Synway AST Series

SynAST Application Platform-Asterisk Installation Manual

Synway Information Engineering Co., Ltd

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Copyright Declaration

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Preface

When you use the Synway AST series boards to set up an Asterisk application system, this file provides the help for software installation, configuration and test. It aims at those people who use the Synway AST series boards in Asterisk for the first time, and takes the use of TEJ-4A/PCI, FXM-16A/PCIe and BRI1611P in Asterisk-1.8.5.0 for example.

Chapter 1 introduces how to install and automatically configure the driver of Synway AST series boards in Asterisk.

Chapter 2 tells how to manually configure the system.

Chapter 3 shows how to test the Synway AST series boards in Asterisk.

Appendix A unfolds the installation of Openr2 in Asterisk.

Appendix B provides answers to some problems that may occur.

Appendix C gives the contact way of technical support and sales department in Synway.

Although Synway has scrupulously checked through this manual, but cannot guarantee the absence of errors and omissions. We sincerely apologize for any consequent inconvenience brought to you and will be very grateful if you kindly give your advice regarding amendments to this book.

Chapter 1 Installation & Automatic Configuration

1.1 Asterisk

For detailed information about Asterisk, visit the official website of Asterisk: http://www.asterisk.org.

Note: The BRI series boards can work normally only in Asterisk-1.8.x.x and above versions.

1.1.1 Preparation

1) **Install the Linux OS. Note: Almost all issued Linux operating systems, such as RED HAT, FC4, DEBAIN, support Asterisk. For more exact information, refer to Asterisk official website.**

2) **Obtain the resource package you need for Asterisk installation. See Table 1-1 below for details.**

Resource Package	Version Recommendation	Address	Description
asterisk-1.8.x.tar.gz	1. 8 or above	http://downloads.digium.com/pub/asterisk/releases/	None
zaptel-1.4.x.tar.gz	1.4.8 or above	http://downloads.digium.com/pub/zaptel/releases/	None
dahdi-linux-complete	2.1.0.4 + 2.1.0.2 or above (BRI series boards require 2.4 or above)	http://downloads.asterisk.org/pub/telephony/dahdi-linux-complete/releases/	None
libss7-1.0.2.tar.gz	1.0.2 or above	http://downloads.digium.com/pub/libss7/releases/	LIBSS7 library for TEJ series
libpri-1.4.11.x.tar.gz	1.4.11 or above	http://downloads.digium.com/pub/libpri/releases/	ISDN library for TEJ, BRI series
openr2-1.3.1.tar.gz	1.3.1 or above	http://code.google.com/p/openr2/downloads/list	SS1 library for TEJ series
astunicall-1.4.18-0.2.tar.gz	1.4.8 or above	http://www.moythreads.com/astunicall/downloads/	SS1 library for TEJ series
SynAST-x.x.x.x.tar.gz	1.1.0.0 or above	http://www.synway.net	None

Table 1-1 Resource Packages for Asterisk Installation

1.1.2 Driver Installation

Step 1: Install the zaptel/dahdi driver and the SynAST driver.

Refer to Section 3.1 of the file *SynAST_UserManual.pdf*.

1.1.3 Asterisk Installation

Step 1: Install the library libpri.

Note: Skip to Step 2 if you do not use the TEJ series boards or ISDN.

```
#cd /opt                                  # enter the directory to libpri
#tar –zxvf libpri-1.4.11.2.tar.gz         # decompress libpri
#cd libpri-1.4.11.2
#make
#make install
```

Step 2: Install the library astunicall.

Notes:

1) ***Skip to Step 3 if you do not use the TEJ series boards or SS1.***
2) ***Skip to Step 3 if you use SS1 in T1 mode on a TEJ series board.***
3) ***The following steps may slightly differ on versions. You can check the file README in the package astunicall for help.***
4) ***The zaptel or dahdi used in driver installation as mentioned in Chapter 3 Driver Installation & Configuration in the file SynAST_UserManual.pdf is just the zaptel or dahdi contained in the astunicall package.***
5) ***To use SS1, follow the SS1 configuration method mentioned in Section 3.2.2 Manual Configuration under Chapter 3 Driver Installation & Configuration in the file SynAST_UserManual.pdf or use the automatic configuration script.***
6) ***See Appendix A for the usage of OpenR2.***

```
#cd /opt                                  # enter the directory to astunicall
#tar –zxvf astunicall-1.4.18-0.2.tar.gz
#cd astunicall-1.4.18-0.2
#cd spandsp-0.0.4                         # install the library spandsp
#./configure    --prefix=/usr
#make
#make install
#cd ../unicall-0.0.5pre1/libsupertone-0.0.2    # install the library libsupertone
#./configure    --prefix=/usr
#make
```

```
#make install
#cd ../ libunicall-0.0.3                    # install libunicall
#./configure –prefix=/usr
#make
#make install
#cd ../ libmfcr2-0.0.3                      # install libmfcr2
#./configure –prefix=/usr
#make
#make install
```

Step 3: Install libss7

Skip to Step4 if you do not use TEJ series boards or LIBSS7

```
#cd /opt
# tar zxvf libss7-1.0.2.tar.gz
#cd libss7
#make
#make install
```

Step 4: Install Asterisk.

Note:

To use SS1 in E1 mode on a TEJ series board, please install the 'asterisk' in the package 'astunicall' mentioned in Step 2.

```
#cd /opt                                    # enter the directory to Asterisk source codes by individual situation
#tar –zxvf asterisk-1.8.5.0.tar.gz          # decompress Asterisk source codes
#cd asterisk-1.8.5.0                        # enter the directory to decompressed Asterisk source codes
#./configure
#make
#make install
#make samples
```

Note: Execute the following command if you install the astunicall package without using the automatic configuration script.

#cp ../unicall.conf.sample /etc/asterisk/unicall.conf *# copy the configuration file unicall.conf*

Skip this operation if you use the automatic configuration script.

1.1.4 Configuration

Refer to the Section 3.1.4 in *SynAST_UserManual.pdf*.

1.1.5 Asterisk Startup

dahdi_cfg -vv

#asterisk –vvvvc

1.1.6 Asterisk Removal

make uninstall-all

Chapter 2 Manual Configuration

2.1 Zaptel/Dahdi Configuration

Refer to Section 3.2.2 Manual Configuration in the document *SynAST_UserManual.pdf*.

2.2 Asterisk Configuration

Modify the configuration file according to Table 2-1 and Table 2-2 below.

Board Config File	TEJ-4A/PCI			FXM-16A/PCIe (top 4 slots: trunk; bottom 4 slots: station)	BRI1611P (one NT module & one TE module)
	E1 Mode				
	ISDN	SS7	SS1		
/etc/asterisk/chan_dahdi.conf	[trunkgroups] [channels] usecallerid=yes hidecallerid=no callwaiting=no threewaycalling=yes transfer=yes rxgain=0.0 txgain=0.0 echocancel=yes echocancelwhenbridged=yes busydetect=yes busycount=7 relaxdtmf=yes				
	Context=from-pstn signalling=pri_cpe switchtype=euroisdn channel=>1-15,17-31 channel=>32-46,48-62 channel=>63-77,79-93 channel=>94-108,110-124 Note: To support channel bank, Context=channelbanktest signalling=fxo_rx channel => 1-15,17-31	signalling = ss7 ss7type = itu ss7_called_nai=dynamic ss7_calling_nai=dynamic networkindicator=national ; port 1 linkset = 1 group = 1 signalling=ss7 ss7type = itu context = default pointcode =2057 adjpointcode = 4114 defaultdpc = 4114 networkindicator = national cicbeginswith = 1 channel => 1-15 cicbeginswith = 17 channel => 17-31 sigchan = 16		Fxo module context=from-pstn signalling=fxs_ks channel=>1-8 ;fxs module context=from-internal signalling=fxo_ks channel=>9-16	group=1 signalling=bri_net_ptmp switchtype=euroisdn channel=>1-2 group=2 signalling=bri_net_ptmp switchtype=euroisdn channel=>4-5 group=3 signalling=bri_cpe_ptmp switchtype=euroisdn channel=>7-8

