

38. Asterisk Application

We offer the application shows that it is convenient and cost saving to implement the free IP-PBX using Asterisk and Vigor 3300V when users want to use the Soft Phone or IP Phone instead of traditional telephone in the company.

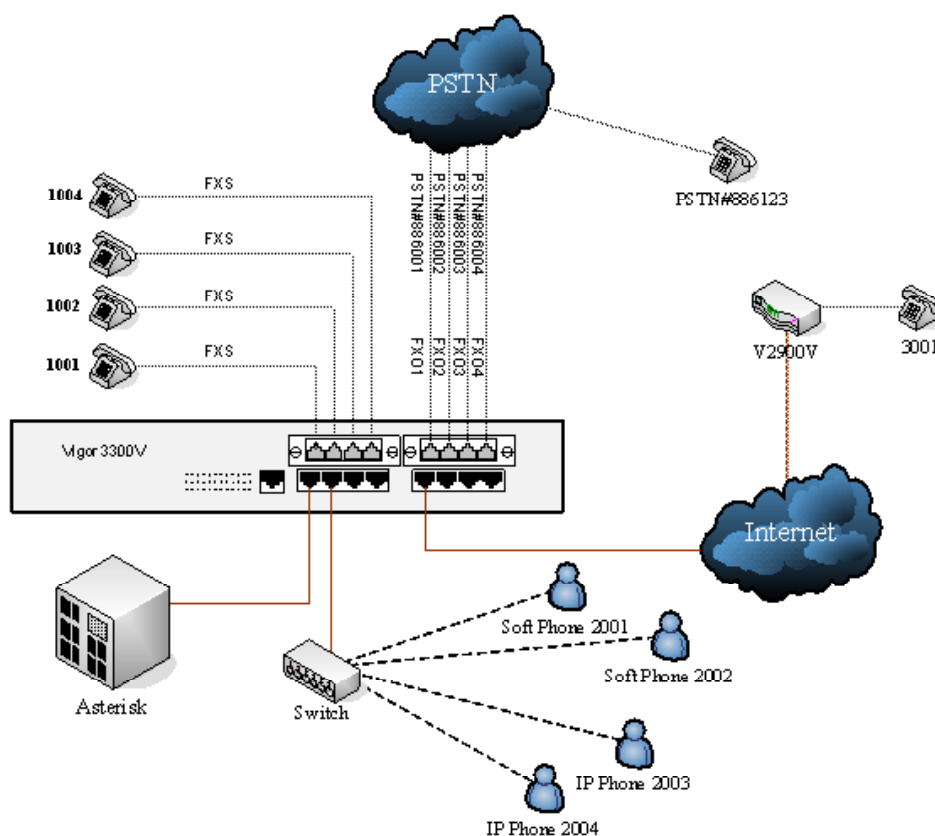


Figure 38-1. The scenario

In the figure using FXO port of Vigor 3300V to connect to PSTN. So, users do not need the other equipment to as the IP-PBX. The way that we work normally with the FXO of Vigor 3300V is that we could make a call from extension of IP-PBX which is the telephones connected with FXS of Vigor 3300V, Soft Phone or IP Phone to PSTN Network. We also could make a call from PSTN to the IP-PBX, there are four PSTN line (the maximum is 8), then forwarding the call to any extension of IP-PBX, or make a call from extensions to remote peer user through VPN in the Internet, or reverse direction call. Another application is workable that putting the Asterisk to the Internet for branch office communication.

38.1 Configuration

IP Address List:

Asterisk – 172.16.2.234

Vigor 3300V – 172.16.2.237

SoftPhone 2001 – 172.16.2.201

SoftPhone 2002 – 172.16.2.202

SoftPhone 2003 – 172.16.2.203

SoftPhone 2004 – 172.16.2.204

V2900V (VPN) – 172.16.2.205

38.2 Installing Asterisk

Download Asterisk from the Asterisk website page <http://www.asterisk.org/>.

Install the Asterisk and refer to the installation guide from the Asterisk website.

38.3 Configuring Asterisk

sip.conf

Modify the sip.conf, the file is usually placed on the location /etc/asterisk.

