



## OpenVox A400P/A400E

### F A Q





## ***OpenVox-Best Cost Effective Asterisk Cards***

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# 1. Installation

Common problems with Zaptel and CentOS:

## 1) Common errors when compiling Zaptel with CentOS 5.2

```
make[4]: *** [/usr/src/zaptel-1.4.11/kernel/xpp/card_fxo.o] 错误 1
make[3]: *** [/usr/src/zaptel-1.4.11/kernel/xpp] 错误 2
make[2]: *** [_module_/usr/src/zaptel-1.4.11/kernel] 错误 2
make[2]: Leaving directory `/usr/src/kernels/2.6.18-92.el5-i686'
make[1]: *** [modules] 错误 2
make[1]: Leaving directory `/usr/src/zaptel-1.4.11'
make: *** [all] 错误 2
```

vi /usr/src/zaptel-xxxx/kernel/xpp/xdefs.h (-xxxx is zaptel's version)

please edit the file at line 114:

```
##if LINUX_VERSION_CODE < KERNEL_VERSION(2,6,19)
```

```
typedef int bool;
```

```
##endif
```

by:

```
##if LINUX_VERSION_CODE < KERNEL_VERSION(2,6,19)
```

```
##if LINUX_VERSION_CODE != KERNEL_VERSION(2,6,18) || !defined(hex_asc)
```

```
typedef int bool;
```

```
##endif
```

```
##endif
```

## 2) Common errors when compiling Zaptel 1.2.17.1, 1.4.2.1 or 1 with Centos 5

if you want to use centos 5 to compile Zaptel 1.2.17.1, 1.4.2.1 or 1.4.1, you may meet with the following errors:

```
/usr/src/zaptel-1.2.17.1/xpp/xbus-core.c: function 'debugfs_open':
```

```
/usr/src/zaptel-1.2.17.1/xpp/xbus-core.c:171: error: 'struct inode'
```

```
hen in line 164 approximately of xbus_core.c file, we must change
```

please edit the xbus\_core.c at line 164 :

```
##if LINUX_VERSION_CODE < KERNEL_VERSION(2,6,19)
```

by

```
##if LINUX_VERSION_CODE < KERNEL_VERSION(2,6,18)
```

## 3) Common Error when compiling zaptel

```
config.status: creating build_tools/make_firmware_object
configure: *** Zaptel build successfully configured ***
****
**** The configure script was just executed, so 'make' needs to be
**** restarted.
****
make: *** [config.status] 错误 1
```

The screen shot shows that you forget to edit `./configure` command when you compile zaptel. Please run the following command one by one:

*make clean*  
*./configure*

#### 4) The LED of A400 is not on after compiled zaptel & asterisk

This is because you haven't configured `/etc/zaptel.conf` and `/etc/asterisk/zapata.conf`, please go to `/usr/src/zaptel-xxxx/kernel/xpp/utis`, run the command:

*./genzaptelconf -sdvM*

for auto-configuration. If the LED is on after you run the command, it means your hardware is working fine. Otherwise, please check if you have connected the system power to your card or run the command :

*dmesg*

After that, if you see the output same as the underline one below, it looks like that you really forget to plug in the power cable to the A400P. Please shutdown your computer and plug the power cable to the card.

*-- DID YOU REMEMBER TO PLUG IN THE HD POWER CABLE TO THE A400P??*

*Unable to do INITIAL ProSLIC powerup on module 0*

*Init ProSlic with Manual Calibration*

*Unable to do INITIAL ProSLIC powerup on module 0*

*Module 0: FAILED FXS (FCC)*

#### 5) What are the differences between FXS and FXO interfaces?

FXS (Foreign eXchange Station) is an interface which drives a telephone. FXS interfaces get phones plugged into them, delivery battery, and provide ringing. FXS interfaces are signalled with FXO signalling.

FXO (Foreign eXchange Office) is an interface that connects to a PSTN line. It enables your PBX to have the access to the public telephone network. FXO interfaces use FXS signalling. FXS interfaces allow you to hook telephones to your PBX, and FXO interfaces allow you to connect your PBX to real PSTN lines.

