



OpenVox GSM Gateway Function Manual

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OpenVox VoxStack GSM Gateway is a feature-rich, highly available and flexible modular gateway product. This manual introduces some major functions of the gateway, so that our customers could configure it easily.

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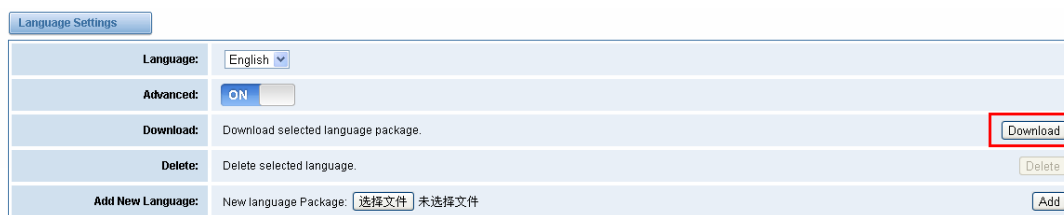
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Chapter 1: How to Change the System Language

OpenVox GSM Gateway supports user custom language package. You can change the Web language of gateways. The default language is English and you could change it to Russian, Spanish, and so on. Now let me show you how to use the function.

Step 1: Download the Language Package

Please click “SYSTEM→General”, and enable the Language Advance Settings. Then download the language package to your PC.



Language Settings	
Language:	English
Advanced:	<input checked="" type="checkbox"/>
Download:	Download selected language package. Download
Delete:	Delete selected language. Delete
Add New Language:	New language Package: 选择文件 未选择文件 Add

Step 2: Change the Language Package

After you download the package, you can edit your own language. For example:
Now I will change the home page “GSM Information” (that displays with English) to Chinese.



GSM Information									
Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	GSM Status	Remain Time

First, please edit the Language package file. You could see this.

```

1 language#english#English
2
3 ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
4 ; Language Package Description
5 ;
6 ; Format:
7 ; key="value"
8 ;
9 ; Notice:
10 ; 1.Package head line format: language#xxx#XXXX
11 ; xxx is the Internal identification.
12 ; XXX is the web view.
13 ; 2.Package must save format: UTF-8(No BOM)
14 ; 3.If there is [""] in value, please use [\\]. For example:
15 ; language help="This is \\\"Portuguese\\\""
16 ;
17 ; 2013.8.23, OpenVox Co.
18 ;
19 ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
20
21
22
23
24
25 ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
26 ;
27 ; general
28 ;
29 ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
30 [general]

```

Please edit the first line as follows:

```

-----1-----2-----3-----4-----5-----6-----7-----8-----9-----0-----
1 language#chinese#Chinese
2
3 ::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
4 ; Language Package Description
5 ;
6 ; Format:
7 ; key="value"
8 ;

```

Now, please find the "system--status" segment.

```

::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
;
; system-status.php
;
::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
[system-status]
;GSM status
GSM Information="GSM Information"
;Port="Port"
Signal="Signal"
BER="BER"
Carrier="Carrier"
Registration Status="Registration Status"
PDD="PDD"
ACD="ACD"
ASR="ASR"
GSM Status="GSM Status"

```

Then change the "GSM Information" to "GSM 信息" and save it.

```
.....  
[system-status]  
;GSM status  
GSM Information="GSM 信息"  
;Port="Port"
```

Step 3: Upload Your New Language Package

Please upload your file, and then click "Add" button.

Language:	English
Advanced:	ON
Download:	Download selected language package. Download
Delete:	Delete selected language. Delete
Add New Language:	New language Package: 选择文件 chinese Add

After uploading successfully, you have achieved to change the language.

Step 4: Choose the Language for Your Gateway

Click the Language drop-down list and choose your language. Don't forget to save and apply it.

Language:	Chinese
Advanced:	OFF

Enable:	OFF
Reboot Type:	By Running Time
Running Time:	Day: 0 Hour: 0 Minute: 0

Save

Now you can see the language has been changed to Chinese.

Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	GSM Status	Remain Time
------	--------	-----	---------	---------------------	--------	--------	--------	------------	-------------

GSM 信息

Notice: If your gateway doesn't support to change language, please upgrade your firmware. The link is shown as below:

<http://downloads.openvox.cn/pub/firmwares/GSM%20Gateway/wq400-current.img>

Chapter 2: What Is the Difference between ETH1 and ETH2

OpenVox GSM Gateway has two RJ45 Network ports, ETH1 and ETH2. If you choose ETH1, you can access Board 1 immediately, and access other boards with the same IP address but different port numbers. This will help to avoid IP conflict. If you choose ETH2, you can access different Boards with different IP addresses.

VoxStack provides 2 kinds of working modes, **Stand-alone** and **Cluster**.

⇒ Stand-alone: A single IP address manages one GSM modules(4 ports)

Stack Num	IP	Username	Password
1	172.16.99.1	admin	admin
2	172.16.99.2	admin	admin
3	172.16.99.3	admin	admin
4	172.16.99.4	admin	admin
5	172.16.99.5	admin	admin

⇒ Cluster: A single IP address manages up to 5 GSM modules (up to 20 ports).

Default IP: 172.16.99.1
User Name: admin
Password: admin

Chapter 3: How to Use Call Duration Limit Function

OpenVox GSM Gateway could support Call Duration Limit. You can control every port Total Call Time, and control Single Call Time. It will be useful for the customers working as operators.

Step 1: Choose One Port

For example, we want to set the first port. Please click GSM→ GSM Settings. Then click “Action”.



Step 2: Set Call Duration Limit

▼ Call Duration Limit Settings

Enable Single Call Duration Limit:	<input type="checkbox"/> OFF
Enable Call Duration Limitation:	<input type="checkbox"/> OFF

Enable Single Call Duration Limit	Define maximum call duration for single call. Example: If Time of single call set to 10, the call will be disconnected after talking 10*step second.
Enable call Duration Limitation	This function is limit the total call duration of GSM channel. The max call duration is between 1 to 999999 steps.

Example:

▼ Call Duration Limit Settings

Step:	<input type="text" value="60"/>	Second
Enable Single Call Duration Limit:	<input checked="" type="checkbox"/> ON	
Single Call Duration Limitation:	<input type="text" value="1"/>	
Enable Call Duration Limitation:	<input type="checkbox"/> OFF	

Step	Step length value range is 1-999 seconds, step length multiplied by time of single call just said a single call duration time allowed
Single call duration Limitation	The value of limitation single call, this value range is 1-999999.step length multiplied by time of single call just said a single call duration time allowed

Means: The single call will be disconnected after session lasts 59 seconds.

Definition of Setting Call Duration Limit

Enable call duration Limitation	This function is limit the total call duration of GSM channel. The max call duration is between 1 to 999999 steps
Call Duration Limitation	This function is to limit the total call duration of GSM channel. The max call duration is between 1 to 999999 steps.
Minimum Charging Time	A single call over this time, GSM side of the operators began to collect fees, unit for seconds
Alarm Threshold	Define a threshold value of call minutes, while the call minutes less than this value, the gateway will send alarm information to designated phone number via SMS
Alarm Phone Number	Receiving alarm phone number, user will received alarm message from gateway
Remain Time	This value is multiplied by to step length is a reset call call time
Enable Auto Reset	Automatic restore remaining talk time, that is, get total call minutes of GSM channel
Auto Reset Type	Reset call minutes by data, by week, by month

➤ Example 1:

▼ Call Duration Limit Settings

Step:	<input type="text" value="60"/>	Second
Enable Single Call Duration Limit:	<input type="checkbox"/> OFF	
Enable Call Duration Limitation:	<input checked="" type="checkbox"/> ON	
Call Duration Limitation:	<input type="text" value="10"/>	
Minimum Charging Time:	<input type="text" value="30"/>	Second
Alarm Threshold:	<input type="text" value="0"/>	
Alarm Phone Number:	<input type="text"/>	
Alarm Description:	<input type="text"/>	
Remain Time:	<input type="text" value="10"/>	<input type="button" value="Reset"/>
Enable Auto Reset:	<input checked="" type="checkbox"/> ON	
Auto Reset Type:	Day(1Day) ▼	
Next Reset Time:	<input type="text" value="2013-12-04 12:58:34"/>	

Means:

Scenario 1:

If the talk time is less than 30'', the Remain Time won't do any change. For example, if your talk time is 28'', the Remain Time is still 10''.

Scenario 2:

If your talk time is bigger than 30'', for example 38 seconds, the Remain Time will reduce 1, and become 9.

Scenario 3:

If your talk time is 62 seconds (one minute and 2 seconds), the Remain Time will be reduce 2 and become 8. That means if your talk time less than a step, it will be regard as a step.

Notice:

The Remain Time will be shown on SYSTEM Status interface dynamically.

GSM Information									
Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	GSM Status	Remain Time
Board-1-gsm-1		0	CHINA MOBILE	Registered (Home network)	7	22	100	READY	10
Board-1-gsm-2		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	1
Board-1-gsm-3		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
Board-1-gsm-4		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit

➤ Example 2:

Call Duration Limit Settings	
Step:	60 Second
Enable Single Call Duration Limit:	<input type="checkbox"/> OFF
Enable Call Duration Limitation:	<input checked="" type="checkbox"/> ON
Call Duration Limitation:	10
Minimum Charging Time:	30 Second
Alarm Threshold:	1
Alarm Phone Number:	13632919026
Alarm Description:	test alarm
Remain Time:	5 <input type="button" value="Reset"/>
Enable Auto Reset:	<input checked="" type="checkbox"/> ON
Auto Reset Type:	Day(1Day) <input type="button" value="v"/>
Next Reset Time:	2013-12-04 12:58:34

Means:

When your Remain Time is 1 (Alarm Threshold), the gateway will send a message to the phone 13632919026(Alarm Phone Number).The message content is “test alarm” (Alarm Description content).

The Remain Time will be reset at 2013-12-04 12:58:34.Because I enable the Auto Reset Type, and set it to 1 Day.

If you want to enable the Call Waiting features. Please enable it.

VoxStack SYSTEM | **GSM** | SIP | ROUTING | NETWORK | ADVANCED | LOGS

(WIRELESS GATEWAY)

GSM Settings | SMS Settings | SMS Sender | SMS Inbox | DTMF | Toolkit

GSM
DETAILS



Free Communication

OpenVox Solution



Port 1

Name:	<input type="text"/>
Speaker Volume:	70
Microphone Volume:	1
DAC Gain:	-15
ADC Gain:	-3
Dial Prefix:	<input type="text"/>
Pin Code:	<input type="text"/> <input type="checkbox"/> On
CLIR:	<input type="checkbox"/> OFF
Call Waiting:	<input checked="" type="checkbox"/> ON

Chapter 4: How to Use the Function of Modify IMEI

OpenVox GSM Gateway allows you to modify IMEI. We provide 2 ways, one is manual and the other is automatic. It will be useful when you need to change the IMEI of modules.

