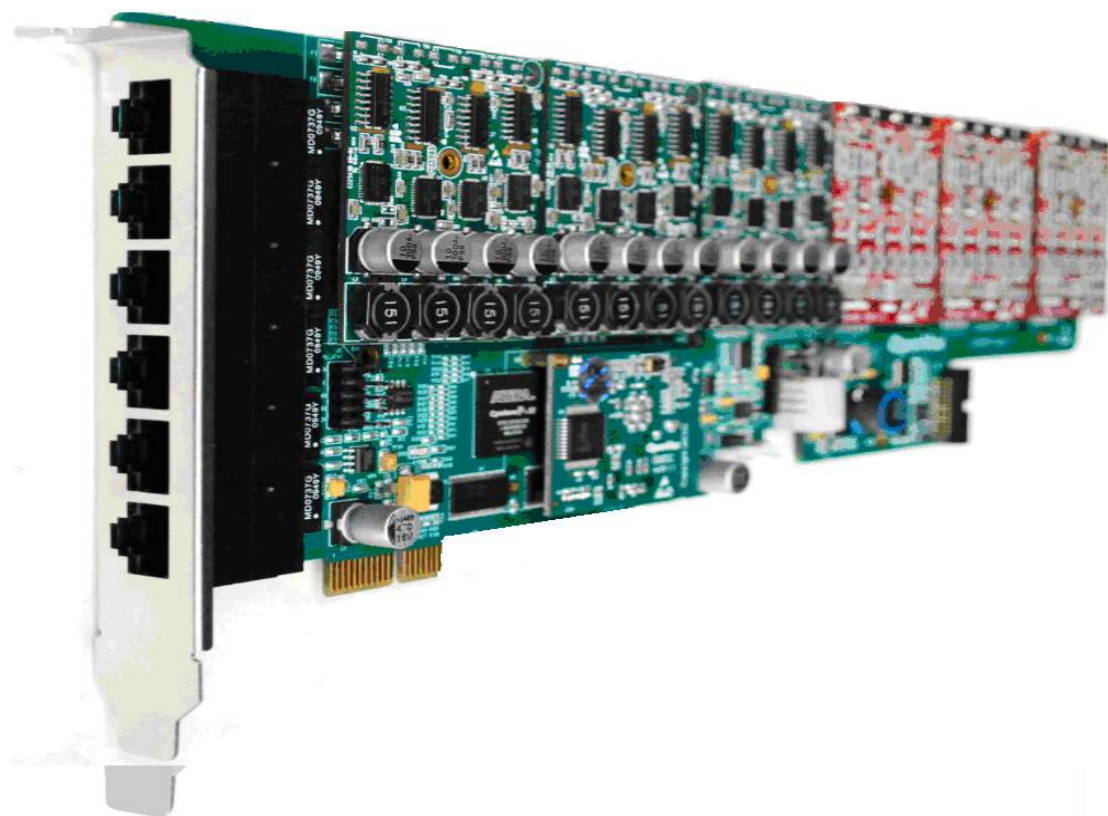




深圳开源通信有限公司

OpenVox A2410E/AE2410E Base on Elastix User Manual



A2410E

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深圳开源通信有限公司

OpenVox-Best Cost Effective Asterisk Cards

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Test environments

CentOS-5.6

Kernel version: 2.6.18-238.12.1.el5

DAHDI: dahdi-linux-complete-2.4.0+2.4.0

Asterisk: 1.8.4.4

Elastix 2.0.4

Hardware: OpenVox A2410E/AE2410E

1. Overview

1.1 What is A2410E/AE2410E

A2410E is an independent research and development modular analog telephony interface product by OpenVox Communication Co. LTD. AE2410E is A2410E with an EC module, they are designed to build SMB PBX. A2410E/AE2410E must be made up with FXO-400 and FXS-400 together to build a workable system.

1.2 What is asterisk

The Definition of Asterisk is described as follows:

Asterisk is a complete PBX in software. It runs on Linux, BSD, Windows (emulated) and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in four protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware. Asterisk provides Voicemail services with Directory, Call Conferencing, Interactive Voice Response, Call Queuing. It has support for three-way calling, caller ID services, ADSI, IAX, SIP, H323 (as both client and gateway), MGCP (call manager only) and SCCP/Skinny (voip-info.org).

