Install B100P/B200P/B400P/B800P with mISDN/trixbox-2.2.-12

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trixbox is very popular PBX. Here, an instruction is given to those who want to install OpenVox B100P/B200P/B400P/B800P in trixbox-2.2.12. Before installing the cards, you should have basic knowledge of Asterisk, mISDN, Linux and trixbox. If you do not know about those stuffs, please google those areas. Of course, it does mean that you can not play OpenVox cards with trixbox. Go ahead with me , and this instruction will manage to help you to make it happen！ There are few steps users must go through：

1. Check the hardware. You must make sure the card can be found in the system. Please run the command to check that:

```
02:04.0 Network controller: Cologne Chip Designs GmbH ISDN network controller [HFC-PCI] (rev 02)
        Subsystem: Cologne Chip Designs GmbH ISDN Board
        Control: I/O- Mem+ BusMaster+ SpecCycle- MemWINV- VGASnoop- ParErr- Stepping- SERR- FastB2B-
        Status: Cap+ 66Mhz- UDF- FastB2B- ParErr- DEVSEL=medium >TAbort- <TAbort- <MAbort- >SERR- <PERR-
        Latency: 16 (4000ns max)
        Interrupt: pin A routed to IRQ 185
        Region 0: I/O ports at a000 [disabled] [size=8]
        Region 1: Memory at f7004000 (32-bit, non-prefetchable) [size=256]
        Capabilities: [40] Power Management version 1
                Flags: PMEClk- DSI+ D1+ D2+ AuxCurrent=0mA PME(D0+,D1+,D2+,D3hot+,D3cold-)
                Status: D0 PME-Enable- DSel=0 DScale=0 PME+
```

2. Update trixbox and install right version of kernel source.
   - Please get the trixbox-update.sh from trixobx. run the command to update trixbox:
   - You must make sure the version of kernel. Please run this command to check it:

     uname -a

```
[root@asterisk1 ~]# uname -a
Linux asterisk1.local 2.6.9-34.0.2.ELsmp #1 SMP Fri Jul 7 19:52:49 CDT 2006 i686 i686 i386 GNU/Linux
```

Please run the command to install the kernel-source:

yum -y install kernel-smp-devel, be sure the version of kernel,

if system kernel is without smp, please run this:

yum -y install kernel-devel

3. Get and run the script. change the directory to /usr/src and run the script to install mISDN:

   wget http://www.beronet.com/downloads/install-misdn-mqueue.tar.gz

   tar xzf install-misdn-mqueue.tar.gz

   cd install-misdn-mqueue

   make

   make install

   After installing the script, there are few packages under the /usr/src directory:

```
[root@asterisk1 src]# cd install-misdn-mqueue
[root@asterisk1 install-misdn-mqueue]# ls
app_bundle          asterisk-1.2-includes  chan_misdn.tar.gz  mISDN-1_1_7_2         mISDNuser-1_1_7_2         README
app_bundle.tar.gz   chan_misdn             Makefile           mISDN-1_1_7_2.tar.gz  mISDNuser-1_1_7_2.tar.gz
[root@asterisk1 install-misdn-mqueue]#
```

