VoxStack GSM Gateway IVR settings

Rev: 1.0
Date: January 2, 2014
From: OpenVox support group
Contact Info: support@openvox.com.cn

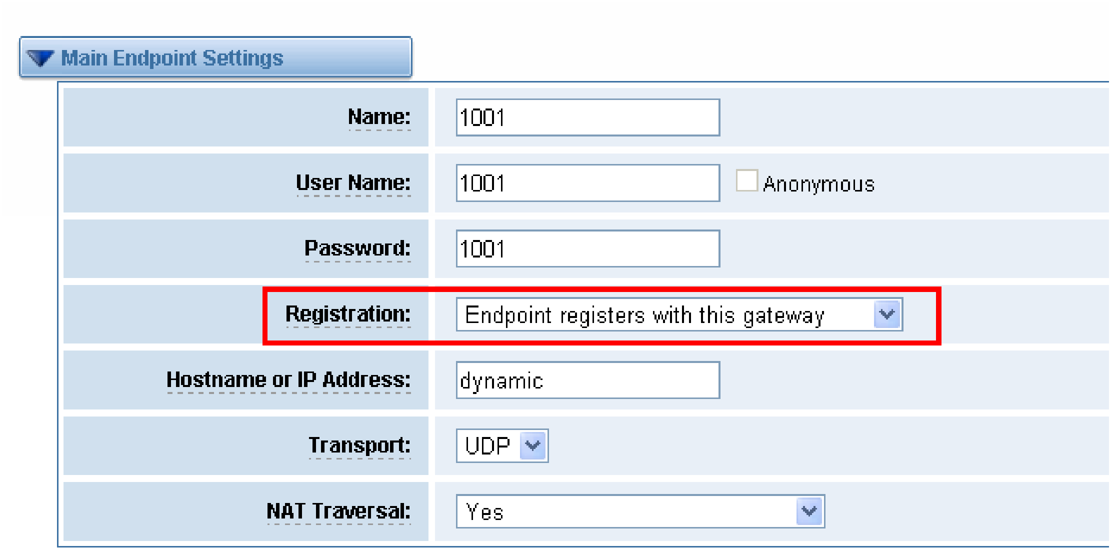
OpenVox VoxStack GSM Gateway is a *feature-rich, high availability, flexible* modular gateway product. This sample manual introduces some methods about "How to set an IVR in the OpenVox GSM Gateway".
OpenVox GSM Gateway can be used a sample PBX, you can use it directly, needn't any PBX. Next I'll show you the basic function of IVR,

Step 1: Configure the SIP in the Gateway .

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

Step 2: Edit SIP Endpoint in the Gateway .

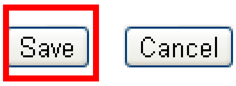
Now the OpenVox GSM Gateway support 3 connected ways via SIP protocol. One is the Gateway as a SIP server; One is the Gateway as a SIP peers registered to PBX; Last is the IP to IP.
I choice the first way to show you how to set the IVR. Follows:

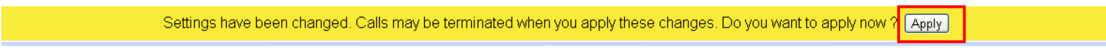


Notice:

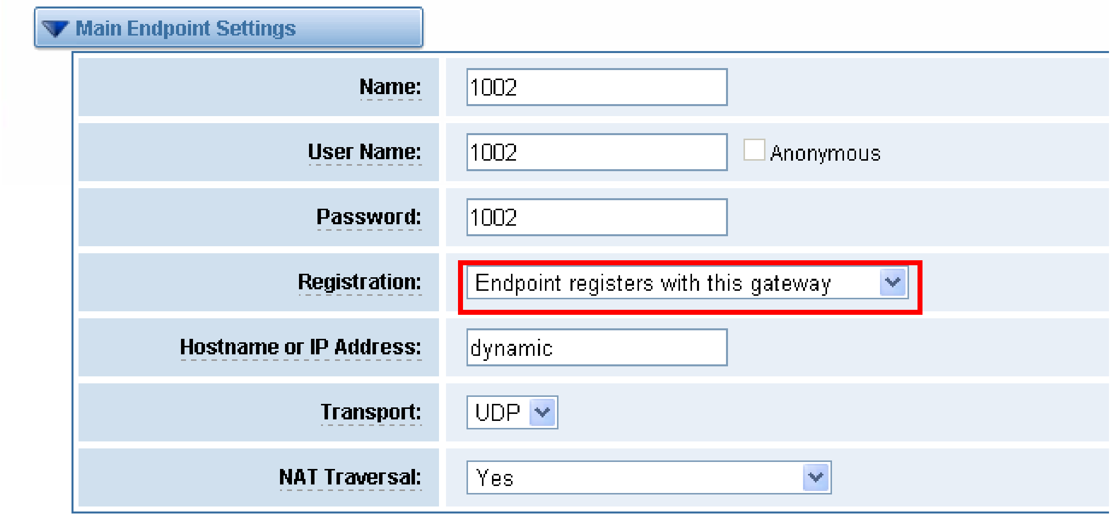
You can choice different connected ways by select the "Registration" options, I select the "Endpoint registers with this gateway", it means the Gateway will be as a SIP server. You can registered your softphone to the gateway directly.

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

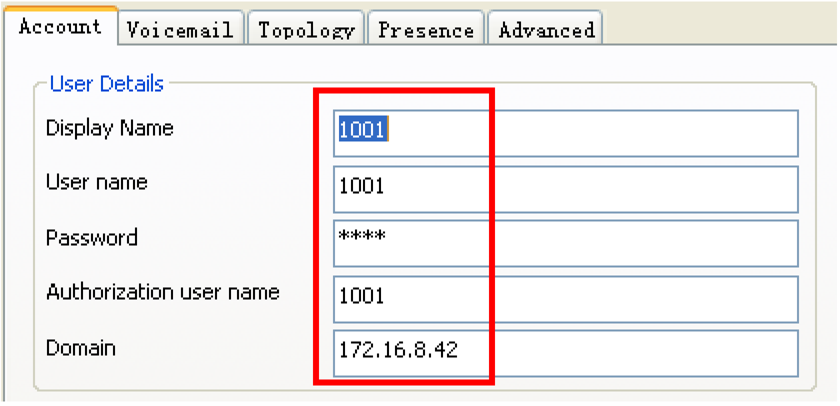




Now, let's set the second SIP server.



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



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

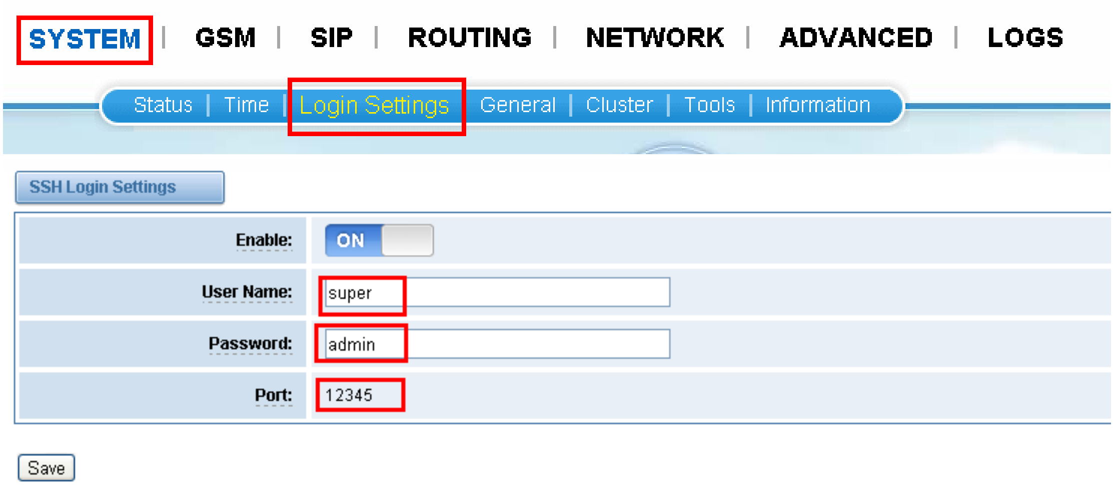
SIP Information

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)