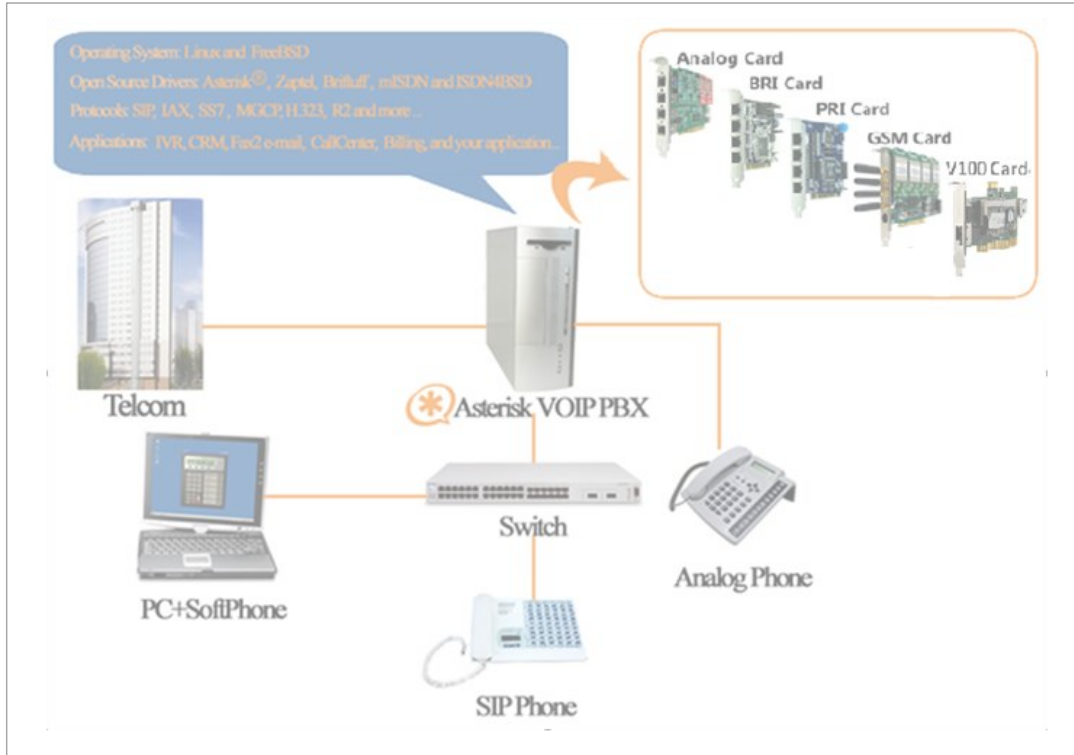


Figure 1 Topology

2. Hardware setup

The following matters need your attention before using A2410E, please check that:

