Synway AST Series

User Manual

Version 1.0.0.0

Synway Information Engineering Co., Ltd

www.sanhuid.com

Contents

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Revision History

Version	Date	Comments
Version 1.0.0.0	2009.02	Initial publication.

Note: Only major revisions to this manual itself recorded herein.

Preface

Thank you for choosing the Synway AST series boards which are designed to fully support open source IP PBX platform, such as Asterisk , Trixbox , FreeSwitch and so on. Synway AST series includes three subseries.

Subseries	Board Model	Description
Synway FXM	FXM-8A/PCI	Use of modular structure and analog lines
	FXM-16A/PCIe	
Synway TEJ	TEJ-1A/PCI	Use of digital trunks (E1/T1/J1)
	TEJ-2A/PCI	
	TEJ-4A/PCI	
Synway CDC	CDC-1522A/PCI	CODEC board
	CDC-2522A/PCI	
	CDC-3522A/PCI	

The Synway FXM series board is a perfect substitute for the analog voice cards such as AEX and TDM from Digium Inc. The Synway TEJ series board can perfectly substitute for the digital trunk cards such as TE from Digium Inc. The Synway CDC series board can perfectly substitute for the TC400B CODEC card from Digium Inc.

Since the FXM and TEJ boards integrate the capability of echo cancellation, there is no need to have the supports of other extra modules and devices. Compared with other products in Synway AST series, they have higher integration and bring an obviously better effect of echo cancellation than other subseries in AST. So there is no need to purchase an extra module of VPMADT032 (to enable echo cancellation) from Digium Inc.

The Synway AST driver program, as a link between the board hardware and open source IP PBX platform, is only used to set up the open source IP PBX platform, but not applicable to secondary development. This manual, as the help file for hardware installation, software installation, test and configuration of the Synway AST series, aims at those installation and maintenance technicians as well as the salesmen who are using the AST boards to set up open source IP PBX application systems.

This document consists of the following chapters.

Chapter 1 introduces the Asterisk , Trixbox and FreeSwitch application model and scenario set up by the Synway AST series boards .

Chapter 2 introduces how to install the hardware of Synway AST series boards.

Chapter 3 introduces how to install the driver of Synway AST series boards in the Asterisk, Trixbox and FreeSwitch application environment.

Chapter 4 introduces how to manually install the driver of Synway AST series boards.

Chapter 5 introduces how to manually configure the system.

Chapter 6 introduces how to test the Synway boards in the Asterisk , Trixbox and FreeSwitch application environment.

Appendix A provide answers to some problems that may occur when using the Synway AST series boards.

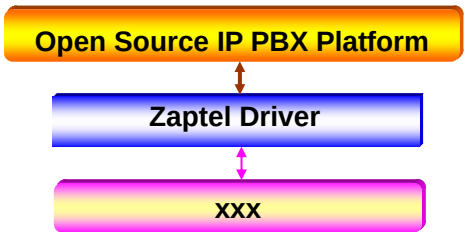
Appendix B shows the contact way of technical support and sales department in Synway.

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 Overview

The driver for the Synway AST series boards (hereinafter referred to as ‘AST), which is compatible and must be used with the Zaptel driver, can smoothly and seamlessly support open source IP PBX platform.

When using the AST board to establish open source IP PBX platform, all that you should do is configure and operate the open source IP PBX platform, without secondary development by using AST. See Figure 1-1 below for the set-up of the open source IP PBX application system with the AST board.

