Integration B400P with mISDN + Trixbox-OpenVox

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Trixbox-OpenVox. can support openvox B400P without any modification. There are few steps to make system work..

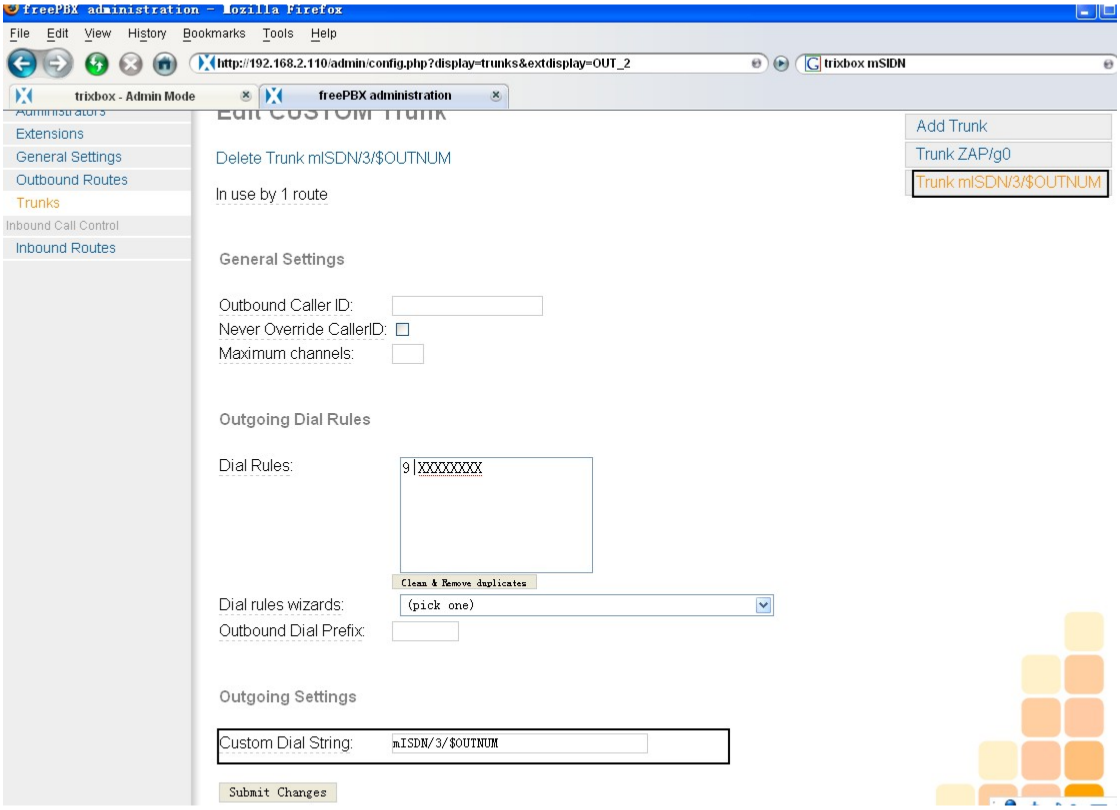
1) Check the driver and hardware：lspci –vvvvvv and dmesg.

```
02:04.0 ISDN controller: Cologne Chip Designs GmbH ISDN network Controller [HFC-4S] (rev 01)
        Subsystem: Cologne Chip Designs GmbH: Unknown device e888
        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-
        Interrupt: pin A routed to IRQ 11
        Region 0: I/O ports at a000 [size=8]
        Region 1: Memory at f7004000 (32-bit, non-prefetchable) [disabled] [size=4K]
        Capabilities: [40] Power Management version 2
                Flags: PMEClk- DSI+ D1+ D2+ AuxCurrent=0mA PME(D0+,D1+,D2+,D3hot+,D3cold-)
                Status: D0 PME-Enable- DSel=0 DScale=0 PME+
```

```
i2c /dev entries driver
CAPI Subsystem Rev 1.1.2.8
capifs: Rev 1.1.2.3
capi20: Rev 1.1.2.7: started up with major 68 (middleware+capifs)
Modular ISDN Stack core version (1_1_5) revision ($Revision: 1.40 $)
mISDNd: kernel daemon started (current:f78b2680)
mISDNd: test event done
ISDN L1 driver version 1.20
ISDN L2 driver version 1.32
mISDN: DSS1 Rev. 1.47
mISDN Capi 2.0 driver file version 1.21
mISDN: HFC-multi driver Rev. 1.68
HFC-multi: card manufacturer: 'Cologne Chip AG' card name: 'HFC-4S OpenVox Card' clock: double
PCI: Found IRQ 11 for device 0000:02:04.0
PCI: Sharing IRQ 11 with 0000:00:1d.2
PCI: Sharing IRQ 11 with 0000:00:1f.1
PCI: Sharing IRQ 11 with 0000:00:1f.2
HFC-4S#1: defined at IOBASE 0xa000 IRQ 11 HZ 1000 leds-type 2
HFC_multi: resetting HFC with chip ID=0xc revision=1
hfcpci_probe: DIPs(0x0) jumpers(0x0)
1 devices registered
mISDN_dsp: Audio DSP  Rev. 1.29 (debug=0x0) EchoCancellor MG2 dtmfthreshold(100)
mISDN_dsp: DSP clocks every 128 samples. This equals 16 jiffies.
NET: Registered protocol family 10
Disabled Privacy Extensions on device c0378f60(lo)
IPv6 over IPv4 tunneling driver
divert: not allocating divert_blk for non-ethernet device sit0
Zapata Telephony Interface Registered on major 196
Zaptel Version: 1.2.19
Zaptel Echo Canceller: KB1
eth0: no IPv6 routers present
Registered PhonicEQ Inc. T100P driver
Registered PhonicEQ Inc. Tormenta 3 driver
Zaptel Transcoder support loaded
Registered tone zone 0 (United States / North America)
Registered tone zone 0 (United States / North America)
```