1. Power supply: Plug 12V power line into the connector according to figure showed.

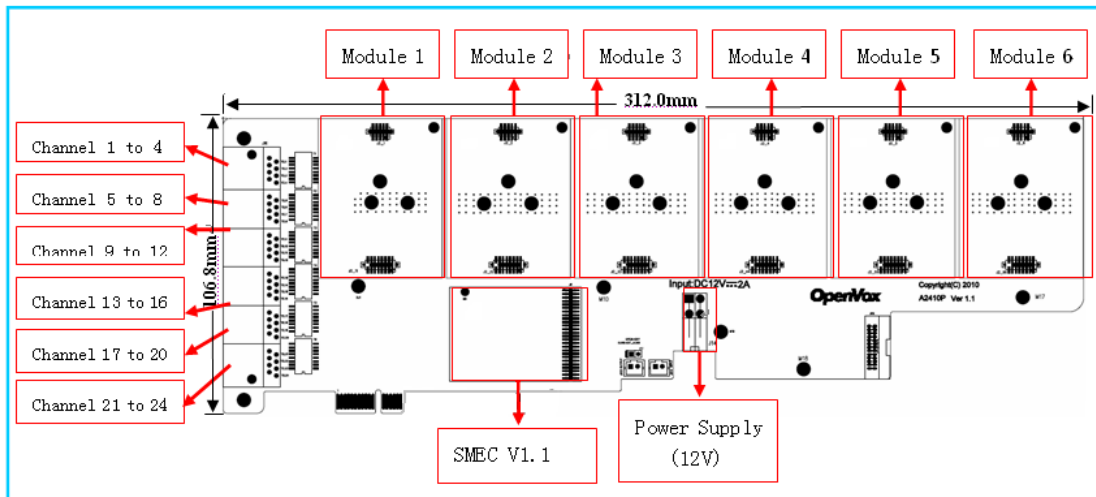


Figure 2 Hardware setup

2. Pin assignment: There are up to 6 FXS-400/FXO-400 modules on every A2410E board, a module corresponds to a RJ45 port which A2410E takes 2 of 8 pins for a pair connect to your 2-wire telephone line, so each RJ45 socket is divided into 4 telephone lines by a splitter.

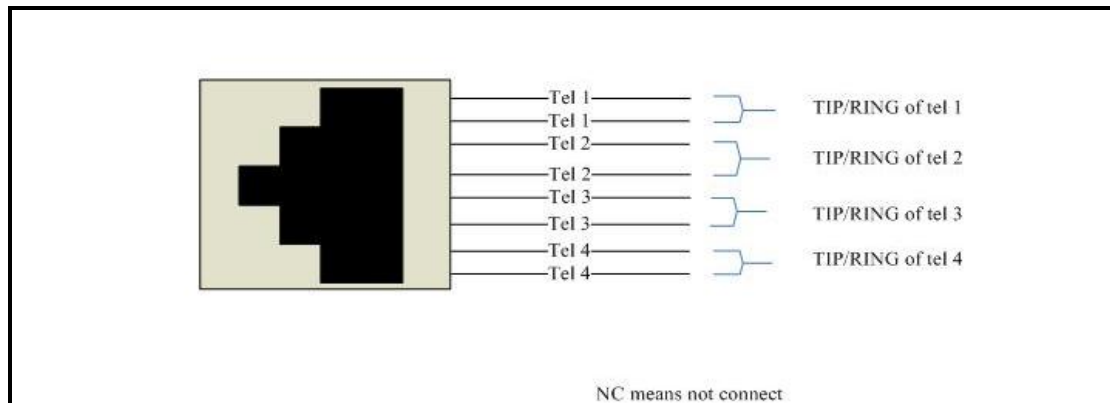


Figure 3 Pin assignment

3. A2410E splitter: It can divide RJ45 port into four ordinary telephone lines, please plug PSTN line into FXO port and normal telephone line corresponds to FXS port.

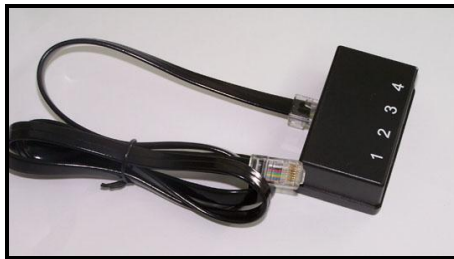


Figure 4 A2410E splitter

3. Software installation and configuration

A2410E/AE2410E support DAHDI software device driver on Linux, to make full use of A2410E/AE2410E, you should download, compile, install and configure DAHDI and asterisk.

3.1 Download

Download DAHDI package to the directory of /usr/src/ from openvox official website
http://downloads.openvox.cn/pub/drivers/dahdi-linux-complete/openvox_dahdi-linux-complete-current.tar.gz

```
# wget http://downloads.openvox.cn/pub/drivers/dahdi-linux-complete/openvox_dahdi-linux-complete-current.tar.gz
# tar -xvzf openvox_dahdi-linux-complete-current.tar.gz
```

