Test chan_ss7 with two OpenVox D110P cards

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Disclaimer

SS7 is a very important protocol in telecommunication. Many users use in their business. We know that SS7 environment is easy to get, if we want to test ss7. Thanks, Knielsen, he has published the reference from voip-info.org. Here we give a more details simple test environment to test ss7 with two OpenVox D110P cards. Some steps have to taken in the two servers:

1. Install chan_ss7(we use chan_ss7, not libss7), Asterisk and Zaptel

- Check the support packages, if not installed, please install that.

  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 kernel-devel

rpm -q openssl-devel

rpm -q gnutls-devel

rpm -q gcc

rpm -q gcc-c++

- Download chan_1.0.0, asterisk-1.4.15 and zaptel-1.4.7.1

```
[root@new-host-3 src]# ls
asterisk-1.2.13.tar.gz  asterisk-1.4.15.tar  chan_ss7-1.0.0.tar.gz  redhat          zaptel-1.4.7.1.tar
asterisk-1.4.15         chan_ss7-1.0.0       kernels                zaptel-1.4.7.1
[root@new-host-3 src]# _
```

2. Modify the Makefile in ss7

You have to edit the Makefile in chan_ss7. Make sure the “INCLUDE” points to your zaptel and asterisk source files.

```
# INCLUDE may be overridden to find asterisk and zaptel includes in
# non-standard places.
INCLUDE+=-I../zaptel-1.4.7.1 -I../asterisk-1.4.15/include
```

3. Compile zaptel, asterisk and chan_ss7

- Compile zaptel->*./configure->make->make install*
- Compile Asterisk->*./configure->make->make install*
- Compile chan_ss7->*make->make install*
- Copy the chan_ss7.so to */usr/lib/asterisk/modules*

```
app_system.so                chan_sip.so        format_wav_gsm.so
app_talkdetect.so            chan_skinny.so     format_wav.so
app_test.so                  chan_ss7.so        func_base64.so
app_transfer.so              chan_zap.so        func_callerid.so
```

- Copy ss7.conf to /etc/asterisk

4. Configure ss7.conf , zaptel.conf and extensions.conf

- Configure ss7.conf. Please check the PC hostname, if you are not sure, run command: **hostname** to get your PC hostname.

```
[linkset-siuc]
enabled => yes
use_connect => no
enable_st => yes
hunting_policy => even_mru
subservice => auto
context = ss7
language => en
[link-|1]
linkset => siuc
channels => 1-15,17-31
schannel => 16
firstcic => 1
enabled => yes

[link-|2]
linkset => siuc
channels => 1-15,17-31
schannel => 16
firstcic => 1
enabled => yes

[host-new-host-3]   the Asterisk A with hostname -> new-host-3
enabled => yes
opc => 0x1
dpc =>siuc:0X2
links =>|1:1

[host-new-host-4]   the Asterisk B with hostname -> new-host-4
enabled => yes
opc =>0x2
dpc =>siuc:0X1
links =>|2:1

~
```

- Configure zaptel.conf

```
# Autogenerated by ./genzaptelconf -- do not hand edit
# Zaptel Configuration File
#
# This file is parsed by the Zaptel Configurator, ztcfg
#

# It must be in the module loading order


# Span 1: WCTDM/0 "Wildcard TDM400P REV E/F Board 1"
span=1,1,0,ccs,hdb3
bchan=1-31
```

- Configure extensions.conf. Here, to test the ss7, we create sip account 500 to dial ss7.

```
[from-internal]
exten => 500,1,Dial(ss7/00453377)       ; Call the Asterisk demo
exten => 500,n,hangup          ; Return to the start over message.
```

