Adit 600 setup

I found this on one of the forums and used it to configure the ADIT 600. I cleaned up a few typos but can't attribute it back to the original owner as the web site mentioned at the end is no longer on-line. This configuration is not exactly like mine. My channel bank only has 3 FXS station cards and no FXO central office analog dial tone cards.

However, for my test server, I also have a Digium TDM400 card with four daughter boards. Two support FXS stations and two support FXO analog dial tone from the central office.

See my setup notes farther down.

The purpose of this document is to help in the configuration of an Adit 600 Channel bank with Asterisk.

Equipment setup and tested with:

1). (1) Adit 600 (eBay)
2). (1) Dual TDM T1 controller (Included with the Adit)
3). (2) FXO 8 port PSTN cards (Again eBay)
4). (1) FXS 8 port POTS card (eBay)
5). Personal laptop with serial port
6). Male to Female serial cable
7). HyperTerminal

Initial setup

For my testing, I've placed the 2 FXO cards into slot 1 and 2 of the adit. The FXS card is in slot 3.

Connect the serial cable from the Adit to the serial port on your laptop (Or desktop for that matter)

Set up serial connection to speed 9600bps. 8N1

Pressing enter should get you to the interface. My units are new so there was no username or password.

To start fresh with this Adit, I wanted to restore it to factory defaults. I did this by typing **restore defaults** and then **reset**. It will prompt you with the (are you sure) prompt. Press Y.

If you prefer to configure the channel bank via the console interface, type **set local off**, otherwise the dipswitches on each card will configure it. I decide to configure via the console.

The following is my default configuration, after resetting my Adit.

show a:1

SLOT A:
Settings for DS1 1:
Circuit ID: CAC DS1# A:1
Up/Down: UP
Framing: ESF
Line Coding: B8ZS
Line Build Out: DSX-1 EQUALIZATION FOR 0-133 ft. (CSU 0dB)
Loop Code Detection: ON
Loopback: OFF
FDL Type: None

Since my default framing and line coding for Asterisk is identical to the defaults, I had very little to change.

I am setting up 16 FXO channels as ground start and 8 FXS channels as loop start. I discovered via the Asterisk mailing list that if you set your FXO channels as ground start, you would have fewer difficulties with hang-up detection. (NOTE: You must have Ground Start service on the lines you want this on.)

The following needs to be typed to shut down both the primary and secondary T1 interfaces on the TDM controller.

Set a:1 down and **Set a:2 down**

The TDM controller is in slot a, hence a:1 and a:2

Next, we need to disconnect all the channel mapping that may be in place.

Disconnect a

Now the channel types need to be defined. We'll set the all 24 channels as voicewith the following:

Set a:1:1-24 type voice

Set the signaling for the 16 FXO channels to ground start:

Set a:1:1-16 signal gs

And set the 7 FXS channels to loop start:

Set a:1:17-24 signal ls

Next, we set each 8-channel card to the proper signaling. We have cards in slots 1, 2 and 3. FXO,

FXO, FXS. So, 16 channels as ground start with the following:

Set 1:1-8 signal gs
Set 2:1-8 signal gs
And, the last 8 with loop start:
Set 3:1-8 signal ls

Since this Adit can have 2 T1 lines attached, we need a way to map each 8-port card to the proper T1 interface. This is done with the connect command.

Since I only have one T1, I'll only be using a:1 (Slot A, 1st controller). We need to connect 8 channels from the 1st card, 8 channels from the 2nd card and 8 channels from the 3rd card (24 voice). Type the following:

Connect a:1:1-8 1:1-8
Connect a:1:9-16 2:1-8
Connect a:1:17-24 3:1-8

I will also be getting Asterisk's timing from the Adit, so I need to make sure the clock source is internal. Type the following:

Set clock1 internal

And, finally we bring the interface online with:

Set a:1 up

If you have your signaling setup properly between the Asterisk box and the Adit, you should see no lights on the 1st 16 channels and green lights on the last 8 channels. The 1st 16 channels will light up green when you have punched down your PSTN lines to the punch block.

The last 8 will turn yellow when a POTS line is in use.