3.2 Installtion

1. Detect hardware by execute command: `lspci -vvvv`

Check the outcome and confirm your system has recognized A2410E. If identified, outputs are like that:

```

01:01.0 Communication controller: Unknown device 1b74:2410 (rev 01)
    Subsystem: Unknown device 1b74:0001
    Control: I/O+ Mem+ BusMaster+ SpecCycle- MemWINV+ VGASnoop- ParErr-
Stepping- SERR- FastB2B-
    Status: Cap- 66MHz- UDF- FastB2B- ParErr- DEVSEL=slow >TAbort- <TAbort-
<MAbort- >SERR- <PERR-
    Latency: 64, Cache Line Size: 16 bytes
    Interrupt: pin A routed to IRQ 50
    Region 0: Memory at dcd00000 (32-bit, non-prefetchable) [size=1M]

```

Figure 5 Hardware detection

2. Modify the environment variables

Edit the file named modules under /etc/dahdi/. You are able to comment out drivers unnecessary to load, add opvxa24xx.

```

# X100P - Single port FXO interface
# X101P - Single port FXO interface
# opvxa1200      #comment out the unnecessary driver
# ystdm8xx
# ystdm16xx
... ..
# Rhino 4/8/12/24 Channel Analog PCI Interface Card
# rcbfx
opvxa24xx      #add opvxa24xx driver

```

Figure 6 Modules modification

3. Compile

Unzip and change directory to dahdi-linux-complete-XX, perform command below one by one.

```

# cd /usr/src/dahdi-linux-complete-XX
# make
# make install
# make config

```

If there is something wrong after “make”, please refer to <http://bbs.openvox.cn/viewthread.php?tid=1557&extra=page%3D1>

If successfully, reboot your PC please.

3.3 Configuration

1. Load opvxa24xx driver

```

# modprobe dahdi
# modprobe -r opvxa24xx
# modprobe opvxa24xx opermode=CHINA

```

openvox_dahdi-linux-complete 2.2.0 or higher versions allow users to adjust IRQ per millisecond. You are able to modify IRQ by the following way:

```
# modprobe opvxa24xx opermode=CHINA ms_per_irq=2
```

ms_per_irq=2 means every 2 milliseconds initiate once IRQ. You may select a valid value of ms_per_irq from 1, 2, 4, 8, 16 according to requirement, the default value is 1. While you download DAHDI from digium official website:

<http://downloads.asterisk.org/pub/telephony>

DAHDI version above **dahdi-linux-complete-2.4.0+2.4.0** supports IRQ adjustment function, and the same method to modify interrupt as described before. After IRQ adjustment, please execute command “dmesg” to check whether you have made the EC module worked. The following figure means EC module has been detected.

```
OpenVox A2410E version: 1.3

Module 0: Installed -- AUTO FXS/DPO
Module 1: Installed -- AUTO FXS/DPO
Module 2: Installed -- AUTO FXS/DPO
Module 3: Installed -- AUTO FXS/DPO
Module 4: Installed -- AUTO FXS/DPO
Module 5: Installed -- AUTO FXS/DPO
Module 6: Installed -- AUTO FXS/DPO
Module 7: Installed -- AUTO FXS/DPO
Module 8: Installed -- AUTO FXS/DPO
Module 9: Installed -- AUTO FXS/DPO
Module 10: Installed -- AUTO FXS/DPO
Module 11: Installed -- AUTO FXS/DPO
Module 12: Installed -- AUTO FXO (FCC mode)
Module 13: Installed -- AUTO FXO (FCC mode)
Module 14: Installed -- AUTO FXO (FCC mode)
Module 15: Installed -- AUTO FXO (FCC mode)
Module 16: Installed -- AUTO FXO (FCC mode)
Module 17: Installed -- AUTO FXO (FCC mode)
Module 18: Installed -- AUTO FXO (FCC mode)
Module 19: Installed -- AUTO FXO (FCC mode)
Module 20: Installed -- AUTO FXO (FCC mode)
Module 21: Installed -- AUTO FXO (FCC mode)
Module 22: Installed -- AUTO FXO (FCC mode)
Module 23: Installed -- AUTO FXO (FCC mode)
VPM450: echo cancellation supports 32 channels
VPM450: echo cancellation for 32 channels
VPM450: hardware DTMF disabled.
VPM450: Present and operational servicing 1
span(s)
```

Figure 7 EC module detections

2. Check configuration files

Run command "**vim /etc/dahdi/genconf_parameters**". If the hardware is AE2410E, please set echo_can to none as following:

```
echo_can none
```

While it is A2410E, just ignore that step and keep default.