		; port 2 cicbeginswith = 33 channel => 32-46 cicbeginswith = 49 channel => 48-62 sigchan = 47 ; port 3 cicbeginswith = 65 channel => 63-77 cicbeginswith = 81 channel => 79-93 sigchan =78 ; port 4 cicbeginswith = 97 channel => 94-108 cicbeginswith = 113 channel => 110-124 sigchan =109			group=4 signalling=bri_cpe_ptmp switchtype=euroisdn channel=>10-11
/etc/asterisk/unicall.conf			[channels] language=en usecallerid=yes echocancel=yes rxgain=0 txgain=0 group=1 callgroup=0 pickupgroup=0 amaflags=default accountcode=avantel musiconhold=default context=pstn-incoming loglevel=255 protocolclass=mfcr2 protocolvariant=[see Table 2-3] category=NATIONAL_SUBSCRIBER channel=>1-15,17-31 channel=>32-46,48-62 channel=>63-77,79-93 channel=>94-108,110-124		

Table 2-1 Asterisk/Trixbox Configuration for E1 Mode

<table>
<tr><td rowspan="3">Board Config File</td><td colspan="3">TEJ-4A/PCI</td><td rowspan="3">FXM-16A/PCIe
(top 4 slots: trunk;
bottom 4 slots: station)</td></tr>
<tr><td colspan="3">T1/J1 Mode</td></tr>
<tr><td>ISDN</td><td>SS7</td><td>SS1</td></tr>
<tr><td>/etc/
asterisk/
chan_dahdi.conf</td><td colspan="4">[trunkgroups]
[channels]
usecallerid=yes
hidecallerid=no
callwaiting=no
threewaycalling=yes
transfer=yes
rxgain=0.0
txgain=0.0
echocancel=yes
echocancelwhenbridged=yes
busydetect=yes
busycount=7
relaxdtmf=yes</td></tr>
<tr><td>context=from-pstn
signalling=pri_cpe
switchtype=national
channel=>1-23
channel=>25-47
channel=>49-71
channel=>73-95</td><td>signalling = ss7
ss7type = itu
ss7_called_nai=dynamic
ss7_calling_nai=dynamic
networkindicator=national

; port 1
linkset = 1
group = 1
signalling=ss7
ss7type = itu
context = default
pointcode =2057
adjpointcode = 4114
defaultdpc = 4114
networkindicator = national
cicbeginswith = 1
channel => 1-23
sigchan = 24
; port 2
cicbeginswith = 33
channel => 25-47
sigchan = 48

; port 3
cicbeginswith = 65
channel => 49-71
sigchan =72
; port 4
cicbeginswith = 97
channel => 73-95
sigchan =96</td><td>context=from-pstn
signalling=em_w
switchtype=national
channel=>1-23
channel=>25-47
channel=>49-71
channel=>73-95</td><td>;fxo Module
context=from-pstn
signalling=fxs_ks
channel=>1-8
;fxs Module
context=from-internal
signalling=fxo_ks
channel=>9-16</td></tr>
</table>

Table 2-2 Asterisk/Trixbox Configuration for T1/J1 Mode

Notes:

1) ***Change 'pri_cpe' to 'pri_net' if using the network side in ISDN.***
2) ***In E1+SS1, the value of the field protocolvariant in the configuration file unicall.conf should be set according to the country or the communication operator.***

See Table 2-3 below for details.

Country/Operator	protocolvariant
China	protocolvariant=cn,20,7
Argentina/Telecom E1	protocolvariant=ar,10,4
Brazil/ Embratel	protocolvariant=br,20,4,8
Brasil/ Telecom	protocolvariant=br,20,4
Brasil/ Telefonica	protocolvariant=br,20,20
GVT	protocolvariant=br,20,20
Telemar	protocolvariant=br,20,20
Colombia/ ETB	protocolvariant = ar,20,4
Telefónica /Telecom	protocolvariant = br,10,7,7
Mexico/ Telmex and Avantel	protocolvariant=mx,10,4
Phillippines/ Nextel	protocolvariant=ph,12,18,1