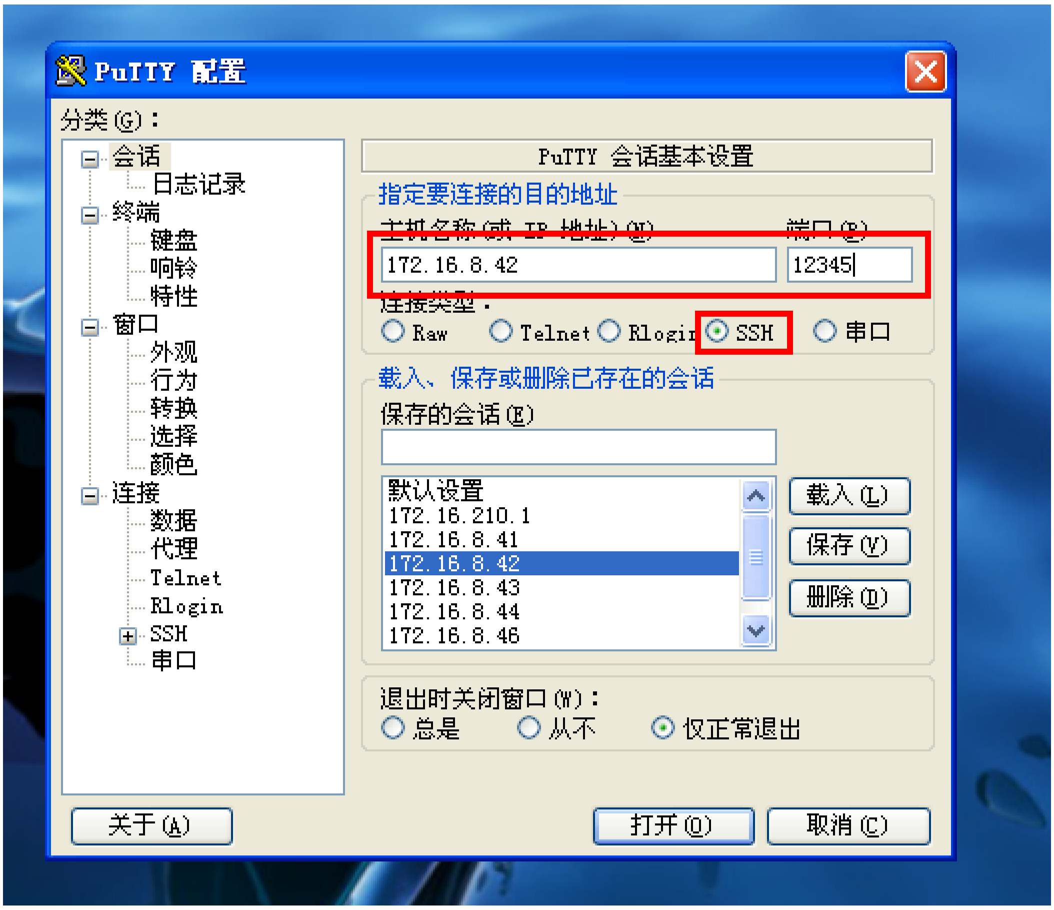
Notice: you can try to set it in our “online demo”. The link be shown as below:
http://demo.openvox.cn:65321/

Step 3: Set SSH Login in the Gateway .

OpenVox Gateway can support SSH login, so that you can know more about the OpenVox Gateway details, and expand your own application.
Please select the “SYSTEM→Login Settings”



Login the Gateway via SSH, follows:



Step 4: Edit the GSM port context.

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

```
; Span 1: opvxg4xx/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: opvxg4xx/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: opvxg4xx/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 received ring, 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. Follows:

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

Notice: you can import your own recording file to the OpenVox Gateway. I have imported my recording file to the gateway.

The dialplan sample:

```
[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 some call from GSM-Port1. The gateway will play a voice. And then the custom will choice the different service by press different digits. The Gateway will detected the DTMF, and execute different operations.

For example, when you hear the sound, and press the 1 digits, the extension 1001 will ring.

Please contact us if you have any question, our contact info be shown as below:

Web Site: www.oepnvox.cn
Support: support@openvox.cn
Sales: sales@openvox.com.cn