.5 sip.conf

```
; GSM VOIP Gateway MV-370
[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

21.6 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-370 sim card mail box thru GSM
exten => _888,1,SetCallerID("xxxxxxxxx")
exten => _888,2,Dial(SIP/${EXTEN}@103,60,r)
exten => _888,3,Hangup()
```

22. How to setup Asterisk to receive Caller ID from

```
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-370: address=192.168.66.203:5060; username=1002, displayname=user_1002



test1

pstn → call 0928492911(mobile number) → MV-370 → 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 - - - -

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:3 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-370 \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