[general] Setting in sip.conf

Modify the realm value to 172.16.2.234 for digest authentication.

```
realm=172.16.2.234 ; Realm for digest authentication
```

Modify the tos value, users could choose one kind of type. There are “lowdelay”, “throughput”, “reliability”, “mincost” and “none”. It is depend on the network status.

```
Tos=lowdelay
```

Modify the defaultexpiry value for registration.

```
Defaultexpiry=300 ; Default length of incoming/outgoing registration
```

Modify the codec settings.

```
disallow=all ; First disallow all codecs  
allow=ulaw ; Allow codecs in order of preference  
allow = alaw  
allow=g729  
allow=g726
```

Modify the language value for all users.

```
language=en ; Default language setting for all users/peers
```

Modify the rtptimeout value for RTP activity.

```
rtptimeout=60 ; Terminate call if 60 seconds of no RTP activity
```

Modify the dtmfmode value.

```
dtmfmode = rfc2833 ; Set default dtmfmode for sending DTMF. Default:
rfc2833

; Other options:
; info : SIP INFO messages
; inband : Inband audio (requires 64 kbit codec -alaw, ulaw)
; auto : Use rfc2833 if offered, inband otherwise
```

Add Phone Number

Add phone setting for each phone number.

```
[1001]
type=friend
nat=no
canreinvite=yes
host=dynamic
defaultip=172.16.2.237
username=1001
secret=0000
dtmfmode=info ; Choices are inband, rfc2833, or info
call-limit=1
mailbox=1000 ; Mailbox for message waiting indicator
context=sip
callerid="1001" <1001>
disallow=all
allow=ulaw
allow=g729
allow=g723.1

[1002]
type=friend
nat=no
canreinvite=yes
host=dynamic
```

```
defaultip=172.16.2.237
username=1002
secret=0000
dtmfmode=info ; Choices are inband, rfc2833, or info
call-limit=1
mailbox=1000 ; Mailbox for message waiting indicator
context=sip
callerid="1002" <1002>
disallow=all
allow=ulaw
allow=g729
allow=g723.1
```

```
[1003]
type=friend
nat=no
canreinvite=yes
host=dynamic
defaultip=172.16.2.237
username=1003
secret=0000
dtmfmode=info ; Choices are inband, rfc2833, or info
call-limit=1
mailbox=1000 ; Mailbox for message waiting indicator
context=sip
callerid="1003" <1003>
disallow=all
allow=ulaw
allow=g729
allow=g723.1
```

```
[1004]
type=friend
nat=no
canreinvite=yes
host=dynamic
```

```
defaultip=172.16.2.237
username=1004
secret=0000
dtmfmode=info ; Choices are inband, rfc2833, or info
call-limit=1
mailbox=1000 ; Mailbox for message waiting indicator
context=sip
callerid="1004" <1004>
disallow=all
allow=ulaw
allow=g729
allow=g723.1

[2001]
type=friend
nat=no
canreinvite=yes
host=dynamic
defaultip=172.16.2.201
username=2001
secret=2001
dtmfmode=info ; Choices are inband, rfc2833, or info
call-limit=1
mailbox=1000 ; Mailbox for message waiting indicator
context=sip
callerid="2001" <2001>
disallow=all
allow=ulaw
allow=g729
allow=g723.1

[2002]
type=friend
nat=no
canreinvite=yes
host=dynamic
```

```
defaultip=172.16.2.202
username=2002
secret=2002
dtmfmode=info ; Choices are inband, rfc2833, or info
call-limit=1
mailbox=1000 ; Mailbox for message waiting indicator
context=sip
callerid="2002" <2002>
disallow=all
allow=ulaw
allow=g729
allow=g723.1

[2003]
type=friend
nat=no
canreinvite=yes
host=dynamic
defaultip=172.16.2.203
username=2003
secret=2003
dtmfmode=info ; Choices are inband, rfc2833, or info
call-limit=1
mailbox=1000 ; Mailbox for message waiting indicator
context=sip
callerid="2003" <2003>
disallow=all
allow=ulaw
allow=g729
allow=g723.1

[2004]
type=friend
nat=no
canreinvite=yes
host=dynamic
```

```
defaultip=172.16.2.204
username=2004
secret=2004
dtmfmode=info ; Choices are inband, rfc2833, or info
call-limit=1
mailbox=1000 ; Mailbox for message waiting indicator
context=sip
callerid="2004" <2004>
disallow=all
allow=ulaw
allow=g729
allow=g723.1

[3001]
type=friend
nat=no
canreinvite=yes
host=dynamic
defaultip=172.16.2.205
username=3001
secret=3001
dtmfmode=info ; Choices are inband, rfc2833, or info
call-limit=1
mailbox=1000 ; Mailbox for message waiting indicator
context=sip
callerid="3001" <3001>
disallow=all
allow=ulaw
allow=g729
allow=g723.1

[fx01]
type=friend
secret=1234
context=sip
disallow=all
```

```
allow=ulaw
allow=g729
allow=g723.1
dtmfmode=info
canreinvite=no
host=dynamic
defaultip=172.16.2.237
```

```
[fxo2]
type=friend
secret=1234
context=sip
disallow=all
allow=ulaw
allow=g729
allow=g723.1
dtmfmode=info
canreinvite=no
host=dynamic
defaultip=172.16.2.237
```

```
[fxo3]
type=friend
secret=1234
context=sip
disallow=all
allow=ulaw
allow=g729
allow=g723.1
dtmfmode=info
canreinvite=no
host=dynamic
defaultip=172.16.2.237
```

```
[fxo4]
type=friend
```



```
secret=1234
context=sip
disallow=all
allow=ulaw
allow=g729
allow=g723.1
dtmfmode=info
canreinvite=no
host=dynamic
defaultip=172.16.2.237
```