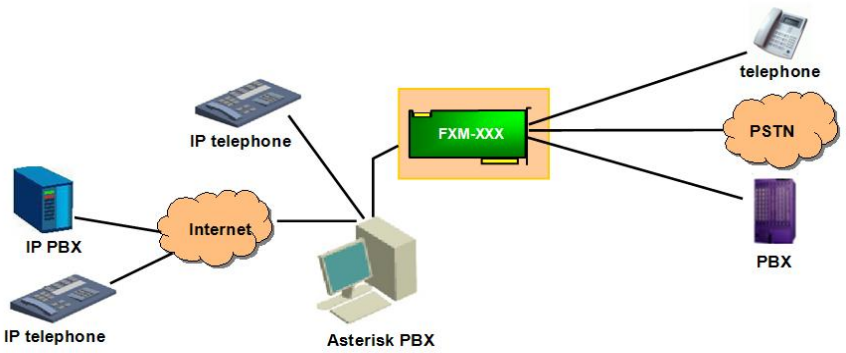


xxx: represents the existing and future board models in the AST series.

Figure 1-1 FXM Driver Set-up

1.1 FXM

The Synway FXM series boards (hereinafter referred to as ‘FXM’) are analog voice boards newly developed for open source IP PBX system. FXM are high-performance and cost-effective , which support FXO(Foreign Exchange Office), FXS(Foreign Exchange Station) and FXC(composite module) and allow various configurations by using these three different modules to provide all features and functionalities of common voice boards. The FXM boards improve I/O speed by using DMA technique for data reading and writing on the basis of PCI bus, resulting in reduced CPU usage and increased board density per server. Figure 1-2 below is typical application models with FXM.



xxx: represents the existing and future board models in the FXM series.

Figure 1-2 Application Model

1.2 TEJ

The Synway TEJ series boards (hereinafter referred to as 'TEJ') are digital trunks boards newly developed for open source IP PBX system. The TEJ are high-performance, cost-effective digital telephony interface boards and support E1, T1 and J1 environments. The environments are selectable on a per-board basis via software configuration. The TEJ boards improve I/O speed by using DMA technique for data reading and writing on the basis of PCI bus, resulting in reduced CPU usage and increased board density per server.

The excellent combination of the TEJ driver and the Asterisk platform brings the following features to the TEJ series boards.

Supported Data Modes: Cisco HDLC, HDLC, PPP, Multi-link PPP, Frame Relay.

Supported Voice Modes:

- PRI CPE and PRI NET
  - NI1
  - NI2
  - EuroISDN
  - 4ESS(AT&T)
  - 5ESS(Lucent)
  - DMS100
- E&M
  - Wink
  - Feature Group B
  - Featrue Group D
- FXO and FXS
  - Ground Start
  - Loop Start
  - Loop Start with Disconnect Detect

The TEJ series connect Asterisk Server with PSTN, Channel Bank or PBX via T1, E1 or J1 interface. The boards and the corresponding drivers can be used to establish a professional telephony network environment. Figure 1-3 and Figure 1-4 below are typical application models with TEJ.

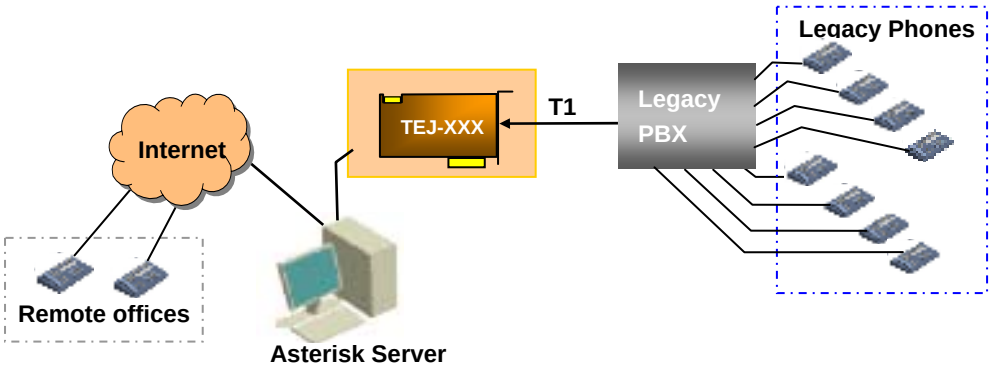
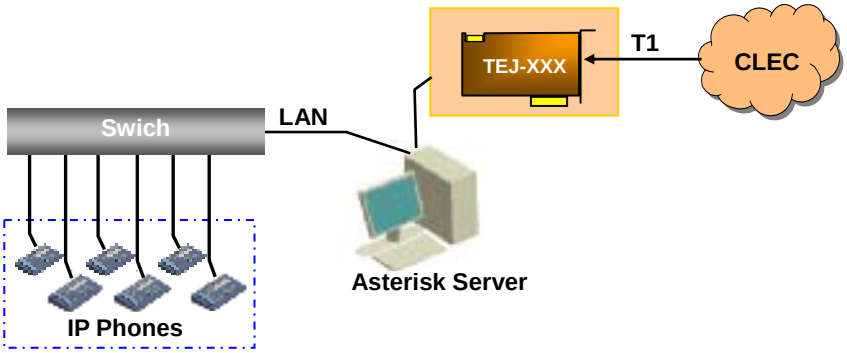