Execute those commands:

```
# dahdi_genconf
# dahdi_cfg -vvvv
```



```
[root@localhost ~]# dahdi_cfg -vvvv
DAHDI Tools Version - 2.4.0
DAHDI Version: 2.4.0
Echo Cancellor(s):
Configuration
=====
Channel map:
Channel 01: FXO Kewlstart (Default) (Echo Canceler: none) (Slaves: 01)
Channel 02: FXO Kewlstart (Default) (Echo Canceler: none) (Slaves: 02)
Channel 03: FXO Kewlstart (Default) (Echo Canceler: none) (Slaves: 03)
Channel 04: FXO Kewlstart (Default) (Echo Canceler: none) (Slaves: 04)
Channel 05: FXO Kewlstart (Default) (Echo Canceler: none) (Slaves: 05)
... ..
Channel 20: FXS Kewlstart (Default) (Echo Canceler: none) (Slaves: 20)
Channel 21: FXS Kewlstart (Default) (Echo Canceler: none) (Slaves: 21)
Channel 22: FXS Kewlstart (Default) (Echo Canceler: none) (Slaves: 22)
Channel 23: FXS Kewlstart (Default) (Echo Canceler: none) (Slaves: 23)
Channel 24: FXS Kewlstart (Default) (Echo Canceler: none) (Slaves: 24)

24 channels to configure.

Setting echocan for channel 1 to none
Setting echocan for channel 2 to none
Setting echocan for channel 3 to none
Setting echocan for channel 4 to none
Setting echocan for channel 5 to none
... ..
Setting echocan for channel 20 to none
Setting echocan for channel 21 to none
Setting echocan for channel 22 to none
Setting echocan for channel 23 to none
Setting echocan for channel 24 to none
```

Figure 8 Channel map

The command **dahdi_genconf** will automatically generate files `/etc/dahdi/system.conf` and `/etc/asterisk/dahdi-channels.conf`. Confirm `dahdi-channels.conf` is included in `chan_dahdi.conf`, otherwise, run command:

```
# echo "#include dahdi-channels.conf" >>
/etc/asterisk/chan_dahdi.conf
```

FXO ports use FXS signaling, while FXS ports adopt FXO signaling. A part of `system.conf`, which is the basic channel configuration file of A2410E, is displayed.

```
# Span 1: OPVXA24XX/24 "OpenVox AE2410E Board 25" (MASTER)
fxoks=1
fxoks=2
fxoks=3
fxoks=4
...
fxsks=21
fxsks=22
fxsks=23
fxsks=24

# Global data

Loadzone= us
defaultzone= us
```

Figure 9 A part of system.conf



In order to match your country pattern, you need to change parameters `loadzone` and `defaultzone` to your country. For example, your system is in CHINA, then, you would like them change to:

```
loadzone = cn
defaultzone = cn
```

Meanwhile, you also need to modify another parameter, which is in file `/etc/asterisk/indications.conf`:

```
country=cn
```

A part of file `/etc/asterisk/dahdi-channels.conf` is showed as below. (Modification, if it is not agree with the hardware setup)

```
; Span 1: OPVXA24XX/24"OpenVox A1610 Board 25" (MASTER)
;;; line="1 OPVXA24XX/24/0 FXOKS"
signalling=fxo_ks //FXS ports use FXO signaling
callerid="Channel 1" <4001>
mailbox=4001
group=5
context=from-internal
channel => 1
callerid=
group=
context=default

;;; line="2 OPVXA24XX/24/1 FXOKS"
signalling=fxo_ks
callerid="Channel 2" <4002>
mailbox=4002
group=5
context=from-internal
channel => 2
callerid=
group=
context=default

.....

;;; line="13 OPVXA24XX/24/12"
signalling=fxs_ks //FXO ports use FXS signaling
callerid=asreceived
group=0
context=from-pstn
channel => 13
callerid=
group=
context=default

;;; line="14 OPVXA24XX/24/13"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 14
callerid=
group=
context=default
```

Figure 10 A part of `dahdi-channels.conf`



Check automatically generated files information is agree with your hardware setup, if not, you should modify to your requirements.