Step 1: How to Modify the IMEI

What does IMEI manually-modify mean:

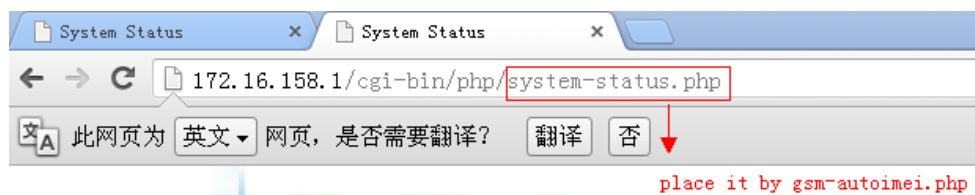
You could provide new IMEI numbers and choose the module you want to modify at any time. The original IMEI will be changed by your new IMEI.

What is IMEI automatically-modify:

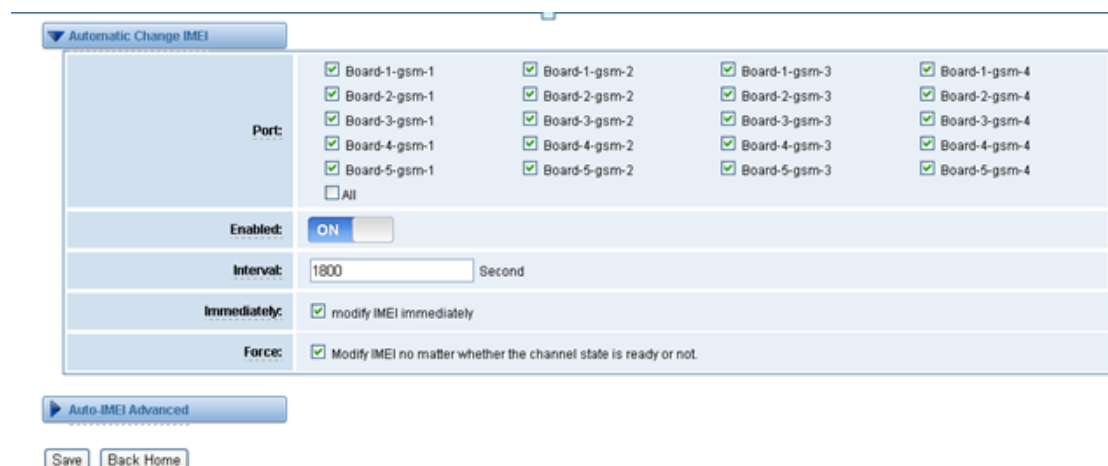
The system will randomly create some new IMEI by the rules you set. And modify the IMEI in a cycle. For example, modify all modules every 30 minutes.

Next we will guide you how to modify the IMEI automatically.

At first please login your system via Web, and look at your address bar. You will see like this "IP/cgi-bin/php/system-status.php". Then delete the "system-status.php", replace it with "gsm-autoimei.php", press "Enter" key.



You will see following figures.



Parameters:

Port	you can select the port you want to modify
Enable	“on” mean enable auto modify IMEI; “off” mean disable auto modify IMEI.
Interval	The time interval that you want to modify. For example, you set “1800” means once every 1800 seconds will be modify IMEI.
Immediately	Immediately begin to modify the IMEI.If you disable it, the system will be modify IMEI after some time, the time is “interval” setting.
Force	Modify IMEI no matter whether the channel state is ready or not

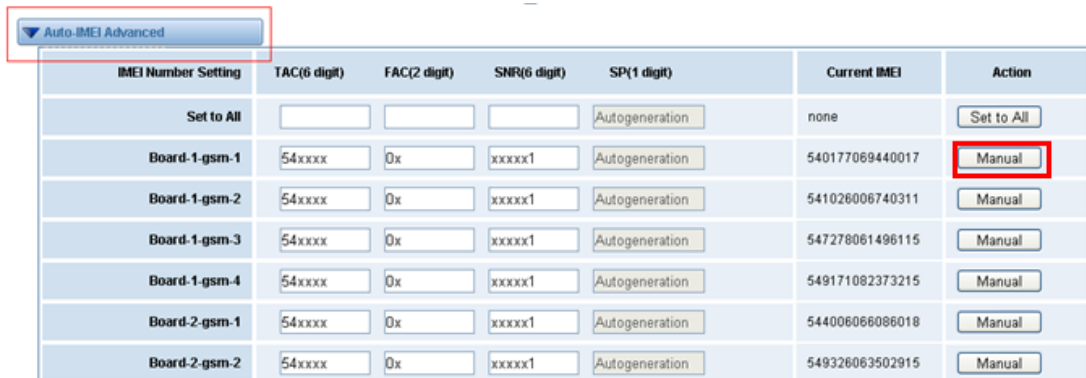
About “Auto-IMEI Advanced”, please click the button “Auto-IMEI Advanced”.

The x means that system will randomly create number to replace it. Set to all: apply the rule to all ports.

Just Click save and apply buttons. The system will begin to change the IMEI automatically according to your settings.

Of course, if you want to modify manually, please click the “Auto-IMEI Advanced” button. At first, please keep enable options “OFF”, like this:

And then, please click the “Auto-IMEI Advanced” button.



IMEI Number Setting	TAC(6 digit)	FAC(2 digit)	SNR(6 digit)	SP(1 digit)	Current IMEI	Action
Set to All	<input type="text"/>	<input type="text"/>	<input type="text"/>	Autogeneration	none	Set to All
Board-1-gsm-1	54xxxx	0x	xxxxx1	Autogeneration	540177069440017	Manual
Board-1-gsm-2	54xxxx	0x	xxxxx1	Autogeneration	541026006740311	Manual
Board-1-gsm-3	54xxxx	0x	xxxxx1	Autogeneration	547278061496115	Manual
Board-1-gsm-4	54xxxx	0x	xxxxx1	Autogeneration	549171082373215	Manual
Board-2-gsm-1	54xxxx	0x	xxxxx1	Autogeneration	544006068086018	Manual
Board-2-gsm-2	54xxxx	0x	xxxxx1	Autogeneration	549326063502915	Manual

For example, if you need to modify the “board-1-gsm-1”, just to click the “manual” button. And input your new IMEI, click “sure” button.



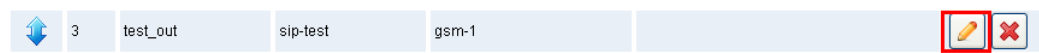
Chapter 5: Change the Callee ID

Step 1: Choose your Call Routing Rules

Please login your GSM Gateway and choose “ROUTING→Call Routing Rules”.



Choose your routing rules, for example, I choose my “Test_out” rule.



Step 2: Configure the Dial Patterns of Rules

Please Click “Advance Routing Rule”.

Advance Routing Rule

Dial Patterns that will use this Route

(prepend) + prefix | [match pattern / CallerId] ❌

+ Add More Dial Pattern Fields

Now I will give you some examples to help you understand how to use it flexibly.

➤ Example 1:

Dial Patterns that will use this Route

(prepend) + 9 | . / CallerId] ❌

+ Add More Dial Pattern Fields

At first the system will detect your callee ID, if it starts with "9", it will delete the number "9", then try to match "." (that means any number). If it could match, the system will send out the call.

For example, I try to call "10086", but I need to dial "910086". The system finds the first num could match the num "9", and delete "9", then match the ".", it will match any numbers. So the system will try to call "10086".

➤ Example 2:

Dial Patterns that will use this Route

(prepend) + 9 | 910086 / CallerId] ❌

If I try to call "910086", the system will find the number begin number "9", it will delete the number "9", and then try to match "910086". But it can't match, because the system has deleted the number 9, so 10086 can't match the 910086. So the system will not send the call.

Notice: About the match pattern, please refer the user manual page 44.

➤ Example 3:

Advance Routing Rule

Dial Patterns that will use this Route

(0755) + 9 | 10086 / CallerId] ❌

I try to call "910086". At first the system will detect the begin number, it can match "9". So the system will delete the number "9", now the number become "10086", not 910086. Now the system will continue, it will add a prefix 0755 for the number. At last send the call "075510086".

➤ Example 4:

Dial Patterns that will use this Route				
(0755)	+	9		(10086) / 1001

Now I try to use my extension 1001 to call "910086". At first the system will try to detect the begin num, it can match "9", so it deletes the number, now the number become "10086" and caller ID is 1001. Then the system will continue to match if it is 10086 and caller ID is 1001. Last the system will add a prefix 0755, then send the call to "075510086".

➤ Example 5:

Dial Patterns that will use this Route				
1	(0755)	+	9	(10086) / 1002
2	(027)	+	9	(10086) / 1001

+ Add More Dial Pattern Fields

Now I use the extension 1001 to call "910086". At first the system will try to match the first rule (table 1). It can match "9", so it will delete the number, the callee number become 10086. It will continue to match if the callee number is 10086 and the caller is 1001. The system finds the 1002 can't match extension (caller) 1001. So it can't match the rule 1. The system will try to match rule 2, it can match. At last call number is "02710086".

Of course, if you try to use the extension 1002 to make a call to "910086", it will send the number "075510086".

The whole process is as follows:

Dial Patterns that will use this Route				
4	(0755)	+	9	(10086) / 1002
8	(027)	+	9	(10086) / 1001

The red label is the order of Gateway try to match.

Chapter 6: How to Use the Time Routing Function

OpenVox GSM Gateway supports routing according to TIME. It can judge if execute the routing according to time. For example, you have a SIM card, it will cost lower during the time 0:00-2:00, I think maybe you need this function.

Step 1: Set a Routing According to Your Need

I set a call out routing as an example (SIP TRUNK → GSM port).

SYSTEM | GSM | SIP | **ROUTING** | NETWORK | ADVANCED | LOGS

Call Routing Rules | Groups

Modify a Call Routing Rule

Call Routing Rule

Routing Name:	Test_out
Call Comes in From:	1001
Send Call Through:	GSM_ALL

Advance Routing Rule

Save Cancel

The GSM_ALL is a "GSM port" group, the routing means that if 1001(SIP trunk) receives calls it will send the call to an available GSM port. Without any limit, the calls will be sent at once. We can set it just within a specified time horizon. Please click "Advance Routing Rule" button.

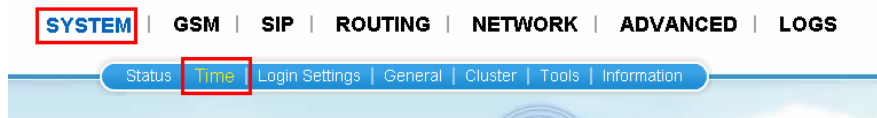
Send Call Through: GSM_ALL

Advance Routing Rule

Save Cancel

Step 2: Set Server Time

Before we set time routings, we need to set the server time (Gateway time). Because the gateway will be in their own time.



Please choose the Time menu.

Time Settings

System Time:	2014-1-2 09:33:54
Time Zone:	Chongqing
POSIX TZ String:	CST-8
NTP Server 1:	us.pool.ntp.org
NTP Server 2:	64.236.96.53
NTP Server 3:	time.nist.gov
Auto-Sync from NTP:	ON

Sync from NTP Sync from Client

Choose your local time zone in the “Time Zone” drop-down list. And click the Sync from NTP. You will see the server time on the top right corner once you set the gateway time.