Figure 1-3 Application Model I: Traditional Telephony System



Note: XXX in Figure 1-3 and Figure 1-4 represents the existing and future board models in the TEJ series.

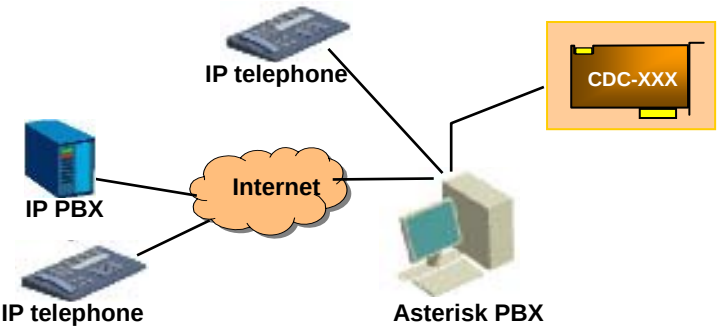
Figure 1-4 Application Model II: VoIP Telephony System

1.3 CDC

The Synway CDC series boards (hereinafter referred to as 'CDC) are CODEC boards newly developed for open source IP PBX system. CDC are multi-channel codec boards with PCI 2.2 bus, supporting the hardware-based transformations between voice codecs. With the burst data transmission rate up to 132 MB/s and the PNP (plug and play) feature which eliminates the need for jumper leads, this series can be used on 3.3V/5V PCI slot and PCI-X slot.

Asterisk, in software and with Digium G.729a licensing, is capable of transforming the G.729a codec into other codecs for the purposes of call origination or termination, bridging disparate calls, or VoIP to TDM connectivity. These transformations in software are very expensive, in terms of MIPS, and require a substantial amount of CPU time to accomplish. The Synway CDC series uses the PCI-based DMA technique for data reading and writing, minimizing the CPU cost. It is fully compatible with the API functions of Asterisk and supports the transformations among G.711 A-Law, μ-Law and G.729A, which ideally cover the shortage of Asterisk. Besides, the CDC board has many DSPs on it to provide powerful voice processing, which brings a nearly real-time voice effect to users.

Figure 1-5 below is typical application models with CDC.



xxx: represents the existing and future board models in the CDC series.

Figure 1-5 Application Example

Chapter 2 Hardware Installation

Refer to the board model to find the relevant hardware manual. After hardware completely installed, you can follow the chapter below to continue the software installation. The reference hardware manual are under the directory '/doc' of SynAST-Ver.x.x.x.x resource package. Table 2-1 shows the corresponding relationship between the board model and the reference hardware manual.

Subseries	Board Model	Reference Hardware Manual
FXM	FXM-8A/PCI	AST FXM_hardwareManual
	FXM-16A/PCIe	
TEJ	TEJ-1A/PCI	AST TEJ_hardwareManual
	TEJ-2A/PCI	
	TEJ-4A/PCI	
CDC	CDC-1522A/PCI	AST CDC_hardwareManual
	CDC-2522A/PCI	
	CDC-3522A/PCI	

Table 2-1 corresponding relationship table between board model and reference hardware manual

Chapter 3 Software Installation

Synway AST series boards support open source IP PBX platform, such as Asterisk, Trixbox, FreeSwitch and so on. How to construct these software platform are separately introduced by the following sections.