2) Configure the trunk and mISDN

Open browser and go to Setup->basic->trunk to add custom trunk to add the outbound trunk. The mISDN trunk is set for mISDN. Here, the example is: **mISDN/3/$OUTNUM**. **3** is port 3 as shown in asterisk console. It indicates up. It will be used to make outbound calls..




```
asterisk1*CLI> misdn show config
Misdn General-Config:
 -> Version: chan_misdn-0.3.1-rc34
 -> misdn_init: /etc/misdn-init.conf -> debug: 0
 -> tracefile: /var/log/asterisk/misdn.log -> bridging: no
 -> stop_tone_after_first_digit: yes -> append_digits2exten: yes
 -> dynamic_crypt: no                 -> crypt_prefix: **
 -> crypt_keys: test,muh              -> ntdebugflags: 0
 -> ntdebugfile: /var/log/misdn-nt.log

[PORT 1]
 -> name: group                       -> allowed_bearers: all
 -> far_alerting: no                  -> rxgain: 0
 -> txgain: 0                         -> te_choose_channel: no
 -> pmp_l1_check: no                  -> reject_cause: 16
 -> block_on_alarm: no                -> hdlc: no
 -> context: from-pstn                -> language: en
 -> musicclass: default               -> callerid:
 -> method: standard                  -> dialplan: 0
 -> localdialplan: 0                  -> cpndialplan: 0
 -> nationalprefix: 0                 -> internationalprefix: 00
 -> presentation: -1                  -> screen: -1
 -> always_immediate: no              -> nodialtone: no
 -> immediate: no                     -> senddtmf: yes
 -> hold_allowed: no                  -> early_bconnect: yes
 -> incoming_early_audio: no          -> echocancel: 128
 -> need more infos: no               -> noautorespond on setup: no
```

```
[PORT 3]
-> name: outcall                     -> allowed_bearers: all
-> far_alerting: no                  -> rxgain: 0
-> txgain: 0                         -> te_choose_channel: no
-> pmp_l1_check: no                  -> reject_cause: 16
-> block_on_alarm: no                -> hdlc: no
-> context: from-internal            -> language: en
-> musicclass: default               -> callerid:
-> method: standard                  -> dialplan: 0
-> localdialplan: 0                  -> cpndialplan: 0
-> nationalprefix: 0                 -> internationalprefix: 00
-> presentation: -1                  -> screen: -1
-> always_immediate: no              -> nodialtone: no
-> immediate: no                     -> senddtmf: yes
-> hold_allowed: no                  -> early_bconnect: yes
-> incoming_early_audio: no          -> echocancel: 128
-> need_more_infos: no               -> noautorespond_on_setup: no
-> overlapdial: 0                    -> nttimeout: no
-> bridging: yes                     -> jitterbuffer: 4000
-> jitterbuffer_upper_threshold: 0   -> callgroup:
-> pickupgroup:                      -> msns: *
-> ptp: no

[PORT 4]
-> name: group                       -> allowed_bearers: all
-> far_alerting: no                  -> rxgain: 0
-> txgain: 0                         -> te_choose_channel: no
-> pmp_l1_check: no                  -> reject_cause: 16
-> block_on_alarm: no                -> hdlc: no
-> context: from-pstn                -> language: en
-> musicclass: default               -> callerid:
-> method: standard                  -> dialplan: 0
-> localdialplan: 0                  -> cpndialplan: 0
-> nationalprefix: 0                 -> internationalprefix: 00
-> presentation: -1                  -> screen: -1
-> always_immediate: no              -> nodialtone: no