Here is the status of my 3 cards:

> status 1

FXO Rx AB Tx AB Signal=>T1 Sig T1 TP

1:1 01 01 GS => GS Traffic N
1:2 01 11 GS => GS Traffic N
1:3 01 11 GS => GS Traffic N
1:4 01 11 GS => GS Traffic N
1:5 01 11 GS => GS Traffic N
1:6 01 11 GS => GS Traffic N
1:7 01 11 GS => GS Traffic N

1:8 01 11 GS => GS Traffic N

> status 2

FXO Rx AB Tx AB Signal=>T1 Sig T1 TP

2:1 01 11 GS => GS Traffic N
2:2 01 11 GS => GS Traffic N
2:3 01 11 GS => GS Traffic N
2:4 01 11 GS => GS Traffic N
2:5 01 11 GS => GS Traffic N
2:6 01 11 GS => GS Traffic N
2:7 01 11 GS => GS Traffic N
2:8 01 11 GS => GS Traffic N

> status 3

FXS Rx AB Tx AB Signal=>T1 Sig T1 TP

3:1 01 01 LS => LS Traffic N
3:2 01 01 LS => LS Traffic N
3:3 01 01 LS => LS Traffic N
3:4 01 01 LS => LS Traffic N
3:5 01 01 LS => LS Traffic N
3:6 01 01 LS => LS Traffic N
3:7 01 01 LS => LS Traffic N
3:8 01 01 LS => LS Traffic N

My Zaptel.conf is:

span=1,1,0,esf,b8zs
fxsgs=1-16
fxols=17-24
defaultzone=us
loadzone=us

And, Zapata.conf:

switchtype = national
context = incoming
nsf = megacom
signalling = fxs_gs
channel => 1-16
busydetect = yes
overlapdial = yes

group = 1

switchtype = national
context = analog_phones
nsf = megacom
signalling = fxo_ls
busydetect=yes
channel => 17-24
overlapdial = yes

I have this system currently running in one of our facilities and will be setting up another some time in December of 2006, please contact me at:
support (AT) drdos (DOT) info for comments, questions or corrections.
Posted:Thu 16 of Jun, 2005 (14:08), Last modification by:lytledd, Sun 15 of Oct, 2006 (09:13)

Note: drdos.info is no longer a live web site.

Configuration of Asterisk with ADIT 600 Channel Bank Notes

This is what I did to Asterisk to get the T1 to talk to the ADIT 600 and for the extension.conf for a couple of test extensions (3101, 3102) to confirm proper operation.

Note: I also have a Digium TDM400 card with two FXS and FXO daughter boards but their settings are not shown here.

/etc/dahdi/system.conf

```
# Autogenerated by /usr/sbin/dahdi_genconf on Tue Jan 28 23:17:41 2014
# If you edit this file and execute /usr/sbin/dahdi_genconf again,
# your manual changes will be LOST.
# Dahdi Configuration File
#
# This file is parsed by the Dahdi Configurator, dahdi_cfg
#
# Span 1: WCT1/0 "Digium Wildcard TE110P T1/E1 Card 0"

span=1,0,0,esf,b8zs
# termtype: te
fxols=1-24

echocanceller=mg2,1-24

# Span 2: WCTDM/4 "Wildcard TDM400P REV E/F Board 5" (MASTER)
fxoks=25
echocanceller=mg2,25
fxoks=26
```

echocanceller=mg2,26
fxsks=27
echocanceller=mg2,27
fxsks=28
echocanceller=mg2,28

Global data

loadzone   = us
defaultzone   = us

/etc/asterisk/chan_dahdi.conf

[trunkgroups]

[channels]

usecallerid=yes
callwaiting=yes
usecallingpres=yes
callwaitingcallerid=yes
threewaycalling=yes
transfer=yes
canpark=yes
cancallforward=yes
callreturn=yes
echocancel=yes
echocancelwhenbridged=yes

group=1   ;tdm400 analog card
callgroup=1
pickupgroup=1
ecallerid = yes
hidecallerid = no
callwaiting = yes
usecallingpres = yes
callwaitingcallerid = yes
threewaycalling = yes
transfer = yes
canpark = yes
cancallforward = yes
callreturn = yes
echocancel = yes
echocancelwhenbridged = yes
relaxdtmf = yes
rxgain = 0.0
txgain = 0.0

immediate = no

```
group = 0		;tdm110 single port T1 caed
switchtype = national
context = default
busydetect = yes
echocancel = yes
signalling = fxo_ls
channel => 1-24
overlapdial=yes
```

/etc/asterisk/extensions.conf

```
[channel_bank]

exten => 3101,1,Dial(Dahdi/1/3101,20,rt);
exten => 3101,n,Playback(nbdy-avail-to-take-call) ; No one home
exten => 3101,n,Playback(goodbye)
exten => 3101,n,Hangup
```