3.1 Asterisk

Refer to Asterisk official website http://www.asterisk.org to obtain the detail information about Asterisk.

3.1.1 Preparation

1) install the Linux OS. Note: in general, issued Linux OS are support the Asterisk, such as RED HAT, FC4, DEBAIN, etc. the exact information please refer to Asterisk official website.

2) Obtain the needed resource package for Asterisk installation. Table 3-1 show the detail information about the resource package.

Resource Name	Required version	address	Description
asterisk-1.4.x.tar.gz	1.4.18 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
libpri-1.4.x.tar.gz	1.4.7 or above	http://downloads.digium.com/pub/libpri/releases/	TEJ series board ISDN library.
astunicall-1.4.18-0.2.tar.gz	1.4.8 or above	http://www.moythreads.com/astunicall/downloads/	TEJ series board SS1 library.
Synast-x.x.x.x.tar.gz	1.0.0.0 or above	http://www.sanhuid.com	None

Table 3-1 the detail information table of resource package for Asterisk installation

3.1.2 Driver Installation

Step1： Install the zaptel driver and the SynAST driver

```
#cd /opt                          # enter into the source code directory
#tar –zxvf SynAST-1.0.0.0.tar.gz  # decompress SynAST driver
#tar –zxvf zaptel-1.4.8.tar.gz    # decompress zaptel driver
#cd SynAST-1.0.0.0                # enter into the directory of SynAST driver
#./Setup install                  # install the driver
Would you like to install SynAST AST package now? y
                                  # input 'y' to confirm the driver installation operation
```

Please enter working zaptel directory [q](exit install) : /opt/zaptel-1.4.8

 # input the directory of zaptel code

SELECT TEJ MODE [t, e, j] :e *# chose the mode of TEJ board, if you don't execute this operation, the mode of TEJ board default E1.*

Note: users also can manually install the driver, please refer to 'Manual Driver Installation'。

3.1.3 Asterisk Installation

Step1： Install libpri Library

Note: the user, who do not use TEJ boards or ISDN signal, can immediately jump to Step2.

#cd /opt *# enter into the directory of the libpri library*

#tar –zxvf libpri-1.4.7.tar.gz *# decompress the libpri*

#cd libpri-1.4.7

#make

#make install

Step2： Install astunicall Library

Note:

1) the user, who do not use TEJ boards or SS1 signal, can immediately jump to Step3.

2) The following steps are correlated with Specific version. Please refer to the file README in astunicall package for help.

#cd /opt *# enter into the directory of astunicall library*

#tar –zxvf astunicall-1.4.18-0.2.tar.gz

#cd astunicall-1.4.18-0.2

#cd spandsp-0.0.4 *# install the spandsp library*

#./configure --prefix=/usr

#make

#make install

#cd ../unicall-0.0.5pre1/libsupertone-0.0.2 *# install the libsupertone library*

#./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
```

Step3： Install Asterisk

```
#cd /opt                                    # enter into the Asterisk source code directory based on the actual situation.
#tar –zxvf Asterisk-1.4.18.tar.gz           # decompress the Asterisk source code package
#cd Asterisk-1.4.18                         # enter into the decompress library
#./configure
#make
#make install
#make samples
```

Note: if the user install the astunicall package, please execute the following command.

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

3.1.4 Configuration

Note: only the 'Manual System Configuration' can be consulted by the user who using SS1. Otherwise, user can use the method below or 'Manual System Configuration' to complete the configuration based on actual requirements.