After you done works above, reboot your PC please.

3. Start asterisk by executing command: asterisk -vvvvvvvvvgc

If asterisk is already activate, run “**asterisk -r**” instead.

After entering CLI, run command “**dahdi show channels**”. If DAHDI channels are found, it means DAHDI channels have been loaded into asterisk.

3.4 Call test

1. Log in Elastix

Type IP address of Elastix operation system in browser, next come to “Welcome to Elastix” interface, type your username and password. Elastix login interface is like that



Figure 11 Elastix login interface

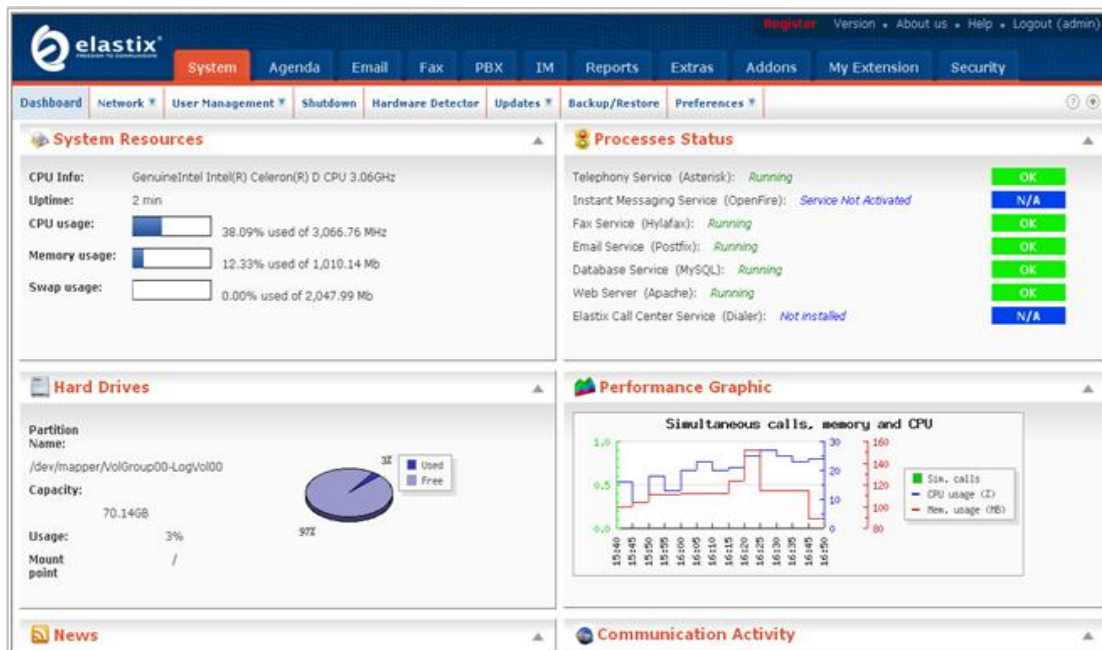


Figure 12 Elastix interface

2. Hardware detection

Click “system” option, then you will see “hardware detection”, choose it you will see the following outcome.

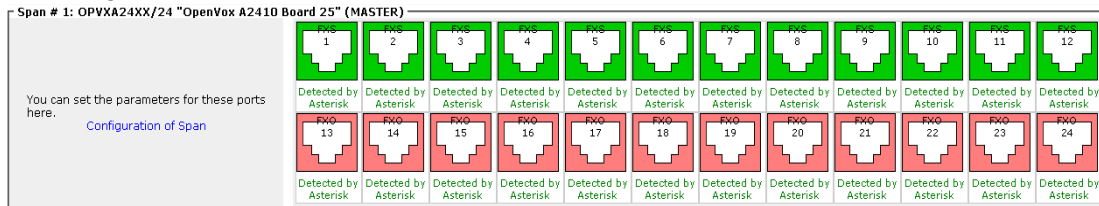


Figure 13 A2410E hardware detection

3. Add SIP extensions

1) Click PBX, extension, choose Generic SIP Device, and finally submit it. You also can refer to the following figure.