5. Check connection cables and make call to test.

Before making calls, please check the cable connection. It should be RJ48 connector. If you are not sure that, please visit the website to know how to make RJ48 connector (http://www.chebucto.ns.ca/Chebucto/Technical/Manuals/Max/max6000/gs/cables.htm#17372). It everything is ok. Starting zaptel and asterisk, the LED will in green color. You also can check the ss7 in asterisk console and make sure it is there. If not loaded, please run: load chan_ss7.so to make it be loaded. After dialing 500, the system will forward to ss7. The results are shown in both of host-name-3 and host-name-4.

```
             ss7 block  Set circuits in local maintenance blocked mode
     ss7 cluster start  Start cluster
    ss7 cluster status  Show status of the cluster
      ss7 cluster stop  Stop cluster
        ss7 dump start  Start MTP2 dump to a file
       ss7 dump status  Stop what dumps are running
         ss7 dump stop  Stop a running MTP2 dump
          ss7 linestat  Show line states
         ss7 link down  Stop the MTP2 link(s) [logical-link-no]...
       ss7 link status  Show status of the MTP2 links
           ss7 link up  Start the MTP2 link(s) [logical-link-no]...
          ss7 mtp data  Copy hex encoded string to MTP
             ss7 reset  Reset all circuits
     ss7 show channels  Show channel states
            ss7 status  Show status of ss7
           ss7 unblock  Remove local maintenance blocked mode from circuits
           ss7 version  Show current version of chan_ss7
       stop gracefully  Gracefully shut down Asterisk
              stop now  Shut down Asterisk immediately
   stop when convenient  Shut down Asterisk at empty call volume
```

host-name-3:

```
   -- SS7 hangup 'SS7/siuc/30' CIC=30 Cause=0 (state=3)
 == Spawn extension (from-internal, 500, 1) exited non-zero on 'SIP/500-08ec9310'
   -- Executing [500@from-internal:1] Dial("SIP/500-08ec9310", "ss7/00453377") in new stack
ew-host-3*CLI>      -- Executing [500@from-internal:1] Dial("SIP/500-08ec9310", "ss7/00453377") in new stack
   -- Sent IAM CIC=30   ANI=500 DNI=00453377 RNI=
ew-host-3*CLI>      -- Sent IAM CIC=30   ANI=500 DNI=00453377 RNI=
   -- Called 00453377
ew-host-3*CLI>      -- Called 00453377
```

host-name-4:

```
Connected to Asterisk 1.4.8 currently running on new-host-4 (pid = 2489)
Verbosity is at least 9
    -- Remote UNIX connection
    -- Recv IAM CIC=30   ANI=500 DNI=00453377 RNI= redirect=no/0 complete=1
  == Starting SS7/siuc/30 at ss7,00453377,1 failed so falling back to exten 's'
  == Starting SS7/siuc/30 at ss7,s,1 still failed so falling back to context 'default'
    -- Executing [s@default:1] Wait("SS7/siuc/30", "1") in new stack
    -- Executing [s@default:2] Answer("SS7/siuc/30", "") in new stack
    -- Executing [s@default:3] Set("SS7/siuc/30", "TIMEOUT(digit)=5") in new stack
    -- Digit timeout set to 5
    -- Executing [s@default:4] Set("SS7/siuc/30", "TIMEOUT(response)=10") in new stack
    -- Response timeout set to 10
    -- Executing [s@default:5] BackGround("SS7/siuc/30", "demo-congrats") in new stack
    -- <SS7/siuc/30> Playing 'demo-congrats' (language 'en')
    -- Executing [s@default:6] BackGround("SS7/siuc/30", "demo-instruct") in new stack
    -- <SS7/siuc/30> Playing 'demo-instruct' (language 'en')
new-host-4*CLI>
```

Reference:

http://www.voip-info.org/wiki/index.php?page=Asterisk+ss7+setup

http://www.chebucto.ns.ca/Chebucto/Technical/Manuals/Max/max6000/gs/cables.htm#17372

http://www.sifira.com/chan-ss7/

http://lists.digium.com/pipermail/asterisk-ss7/

www.openvox.com.cn

Test environment:

- ✓ Centos 5.0
- ✓ Zaptel-1.4.7.1
- ✓ Asterisk-1.4.15
- ✓ Chan_ss7-1.0.0
- ✓ Kernel 2.6
- ✓ OpenVox D110P PRI card

Notes: if you have any problems, please report to asterisk-ss7 email list.