Server time: 2014-1-2 09:54:34

VOXSTACK WIRELESS GATEWAY

SYSTEM | GSM | SIP | ROUTING | NETWORK | ADVANCED | LOGS

Call Routing Rules | Groups

ROUTING

Free Communication

OpenVox Solution

Modify a Call Routing Rule

Call Routing Rule

Routing Name: Test_out

Call Comes in From: 1001

Send Call Through: GSM_ALL

Advance Routing Rule

Dial Patterns that will use this Route

Time Patterns that will use this Route

Step 3: Set Sample Time Routing

➤ Example 1:

If you allow the routing send the calls just during 09:00—10:00 every day. You can set it as below:

Time Patterns that will use this Route

Time to start: 09 : 00

Time to finish: 10 : 00

Week Day start: -

Week Day finish: -

Month Day start: -

Month Day finish: -

Month start: -

Month finish: -

+ Add More Time Pattern Fields

The call will be sent successfully, because now the server time is “2014-1-2 09:51:32” .It meets the current setting of the time. If you try to make a call after 10 minutes, the call will be cut, because the time is not during 09:00--10:00.

➤ Example 2:

If you want the routing to send the call just during 09:00—10:00 from Monday to Friday, you can set it like this:

Time Patterns that will use this Route

Time to start: 09 : 00

Time to finish: 10 : 00

Week Day start: Monday

Week Day finish: Friday

Month Day start: -

Month Day finish: -

Month start: -

Month finish: -

+ Add More Time Pattern Fields

Now the call have been established, because now the gateway server time is 09:58 and today is Thursday, it can meet current settings of the time.

➤ Example 3:

Let's set a time rule as below:

Time Patterns that will use this Route			
Time to start:	09 : 10	Week Day start:	Thursday
Time to finish:	10 : 30	Week Day finish:	Thursday
		Month Day start:	02
		Month Day finish:	02
		Month start:	January
		Month finish:	January

+ Add More Time Pattern Fields

Today is the 2014-1-2 10:02, and Thursday. There is no doubt that the call will be established smoothly, because the server time now can match the time rule.

The process seems to be like this. When you have a call coming from 1001(SIP trunk), it will find the routing (1001-→GSM_ALL), and try to match the routing. If you set the time rule, it will try to match the time rule.

At first it will try to match the time. Now the server time is 10:02, it can match your setting (09:10—10:30). Then it will continue, try to match the Week Day, today is Thursday, It can match your setting (Thursday--Thursday). It doesn't finish, and will try to match other rules. Today is 2th that can match Month Day (02--02), and today is January. It matches all time rules. So the call will be established.

Step 4: Set Complex Time Routing

Now I want the call to be sent just from 2013-12-15 to 2014-1-3 and from Monday to Friday during 09:00-18:00. You can set it as follow:

At first, set the "Time to start" and "Time to finish", it is 09:00—18:00.

Time Patterns that will use this Route	
Time to start:	09 : 00
Time to finish:	18 : 00

Then we set the "Week Day start" and "Week Day finish". We need to set it is Monday to Friday.

Time Patterns that will use this Route	
Time to start:	09 : 00
Time to finish:	18 : 00
Week Day start:	Monday
Week Day finish:	Friday

+ Add More Time Pattern Fields

Then let's set the "Month Day start" and "Month Day finish". It is 12.15 to 1.3, how can we set it? Maybe you will set it as following:

Time Patterns that will use this Route			
Time to start:	09 : 00	Week Day start:	Monday
Time to finish:	18 : 00	Week Day finish:	Friday
		Month Day start:	15
		Month Day finish:	03
		Month start:	December
		Month finish:	January

But it is **wrong**. The right settings should be set as below:

At first, we set the “Month Day start” and “Month Day finish”, it should be 15th to next month 3th. That means it is 12.15---12.31 and 1.1—1.3. Please click the “+ Add More Time Pattern Fields”, then set it.

Time Patterns that will use this Route			
Time to start: 09 : 00	Week Day start: Monday	Month Day start: 15	Month start: -
Time to finish: 18 : 00	Week Day finish: Friday	Month Day finish: 31	Month finish: -
Time to start: 09 : 00	Week Day start: Monday	Month Day start: 01	Month start: -
Time to finish: 18 : 00	Week Day finish: Friday	Month Day finish: 03	Month finish: -

+ Add More Time Pattern Fields

At last, we set the “Month start” to “Month finish”. It is December to January.

Time Patterns that will use this Route					
1	Time to start: 09 : 00	Week Day start: Monday	Month Day start: 15	Month start: December	✘
	Time to finish: 18 : 00	Week Day finish: Friday	Month Day finish: 31	Month finish: December	
2	Time to start: 09 : 00	Week Day start: Monday	Month Day start: 01	Month start: January	✘
	Time to finish: 18 : 00	Week Day finish: Friday	Month Day finish: 03	Month finish: January	

+ Add More Time Pattern Fields

Today is the 2014-1-2 10:02, and Thursday. At first it will try to match the label “1” rule, but it can’t match it, so it continues to match label “2” rule. It can match correctly. So the call will be established.

Notice: Don’t forget to click the save button and apply it after you change the settings.

Settings have been changed. Calls may be terminated when you apply these changes. Do you want to apply now?

Chapter 7: The SIP Connection Ways.

Now the OpenVox GSM Gateway supports 3 ways to connect via SIP protocols. One is the GSM Gateway as a SIP server, one is the GSM Gateway as a SIP endpoint and register, the other is the Gateway as a SIP endpoint (IP to IP).

- The first way that GSM Gateway works as a SIP server.

Step 1: Configure SIP Server in Gateway

Please click the “SIP→SIP Endpoints→Add New SIP Endpoint” to set SIP server. The following figure shows detail information about how to set it.



[Add New SIP Endpoint](#)

Edit SIP Endpoint "1001"

▼ Main Endpoint Settings

Name:	1001
User Name:	1001 <input type="checkbox"/> Anonymous
Password:	1001
Registration:	Endpoint registers with this gateway
Hostname or IP Address:	dynamic
Transport:	UDP
NAT Traversal:	Yes

▶ Advanced:Registration Options

Other parameters about SIP, please set according to your requirements because there is no need to set them in simple calls.

Now I will let my softphone register to the Gateway. Of course, if you need call out or call in, also you need to make the routing. I configure it like follows.

SYSTEM | GSM | SIP | **ROUTING** | NETWORK | ADVANCED | LOGS

[Call Routing Rules](#) | Groups

▼ Call Routing Rule

Routing Name:	Test_out
Call Comes in From:	1001
Send Call Through:	gsm-1

Step 2: Configure SIP Endpoint in Softphone

Please run your softphone, I use the X-lite as my SIP endpoint, and register it to the Gateway.

Account Voicemail Topology Presence Advanced

User Details

Display Name: 1001

User name: 1001

Password: ****

Authorization user name: 1001

Domain: 172.16.8.42

Domain Proxy

Register with domain and receive incoming calls

Send outbound via:

domain

proxy Address: []

target domain

Dialing plan: #1\|a.T;match=1;prestrip=2;

确定 取消 应用 (A)

Please input the correct username, password and Domain. You will see the register information in the WEB of Gateway once you register.

SIP Information

Endpoint Name	User Name	Host	Registration	SIP Status
1025	1025	172.16.2.209	client	Registered
1001	1001	172.16.8.60	server	OK (103 ms)

Now we can try to make a call via softphone.



gsm-1	0	CHINA MOBILE	Registered (Home network)	6	18	100	CALL ACTIVE Called to 10086 00:00:02	No Limit
-------	---	--------------	---------------------------	---	----	-----	--------------------------------------	----------

As you see, the call has been established.

- The second way that GSM Gateway works as a SIP endpoint and register.

Step 1: Configure SIP Endpoint in Gateway

We register the SIP endpoint to my PBX via SIP protocol. This is the Gateway SIP setting.

▼ Main Endpoint Settings

Name:	1025
User Name:	1025 <input type="checkbox"/> Anonymous
Password:	1025
Registration:	This gateway registers with the endpoint ▼
Hostname or IP Address:	172.16.2.209
Transport:	UDP ▼
NAT Traversal:	Yes ▼

Step 2: Configure SIP Peer in PBX

The PBX I use is Elastix, you can set it in “PBX→Trunk→SIP Trunk”

```
host=dynamic
username=1025
secret=1025
type=friend
fromuser=1025
disallow=all
allow=g729,g723,ulaw,alaw
dtmfmode=rfc2833
insecure=port,invite
context=from-playback
```

Step 3: Check the Register Status in Gateway

Please click the “SYSTEM→Status→SIP Information”.

SYSTEM	GSM	SIP	ROUTING	NETWORK	ADVANCED	LOGS
Status	Time	Login Settings	General	Cluster	Tools	Information
1025	1025	172.16.2.209	client	Registered		
1001	1001	172.16.8.60	server	OK (103 ms)		

As you see, it can register successfully.

- The last way (IP to IP)

At first, please configure a SIP peer in OpenVox Gateway.

Add New SIP Endpoint

▼ Main Endpoint Settings

Name:	9999
User Name:	<input type="checkbox"/> Anonymous
Password:	<input type="password"/>
Registration:	None
Hostname or IP Address:	172.16.2.209
Transport:	UDP
NAT Traversal:	Yes

Then please configure it in your PBX, I configure it in Elastix. For example:

Trunk Name: 9999

PEER Details:

```
host=172.16.8.42  
type=friend  
fromuser=9999
```

You can see the status:

Endpoint Name	User Name	Host	Registration	SIP Status
1025	1025	172.16.2.209	client	Registered
1001	1001	172.16.8.60	server	OK (131 ms)
9999	anonymous	172.16.2.209	none	Unmonitored

Chapter 8: How to Use the Cluster Function in OpenVox Gateway

OpenVox GSM Gateway provides the cluster function to manage every board, it will quite flexible. Users can combine the boards at random. For example you can configure your Gateway with 4/8/12/16/20 GSM ports via cluster function.

Before we configure the cluster, we need to know the OpenVox GSM Gateway IP address. Every OpenVox GSM Board have 2 IP addresses (one is a Reserved Address). It means that if you have 5 boards (20 GSM ports) in the Box. You will have 10 IP addresses, every board have 2 IP addresses.

Step 1: Configure the IP Address in Gateway

Please click the menu of "NETWORK→LAN Settings".

IPv4 Settings

Address:	172.16.8.42
Netmask:	255.255.0.0
Default Gateway:	172.16.0.1

Reserved Access IP

Enable:	ON <input type="checkbox"/>
Reserved Address:	192.168.99.1
Reserved Netmask:	255.255.255.0

You can configure the "IPv4 Settings", but about "Reserved Access IP", you just can enable or disable it. If you enable it, you can visit gateway with this address, the same as IPv4 Address. Now, let us use the "cluster" function.