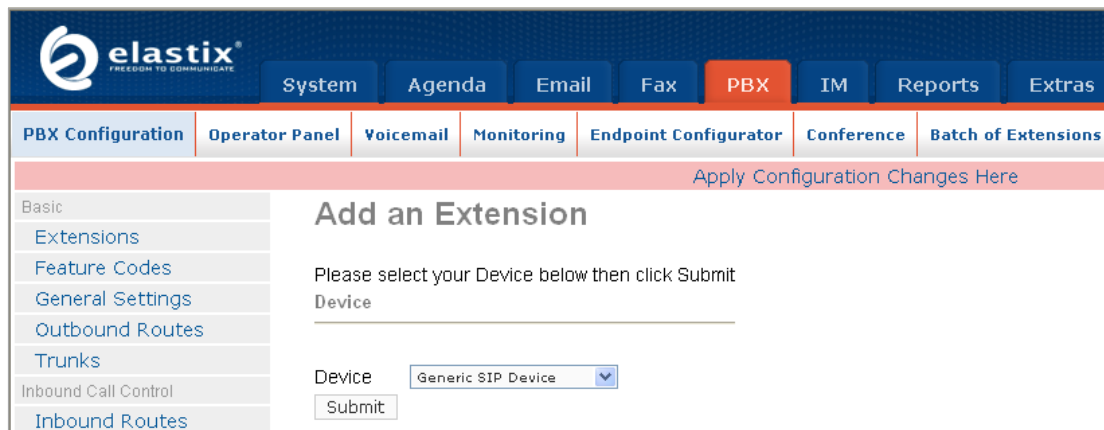


Figure 14 Add a SIP

2) Configure “User Extension”, “Display Name”, “Secret” these three options, keep others default, and submit your configurations.

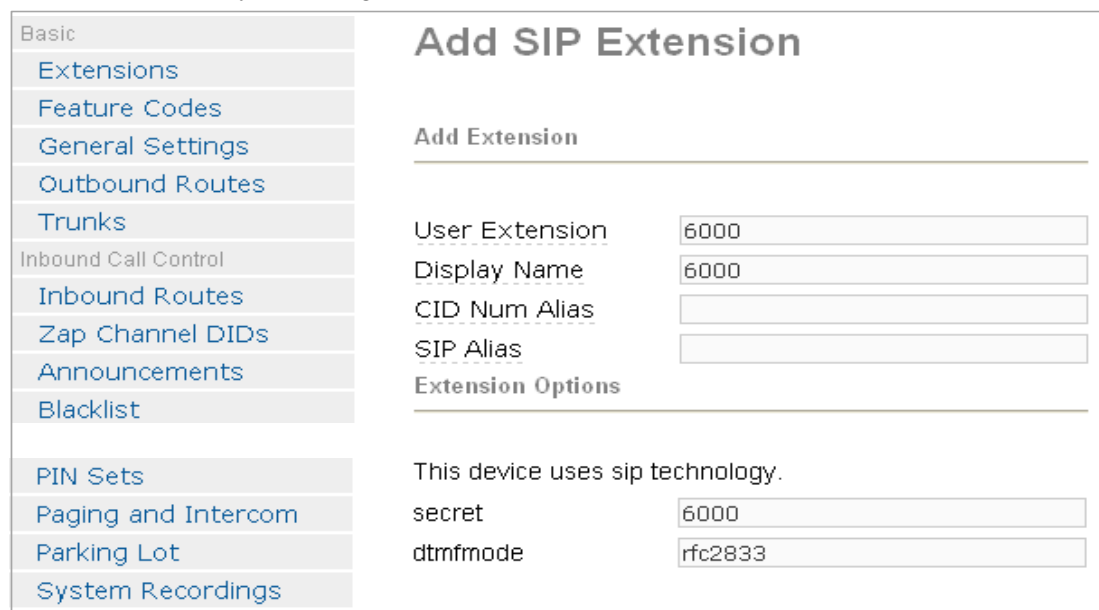


Figure 15 SIP extension parameters

3) After successfully adding, click “Apply Configuration Changes Here” button to take your configurations effect. Also you are able to add another SIP by click “Add Extension”.