#### 6) Do I have to install OpenVox card when installing Zaptel? What is Zaptel?

No, you don't have to. Zaptel is the driver for asterisk hardware. It can detect the hardware by read the configuration file in `etc/zaptel.conf`. The `zaptel.conf` can be set by hand or automatically set by running the `genzaptelconf -sdvM` under `/usr/src/zaptel-1.4.11/kernel/xpp/utis/`  
(The zaptel is supposed to be installed in `zaptel-1.4.11`)



## 2. Debug

### 1) No Channels are found when runs the Command “Zap Show Channels” in CLI

Environment:

CentOS 5.0

zaptel-1.4.12.1

asterisk-1.4.21.1

\*CLI> *zap show channels*

Chan Extension	Context	Language	MOH Interpret
pseudo	default		default

\*CLI>

1>

edit command ztcfg -vvvvv

root@bogon ~]# *ztcfg -vvvvv*

Zaptel Version: 1.4.12.1

Echo Cancellor: 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.

When your card's FXS+FXO channels are corresponding to the output, it means your zaptel drive works fine.

2>

If you use “genzaptelconf -sdvM” to automatically generated profile, then edit “asterisk -r”, input “reload” and check the output like:

*/etc/asterisk/zapata-channels.conf: Found*

And enter /etc/asterisk/zapata.conf

Add a new line “*#include zapata-channels.conf*” at last line of this file, save and quit.

-- Reloading module 'chan\_zap.so' (Zapata Telephony)

== Parsing '/etc/asterisk/zapata.conf': Found



3>

Then restart the asterisk, and use command “zap show channels” to check the output

\*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

## 2) Why my card gets an IRQ miss?

Each piece of hardware takes 1,000 interrupts per second. When, for some reason the cards get less than this, an IRQ miss occurs. You can see if the card is missing interrupts using 'zttool.'

IRQ misses can cause different problems with Asterisk. Symptoms of IRQ misses are bad audio quality or perhaps PRI errors, although IRQ misses will not cause alarms. Also DTMF detection not working is something that can be caused by IRQ misses as well.

Several common things that contribute to IRQ misses are:

- Running the X window system
- Shared IRQs
- No hard drive DMA
- Hard drive DMA too high (shoot for udma3)
- Running serial terminals or frame buffers

To check for shared IRQs you can run:

# *cat /proc/interrupts*

```

CPU0
0      10756672      XT-PIC  timer
2       0           XT-PIC  cascade
5      10812879      XT-PIC  uhci_hcd, uhci_hcd, wctdm
10     226219        XT-PIC  t1xpc, CS46XX
11     1550046       XT-PIC  eth0, nvidia
12     387234        XT-PIC  i8042
14     32641         XT-PIC  ide0
15     18           XT-PIC  ide1
NMI    0
LOC 10757616
ERR 40481

```



## MIS 0

Notice the A100P card sharing with the sound card, and the A400P card is sharing with the USB controller. This is likely to cause problems at most of the time. If you are not using any USB devices, then it would probably be fine. But it would be better to disable USB or get the card on its own IRQ.

There are several ways to move cards to their own IRQ.

- Turn on APIC
- Tweak BIOS settings
- Try a different PCI slot
- Use setpci

### 3) Connect analog telephone to FXS channel but find no dialing tone

It's probably because your channels haven't been configured correctly. Please use command

`“ztcfg -vvvv”`

Check the output , it should like:

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

4 channels to configure.

Changing signalling on channel 1 from Unused to FXS Kewlstart
ZT_CHANCONFIG failed on channel 1: Invalid argument (22)
Did you forget that FXS interfaces are configured with FXO signalling
and that FXO interfaces use FXS signalling?
```

It means your configuration in `/etc/zaptel.conf` have some error, FXS channels use the FXO signaling, FXO channel use the FXS signaling, so edit `zaptel.conf`:

```
fxsks=1
```

```
fxsks=2
```

```
.....
```

Run command `“ztcfg -vvvv”` again to check

```
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.
```

Then check `/etc/asterisk/zapata.conf` , if you use command `“./genzaptelconf -sdvM”`, please check `/etc/asterisk/zapata-channels.conf`, make sure the channels config is the same as `zaptel.conf` (**FXS channels use the FXO signaling, FXO channel use the FXS signaling**), restart asterisk.



#### 4) How to remove the old asterisk before install a new one?

First, please stop asterisk module and uninstall it.

The red color command will stop asterisk and close all.