-> immediate: no                     -> senddtmf: yes
-> hold_allowed: no                  -> early_bconnect: yes
-> incoming_early_audio: no          -> echocancel: 128
-> need_more_infos: no               -> noautorespond_on_setup: no
-> overlapdial: 0                    -> nttimeout: no
-> bridging: yes                     -> jitterbuffer: 4000
-> jitterbuffer_upper_threshold: 0   -> callgroup:
-> pickupgroup:                      -> msns: *
-> ptp: no
asterisk1*CLI>
```

```
[root@asterisk1 asterisk]# vi misdn.conf
[root@asterisk1 asterisk]# asterisk -r
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=========================================================================
Connected to Asterisk 1.2.23 currently running on asterisk1 (pid = 4248)
Verbosity is at least 40
asterisk1*CLI> misdn show stacks
BEGIN STACK_LIST:
  * Port 1 Type TE Prot. PMP L2Link DOWN L1Link:DOWN Blocked:0  Debug:0
  * Port 2 Type TE Prot. PMP L2Link DOWN L1Link:DOWN Blocked:0  Debug:0
  * Port 3 Type TE Prot. PMP L2Link DOWN L1Link:UP Blocked:0  Debug:0
  * Port 4 Type TE Prot. PMP L2Link DOWN L1Link:DOWN Blocked:0  Debug:0
```

```
-- Goto (macro-outbound-callerid,s,3)
-- Executing NoOp("SIP/600-09cfb218", "REALCALLERIDNUM is 600") in new stack
-- Executing GotoIf("SIP/600-09cfb218", "1?normcid") in new stack
-- Goto (macro-outbound-callerid,s,9)
-- Executing Set("SIP/600-09cfb218", "USEROUTCID=") in new stack
-- Executing Set("SIP/600-09cfb218", "EMERGENCYCID=") in new stack
-- Executing Set("SIP/600-09cfb218", "TRUNKOUTCID=") in new stack
-- Executing GotoIf("SIP/600-09cfb218", "1?trunkcid") in new stack
-- Goto (macro-outbound-callerid,s,16)
-- Executing GotoIf("SIP/600-09cfb218", "1?usercid") in new stack
-- Goto (macro-outbound-callerid,s,18)
-- Executing GotoIf("SIP/600-09cfb218", "1?report") in new stack
-- Goto (macro-outbound-callerid,s,22)
-- Executing NoOp("SIP/600-09cfb218", "CallerID set to "600" <600>") in new stack
-- Executing GotoIf("SIP/600-09cfb218", "1?nomax") in new stack
-- Goto (macro-dialout-trunk,s,16)
-- Executing DeadAGI("SIP/600-09cfb218", "fixlocalprefix") in new stack
-- Launched AGI Script /var/lib/asterisk/agi-bin/fixlocalprefix
   >  fixlocalprefix: Using pattern 9|XXXXXXXX
-- AGI Script fixlocalprefix completed, returning 0
-- Executing Set("SIP/600-09cfb218", "OUTNUM=82535362") in new stack
-- Executing Set("SIP/600-09cfb218", "custom=AMP") in new stack
-- Executing GotoIf("SIP/600-09cfb218", "1?customtrunk") in new stack
-- Goto (macro-dialout-trunk,s,22)
-- Executing Set("SIP/600-09cfb218", "pre_num=AMP:mISDN/3/") in new stack
-- Executing Set("SIP/600-09cfb218", "the_num=OUTNUM") in new stack
-- Executing Set("SIP/600-09cfb218", "post_num=") in new stack
-- Executing GotoIf("SIP/600-09cfb218", "1?outnum:skipoutnum") in new stack
-- Goto (macro-dialout-trunk,s,26)
-- Executing Set("SIP/600-09cfb218", "the_num=82535362") in new stack
-- Executing Dial("SIP/600-09cfb218", "mISDN/3/82535362|300|") in new stack
-- Called 3/82535362
-- mISDN/3-u8 is proceeding passing it to SIP/600-09cfb218
-- mISDN/3-u8 is ringing
-- mISDN/3-u8 answered SIP/600-09cfb218
```

Notes:

1) LEDs will turn into red and blink if the drivers are loaded.
2) Misdn show channels commands show nothing.
3) When calls coming, the LED will be turned into green status for very short while
4) If ISDN plugs into the port, the LED will not blink, but in red color.