深圳开源通信有限公司

OpenVox-Best Cost Effective Asterisk Cards

OpenVox A400P/A400E User Manual

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Chapter 1 Overview

1. What is A400P/A400E

A400P/A400E is a modular analog telephony interface product. It is designed to be hardware compatible with Digium' s TDM400p. A400P/A400E must be used with FXO-100 or FXS-100 together to build a workable system. The FXO-100 and FXS-100 modules are also pin to pin compatible with Digium' s X100M and S100M。

Key Benefits:
Scalable: just add additional cards to extend system
Support PCI for A400P and PCI Express for A400E
Be easy to use: full software and hardware compatible with Digium's TDM400P. User can use Diguim's X100M/S100M module on this card, or use our FXO-100/ FXS-100 Module on TDM400P

RoHS Compliant
Certificates: A-Tick, CE and FCC
trixbox Officially Certified

Disclaimers
Asterisk® is a registered trademark of Digium, Inc.

2. What is Asterisk:

The Definition of Asterisk is described as follow:
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.

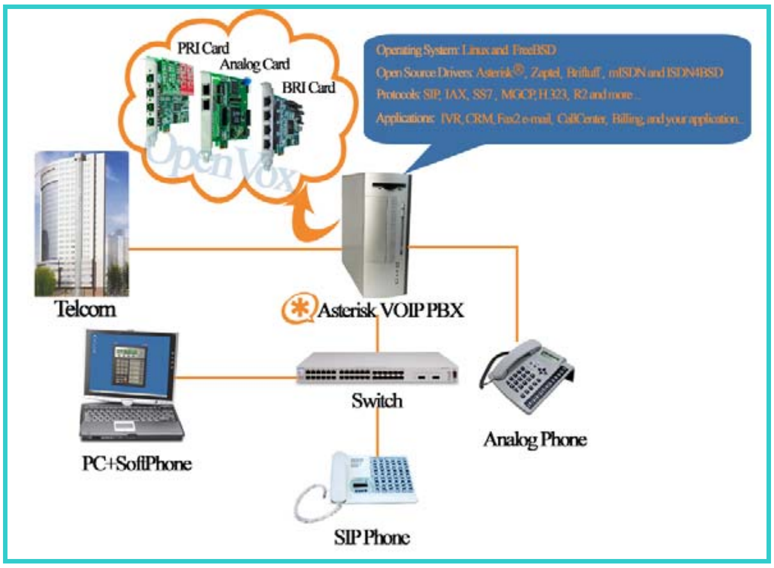

Figure 1: Asterisk Setup

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, H.323 (as both client and gateway), MGCP (call manager only) and SCCP/Skinny(voip-info.org).

Chapter 2 Software Installation and Configuration

Before installing the software, user should have some fundamental knowledge of Linux. Some commands are needed such as cd, tar and vi. Let' s go through the steps:

1. Hardware Installation and Setup

   1. Power off your pc, remember unplug the AC power cable
   2. Insert A400P/A400E into a 3.3V or 5.0V PCI slot or PCI express slot
   3. If the card equips FXS modules (green color), plugging the power supply cable (D style) into A400P.
   4. Plug back the AC power cable, and power on PC.

2. Software Installation and Setup

   1) Checking the A400P/A400E hardware by command: lspci –vvvv
   From figure 2, user can see that that one device called communication controller be found.

```
02:05.0 Ethernet controller: Marvell Technology Group Ltd. 88E8001 Gigabit Ethernet Controller (rev 13)
        Subsystem: ASUSTeK Computer Inc. Marvell 88E8001 Gigabit Ethernet Controller (Asus)
        Flags: bus master, 66Mhz, medium devsel, latency 64, IRQ 209
        Memory at feafc000 (32-bit, non-prefetchable) [size=16K]
        I/O ports at d800 [size=256]
        Expansion ROM at feac0000 [disabled] [size=128K]
        Capabilities: [48] Power Management version 2
        Capabilities: [50] Vital Product Data

02:0d.0 Communication controller: Tiger Jet Network Inc. Tiger3XX Modem/ISDN interface
        Subsystem: Unknown device b100:0003
        Flags: bus master, medium devsel, latency 64, IRQ 217
        I/O ports at d400 [size=256]
        Memory at feafb000 (32-bit, non-prefetchable) [size=4K]
        Capabilities: [40] Power Management version 2
```

Figure 2: Hardware detection

   2) Install supporting packages
   TO install A400P/A400E, user needs install the following packages before compiling asterisk and zaptel driver:
   - kernel-devel
   - zlib
   - zlib-devel
   - openssl
   - openssl-devel

   3) Download zaptel and asterisk
   User can download the source code from asterisk.org. Unzip those packages under /usr/src

   4) Compile zaptel-XXX and asterisk-XXX
   Under /usr/src, execute the commands:

```
cd zaptel-XXX
./configure
make
make install
make config

cd asterisk-XXX
./configure
make
make install
make samples
```