Step 2: Know the IP Address of Gateway

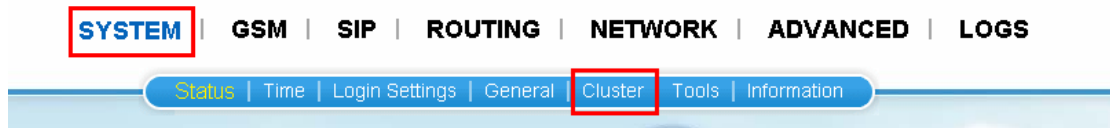
Now I have one 5 boards (20 ports) GSM Gateway, the first Board IP is (172.16.99.1 and 192.168.99.1), the other in sequence are (172.16.99.2 and 192.168.99.2).....(172.16.99.5 and 192.168.99.5). That is default IP address.

Now I login the 172.16.99.1 or 192.168.99.1, you will see like that:

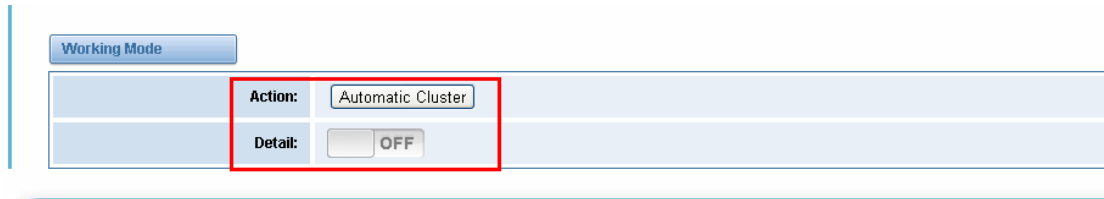
Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	GSM Status	Remain Time
gsm-1		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-2		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-3		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-4		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit

Step 2: Configure the Cluster

Please click the "SYSTEM→Cluster" menu.



You will see follow:



Now OpenVox GSM Gateway can support 2 ways to configure the cluster. One is manual, the other is Automatic. At first, we cluster manual.

➤ Manual:

Please enable the "Detail", you will see as follows:

Action:	Automatic Cluster
Detail:	ON <input type="checkbox"/>
Mode:	Stand-alone <input type="button" value="v"/>
Action:	Manual Cluster

Now I want the first board to work as a master board and manage other 4 boards. So I choose the mode is "Master".

Action:	Automatic Cluster
Detail:	ON <input type="checkbox"/>
Mode:	Master <input type="button" value="Set Default"/>
Password:	<input type="text"/>
Master IP(Local IP):	<input type="text"/>
Slaves IP List:	Board-2 Original IP: <input type="text"/> Target IP: <input type="text"/>
	Board-3 Original IP: <input type="text"/> Target IP: <input type="text"/>
	Board-4 Original IP: <input type="text"/> Target IP: <input type="text"/>
	Board-5 Original IP: <input type="text"/> Target IP: <input type="text"/>
Remain Original IP address:	ON <input type="checkbox"/>
Action:	Manual Cluster

Parameters :

Password	The Master mode password, must be 4-16 bits and 0-9
Master IP	The Master's target IP address
Slaves IP list	The slaves original IP and target IP address
Remain Original address	You can enable or disable it and decide that if you can visit the Original IP address.

Now I set like that:

Mode:	Master <input type="button" value="Set Default"/>
Password:	9999
Master IP(Local IP):	192.168.9.1
Slaves IP List:	Board-2 Original IP: 172.16.99.2 Target IP: 192.168.9.2
	Board-3 Original IP: 172.16.99.3 Target IP: 192.168.9.3
	Board-4 Original IP: 172.16.99.4 Target IP: 192.168.9.4
	Board-5 Original IP: 172.16.99.5 Target IP: 192.168.9.5
Remain Original IP address:	<input checked="" type="checkbox"/> ON
Action:	<input type="button" value="Manual Cluster"/>

At first, I set the Master IP(Local IP) is 192.168.9.1, I give another IP to the first board(172.16.99.1), now the first Board have 3 IP address, one is 172.16.99.1, one is 192.168.9.1(Reserved IP address) and the other is 192.168.9.1.

Slaves IP List:

Board-2	Original IP:	172.16.99.2	Target IP:	192.168.9.2
---------	--------------	-------------	------------	-------------

My original IP is 172.16.99.2, and I give it another IP 192.168.9.2. It has 3 IP addresses.

Remain Original IP address:	<input checked="" type="checkbox"/> ON
------------------------------------	--

That means if you need to remain original IP address, but you disable it, you couldn't visit the Gateway with IP 172.16.99.2/172.16.99.3/172.16.99.4/172.16.99.5.

When you ensure your settings are right, please click the "Manual Cluster" button, you will see follows:

Manual Cluster

Report
172.16.99.2 is alive... set 172.16.99.2 to 192.168.9.2 Set 192.168.9.2 OK
172.16.99.3 is alive... set 172.16.99.3 to 192.168.9.3 Set 192.168.9.3 OK
172.16.99.4 is alive... set 172.16.99.4 to 192.168.9.4 Set 192.168.9.4 OK
172.16.99.5 is alive... set 172.16.99.5 to 192.168.9.5 Set 192.168.9.5 OK

That means you have cluster successfully.

You can login with the IP address 172.16.99.1 and check if you can see 20 ports.

172.16.99.1/cgi-bin/php/system-status.php

GSM Information

Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	GSM Status	Remain Time
gsm-1		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-2		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-3		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-4		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-5		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-6		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-7		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-8		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-9		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-10		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-11		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-12		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-13		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-14		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-15		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-16		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-17		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-18		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-19		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-20		-1		Undetected SIM Card	0	0	0		No Limit

You can use it as a 20-port Gateway. For example, I try to change the IP address.

SYSTEM | GSM | SIP | ROUTING | **NETWORK** | ADVANCED | LOGS

LAN Settings | DDNS Settings | Toolkit

IPv4 Settings	
Address:	172.16.99.1
Netmask:	255.255.0.0
Default Gateway:	172.16.0.1

LAN IPv4	
Interface:	eth0
Type:	Static
MAC:	A0:98:05:01:08:63

IPv4 Settings	
Address:	172.16.8.42
Netmask:	255.255.0.0
Default Gateway:	172.16.0.1

Now I use 172.16.8.42 to login the gateway, you will see that:

← → ↻ 📄	172.16.8.42/cgi-bin/php/system-status.php
---------	---

Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	GSM Status	Remain Time
gsm-1		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-2		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-3		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-4		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-5		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-6		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-7		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-8		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-9		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-10		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-11		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-12		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-13		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-14		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-15		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-16		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-17		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-18		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-19		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-20		-1		Undetected SIM Card	0	0	0		No Limit

Maybe you will ask me "I have a 20-port gateway, now I want to set it two gateways, one is 8 ports, the other is 12 ports. How can I do that?"

Ok, let us set it, in order to describe more clearly, I first factory reset my gateway. Now that gateway is in factory model. It has 5 boards in box, and every board has 2 IP addresses: (172.16.99.1/192.168.99.1).....(172.16.99.5/192.168.99.5).

At first, we login 172.16.99.1 and configure 8-port Gateway. Please choose the menu "SYSTEM→Cluster", and set like follows:

Working Mode

Action:

Detail:

Mode:

Password:

Master IP(Local IP):

Slaves IP List:

Board-2	Original IP:	<input type="text" value="172.16.99.2"/>	Target IP:	<input type="text" value="192.168.9.2"/>
Board-3	Original IP:	<input type="text"/>	Target IP:	<input type="text"/>
Board-4	Original IP:	<input type="text"/>	Target IP:	<input type="text"/>
Board-5	Original IP:	<input type="text"/>	Target IP:	<input type="text"/>

Remain Original IP address:

Action:

And then click the button “Manual Cluster”. You will see follow output:

Manual Cluster

Report
172.16.99.2 is alive... set 172.16.99.2 to 192.168.9.2 Set 192.168.9.2 OK

Notice: If you can't see any output in Report, please enable the “Detail” and set again.

Now, let's login 172.16.99.1 and check it.

As you see, you have got an 8-port GSM Gateway.

GSM Information

Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	GSM Status	Remain Time
gsm-1		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-2		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-3		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-4		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-5		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-6		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-7		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-8		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit

We continue to configure the 12-port GSM Gateway.

Please login 172.16.99.3 (it is the third board IP, because the second board has been used as a slave board of 8-port GSM Gateway (172.16.99.1)).



GSM Information

Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	GSM Status	Remain Time
gsm-1		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-2		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-3		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-4		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit

Please set the “Cluster” like follows:

Working Mode

Action: Automatic Cluster

Detail: ON

Mode: Master

Password: 9999

Master IP(Local IP): 192.168.9.3

Slaves IP List:

Board-2	Original IP:	172.16.99.4	Target IP:	192.168.9.4
Board-3	Original IP:	172.16.99.5	Target IP:	192.168.9.5
Board-4	Original IP:		Target IP:	
Board-5	Original IP:		Target IP:	

Remain Original IP address: ON

Action: Manual Cluster

When ensure the settings are right, please click the “Manual Cluster” button, you will see follows output.

Manual Cluster

Report

```

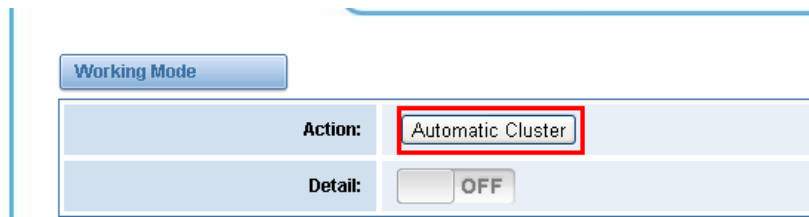
172.16.99.4 is alive...
set 172.16.99.4 to 192.168.9.4
Set 192.168.9.4 OK
172.16.99.5 is alive...
set 172.16.99.5 to 192.168.9.5
Set 192.168.9.5 OK
    
```

Now, let’s see if we get the 12-port GSM Gateway.

GSM Information

Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	GSM Status	Remain Time
gsm-1		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-2		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-3		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-4		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-5		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-6		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-7		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-8		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-9		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-10		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-11		0	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
gsm-12		-1		Undetected SIM Card	0	0	0		No Limit

Above it is the manual cluster function. The automatic cluster function means “one key cluster”. You just need to click one button and the system will find all OpenVox GSM Gateways via MAC address then make them work as slave boards.



Chapter 9: How to expand functions of OpenVox GSM Gateway

You can setup your personalized dialplan like setting in asterisk. The manual will refer you how to setup the IVR (include DISA) and Callback.

➤ IVR

OpenVox GSM Gateway can be used as a sample PBX and you can use it directly without any PBX. Next I'll show you the basic functions of IVR.

Step 1: Configure the SIP in the Gateway

Please login your GSM gateway, and select the "SIP→SIP Endpoints", and then click t "Add New SIP Endpoint" button.