   chan_misdn.so should be in /usr/lib/asterisk/modules.

```
[root@asterisk1 src]# cd /usr/lib/asterisk/modules/
[root@asterisk1 modules]# ls
app_addon_sql_mysql.so  app_image.so               app_setcidname.so             chan_alsa.so      format_mp3.so
app_adsiprog.so         app_lookupblacklist.so     app_setcidnum.so              chan_features.so  format_ogg_vorbis.so
app_alarmreceiver.so    app_lookupcidname.so       app_setrdnis.so               chan_h323.so      format_pcm_alaw.so
app_authenticate.so     app_macro.so               app_settransfercapability.so  chan_iax2.so      format_pcm.so
app_cdr.so              app_math.so                app_sms.so                    chan_local.so     format_sln.so
app_chanisavail.so      app_md5.so                 app_softhangup.so             chan_mgcp.so      format_vox.so
app_chanspy.so          app_meetme.so              app_speech_utils.so           chan_misdn.so     format_wav_gsm.so
app_controlplayback.so  app_milliwatt.so           app_stack.so                  chan_oss.so       format_wav.so
app_curl.so             app_mixmonitor.so          app_system.so                 chan_phone.so     func_callerid.so
app_cut.so              app_mp3.so                 app_talkdetect.so             chan_sip.so       func_enum.so
app_db.so               app_nbscat.so              app_test.so                   chan_skinny.so    func_uri.so
app_dial.so             app_nv_backgrounddetect.so app_transfer.so               chan_zap.so       pbx_ael.so
app_dictate.so          app_nv_faxdetect.so        app_txfax.so                  codec_adpcm.so    pbx_config.so
app_directed_pickup.so  app_page.so                app_txtcidname.so             codec_alaw.so     pbx_dundi.so
app_directory.so        app_parkandannounce.so     app_url.so                    codec_a_mu.so     pbx_functions.so
app_disa.so             app_playback.so            app_userevent.so              codec_g726.so     pbx_loopback.so
app_dumpchan.so         app_privacy.so             app_verbose.so                codec_gsm.so      pbx_realtime.so
app_echo.so             app_queue.so               app_voicemail.so              codec_ilbc.so     pbx_spool.so
app_enumlookup.so       app_random.so              app_waitforring.so            codec_lpc10.so    res_adsi.so
app_eval.so             app_readfile.so            app_waitforsilence.so         codec_speex.so    res_agi.so
app_exec.so             app_read.so                app_while.so                  codec_ulaw.so     res_config_mysql.so
app_externalivr.so      app_realtime.so            app_zapateller.so             codec_zap.so      res_config_odbc.so
app_festival.so         app_record.so              app_zapbarge.so               format_au.so      res_crypto.so
app_flash.so            app_rxfax.so               app_zapras.so                 format_g723.so    res_features.so
app_flite.so            app_saycountpl.so          app_zapscan.so                format_g726.so    res_indications.so
app_forkcdr.so          app_sayunixtime.so         cdr_addon_mysql.so            format_g729.so    res_monitor.so
app_getcpeid.so         app_senddtmf.so            cdr_csv.so                    format_gsm.so     res_musiconhold.so
app_groupcount.so       app_sendtext.so            cdr_custom.so                 format_h263.so    res_odbc.so
app_hasnewvoicemail.so  app_setcallerid.so         cdr_manager.so                format_ilbc.so    res_speech.so
app_ices.so             app_setcdruserfield.so     chan_agent.so                 format_jpeg.so
[root@asterisk1 modules]# pwd
/usr/lib/asterisk/modules
```

4. Configure the hardware: change the directory to /usr/sbin and run:

   misdn-init scan; show the card information

   misdn-init config ; set configuration file for you.

   misdn-init start ; start the mISDN

   If all steps are executed successfully, you can go to the next steps. If you have problems with installing mISDN, please refer to Troubleshoot section.

