

OpenVox

深圳开源通信有限公司

OpenVox-Best Cost Effective Asterisk Cards

OpenVox B100P User Manual for mISDN



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OpenVox-Best Cost Effective Asterisk Cards

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Chapter 1 Overview

1. What is B100P

B100P is a PCI 2.2 compliant card supporting one BRI S/T interface, with an onboard multi NT power feeding circuit. NT/TE mode can be independently configured on the port.

B100P can be implemented to build open source Asterisk based systems such as ISDN PBX and VoIP gateway.

Target Applications:

High Performance ISDN PC Cards

ISDN PABX for BRI

VoIP Gateways

ISDN LAN Routers for BRI

ISDN Least Cost Routers for BRI

ISDN Test Equipment for BRI

Main Features:

One S/T interface

ITU-T I.430 and TBR 3 certified and S/T ISDN supporting in TE and NT mode

Integrated PCI bus interface (Spec. 2.2) for 3.3V and 5V signal environments

Port can be independently configured for TE or NE mode

Support mISDN driver

Application ready: use Asterisk to build your IP-PBX/Voicemail system

RoHS compliant

Certificates: CE, FCC

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.

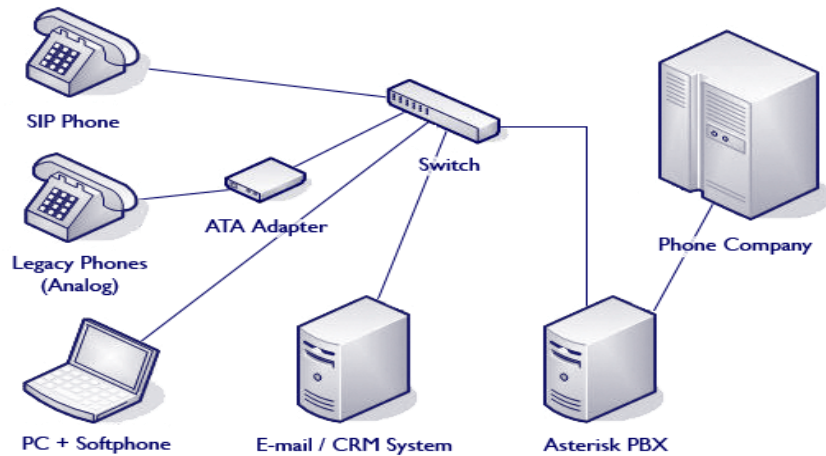


Figure 1: Asterisk Setup

Source (<http://www.siriusit.co.uk/uploads/images/consulting/asteriskSetup.gif>)

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).

command **yum** will install the sources for your current version of the kernel. It is time to check for the availability of some other packages:

```
rpm -q bison
rpm -q bison-devel
rpm -q ncurses
rpm -q ncurses-devel
rpm -q zlib
rpm -q zlib-devel
rpm -q openssl
rpm -q openssl-devel
rpm -q gnutls-devel
rpm -q gcc
rpm -q gcc-c++
```

If any of those packages is not installed, please install those packages by using **yum**

```
yum install bison
yum install bison-devel
yum install ncurses
yum install ncurses-devel
yum install zlib
yum install zlib-devel
yum install openssl
yum install openssl-devel
yum install gnutls-devel
yum install gcc
yum install gcc-c++
```

3) Downloading, unzipping and compiling driver

- A. Download the stable version of mISDN, mISDNuser, chan_mISDN and asterisk drivers from http://www.misdn.org/index.php/Installing_mISDN, and copy the tar file to /usr/src/:

```
root@new-host ~]# cd /usr/src
root@new-host src]# ls
asterisk          chan_misdn      install_openvox_a.sh  mISDN-1_1_7      mISDNuser-1_1_7      redhat
asterisk-1.4.18.tar.gz  chan_misdn.tar.gz  kernels              mISDN-1_1_7.tar.gz  mISDNuser-1_1_7.tar.gz
root@new-host src]#
```

Here, we are using mISDN-1_1_7 and mISDNuser_1_1_7. Please check it from mISDN.org for details.