```
killall -9 safe_asterisk
```

```
killall -9 asterisk
```

then please remove zaptel driver and check what modules have loaded:

```
[root@zebru]# lsmod | grep zaptel
zaptel                214820      2 wcfxo,wctdm
crc_ccitt              2113       1 zaptel
```

edit command to remove:

```
modprobe -r wcfxo
```

```
modprobe -r wctdm
```

```
modprobe -r zaptel
```

enter "`lsmod | grep zaptel`" again , the output should be null.

Remove Asterisk:

Run the following command , then you can remove the asterisk system(**Unrecoverable**)。 Before you do this, please backup the file you wish to keep. For example, the log file and asterisk's config。

```
rm -rf /etc/asterisk
rm -f /etc/zaptel.conf
rm -rf /var/log/asterisk
rm -rf /var/lib/asterisk
rm -rf /var/spool/asterisk
rm -rf /usr/lib/asterisk
```

now your asterisk is completely removed。

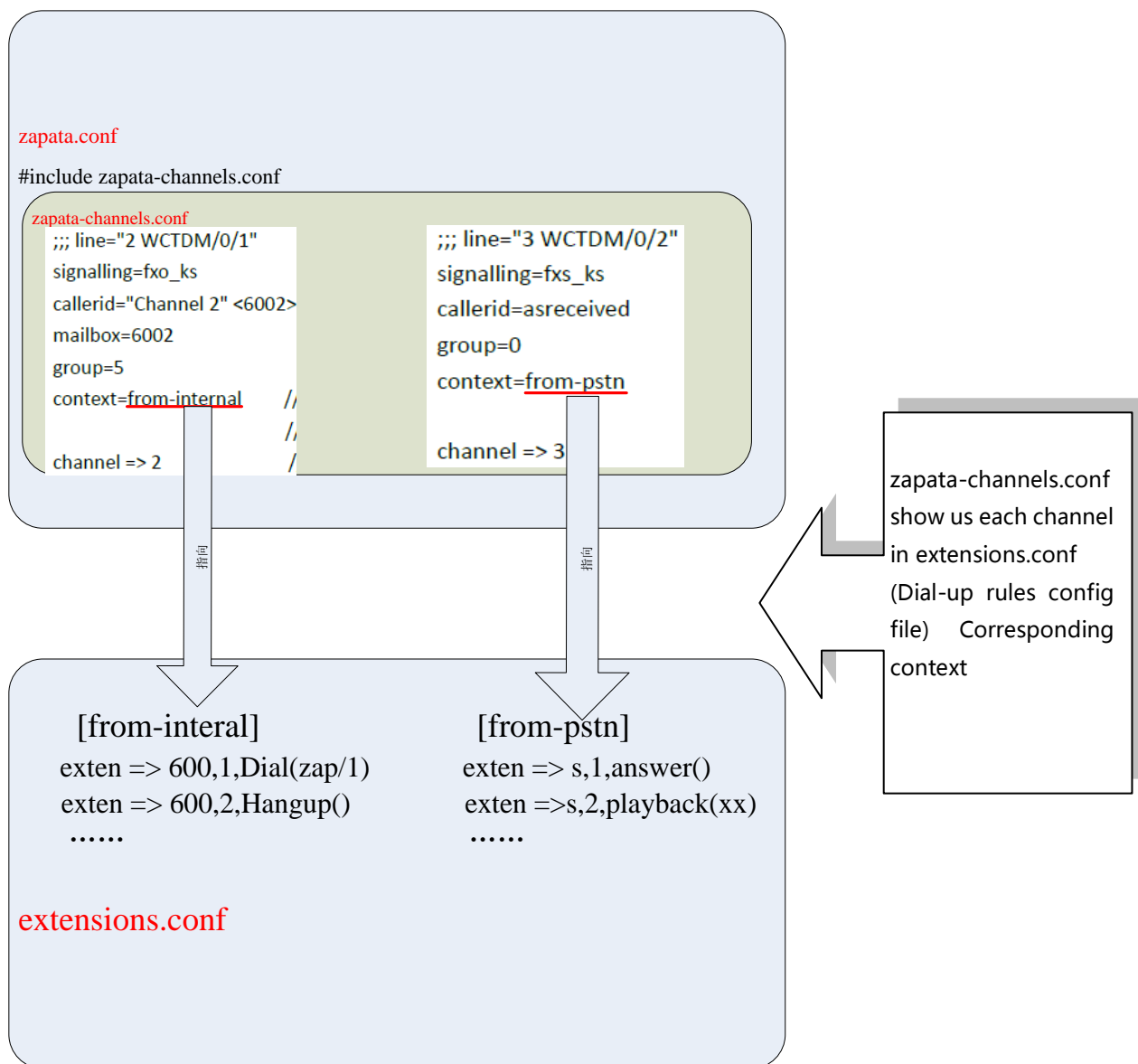


### 3. Voice quality

1) When dialup on the analog telephone, it shows busy tone in CLI