mgcp.conf

Modify the mgcp.conf, the file is usually placed on the location /etc/asterisk.

[general] Setting in mgcp.conf

Modify the Call Agent port value to 2727 for Vigor 3300V.

```
port = 2727
```

Add Endpoint for MGCP

Modify the port value to 2727 for Call Agent.

```
[172.16.2.237]
host = 172.16.2.237
context = mgcp
line => aaln/1
line => aaln/2
line => aaln/3
line => aaln/4
line => aaln/5
line => aaln/6
line => aaln/7
line => aaln/8

[172.16.2.201]
host = 172.16.2.201
```

```
context = mgcp
line => aaln/1

[172.16.2.202]
host = 172.16.2.202
context = mgcp
line => aaln/1

[172.16.2.203]
host = 172.16.2.203
context = mgcp
line => aaln/1

[172.16.2.204]
host = 172.16.2.204
context = mgcp
line => aaln/1

[172.16.2.205]
host = 172.16.2.205
context = mgcp
line => aaln/1
```

extensions.conf

Add extensions for SIP.

```
[sip]
exten => 1001,1,Dial(SIP/1001,20,tr)
exten => 1002,1,Dial(SIP/1002,20,tr)
exten => 1003,1,Dial(SIP/1003,20,tr)
exten => 1004,1,Dial(SIP/1004,20,tr)
exten => 2001,1,Dial(SIP/2001,20,tr)
exten => 2002,1,Dial(SIP/2002,20,tr)
exten => 2003,1,Dial(SIP/2003,20,tr)
exten => 2004,1,Dial(SIP/2004,20,tr)
exten => 3001,1,Dial(SIP/3001,20,tr)
```

```
exten => 1,1,Dial(SIP/fxo1,20,tr)
exten => 2,1,Dial(SIP/fxo2,20,tr)
exten => 3,1,Dial(SIP/fxo3,20,tr)
exten => 4,1,Dial(SIP/fxo4,20,tr)
```