Add New SIP Endpoint

Step 2: Edit SIP Endpoint in the Gateway

Now the OpenVox GSM gateway supports 3 kinds of connecting ways via SIP protocol. One is the Gateway as a SIP server, one is the Gateway as a SIP peers registered to PBX, and the last is the IP to IP.

I choose the first way to show you how to set the IVR.

▼ Main Endpoint Settings	
Name:	1001
User Name:	1001 <input type="checkbox"/> Anonymous
Password:	1001
Registration:	Endpoint registers with this gateway ▼
Hostname or IP Address:	dynamic
Transport:	UDP ▼
NAT Traversal:	Yes ▼

Notice: You can choose different connecting ways by selecting the “Registration” options. I select the “Endpoint registers with this gateway” which means the Gateway will work as a SIP server. You can register your softphone to the gateway directly.

Just need to save it and apply. Be shown as below:

Save	Cancel
------	--------

Settings have been changed. Calls may be terminated when you apply these changes. Do you want to apply now? Apply

Now, let’s set the second SIP server.

▼ Main Endpoint Settings	
Name:	1002
User Name:	1002 <input type="checkbox"/> Anonymous
Password:	1002
Registration:	Endpoint registers with this gateway ▼
Hostname or IP Address:	dynamic
Transport:	UDP ▼
NAT Traversal:	Yes ▼

At last, we register our softphone to the GSM Gateway.

Account | Voicemail | Topology | Presence | Advanced

User Details

Display Name: 1001

User name: 1001

Password: *****

Authorization user name: 1001

Domain: 172.16.8.42

You will see the registration status in the “SYSTEM→Status→SIP Information”.

SIP Information

Endpoint Name	User Name	Host	Registration	SIP Status
1025	1025	172.16.2.209	client	Registered
1001	1001	172.16.8.60	server	OK (108 ms)
9999	anonymous	172.16.2.209	none	Unmonitored
1002	1002	172.16.8.60	server	OK (13 ms)

Step 3: Set SSH in the Gateway

OpenVox Gateway can support SSH login, so that you can know more details, and expand your own applications.

Please select the “SYSTEM→Login Settings”.

SYSTEM | GSM | SIP | ROUTING | NETWORK | ADVANCED | LOGS

Status | Time | Login Settings | General | Cluster | Tools | Information

SSH Login Settings

Enable: ON

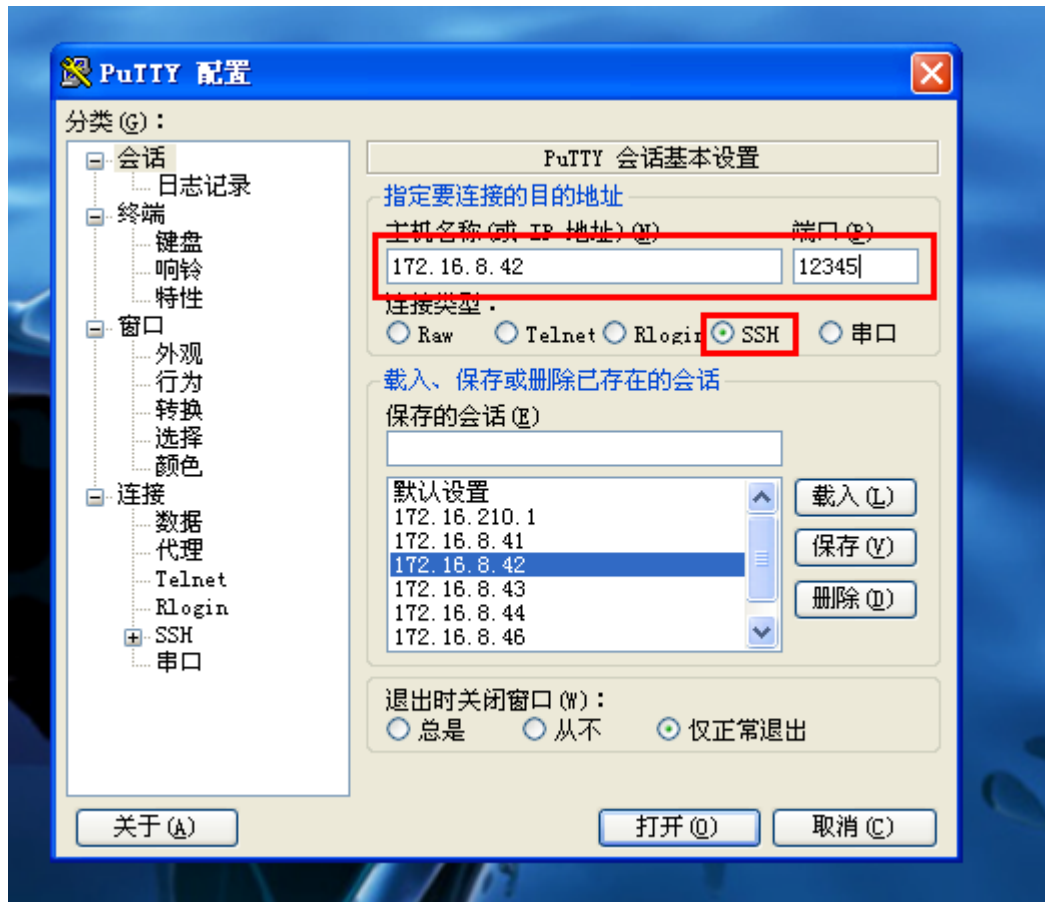
User Name: super

Password: admin

Port: 12345

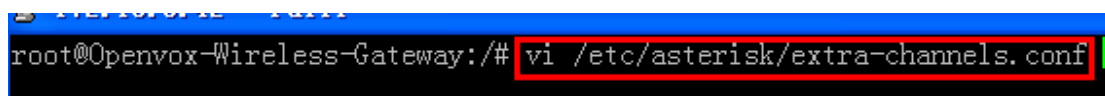
Save

Login the Gateway via SSH.



Step 4: Edit the GSM Port Context

Please edit the file /etc/asterisk/extra-channels.conf, when you open the file, you will see:



```

; Span 1: opvxxg4xx/0/1 "OpenVox G400P GSM/CDMA PCI Card 0" AMI/CCS
group=1
context=IVR
signalling = gsm
vol=70
mic=1
dacgain=-15
adcgain=-3
debugat=on
smscodec=utf-8
;hwdtmfdet=1
anonymouscall=off
call_waiting=off
band=
dialprefix=
switchtype=SIMCOM_SIM840W
channel => 1

; Span 2: opvxxg4xx/0/2 "OpenVox G400P GSM/CDMA PCI Card 0" AMI/CCS
group=1
context=gsm-2
signalling = gsm
vol=70
mic=1
dacgain=-15
adcgain=-3
debugat=on
smscodec=utf-8
;hwdtmfdet=1
anonymouscall=off
call_waiting=off
band=
dialprefix=
switchtype=SIMCOM_SIM840W
channel => 3

; Span 3: opvxxg4xx/0/3 "OpenVox G400P GSM/CDMA PCI Card 0" AMI/CCS

```

Please change the default context. Above I have changed the context of gsm-1. When the first gsm port receives a call, it will go to the context dialplan. You can edit your own dialplan.

Step 5: Edit the IVR Dialplan

Please open the file `/etc/asterisk/extensions_custom.conf`, and edit it.

```

root@Openvox-Wireless-Gateway:/# vi /etc/asterisk/extensions_custom.conf ]

```

Of course, you can edit it via our web interface also.



Configuration Files

File Name
enum.conf
extconfig.conf
extensions.conf
extensions_custom.conf
extensions_macro.conf
extensions_routing.conf
extra-channels.conf
extra-global.conf
features.conf
gw.conf

◀ 1 2 3 4 5 ▶ 2 / 5 go

When you finished editing, please reload the asterisk.

◀ 1 2 3 4 5 ▶ 2 / 5 go

New Configuration File Reload Asterisk

Notice: You can import your own recording file to OpenVox Gateway. I have imported my recording file to the gateway. And you need to add some modules in the gateway. On the other hand, I suggest you to use the gsm codec record.

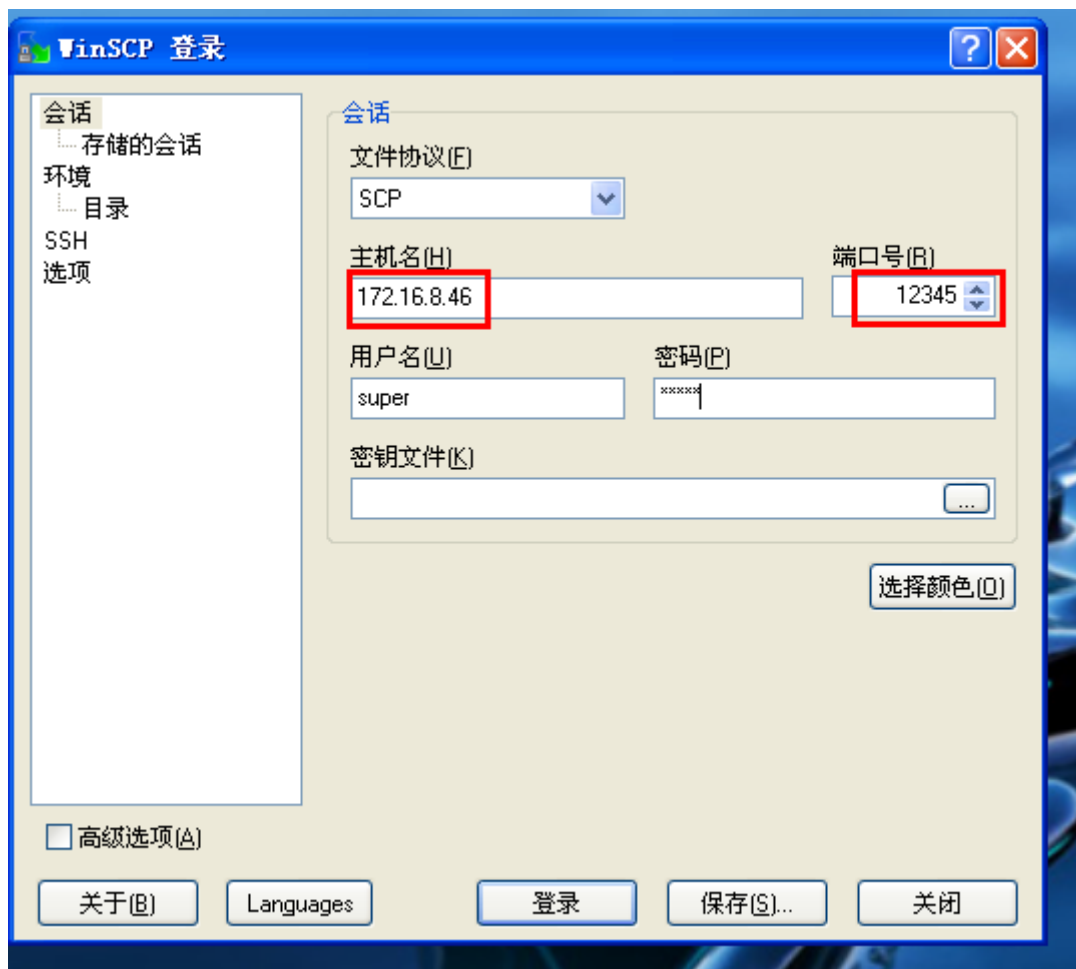
```
172.16.8.46 - PuTTY
root@Openvox-Wireless-Gateway:/etc/asterisk# asterisk -r
```

```
Openvox-Wireless-Gateway*CLI> module load format_gsm.so
```

```
Openvox-Wireless-Gateway*CLI> module load codec_gsm.so
```

How to import recording file to gateway:

You can download a tool to import the file to the gateway, I use the winscp and you can download it from internal.