```
-- Starting simple switch on 'Zap/2-1'
-- Hungup 'Zap/2-1'
```

This problem may be caused by the configuration in your extensions.conf, asterisk doesn't "recognize" the number you dial, then it reports you busy tone. The following picture shows us their relationships:



## 2) SIP softphone can't be registered the asterisk server

1. Please check if the Linux system's firewall and SELinux is close. If not, please close them or it will cause the problem below .



2.cd the sip.conf, check your sip extension to see if the setting is correct .

The follow is a basic configuration :

```
[202]
type=friend
username=202
secret=111111
host=dynamic
context=internal
```

## 3) Why can't I get any Caller IDs?

In some PSTN connections, you have to subscribe to CALLER ID from the telco. In other cases, the caller id is delivered between the first and second ring, so start by waiting a while before answering, i.e. exten => s, 1, wait(2), exten => s, 2, answer

## 4) echo problem

Please verify that your rxgain and txgain are set to 0 in your zapata.conf during testing. High rxgain and txgain settings can cause echo.

Please verify that your echocancelwhenbridged option is set to no. Generally, it is not necessary, and in fact undesirable, to echo cancel when the circuit path is entirely TDM. You may need to enable this only under the rare circumstance that you are unable to completely eliminate echo otherwise.

The echocancel option in the zapata.conf file only enables echo cancellation for Zap channels. Please try an echocancel setting of 64 and an echotraining setting of 800 for testing. Valid settings for echocancel are 32, 64, 128, and 256. Valid settings for echotraining are 10 - 2000. The echocancel and echotraining options may need to be tweaked in order to fully eliminate echo. This will be a trial and error test that must be performed in order to determine the best echocancel and echotraining settings for your system. Start with an echocancel setting of 32. Start with an echotraining setting of 100. Increment the echotraining setting by 100. Do not exceed an echotraining setting of 1200.

Increment the echocancel option to the next highest setting if you still exhibit echo after reaching an echotraining setting of 1200. Repeat this process until you find the settings that completely eliminate echo. Asterisk must be restarted in order for these changes to take effect.

#### 5) Does the openvox card have echo cancel software?

No, zaptel has the echocancel function. You can also use Octasic SoftEcho but you will have to buy the licence first.

#### 6) Common Problems with trixbox CE

**zapata-channels.conf hasn't been included by zapata.conf**

Please refer to chapter 2, question 1

#### 7) Asterisk can't detect hang-up signal properly

There are several settings in zaptel.conf and zapata.conf that you can experiment with. If you are using loopstart, try using kewlstart instead. Kewlstart is loopstart with disconnect supervision. Your telco must have what's called "disconnect supervision" configured on the line. Another name for disconnect supervision is "forward disconnect".

You may also edit the ZT\_KEWLTIME variable in zaptel.h. Navigate to the #define ZT\_KEWLTIME line and change the value from 500 to 1000 (You'll need to recompile zaptel before this takes effect).

You can enable busydetect and busycount to detect when a call has been hungup. Set 'busydetect=yes' and 'busycount=10'. You may experience hangups if your busycount number is too low, so increase the number until you find a variable that you're comfortable with. You can also try enabling callprogress, and set your progzone to your current country code. Callprogress is experimental, so if you experience frequent disconnects, you should disable it. Busydetect and callprogress are ways to try and detect a hangup in software which isn't as reliable as having the actual signalling indicate the hangup.