5. Add mISDN for system booting

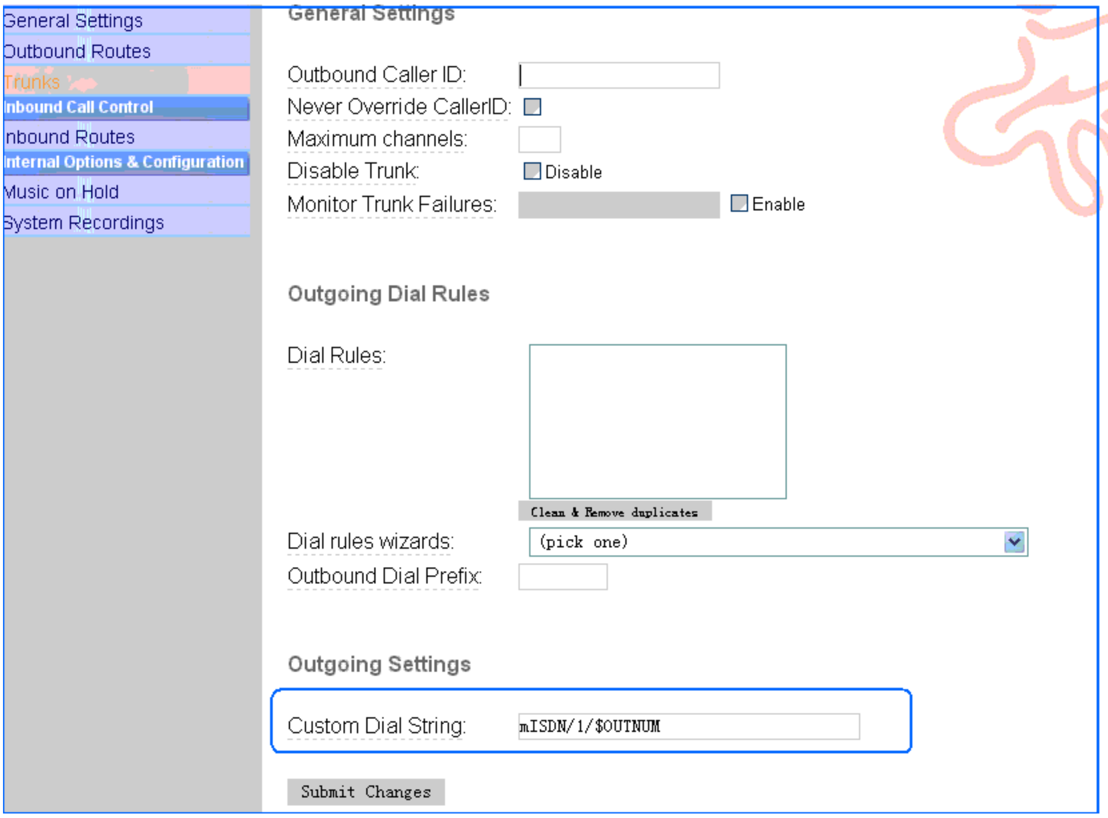
   - *chkconfig --add misdn-init*
   - open the rc.local file: vi /etc/rc.d/rc.local
   - add before "/usr/sbin/amportal start" the following line:

     /usr/sbin/misdn-init start

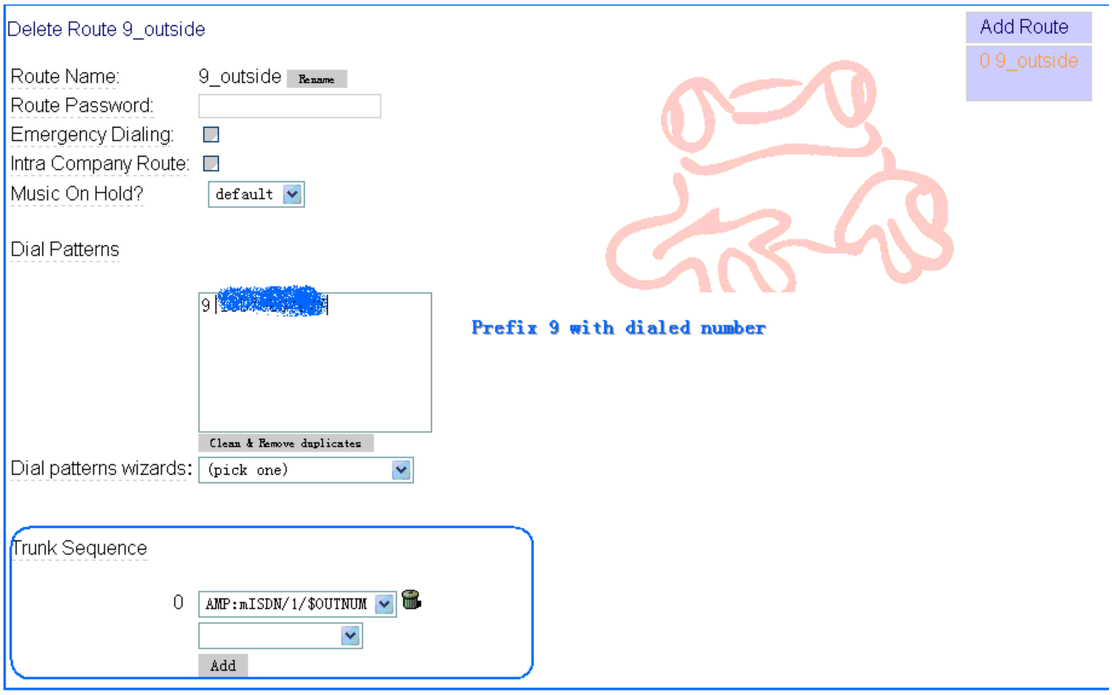
```
#!/bin/sh
#
# This script will be executed *after* all the other init scripts.
# You can put your own initialization stuff in here if you don't
# want to do the full Sys V style init stuff.

touch /var/lock/subsys/local
/etc/trixbox/runonce
/usr/local/sbin/motd.sh > /etc/motd
/sbin/udevstart
/usr/sbin/ztcfg
/usr/sbin/fxotune -s
/usr/sbin/misdn-init start
/usr/sbin/amportal start
~
```