Edit /etc/zaptel.conf file, here the example file likes this:

```
# Span 1: WCTDM/0 "Wildcard TDM400P REV E/F Board 1" (MASTER)
fxoks=1
fxoks=2
fxsks=3
fxsks=4

# Global data

loadzone        = us
defaultzone     = us
~
~
~
```

Figure 3: zaptel.conf

After installing zaptel and asterisk, user executes:

```
modprobe zaptel
modprobe wctdm
ztcfg -vvvvvvvvvvvv
```

After ztcfg command, system will show some information.

```
[root@bogon etc]# modprobe zaptel
[root@bogon etc]# modprobe wctdm
[root@bogon etc]# ztcfg -vvvvvvvv

Zaptel Version: 1.4.12.1
Echo Canceller: MG2
Configuration
======================


Channel map:

Channel 01: FXO Kewlstart (Default) (Slaves: 01)
Channel 02: FXO Kewlstart (Default) (Slaves: 02)
Channel 03: FXS Kewlstart (Default) (Slaves: 03)
Channel 04: FXS Kewlstart (Default) (Slaves: 04)

4 channels to configure.
```

Figure 4: ztcfg

5) Start asterisk

Before starting asterisk, please configure zapata.conf and extensions.conf under /etc/asterisk. zapata-channel.conf is given as bellow:

```
; Span 1: WCTDM/0 "Wildcard TDM400P REV E/F Board 1" (MASTER)
;;; line="1 WCTDM/0/0"
signalling=fxo_ks
callerid="Channel 1" <6001>
mailbox=6001
group=5
context=from-internal
channel => 1
callerid=
mailbox=
group=
context=default

;;; line="2 WCTDM/0/1"
signalling=fxo_ks
callerid="Channel 2" <6002>
mailbox=6002
group=5
context=from-internal
channel => 2
callerid=
mailbox=
group=
context=default

;;; line="3 WCTDM/0/2"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 3
context=default

;;; line="4 WCTDM/0/3"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 4
```

Figure 5: Zapata-channels.conf

Please make sure that zapata-channels.conf has been included into the zapata.conf.

Please edit the extensions.conf, make sure that there is a context called from-pstn. An example is shown as bellow:

```
[from-pstn]
exten => s,1,Dial(zap/1)
exten => s,2,Hangup
[from-internal]
exten => 200,1,Dial(SIP/200)
exten => 200,2,Hangup
```

Figure 6: extensions.conf

Starting asterisk by:

asterisk －vvvvvvvgc

After starting asterisk, user should check status of zap channels first. Under asterisk console, run command: zap show channels:

```
*CLI> zap show channels
   Chan Extension  Context         Language    MOH Interpret
 pseudo            default                     default
      1            from-internal               default
      2            from-internal               default
      3            from-pstn                   default
      4            from-pstn                   default
*CLI>
```

Figure 7: zap-show-channels

If user can see the zap channels, which means that the zap channels are loaded successfully. After then, you can make inbound calls and the call will be forward to first FXS channel:

```
*CLI> [Dec 29 21:06:53] NOTICE[3268]: chan_zap.c:6700 handle_init_event: Alarm cleared on channel 3
[Dec 29 21:06:53] NOTICE[3268]: chan_zap.c:6700 handle_init_event: Alarm cleared on channel 3
    -- Starting simple switch on 'Zap/3-1'
    -- Executing [s@from-pstn:1] Dial("Zap/3-1", "zap/1") in new stack
    -- Called 1
    -- Zap/1-1 is ringing
    -- Zap/1-1 is ringing
[Dec 29 21:07:26] DEBUG[3294]: chan_zap.c:1461 zt_train_ec: No echo training requested
[Dec 29 21:07:26] DEBUG[3294]: chan_zap.c:4045 zt_handle_event: channel 1 answered
    -- Zap/1-1 answered Zap/3-1
[Dec 29 21:07:26] DEBUG[3294]: chan_zap.c:2782 zt_answer: Took Zap/3-1 off hook
[Dec 29 21:07:26] DEBUG[3294]: chan_zap.c:1461 zt_train_ec: No echo training requested
[Dec 29 21:07:26] DEBUG[3294]: chan_zap.c:3275 zt_bridge: master: 3, slave: 1, nothingok: 0
[Dec 29 21:07:26] DEBUG[3294]: chan_zap.c:3290 zt_bridge: Stopping tones on 3/0 talking to 1/0
[Dec 29 21:07:26] DEBUG[3294]: chan_zap.c:3302 zt_bridge: Stopping tones on 1/0 talking to 3/0
[Dec 29 21:07:26] DEBUG[3294]: chan_zap.c:3108 zt_link: Making 1 slave to master 3 at 0
[Dec 29 21:07:26] DEBUG[3294]: chan_zap.c:1263 conf_add: Added 16 to conference 9/3
[Dec 29 21:07:26] DEBUG[3294]: chan_zap.c:1263 conf_add: Added 18 to conference 9/1
    -- Native bridging Zap/3-1 and Zap/1-1
[Dec 29 21:07:32] DEBUG[3294]: chan_zap.c:1421 zt_enable_ec: Echo cancellation already on
[Dec 29 21:07:32] DEBUG[3294]: chan_zap.c:3052 zt_unlink: Unlinking slave 1 from 3
[Dec 29 21:07:32] DEBUG[3294]: chan_zap.c:1295 conf_del: Removed 16 from conference 9/3
[Dec 29 21:07:32] DEBUG[3294]: chan_zap.c:1295 conf_del: Removed 18 from conference 9/1
    -- Hungup 'Zap/1-1'
  == Spawn extension (from-pstn, s, 1) exited non-zero on 'Zap/3-1'
    -- Hungup 'Zap/3-1'
```