Figure 16 SIP Apply Configuration

Once add two or more SIP phones, make them effective and registered, you are able to make the soft phones call each other fluently and conveniently.

4. Add analog phones

1) The way to add an analog phone is similar to SIP phone. The figure below will make you clear.

Figure 17 Add analog phones

2) After finishing works above, interface will come to “Add DAHDI Extension”, please configure “User Extension”, “Display Name”, “channel” these three items, and keep others default, finally click the left bottom “submit”.

Figure 18 Analog extension configurations

3) Click “Add Extension” button to add more phones, and select device type by your requirement. Finally, click “Apply Configuration Changes here” to make your configurations effective.

Once add two or more analog phones, make them effective and registered, you are able to make calls fluently and conveniently.

5. Configure inbound routes

Click “Inbound Routes”, you may like to fill in “Description” which is optional, and then choose “Extensions” in “Set Destination”. After submitting settings, you are also able to select an extension number you need, submit again, finally “Apply Configuration Changes Here”.



The image shows a web form titled "Set Destination". It contains two dropdown menus. The first dropdown is labeled "Extensions" and the second is labeled "<4000> 4000". Below the dropdowns are two buttons: "Submit" and "Clear Destination & Submit".

Figure 19 Inbound routes settings

6. Set outbound routes

Click “Outbound Routes”, set “Route name”, “Dialplan pattern”, “Trunk sequence” these three items to meet your requirements, finally submit changes. The following settings mean all outbound calls through g0 which is an exterior line.

Apply Configuration Changes Here

<ul style="list-style-type: none"> Basic Extensions Feature Codes General Settings Outbound Routes Trunks Inbound Call Control Inbound Routes Zap Channel DIDs Announcements Blacklist CallerID Lookup Sources Day/Night Control Follow Me IVR Queue Priorities Queues Ring Groups Time Conditions Time Groups Internal Options & Configuration Conferences Languages Misc Applications Misc Destinations Music on Hold 	<h2 style="margin: 0;">Add Route</h2> <hr/> <h3 style="margin: 0;">Route Settings</h3> <p>Route Name: <input type="text" value="out"/></p> <p>Route CID: <input type="text"/> <input type="checkbox"/> Override Extension</p> <p>Route Password: <input type="text"/></p> <p>Route Type: <input type="checkbox"/> Emergency <input type="checkbox"/> Intra-Company</p> <p>Music On Hold?: <input type="text" value="default"/></p> <p>Time Group: <input type="text" value="---Permanent Route---"/></p> <p>Route Position: <input type="text" value="Last after out"/></p> <hr/> <h3 style="margin: 0;">Additional Settings</h3> <p>PIN Set: <input type="text" value="None"/></p> <h4 style="margin: 0;">Dial Patterns that will use this Route</h4> <hr/> <p>(<input type="text"/>) + <input type="text"/> [<input type="text"/>X.<input type="text"/> / <input type="text"/>] </p> <p><input type="text" value="+ Add More Dial Pattern Fields"/></p> <p>Dial patterns wizards: <input type="text" value="(pick one)"/></p> <hr/> <h4 style="margin: 0;">Trunk Sequence for Matched Routes</h4> <hr/> <p><input type="text" value="0"/> <input type="text" value="ZAP/g0"/></p> <p style="text-align: right;"><input type="button" value="Submit Changes"/></p>
---	---

Figure 20 Outbound routes configurations

Additional function

Users should run command “**cat /proc/interrupts**” to check A2410E has independent interrupt. If A2410E shares interrupt with other device, it may cause some problems even cannot work normally. While A2410E allows users to modify interrupt pin during firmware upgrade for avoid conflict, please visit the following link for details:

<http://downloads.openvox.cn/pub/misc/opyx-update%20user%20manual.pdf>



4. Reference

www.openvox.cn

www.digium.com

www.asterisk.org

www.voip-info.org

www.asteriskguru.com

www.elastix.org

Tips

Any questions during installation and usage, please consult in our forum or look up for answers from the following websites:

<http://bbs.openvox.cn/>

<http://wiki.openvox.cn/index.php/%E9%A6%96%E9%A1%B5>