Table 2-3 Value of protocolvariant Field

3) ***Do not configure a channel repeatedly in /etc/asterisk/unicall.conf and /etc/asterisk/chan_dahdi.conf; otherwise, errors occur.***
4) ***Use the following command to correct if the system reports error in chan_unicall.so at the start of Asterisk.***
   chcon -t texrel_shlib_t /usr/lib/asterisk/modules/chan_unicall.so
5) ***According to dialing rules, dahdi channels are used for ISDN or for SS1 in T1 mode on a TEJ board, while unicall channels are used for SS1 in E1 mode on a TEJ board.***
6) ***The values of pointcode, adjpointcode and defaultdpc should be determined by actual circumstances when using libss7.***
7) ***Calculation method of CIC value:***
   CIC (Circuit Identification Code) is a 12-bit value, with the high 7 bits indicating the PCM value and the lower 5 bits corresponding to the starting channel number.
   In T1 mode, the PCM value accumulates with the span number while the starting channel number is always 1. The table below shows the CIC value corresponding to each span:

Span Number	*CIC Value in Binary*	*CIC Value in Decimal*
1st T1	*0000000 00001*	*1*
2nd T1	*0000001 00001*	*33*
3rd T1	*0000010 00001*	*65*
4th T1	*0000011 00001*	*97*

For other span numbers followed, the CIC values are 129, 161, 193, 225, and so on. In T1 mode, the signaling slot is the last timeslot. In E1 mode, the default signaling slot is the 16th Timeslot and therefore, a span should be set with two CICs, whose values are:
cicbeginswith = 1 (0000000 00001)
channel => 1-15
cicbeginswith = 17 (0000000 10001)
channel => 17-31
The CIC is used to indicate the PCM value of the current channel and the starting channel number of a series of continuous voice channels. Its value should be determined by the actual circumstances of the signaling link.

Chapter 3 Test

3.1 Preparation

Use an FXM-16A/PCIe board and a TEJ-4A/PCI board for example. The former 4 modules on the FXM-16A/PCIe board are FXO and the latter 4 are FXS. Meanwhile, configure the TEJ-4A/PCI board with E1+ISDN mode.

Examine the configuration of dahdi:

```
#dahdi_cfg –vv
```

3.2 Test Example

3.2.1 Asterisk Environment

Step 1: Examine the configuration of Asterisk.

```
#asterisk –vvvc                    # start Asterisk
*CLI>dahdi show channels           # check the channel state
```

Step 2: Test Example 1 (FXM-16A/PCIe).

a) Add dialing rules to '/etc/asterisk/extensions.conf':

```
[text]
exten => _300X,1,Dial(dahdi/ 13,50)
exten => _300X,n,playback(hello-world)
exten => _300X,n,Hangup()
```

b) Use Station 15 to dial 3000. Then test the call with Channel 13.

Step 3: Test Example 2 (TEJ-4A/PCI).

a) Register sip to Asterisk.

b) Add dialing rules to '/etc/asterisk/extensions.conf':

```
[text]
exten => _300X,1,Dial(dahdi/ 13,50)
exten => _300X,n,playback(hello-world)
exten => _300X,n,Hangup()
```

c) Use sip to dial 3000 out. Then test the call with Channel 13.

Appendix A Openr2 Installation Under Asterisk

Openr2 is an open-source SS1 protocol library that is more convenient than unicall. As OpenR2 requires high version of Asterisk (version 1.6.2 or above, other versions need patches). Our local test environment is as follows: Asterisk1.6.2.7, dahdi-linux-complete-2.3.0.1+2.3.0, OpenR2-1.3.1, SynAST1.7.0.0 driver.

Installation Steps:

1) Install Dahdi and SynAST. As how to install, refer to relevant files in our driver installation package. The configuration file /etc/dahdi/system.conf is the same as that described in *SynAST_UserManual*.

   Take the 401E board for example. You should set system.conf as follows.

   loadzone=us

   defaultzone=us

   span=1,1,0,cas,hdb3

   cas=1-15,17-31:1111

   dchan=16

   span=2,2,0,cas,hdb3

   cas=32-46,48-62:1111

   dchan=47

   span=3,3,0,cas,hdb3

   cas=63-77,79-93:1111

   dchan=78

   span=4,4,0,cas,hdb3

   cas=94-108,110-124:1111

   dchan=109

2) Install OpenR2.