- B. Compiling mISDN, mISDNuser, chan_mISDN and asterisk

```
cd /usr/src/mISDN-1_1_7
make
make install
```

```
cd /usr/src/mISDNuser-1_1_7
make
make install
```

```
cd /usr/src/chan_misdn
make
make install
```

```
cd /usr/src/asterisk
./configure
make menuselect
```

Now you should enable chan_misdn in the Channel Driver Section and reinstall asterisk with "make install".

```
*****
Asterisk Module and Build Option Selection
*****

Press 'h' for help.

[*] 1. chan_agent
[*] 2. chan_alsa
[ ] 3. chan_features
XXX 4. chan_gtalk
XXX 5. chan_h323
[*] 6. chan_iax2
[*] 7. chan_local
[*] 8. chan_mgcp
[*] 9. chan_misdn
XXX 10. chan_nbs
[*] 11. chan_oss
[*] 12. chan_phone
[*] 13. chan_sip
[*] 14. chan_skinny
XXX 15. chan_zap
XXX 16. chan_vpb
```

After that procedure you should have the current mISDN releases installed and the current chan_misdn with asterisk.

C. Modifying and loading modules for mISDN

User can run: *mISDN scan* to check what kind of cards is installed in the system. *If B200P is installed in the system*, please edit the file: mISDN

under /usr/sbin:
cd /usr/sbin/
run mISDN scan and mISDN config to create mISDN.conf
The file mISDN.conf under /etc should like this:

```
mISDNconf>  
  <module poll="128" debug="0" timer="no">hfcmulti</module>  
  <module debug="0" options="0">mISDN_dsp</module>  
  <devnode user="root" group="root" mode="644">mISDN</devnode>  
  <card type="hfcpci">  
    <port mode="te" link="ptmp">1</port>  
  </card>  
</mISDNconf>
```

vi /etc/asterisk/misdn.conf, the part of it looks like this:

```
max_incoming=-1  
;  
; defines the maximum amount of outgoing calls per port for this group  
; exceeding calls will be rejected  
;  
max_outgoing=-1  
[isdn]  
; define your ports, e.g. 1,2 (depends on mISDN-driver loading order)  
ports=1  
; context where to go to when incoming Call on one of the above ports  
context=from-pstn
```

must match the context called from-pstn in extensions.conf

vi /etc/asterisk/extensions.conf, the dialplan shows as the follow:

```
[from-pstn]  
;  
; We start with what to do when a call first comes in.  
;  
exten => s,1,Wait(1) ; Wait a second, just for fun  
exten => s,n,Answer ; Answer the line  
exten => s,n,Set(TIMEOUT(digit)=5) ; Set Digit Timeout to 5 seconds  
exten => s,n,Set(TIMEOUT(response)=10) ; Set Response Timeout to 10 seconds  
exten => s,n(restart),BackGround(demo-congrats) ; Play a congratulatory message  
exten => s,n(instruct),BackGround(demo-instruct) ; Play some instructions  
exten => s,n,WaitExten ; Wait for an extension to be dialed.  
  
exten => 2,1,BackGround(demo-moreinfo) ; Give some more information.  
exten => 2,n,Goto(s,instruct)
```

The dialplan is linked with misdn.conf. The example shows that the port 1 is used for inbound calls.

D. Running mISDN and asterisk:

Execute: mISDN start

Execute: *asterisk -vvvvvvc*, Please make sure that the *chan_misdn.so* is loadable in *modules.conf* under */etc/asterisk*.

If it is not loaded automatically by asterisk server, please run: *load chan_misdn.so* in asterisk console.

The below screens show the asterisk makes inbound calls:

```
Asterisk Ready.
CLI> == Starting mISDN/2-u0 at from-pstn,1 failed so falling back to exten 's'
-- Executing [s@from-pstn:1] Wait("mISDN/2-u0", "1") in new stack
-- Executing [s@from-pstn:2] Answer("mISDN/2-u0", "") in new stack
-- Executing [s@from-pstn:3] Set("mISDN/2-u0", "TIMEOUT(digit)=5") in new stack
-- Digit timeout set to 5
-- Executing [s@from-pstn:4] Set("mISDN/2-u0", "TIMEOUT(response)=10") in new stack
-- Response timeout set to 10
-- Executing [s@from-pstn:5] Background("mISDN/2-u0", "demo-congrats") in new stack
-- <mISDN/2-u0> Playing 'demo-congrats' (language 'en')
== Spawn extension (from-pstn, s, 5) exited non-zero on 'mISDN/2-u0'
```

Notes:

Test environments:

OS: Centos 5

Kernel version: 2.6.18-8.15

Asterisk version: Asterisk-1.4.18

mISDN version: mISDN-1_1_7

Hardware: OpenVox B100P

No LEDs

If want to know more mISDN, please go to misdn.org.

References:

<http://www.openvox.com.cn>

http://www.misdn.org/index.php/Main_Page

<http://www.asterisk.org>

<http://www.voip-info.org>

Chapter 3 Hardware Setting