Figure 8: inbound calls

Chapter 3 Hardware Setting

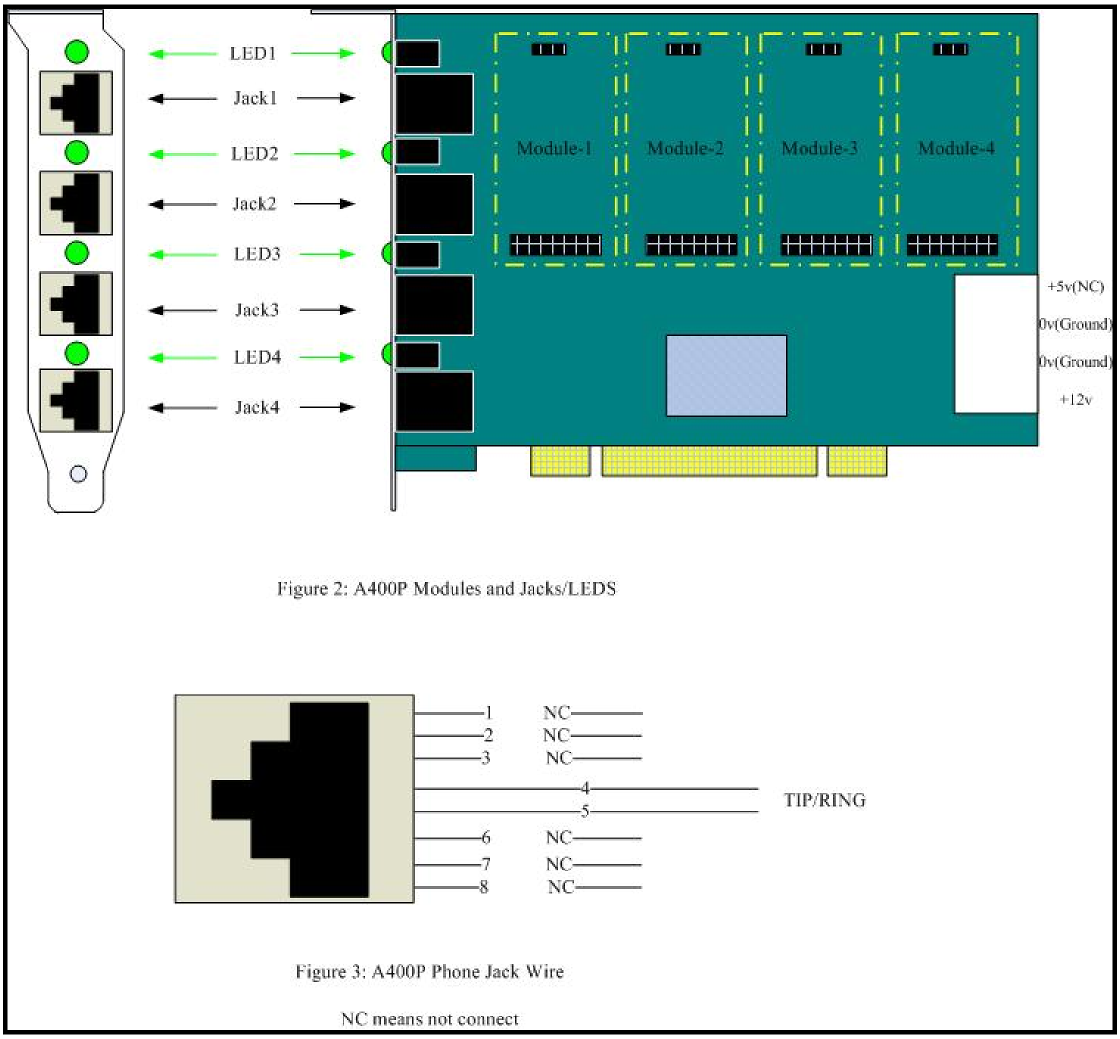

Figure 9: hardware setting

Notes:

Test environments are:

Centos-5.0

Kernel version: 2.6.18-8.el5

Zaptel: 1.4.12.1

Asterisk: 1.4.20.1

Analog Card: OpenVox A400P

Some problems with compiling A400P/A400E have been summarized and documented into FAQ of A400P/A400E; please check that under A400P/A400E categories.

Chapter 4 References

www.openvox.com.cn
www.digium.com
www.asterisk.org
www.voip-info.org
www.asteriskguru.com