Add extensions for MGCP.

```
[mgcp]
exten => 1001,1,Dial(MGCP/aaln/1@172.16.2.237)
exten => 1002,1,Dial(MGCP/aaln/2@172.16.2.237)
exten => 1003,1,Dial(MGCP/aaln/3@172.16.2.237)
exten => 1004,1,Dial(MGCP/aaln/4@172.16.2.237)
exten => 1,1,Dial(MGCP/aaln/5@172.16.2.237)
exten => 2,1,Dial(MGCP/aaln/6@172.16.2.237)
exten => 3,1,Dial(MGCP/aaln/7@172.16.2.237)
exten => 4,1,Dial(MGCP/aaln/8@172.16.2.237)
exten => 2001,1,Dial(MGCP/aaln/1@172.16.2.201)
exten => 2002,1,Dial(MGCP/aaln/1@172.16.2.202)
exten => 2003,1,Dial(MGCP/aaln/1@172.16.2.203)
exten => 2004,1,Dial(MGCP/aaln/1@172.16.2.204)
exten => 3001,1,Dial(MGCP/aaln/1@172.16.2.205)
```

38.4 Configuring Vigor 3300V

38.4.1 SIP Configuration

SIP Proxy

VoIP - Protocol

Select Protocol: SIP MGCP

SIP Configuration | MGCP Configuration

SIP Local Port:

#	Active	Outbound Proxy	Proxy Name	Proxy Address	Proxy Port	Registrar Addr	Registrar Port	Expires (sec)	Domain
1.	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Asterisk	172.16.2.234	5060	172.16.2.234	5060	300	172.16.2.234
2.	<input type="checkbox"/>	<input type="checkbox"/>		0	5060	0	5060	300	0
3.	<input type="checkbox"/>	<input type="checkbox"/>		0	5060	0	5060	300	0

Example: iptel iptel.org iptel.org iptel.org

Figure 38-2. SIP configuration

Port Setting

Configure each port in Vigor 3300V. For example, the setting for port 1 shows as below. Input the correct data to Username, Password, Display Name, Authentication ID and Proxy Server. The VoIP IP Address should be selected to LAN1/VPN in the scenario, because the Asterisk server is placed on LAN.

VoIP - Port Settings - Port1 - Edit

Port 1 (FXS)

Disable Enable

Username:

Password:

Display Name:

Authentication ID:

Proxy Server:

VoIP IP Address:

Figure 38-3. Port setting-edit

Choose the “SIP INFO” for DTMF Mode to meet the Asterisk setting.

DTMF

DTMF Mode: InBand OutBand(RFC2833) SIP INFO

DTMF Volume: (Range: 0 - 31)

Call Forwarding

Disable

Call forwarding all calls

Call forwarding busy

Call forwarding no answer after rings (Range:1-10)

SIP URL: (Example: 8001@iptel.org)

Figure 38-4. DTMF mode

VoIP - Port Settings

Phone Number		Group					
#	Edit	Type	Active	Group	Username	Proxy	Codec
1		FXS	V	1	1001	Asterisk	G.729A-8kbps
2		FXS	V	2	1002	Asterisk	G.729A-8kbps
3		FXS	V	3	1003	Asterisk	G.729A-8kbps
4		FXS	V	4	1004	Asterisk	G.729A-8kbps
5		FXO	V	5	fxo1	Asterisk	G.729A-8kbps
6		FXO	V	6	fxo2	Asterisk	G.729A-8kbps
7		FXO	V	7	fxo3	Asterisk	G.729A-8kbps
8		FXO	V	8	fxo4	Asterisk	G.729A-8kbps

Figure 38-5. Port setting configuration

38.4.2 MGCP Configuration

Configure VoIP IP Address to LAN1/VPN for each port in the scenario, because the Asterisk server is placed on LAN.

VoIP - Port Settings - Port1 - Edit

Port 1 (FXS)

Disable Enable

Username:

Password:

Display Name:

Authentication ID:

Proxy Server:

VoIP IP Address:

Figure 38-6. VoIP IP address

Configuring the Call Agent IP address.

VoIP - Protocol

Select Protocol: SIP MGCP

SIP Configuration **MGCP Configuration**

MGCP Local Port:

MGCP Call Agent Address:

MGCP Call Agent Port:

EndPoint Name Style: aalr#@[ip_addr] mac_addr#@[ip_addr] aalr#@[mac_addr]

aalr#@[]

Wild-carded RSIP: Each endpoint sends its own RSIP Send only one wild RSIP

Figure 38-7. MGCP call agent address