The sample of dialplan:

```
[IVR]
;exten => s,1,Answer()
exten => s,1,Set(LOOPCOUNT=0)
exten => s,n(begin),Set(TIMEOUT(digit)=3)
exten => s,n,Set(TIMEOUT(response)=10)
exten => s,n,Background(/etc/asterisk/sounds/welcome)
exten => s,n,WaitEXTEN(2)
exten => s,n,Goto(t,1)
exten => s,n(dial),Dial(sip/1001)
;exten => s,n(dial),Dial(sip/8899/${Forward_CALLEEID})
;exten => s,n(dial),GrpPolicy(GSM_OUT)
;exten => s,n,Macro(dial-failover,${Forward_CALLEEID},, ${POLICY_GSM_OUT} )
exten => 1,1,Playback(/etc/asterisk/sounds/please_hold_cn)
exten => 1,n,Goto(s,dial)
exten => 2,1,Playback(/etc/asterisk/sounds/please_hold_en)
exten => 2,n,Goto(s,dial)
exten => i,1,Goto(loop,1)
exten => t,1,Goto(loop,1)
exten => loop,1,Set(LOOPCOUNT=${[${LOOPCOUNT} + 1])

exten => loop,n,GotoIf(${[${LOOPCOUNT} > 2]}?hang,1)
exten => loop,n,Goto(s,begin)
exten => hang,1,Hangup
exten => h,1,WriteCDR("${CDR(src)}", "${CDR_CALLEEID}", "gsm-1", "${CDR_TOCHAN}", "$
exten => h,n,Set(SPAN=1)
exten => h,n,Set(SMSTEXT=${CDR(src)} called you at ${CDR(start)},)
exten => h,n,System(sleep 5)
exten => h,n,GotoIf("${CDR(disposition)}" = "ANSWERED"?answered:missed)
exten => h,n(answered),Set(ANSWERED=Please take a short note as a reminder and f
exten => h,n,System(/usr/bin/asterisk -rx 'gsm send sms ${SPAN} ${Forward_CALLEE
exten => h,n,Goto(hangup)
exten => h,n(missed),Set(MISSED=Please call back ASAP in case there is something
exten => h,n,System(/usr/bin/asterisk -rx 'gsm send sms ${SPAN} ${Forward_CALLEE
exten => h,n(hangup),Hangup
```

This dialplan means that when someone calls GSM-Port 1, the gateway will play a voice. And then the custom will choose different services by inputting different digits. The gateway will detect DTMF, and execute different operations.

For example, when you hear sound, and press digit 1, the extension 1001 will ring. Of course you can setup richer dialplan according to your need, just like you setting up it in asterisk. You can also write it via AGI and AMI. It can support PHP and other program languages.

Another example:

Image that, when you have a call from the Trunk. You want to match the extension, if the extension can match your settings, it will ask the caller input the destination number, and then choice a fix port send the call. If not, it will ask the caller input the PIN and match the PIN, If the PIN can match, it will ask input the destination number, and then send the call, if not it will hang up the calls directly.

Step 1: Create a Trunk via the WEB

Edit SIP Endpoint "9999"

Main Endpoint Settings	
Name:	9999
User Name:	9999 <input type="checkbox"/> Anonymous
Password:
Registration:	This gateway registers with the endpoint <input type="button" value="v"/>
Hostname or IP Address:	172.16.8.44
Transport:	UDP <input type="button" value="v"/>
NAT Traversal:	Yes <input type="button" value="v"/>

Step 2: Login your gateway via ssh, and find the Trunk settings in the /etc/asterisk/sip_endpoints.conf, and change the context=sipinbound to context=IVR. Of course, you can change it via the WEB(Advance→Asterisk File Edit)

```
session-refresher=uas
insecure=port,invite
type=friend
context=IVR
setvar=SIPROUTE=sip-9999-172.16.8.44
endpoint_name=9999
```

That means when you have a call from this Trunk, it will choose the IVR dialplan, you can use other text, just need you setup it in dialplan.

Step 3: reload the SIP settings

Run command: asterisk -r

Run command: "sip reload" or "core reload";

```
root@Openvnx-Wireless-Gateway:/etc/asterisk#
root@Openvnx-Wireless-Gateway:/etc/asterisk#
root@Openvnx-Wireless-Gateway:/etc/asterisk# asterisk -r
Cannot read termcap database;
using dumb terminal settings.
Openvnx-Wireless-Gateway*CLI> core reload
```

Step 4: Setup the dialplan:

Run command: vi /etc/asterisk/extensions_custom.conf

Step 5: Edit the Dialplan:

[IVR]

exten => _X.,1,noop(=====IVR=====)

;exten => _X.,n,Answer()

exten => _X.,n,Set(MYEXTEN=\${EXTEN})

```
exten => _X.,n,Set(MYCALLERID=${CALLERID(num)})
exten=>_X.,n,GotoIf($[${MYEXTEN}=442037342594] | ${MYEXTEN}=16474960185] | ${MYEXTEN}=61390880337] | ${MYEXTEN}=6767671000]${MYEXTEN}=442031294012]?CALLOUTDIR
ECTLY,_X.,1)
exten => _X.,n,Answer()
exten => _X.,n,Noop(===== ${MYCALLERID:0:2} =====)
;exten => _X.,n,Goto(AUTHEIN,_X., ${MYCALLERID:0:2})
exten => _X.,n,Goto(CHOICEDID,_X.,1)
exten => _X.,n,Hangup()
```

[CHOICEDID]

```
exten => _X.,1,GotoIf($[${MYEXTEN}=442037342594]?AUTHEIN,_X.,01)
exten => _X.,n,GotoIf($[${MYEXTEN}=390294756396]?AUTHEIN,_X.,02)
exten => _X.,n,GotoIf($[${MYEXTEN}=442031291067]?AUTHEIN,_X.,03)
exten => _X.,n,GotoIf($[${MYEXTEN}=442031291070]?AUTHEIN,_X.,04)
exten => _X.,n,GotoIf($[${MYEXTEN}=442031291071]?AUTHEIN,_X.,05)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348944]?AUTHEIN,_X.,06)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348945]?AUTHEIN,_X.,07)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348946]?AUTHEIN,_X.,08)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348947]?AUTHEIN,_X.,09)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348980]?AUTHEIN,_X.,10)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348981]?AUTHEIN,_X.,11)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348982]?AUTHEIN,_X.,12)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348983]?AUTHEIN,_X.,13)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348984]?AUTHEIN,_X.,14)
exten => _X.,n,GotoIf($[${MYEXTEN}=442034111865]?AUTHEIN,_X.,15)
exten => _X.,n,GotoIf($[${MYEXTEN}=442034682146]?AUTHEIN,_X.,16)
exten => _X.,n,GotoIf($[${MYEXTEN}=442035820783]?AUTHEIN,_X.,17)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037342073]?AUTHEIN,_X.,18)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037342486]?AUTHEIN,_X.,19)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037342552]?AUTHEIN,_X.,20)
exten => _X.,n,Hungup()
```

[CALLOUTDIRECTLY]

```
exten => _X.,1,Noop(=====CALLOUTDIRECTLY=====)
;exten => _X.,n,Progress()
exten => _X.,n,Macro(dial-failover,,0714600800,extra/1,0,gsm-1)
exten => _X.,n,Hangup()
```

[AUTHEIN]

```
;exten=>_X.,01,GotoIf($[${MYCALLERID}=61386093888] | ${MYCALLERID}=441923233214] | ${
MYCALLERID}=447956494166] | ${MYCALLERID}=442070091605] | ${MYCALLERID}=9471460
0800] | ${MYCALLERID}=1001]?destination:pin)
exten=>_X.,02,GotoIf($[${MYCALLERID}=+390294756396] | ${MYCALLERID}=0294756396] | ${
```

{MYCALLERID}=441923233214] | \${MYCALLERID}=44207001605] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=1001] | \${MYCALLERID}=1245]]?destion:pin)
exten=>_X.,03,GotoIf(\${MYCALLERID}=9992] | \${MYCALLERID}=999] | \${MYCALLERID}=447824705843] | \${MYCALLERID}=447502298673] | \${MYCALLERID}=442037342594] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,04,GotoIf(\${MYCALLERID}=447577262411] | \${MYCALLERID}=447455371671] | \${MYCALLERID}=442088101167] | \${MYCALLERID}=447448209394] | \${MYCALLERID}=442037342594] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,05,GotoIf(\${MYCALLERID}=442037342594] | \${MYCALLERID}=442089035600] | \${MYCALLERID}=447782223566] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,06,GotoIf(\${MYCALLERID}=442085909738] | \${MYCALLERID}=447956904363] | \${MYCALLERID}=442037342594] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,07,GotoIf(\${MYCALLERID}=447702080740] | \${MYCALLERID}=442037342594] | \${MYCALLERID}=1004] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,08,GotoIf(\${MYCALLERID}=61386093888] | \${MYCALLERID}=9471444440] | \${MYCALLERID}=1004] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,09,GotoIf(\${MYCALLERID}=61386093888] | \${MYCALLERID}=9471444440] | \${MYCALLERID}=1004] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,10,GotoIf(\${MYCALLERID}=61386093888] | \${MYCALLERID}=9471444440] | \${MYCALLERID}=1004] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,11,GotoIf(\${MYCALLERID}=61386093888] | \${MYCALLERID}=9471444440] | \${MYCALLERID}=1004] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,12,GotoIf(\${MYCALLERID}=61386093888] | \${MYCALLERID}=9471444440] | \${MYCALLERID}=1004] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,13,GotoIf(\${MYCALLERID}=61386093888] | \${MYCALLERID}=9471444440] | \${MYCALLERID}=1004] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,14,GotoIf(\${MYCALLERID}=61386093888] | \${MYCALLERID}=9471444440] | \${MYCALLERID}=1004] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,15,GotoIf(\${MYCALLERID}=61386093888] | \${MYCALLERID}=9471444440] | \${MYCALLERID}=1004] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)
exten=>_X.,16,GotoIf(\${MYCALLERID}=61386093888] | \${MYCALLERID}=9471444440] | \${MYCALLERID}=1004] | \${MYCALLERID}=94777760265] | \${MYCALLERID}=94714600800] | \${MYCALLERID}=1001]]?destion:pin)


```
exten=>_X.,17,GotoIf($[${MYCALLERID}=61386093888]|${MYCALLERID}=9471444440)|${MYCALLERID}=1004)|${MYCALLERID}=94777760265)|${MYCALLERID}=94714600800)|${MYCALLERID}=1001]?destion:pin)
exten=>_X.,18,GotoIf($[${MYCALLERID}=61386093888]|${MYCALLERID}=9471444440)|${MYCALLERID}=1004)|${MYCALLERID}=94777760265)|${MYCALLERID}=94714600800)|${MYCALLERID}=1001]?destion:pin)
exten=>_X.,19,GotoIf($[${MYCALLERID}=61386093888]|${MYCALLERID}=9471444440)|${MYCALLERID}=1004)|${MYCALLERID}=94777760265)|${MYCALLERID}=94714600800)|${MYCALLERID}=1001]?destion:pin)
exten=>_X.,20,GotoIf($[${MYCALLERID}=61386093888]|${MYCALLERID}=9471444440)|${MYCALLERID}=1004)|${MYCALLERID}=94777760265)|${MYCALLERID}=94714600800)|${MYCALLERID}=1001]?destion:pin)
exten => _X.,n(destion),Goto(INPUTNUM,_X.,1)
exten => _X.,n(pin),Goto(INPUTPIN,_X.,1)
exten => _X.,n,Hangup()
```