   #cd /opt

   #tar -zxvf openr2-1.3.1.tar.gz

   #cd openr2-1.3.1

   #./configure --prefix=/usr

   #make

#make install

3) Install Asterisk.

#cd

#cd /opt/

#tar zxvf asterisk-1.6.2.7.tar.gz

#cd asterisk-1.6.2.7

#./configure

#make

#make install
#make samples

Then check if OpenR2 is compiled into the chan_dahdi.so library using the following command.

#ldd channels/chan_dahdi.so | grep openr2

The test machine will show:

libopenr2.so.3 => /usr/lib/libopenr2.so.3 (0x00764000)

If you can see no information on the display, it means OpenR2 is not well installed or OpenR2 is installed after Asterisk. That is, you may not strictly follow the above steps to perform the installation.

4) Configure /etc/asterisk/chan_dahdi.conf.

Add the following fields.

mfcr2_variant=cn ; (country, for example: users in Brazil fill in br)

mfc_max_ani=yes ; (Max amount of ANI to ask for)

mfc_max_dnis=yes ; (Max amount of DNIS to ask for)

protocolclass=mfcr2

protocolvariant=cn,20,7 ; [Read the file Platform(Asterisk)_InstManual_cn, Table 2-3]

category=INTERNATIONAL_PRIORITY_SUBSCRIBER

****

SS1 configuration is more complicated than its installation. You can not use the default settings as you do for ISDN configuration. Therefore, we'd like to sum up the whole process for SS1 configuration here below for your reference.

1) First install SynAST, then OpenR2, then Asterisk. After that, verify OpenR2 components are correctly compiled into Asterisk.

2) Run the command 'astcfg_dahdi asterisk' to restore default settings for all configurations. Take

the TEJ-4A board for example.

Modify the file system.conf to:

```
loadzone=us
defaultzone=us
span=1,1,0,cas,hdb3
cas=1-15,17-31:1111
dchan=16
span=2,2,0,cas,hdb3
cas=32-46,48-62:1111
dchan=47
span=3,3,0,cas,hdb3
cas=63-77,79-93:1111
dchan=78
span=4,4,0,cas,hdb3
cas=94-108,110-124:1111
dchan=109
```

Modify the file chan_dahdi.conf to:

```
[trunkgroups]
[channels]
context=text
usecallerid=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

group=1

callgroup=1

pickupgroup=1

immediate=no

pridialplan=unknown

prilocaldialplan=unknown

mfcr2_variant=cn ; (country, for example: users in Brazil fill in br)

mfc_max_ani=yes ; (Max amount of ANI to ask for)

mfc_max_dnis=yes ; (Max amount of DNIS to ask for)

protocolclass=mfcr2

protocolvariant=cn,20,7 ; [Read the file Platform(Asterisk)_InstManual_cn, Table 2-3]

category=INTERNATIONAL_PRIORITY_SUBSCRIBER

channel=>1-15,17-31

channel=>32-46,48-62

channel=>63-77,79-93

channel=>94-108,110-124

Modify the dialing plans according to your actual line connection.

Run dahdi_cfg –vv to activate all configurations and then run asterisk –vvvvc.

Appendix B FAQ

Q1: Why does the NT module fail to be used in the mode of bri_net_ptmp in Asteriak-1.8.7.0 and above versions?

This is because the macro definition HAVE_PRI_CALL_HOLD in Asterisk has been cancelled by default in Asterisk-1.8.7.0 and above versions, which makes the NT module fail to connect with BRI phones while its signal type is set to bri_net_ptmp.

To solve this problem, modify as follows: add the sentence '#define HAVE_PRI_CALL_HOLD 1' to the file 'include/asterisk/autoconfig.h' under the Asterisk source code directory, then execute the command '#make' and '#make install'.

Note: './configure' command cannot be used here, otherwise the file 'autoconfig.h' will be reset to default settings.

Appendix C Technical/Sales Support

Thank you for choosing Synway. Please contact us should you have any inquiry regarding our products. We shall do our best to help you. However, our technicians and salesmen are mainly responsible for maintaining our boards and providing relative technical support. If there are problems about Asterisk, please keep touch with Digium Inc. for help.

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