If you're telcom uses polarity switching to signal for answer and/or disconnects, you should use 'answeronpolarityswitch=yes' and 'hanguponpolarityswitch=yes'



## 8) Why am I getting a clicking noise?

If a clicking noise is present when dialing through an FXO or when getting dialtone from an FXS, this is caused by echotraining. Turn it off to get rid of the clicking. The click is necessary for the echotraining.

## 9) How to solve the hang-up problems in random calls

If you have `busydetect=yes` or `callprogress=yes` in `zapata.conf` then this is why the problem happens. You might want to try to set `busycount=10` if using `busydetect`. As a last resort you may just change `busydetect` to `no`: `busydetect=no`. Then asterisk won't try to detect the hangup on incoming calls. Notice that if "Loopstart with Remote Disconnection Supervision" (`Kewlstart`) is working for you than you don't need `busydetect=yes` at all. You can know if your current config is set to `Kewlstart` if you have a line `fxsks=channel_no` in `zaptel.conf` and `signalling=fxs_ks` in `zapata.conf`.



## 4. Demo File

Environment:

CentOS 5.2  
zaptel-1.4.11  
asterisk-1.4.21.1  
openvox A400P 2FXS + 2FXO

Some important configure files as below, (please check the annex)

- 1.zaptel.conf
2. zapata.conf
- 3.zapata-channels.conf
- 4.extensions.conf



## 5. Remark

The following combination of Asterisk & zaptel versions are recommended (for CentOS 5.0 or 5.2 )

1. [asterisk-1.4.19-1](#)      [zaptel-1.4.10.1](#)
2. [asterisk-1.4.21.1](#)      [zaptel-1.4.11.1](#)
3. [asterisk-1.4.17-1](#)      [zaptel- 1.4.7.1](#)
4. [asterisk-1.4.22](#)      [zaptel-1.4.11.1](#)

Asterisk Supports All OpenVox Cards:

- [A400E PCI-e Card-TDM compatible](#)
- [A400M Mini-PCI Card](#)
- [A400P PCI Card - TDM400 compatible](#)
- [A800P PCI Card - 8 port FXS/FXO card](#)
- [A1200P PCI Card - 12 port FXS/FXO card](#)
- [D110P PCI Card - T1/E1/J1 TE110P compatible](#)
- [D210P PCI Card - T1/E1/J1 TE210P/TE205P compatible](#)
- [D210E PCI-express Card - T1/E1/J1 TE210P/TE205P compatible](#)
- [D410P PCI Card - T1/E1/J1 TE410/TE405P compatible](#)
- [D410E PCI-express Card - T1/E1/J1 TE410/TE405P compatible](#)
- [DE210E PCI Express card with hardware Echo cancellation - T1/E1/J1](#)
- [DE210P PCI card with hardware Echo cancellation - T1/E1/J1](#)
- [DE410E PCI Express card with hardware Echo cancellation - T1/E1/J1](#)
- [DE410P PCI card with hardware Echo cancellation - T1/E1/J1](#)
- [B100P PCI Card-1 Port ISDN Card](#)
- [B200P PCI Card - 2 Port ISDN BRI Card](#)
- [B200E PCI-e Card- 2 Port ISDN BRI Card](#)
- [B200M Mini PCI card- 2 Port ISDN BRI Card](#)
- [B400P PCI Card - 4 Port ISDN BRI Card](#)
- [B400E PCI Express 1.0 - 4 Port ISDN BRI Card](#)
- [B400M Mini PCI card- 4 Port ISDN BRI Card](#)
- [B800P PCI Card - 8 Port ISDN BRI Card](#)

The Following Telephones and softphone are recommended:

Telephones Hardware and terminal adapter

- [Cisco 79xx series: Configuring Cisco 79xx phones with Asterisk](#)
- [Cisco ATA 18x series: Cisco ATA-18x Series Analog Telephone Adaptor](#)
- [Cisco 12SP+/VIP30 Configuring Cisco 12SP phones with Asterisk](#)



- [D-Link](#) DPH-540: [DPH-540](#) and a [video](#) discussing and promoting the phone.
- [Digitmat GP1266](#) based on [Palmmicro AR1688](#) with native IAX2 support
- [Snom Phones](#) products: [Tweaks to make the SNOMS happier with Asterisk](#) | [Howto configure Asterisk with SNOM phones](#)

softphone

- [CounterPath](#) X-Lite: [How to configure Asterisk and X-Lite](#)

Asterisk useful debug commands:

1. `ztcfg -vvvv`  
To check the channel's config
2. *In CLI, "zap show channels"*  
To check if the asterisk has recognize the channels
3. `dmesg`  
To check self-drive information,





## 6. Reference

[www.openvox.com.cn](http://www.openvox.com.cn)

[www.digium.com](http://www.digium.com)

[www.asterisk.org](http://www.asterisk.org)

[www.voip-info.org](http://www.voip-info.org)

[www.asteriskguru.com](http://www.asteriskguru.com)