[INPUTNUM]

```
exten => _X.,1,Noop(=====inputnumber=====)
exten => _X.,n,Background(number)
exten => _X.,n,Noop(===== ${CALLERID(name)} =====)
exten => _X.,n,DISA(no-password,${MYEXTEN})
;exten => _X.,n,DISA(no-password,CallOut${MYCALLERID:0:2})
```

[INPUTPIN]

```
exten => _X.,1,Noop(=====inputpin=====)
exten => _X.,n,Set(TIMEOUT(digit)=5)
exten => _X.,n,Set(TIMEOUT(response)=15)
;exten => _X.,n,GotoIf($[${MYEXTEN}=442037348107]?pin01)
exten => _X.,n,GotoIf($[${MYEXTEN}=390294756396]?pin02)
exten => _X.,n,GotoIf($[${MYEXTEN}=442031291067]?pin03)
exten => _X.,n,GotoIf($[${MYEXTEN}=442031291070]?pin04)
exten => _X.,n,GotoIf($[${MYEXTEN}=442031291071]?pin05)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348944]?pin06)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348945]?pin07)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348946]?pin08)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348947]?pin09)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348980]?pin10)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348981]?pin11)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348982]?pin12)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348983]?pin13)
exten => _X.,n,GotoIf($[${MYEXTEN}=442037348984]?pin14)
exten => _X.,n,GotoIf($[${MYEXTEN}=442034111865]?pin15)
exten => _X.,n,GotoIf($[${MYEXTEN}=442034682146]?pin16)
```

exten => _X.,n,GotoIf(\$[\${MYEXTEN}=442035820783]?pin17)
exten => _X.,n,GotoIf(\$[\${MYEXTEN}=442037342073]?pin18)
exten => _X.,n,GotoIf(\$[\${MYEXTEN}=442037342486]?pin19)
exten => _X.,n,GotoIf(\$[\${MYEXTEN}=442037342552]?pin20)
exten => _X.,n(pin01),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin02),Authenticate(1951)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin03),Authenticate(2008)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin04),Authenticate(5566)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin05),Authenticate(1997)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin06),Authenticate(208)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin07),Authenticate(1234)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin08),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin09),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin10),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin11),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin12),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin13),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin14),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin15),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin16),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin17),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin18),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin19),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)
exten => _X.,n(pin20),Authenticate(123456)
exten => _X.,n,Goto(inputnumber)

exten => _X.,n(inputnumber),Goto(INPUTNUM,_X.,1)

[442037348107]

```
exten => _X.,1,Noop(=====CallOut 1th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},extra/1,0,gsm-1)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
```

[390294756396]

```
exten => _X.,1,Noop(=====CallOut 2th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},extra/3,0,gsm-1)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
```

[442031291067]

```
exten => _X.,1,Noop(=====CallOut 3th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},extra/5,0,gsm-1)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
```

[442031291070]

```
exten => _X.,1,Noop(=====CallOut 4th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},extra/7,0,gsm-1)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
```

[442031291071]

```
exten => _X.,1,Noop(=====CallOut 5th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},SIP/1583621-192.168.182.138,0,Board-2gsm-1)
```

```

;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
[442037348944]
exten => _X.,1,Noop(=====CallOut 6th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,,${EXTEN},SIP/1583622-192.168.182.138,0,Board-2-gsm-2)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
[442037348945]
exten => _X.,1,Noop(=====CallOut 7th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,,${EXTEN},SIP/1583623-192.168.182.138,0,Board-2-gsm-3)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
[442037348946]
exten => _X.,1,Noop(=====CallOut 8th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,,${EXTEN},SIP/1583624-192.168.182.138,0,Board-2-gsm-4)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
[442037348947]
exten => _X.,1,Noop(=====CallOut 9th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,,${EXTEN},SIP/1583621-192.168.182.139,0,Board-3-gsm-1)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")

[442037348980]
exten => _X.,1,Noop(=====CallOut 10th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)

```

```
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,,${EXTEN},SIP/1583622-192.168.182.139,0,Board-3-gsm-2)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
```

[442037348981]

```
exten => _X.,1,Noop(=====CallOut 11th port=====)
exten => _X.,n,Noop(=====${EXTEN}=====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,,${EXTEN},SIP/1583623-192.168.182.139,0,Board-3-gsm-3)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
```

[442037348982]

```
exten => _X.,1,Noop(=====CallOut 12th port=====)
exten => _X.,n,Noop(=====${EXTEN}=====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,,${EXTEN},SIP/1583624-192.168.182.139,0,Board-3-gsm-4)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
```

[442037348983]

```
exten => _X.,1,Noop(=====CallOut 13th port=====)
exten => _X.,n,Noop(=====${EXTEN}=====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,,${EXTEN},SIP/1583621-192.168.182.140,0,Board-4-gsm-1)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
```

[442037348984]

```
exten => _X.,1,Noop(=====CallOut 14th port=====)
exten => _X.,n,Noop(=====${EXTEN}=====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,,${EXTEN},SIP/1583622-192.168.182.140,0,Board-4-gsm-2)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
```

[442034111865]

```

exten => _X.,1,Noop(=====CallOut 15th  port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},SIP/1583623-192.168.182.140,0,Board-4-gsm-3)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
[442034682146]
exten => _X.,1,Noop(=====CallOut 16th  port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},SIP/1583624-192.168.182.140,0,Board-4-gsm-4)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
[442035820783]
exten => _X.,1,Noop(=====CallOut 17th  port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},SIP/1583621-192.168.182.141,0,Board-5-gsm-1)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
[442037342073]
exten => _X.,1,Noop(=====CallOut 18th  port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},SIP/1583622-192.168.182.141,0,Board-5-gsm-2)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
[442037342486]
exten => _X.,1,Noop(=====CallOut 19th  port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},SIP/1583623-192.168.182.141,0,Board-5-gsm-3)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")

```

[442037342552]

```
exten => _X.,1,Noop(=====CallOut 20th port=====)
exten => _X.,n,Noop(===== ${EXTEN} =====)
;exten => _X.,n,Set(CRD_CALLEEID=${EXTEN})
exten => _X.,n,Macro(dial-failover,, ${EXTEN},SIP/1583624-192.168.182.141,0,Board-5-gsm-4)
;exten => _X.,n,Dial(extra/3/${EXTEN})
exten => _X.,n,Hangup()
;exten=>h,1,WriteCDR("${CALLERID(name)}","${CDR_CALLEEID}","Board-5-gsm-4","${CDR_TOCH
AN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
```

Step 6: Reload the dialplan

Run command: asterisk -r

Run command: dialplan reload

Notic: You need to import some record to the gateway if you need. And you need to add some module in the gateway.

Run command: asterisk -r

Run command: module load app_disa.so

Run command: module load app_authenticate.so

Run command: module load codec_gsm.so

Run command: module load format_gsm.so

On the other hand, I suggest you use the gsm codec record.

➤ Callback Function

Example scenario:

The extension 1001 makes a call "SIP -> GSM" to number 13632919026, but this number doesn't answer the call. After a few minutes, the number "13632919026" calls back "GSM -> SIP" and this call can't go to any other extension, it has to go to the extension who originates the call (SIP extension 1001). The call must be in a buffer so that when the target phone calls back, the OpenVox gateway knows which extension that originates the mobile target.

This example we will use the PHPAGI to setup it, so that you can know how to use the PHPAGI in the gateway.

Step 1: We need to get the call state, whether "Answer, or No Answer". And if the call is "No Answer", we will save the originated extension and the target number (cellphone) to a file. So that we can match the file once some calls from GSM→SIP.

Step 2: Please edit the file extensions_routing.conf. The file content is /etc/asterisk/extensions_routing.conf. Follows:

```
root@Openvox-Wireless-Gateway:/etc/asterisk# vi /etc/asterisk/extensions_routing.conf
```

First, please add a dialplan in your Trunk. I add it here. You can check the dialplan and add it to the right place.

```
[sip-9999-172.16.8.44]
include => rtg-CallOut-1
exten => h,1,WriteCDR("${CDR(src)}","${CDR_CALLEEID}","9999","${CDR_TOCHAN}","${CDR(start)}","${CDR(billsec)}","${CDR(disposition)}")
exten => h,n,AGI(/etc/asterisk/agi-bin/setcallbacklist.php,"${CDR(disposition)}","${CDR(src)}","${CDR_CALLEEID}");
```

I add an AGI dialplan after the Trunk has call hung up, it will run a PHP Script named setcallbacklist.php under the content /etc/asterisk/agi-bin/setcallbacklist.php.

```
[sip-9999-172.16.8.44]
include => rtg-CallOut-1
exten=>h,1,WriteCDR("${CDR(src)}","${CDR_CALLEEID}","9999","${CDR_TOCHAN}","${CDR(start)}
","${CDR(billsec)}","${CDR(disposition)}")
exten=>h,n,AGI(/etc/asterisk/agi-bin/setcallbacklist.php,"${CDR(disposition)}","${CDR(src)}","${C
DR_CALLEEID}");
```

The red line is I add. About the means please refer to the asterisk AGI from below link:

<http://www.voip-info.org/wiki/view/Asterisk+AGI>

Step 3: Create a PHP script.