After done that, please reboot the system.

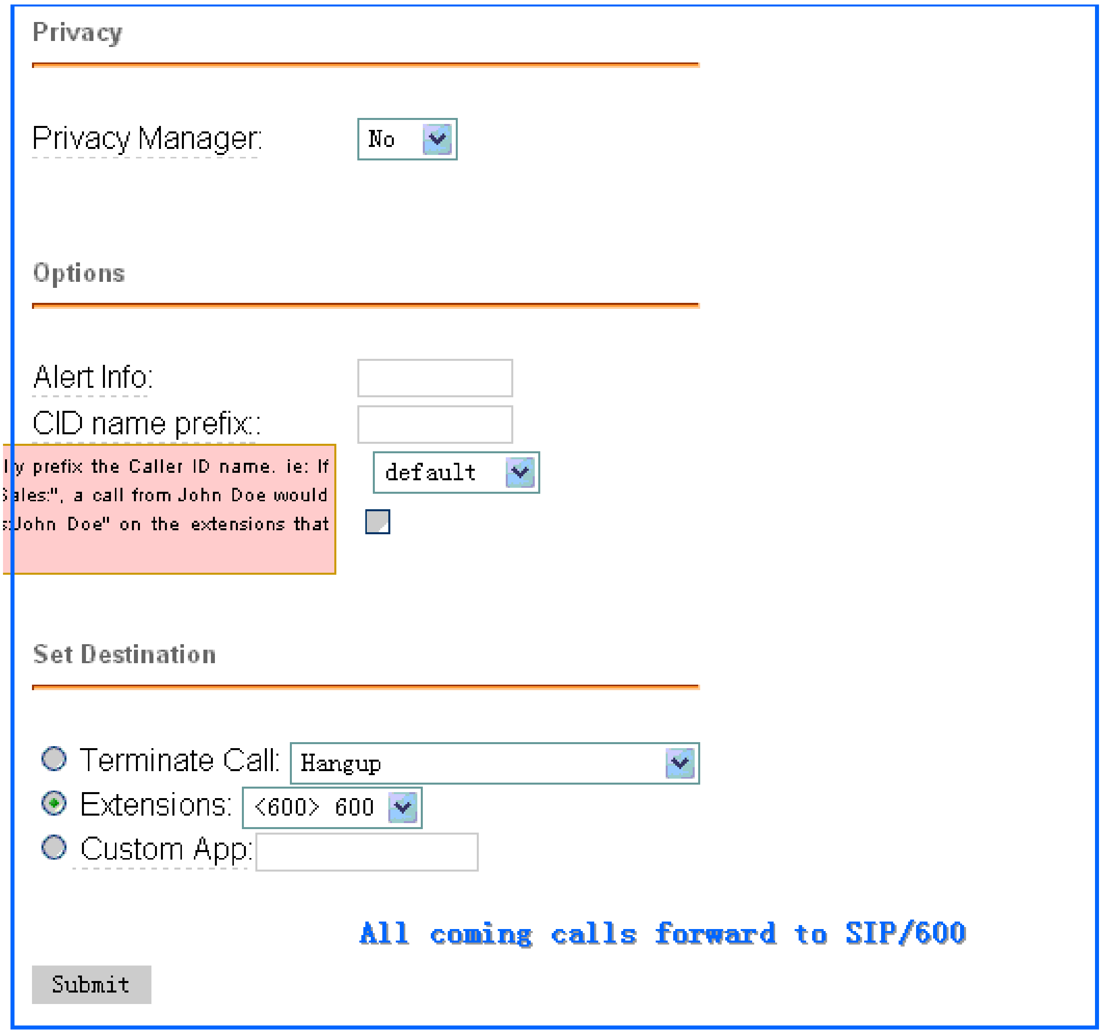
6. Set trunk for outbound and inbound calls. Here, I give an example to make inbound calls and outbound calls. Open the browser and access Freepbx GUI, click trunks->add custom trunks. Here I add mISDN/1/$OUTNUM as a trunk:



Add outbound routes:



Add inbound call



Make calls to test the BRI card:

```
[root@asterisk1 install-misdn-mqueue]# asterisk -r
Asterisk 1.2.26.1 svn rev 79171, Copyright (C) 1999 - 2007 Digium, Inc. and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'show license' for details.
=========================================================================
Connected to Asterisk 1.2.26.1 svn rev 79171 currently running on asterisk1 (pid = 4275)
Verbosity is at least 1
asterisk1*CLI> misdn show stacks
BEGIN STACK_LIST:
 * Port 1 Type TE Prot. PMP L2Link UP L1Link:UP Blocked:0  Debug:0
asterisk1*CLI>
```

```
 == Manager 'admin' logged off from 127.0.0.1
   -- AGI Script dialparties.agi completed, returning 0
   -- Executing Dial("mISDN/1-2", "SIP/600||tr") in new stack
   -- Called 600
   -- SIP/600-092fde00 is ringing
   -- SIP/600-092fde00 answered mISDN/1-2
```

Inbound calls and SIP/600 ring

```
  -- Executing GotoIf("SIP/600-092c3140", "1?outnum:skipoutnum") in new stack
  -- Goto (macro-dialout-trunk,s,31)
  -- Executing Set("SIP/600-092c3140", "the_num=825[redacted]) in new stack
  -- Executing Dial("SIP/600-092c3140", "mISDN/1/82[redacted]|300|") in new stack
  -- Called 1/82[redacted]
  -- mISDN/1-u1 is proceeding passing it to SIP/600-092c3140
  -- mISDN/1-u1 is ringing
  -- mISDN/1-u1 answered SIP/600-092c3140
== Spawn extension (macro-dialout-trunk, s, 32) exited non-zero on 'SIP/600-092c3140' in macro 'dialout-trunk'
== Spawn extension (macro-dialout-trunk, s, 32) exited non-zero on 'SIP/600-092c3140'
```

Make a outbound call from SIP/600

7. Troubleshoot F&Q

- You can not compile the mISDN. If you have problems with compiling mISDN, please make sure the right version of kernel-source has been installed.
- The mISDN can not be found from asterisk console. if the mISDN can start successfully, but mISDN can not be fund in asterisk console, please check these:
    1) Make sure the chan_misdn.so is under /usr/lib/asterisk/modules.
    2) Load chan_misdn.so by adding chan_msidn.so in /etc/asterisk/modules.conf
    3) Recompile asterisk and enable the chan_misdn when you run: *./configure, make menuselect->chan drivers, make, make install*
- Make sure the mISDN is under /usr/sbin, and mISDN has been started before starting asterisk server.
- If you have any problem, please report to OpenVox's forum or mISDN.org

8. Reference

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

www.openvox.com.cn

www.asteriskguru.com

www.voip-info.org

Notes:

Test environment:

1. *trixbox-2.2.12*
2. *OpenVox B100P*
3. *ISDN line*

This instruction is also workable for B200P,B400P and B800P(need patch for B800P)