I create it under the content /etc/asterisk/agi-bin/, so I need to create a content named agi-bin.

```
root@Openvox-Wireless-Gateway:/# cd /etc/asterisk/
root@Openvox-Wireless-Gateway:/etc/asterisk# mkdir agi-bin
```

Then you need to copy at least 2 AGI libs to the content. You can copy it from asterisk or elastix. And then change the permissions of the lib.

```
root@Openvox-Wireless-Gateway:/etc/asterisk/agi-bin# ls
callback.php          phpagi-asmanager.php  test.php
callbackfun.php      phpagi.php
callbacklist.txt     setcallbacklist.php
root@Openvox-Wireless-Gateway:/etc/asterisk/agi-bin#
```

The AGI lib name is phpagi-asmanager.php and phpagi.php


```
/agi-bin# chmod 777 phpagi-asmanager.php
```

```
/agi-bin# chmod 777 phpagi.php
```

Step 4: Write your AGI script, there is a sample PHP script:

```
asterisk/agi-bin# vi setcallbacklist.php
```

```
#!/bin/php -q
```

```
<?php
```

```
include("/etc/asterisk/agi-bin/phpagi.php");
```

```
function WriteTheCallBackList($GSM,$SIP)
```

```
{
```

```
    $MyFile = fopen("/etc/asterisk/agi-bin/callbacklist.txt","a");
```

```
    if(!$MyFile)
```

```
    {
```

```
        echo "Open the callbacklist.txt error\n";
```

```
        return 0;
```

```
    }
```

```
    $List = $GSM."<----->".$SIP;
```

```
    fwrite($MyFile,$List);
```

```
    fwrite($MyFile,"\n");
```

```
    fclose($MyFile);
```

```
}
```

```
$agi = new AGI();
```

```
// WriteTheCallBackList($argv[1],$argv[2]);
```

```
if($argv[1]=="NO ANSWER")
```

```
{
```

```
    WriteTheCallBackList($argv[3],$argv[2]);
```

```
}
```

```
?>
```

This PHP script will create a file record the "No Answer" cellphone and the SIP extensions to a name of callbacklist.txt. Now I will make a call to a cellphone use my extensions 1001, and no answer it. You will see the file will be created and have a "No Answer" list.

```
root@Openvox-Wireless-Gateway:/etc/asterisk/agi-bin# vi callbacklist.txt
```

```
172.16.8.46 - PuTTY  
3632919026<----->1001
```

Step 5: Match the cellphone number if have a call coming in gateway.

Please edit the file /etc/asterisk/extensions_macro.conf

```
root@Openvox-Wireless-Gateway:/etc/asterisk# vi extensions_macro.conf
```

Add the AGI dialplan in this file.

```
[*]; Read only, Don't Edit!  
[macro-dial-failover]  
exten => s,1,AGI(/etc/asterisk/agi-bin/callback.php);  
exten => s,n,Set(ADEV=3)  
exten => s,n,Set(AEXTEN_FLAG=4)  
exten => s,n,Set(ACDR_NAME=5)  
exten => s,n,Set(ARG=ARG)  
exten => s,n,Set(MAX=128)
```

```
exten => s,1,AGI(/etc/asterisk/agi-bin/callback.php);
```

Once have calls coming from GSM port, it will check a file and judge if it is a "No Answer" cellphone call in. If yes, it will call out to the right SIP extension, if no, it will continue (not change anything).

This is the callback.php.

```
#!/bin/php -q
```

```
<?php
```

```
include("/etc/asterisk/agi-bin/phpagi.php");  
include("/etc/asterisk/agi-bin/callbackfun.php");  
function MatchTheCallBackListh($PhoneNum)  
{  
    $MyFile = fopen("/etc/asterisk/agi-bin/callbacklist.txt","r");  
    if(!$MyFile)  
    {  
        echo "Open the callbacklist.txt error\n";  
        return 0;  
    }  
    while(!feof($MyFile))  
    {  
        $buf = fgets($MyFile);  
        $gsmpos = strpos($buf,"<");  
        $GSM = substr($buf,0,$gsmpos);  
        $sippos = strpos($buf,">")+1;  
        $SIP = substr($buf,$sippos);  
        if($PhoneNum==$GSM)  
        {
```

```

        return $SIP;
    }
}
}
// if($argv[1]=="true")
// {
    $agi = new AGI();
    $cid = trim($agi->request['agi_callerid']);
    if(($SIP=MatchTheCallBackListh($cid)))
    {
        echo "SIP =====$SIP";
        $CallOut = "SIP/9999-172.16.8.44/" . $SIP;
        $agi->exec('Dial', $CallOut);
        $agi->hangup();
    }
    echo $cid;
// }
?>

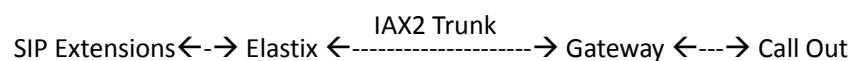
```

Notice: Things above are based on asterisk dialplan, if you want to understand more clearly, please scan more information about asterisk.

Chapter 10: How to Use the IAX2 in Gateway

OpenVox GSM Gateway could support IAX2 protocol, so that the gateway could connect more devices and improve the compatibility of the gateway.

I will setup an IAX2 trunk connected the Elastix Server, and then use the trunk to call out via our gateway.



Notice: In order to meet the personalized requirements of customers, we don't design the IAX2 protocol in WEB interface, so you need to configure it via SSH.

Step 1: Create an SIP Extension in Elastix

Step 2: Configure IAX2 in Elastix

General Settings

Trunk Name:

Outbound Caller ID:


CID Options: 


Maximum Channels:

Disable Trunk: Disable

Monitor Trunk Failures: Enable

Dialed Number Manipulation Rules

() + | 

Dial Rules Wizards: 

Outbound Dial Prefix:

Outgoing Settings

Trunk Name:

PEER Details:

```
host=172.16.8.46  
username=6666  
secret=6666  
type=friend
```

Step 3: Configure the Outbound Rules in Elastix

Route Settings

Route Name:

Route CID: Override Extension

Route Password:

Route Type: Emergency Intra-Company

Music On Hold?:

Time Group:

Route Position:

Additional Settings

PIN Set:

Dial Patterns that will use this Route

Dial patterns wizards:

Trunk Sequence for Matched Routes

Step 4: Configure the IAX2 Endpoint in the Gateway

1. Edit the `iax.conf` in the gateway and configure it.

Command: `vi /etc/asterisk/iax.conf`

```
[6666]
context=iax-elastic
host=172.16.8.44
qualify=yes
secret=6666
type=friend
username=6666
```

2. reload the IAX configuration in gateway.

```
Openvox-Wireless-Gateway*CLI> module load chan_iax2.so
Openvox-Wireless-Gateway*CLI>
Openvox-Wireless-Gateway*CLI>
Loaded chan_iax2.so

Openvox-Wireless-Gateway*CLI>
Openvox-Wireless-Gateway*CLI> iax2 reload
```

Command: module load chan_iax2.so

Command: iax2 reload

Step 5: Endpoint Settings in Web

You can use the routing rules that you've configured in Web, then you need to configure a SIP endpoint. When some calls from/to the IAX trunk, you can use the SIP endpoint.

1. Create a SIP endpoint in the Gateway.



2. Configure it.

Edit SIP Endpoint "1001"

▼ Main Endpoint Settings

Name:	<input type="text" value="1001"/>
User Name:	<input type="text" value="1001"/> <input type="checkbox"/> Anonymous
Password:	<input type="password"/>
Registration:	<input type="text" value="None"/>
Hostname or IP Address:	<input type="text" value="172.16.8.46"/>
Transport:	<input type="text" value="UDP"/>
NAT Traversal:	<input type="text" value="Yes"/>

3. Configure Routing in the gateway.

SYSTEM | **GSM** | **SIP** | **ROUTING** | **SMS** | **NETWORK** | **ADVANCED** | **LOGS**

Call Routing Rules | **Groups** | MNP Settings

(1) At first we set up a gsm group, and use the RoundRobin Policy.

Group Name:	AIIGSM
Type:	GSM
Policy:	Roundrobin
Members	<p>NO. <input type="checkbox"/> All</p> <p>1 <input checked="" type="checkbox"/> gsm-1.1</p> <p>2 <input checked="" type="checkbox"/> gsm-1.2</p> <p>3 <input checked="" type="checkbox"/> gsm-1.3</p> <p>4 <input checked="" type="checkbox"/> gsm-1.4</p> <p>5 <input checked="" type="checkbox"/> gsm-2.1</p> <p>6 <input checked="" type="checkbox"/> gsm-2.2</p> <p>7 <input checked="" type="checkbox"/> gsm-2.3</p> <p>8 <input checked="" type="checkbox"/> gsm-2.4</p> <p>9 <input checked="" type="checkbox"/> gsm-3.1</p> <p>10 <input checked="" type="checkbox"/> gsm-3.2</p> <p>11 <input checked="" type="checkbox"/> gsm-3.3</p> <p>12 <input checked="" type="checkbox"/> gsm-3.4</p> <p>13 <input checked="" type="checkbox"/> gsm-4.1</p> <p>14 <input checked="" type="checkbox"/> gsm-4.2</p> <p>15 <input checked="" type="checkbox"/> gsm-4.3</p> <p>16 <input checked="" type="checkbox"/> gsm-4.4</p> <p>17 <input checked="" type="checkbox"/> gsm-5.1</p> <p>18 <input checked="" type="checkbox"/> gsm-5.2</p> <p>19 <input checked="" type="checkbox"/> gsm-5.3</p> <p>20 <input checked="" type="checkbox"/> gsm-5.4</p>

(2) Set up callout rules.

Routing Name:	CallOut
Call Comes in From:	1001
Send Call Through:	AIIGSM

Step 6: Set up the Dialplan in Gateway

```
[root@Elx asterisk]# vim /etc/asterisk/extensions_custom.conf
```

```
[iax-elastic]  
exten => _X., 1, Dial(SIP/1001-172.16.8.46/${EXTEN})  
exten => _X., n, Hangup()
```

About how to set the Dial (), please check your sip endpoint settings. For example, I use Dial (SIP/1001-172.16.8.46), because I found info in my sip_endpoints.conf, follows:

```
[1001-172.16.8.46]  
username=1001  
host=172.16.8.46  
transport=udp  
nat=yes  
qualify=no  
qualifyfreq=60  
dtmfmode=rfc2833  
trustpid=no  
sendrpid=no  
callingpres=allowed_passed_screen  
progressinband=never  
usereqphone=no  
use_q850_reason=no  
honorsdpversion=yes  
allowtransfer=yes  
promiscredir=no  
max_forwards=70  
registertrying=no  
timert1=500  
timerb=32000  
session-timers=accept  
session-minse=90  
session-expires=1800  
session-refresher=uas  
insecure=port,invite  
type=friend  
context=sipinbound  
setvar=SIPROUTE=sip-1001-172.16.8.46  
endpoint_name=1001
```

Notice: Please reload the dialplan when you finish edit.

```
~
root@Openvox-Wireless-Gateway:/etc/asterisk# asterisk -r
Cannot read termcap database;
using dumb terminal settings.
Openvox-Wireless-Gateway*CLI> dialplan reload
Dialplan reloaded.

Openvox-Wireless-Gateway*CLI> █
```

If you have any questions, please contact us:
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