```
#./astcfg_zaptel    asterisk
```

3.1.5 Startup asterisk

```
#ztcfg -vv
#asterisk –vvvvc
```

3.2 Trixbox

Refer to Trixbox official website http://www.trixbox.org to obtain the detail information about Trixbox.

3.2.1 Preparation

Obtain the needed resource package for Trixbox installation. Table 3-2 show the detail information about the resource package.

Resource Name	Required version	address	Description
zaptel-1.4.x.tar.gz	Downloads corresponding zaptel driver in trixbox	http://downloads.digium.com /pub/zaptel/releases/	Using the command "#rpm –q zaptel" to examine the version of zaptel.
Synast-x.x.x.x.tar.gz	1.0.0.0 or above	http://www.sanhuid.com	None

Table 3-2 the detail information table of resource package for Trixbox installation

3.2.2 Trixbox System Installation

Step1： Install trixbox

Downloads the trixbox ISO file from the web and makes it as a Trixbox System installation disk, then install this system in you computer.

Note: when install the trixbox system, the trixbox will format the whole hard disk.

Step2： Install compiling environment

```
#yum install gcc                          # install gcc compiler
#yum install kernel-devel-`uname –r`      # install kernel source code tree
```

3.2.3 driver Installation

Step1： Close the zaptel service

```
#service asterisk stop                    # close the asterisk service
#service zaptel stop                      # close the zapte service
```

Step2： Install zaptel driver and SynAST driver

```
#cd /opt                                  # enter into the source code directory
#tar –zxvf SynAST-1.0.0.0.tar.gz          # decompress synast driver
#tar –zxvf zaptel-1.4.8.tar.gz            # decompress zaptel driver
#cd SynAST-1.0.0.0                        # enter into the directory of synast driver
#./Setup install                          # install the driver
Would you like to install SynAST AST package now? y
                                          # input 'y' to confirm the driver installation operation
```

```
Please enter working zaptel directory [q](exit install) : /opt/zaptel-1.4.8
                                        # input the zaptel source code directory
SELECT TEJ MODE [t, e, j] :e            # chose the mode of TEJ board, if you don't execute this
                                          operation, the mode of TEJ board default E1.
```

Note: users also can manually install the driver, please refer to 'Manual Driver Installation'。

3.2.4 Configuration

Note: user can use the method below or 'Manual System Configuration' to complete the configuration based on actual requirements.

```
#./astcfg_zaptel    trixbox
```

3.2.5 Startup Trixbox

```
#service zaptel start          # start the zaptel service
#service asterisk start        #start the asterisk service
```

3.3 FreeSwitch

Refer to FreeSwitch official website http://www.freeswitch.org to obtain the detail information about FreeSwitch.

3.3.1 Preparation

Obtain the needed resource package for FreeSwitch installation. Table 3-3 show the detail information about the resource package.

Resource Name	Required version	address	Description
zaptel-1.4.x.tar.gz	1.4.x	http://downloads.digium.com/pub/zaptel/releases/	using the command "#rpm –q zaptel" to examine the version of zaptel.
freeswitch	downloads the latest version from the svn	http://svn.freeswitch.org/svn/freeswitch/trunk	using the command "#svn co [address] freeswitch " to download freeswitch
Synast-x.x.x.x.tar.gz	1.0.0.0 or above	http://www.sanhuid.com	None

Table 3-3 the detail information table of resource package for FreeSwitch installation

3.3.2 Driver Installation

Step1： Install zaptel driver and SynAST driver

```
#cd /opt                          # enter into the source code directory
#tar –zxvf SynAST-1.0.0.0.tar.gz  # decompress synast driver
#tar –zxvf zaptel-1.4.8.tar.gz    # decompress zaptel driver
#cd SynAST-1.0.0.0                # enter into the directory of synast driver
```

#./Setup install *# install driver*

Would you like to install SynAST AST package now? y

 # input 'y' to confirm the driver installation operation

Please enter working zaptel directory [q](exit install) : /opt/zaptel-1.4.8

 # input the zaptel resource code directory

SELECT TEJ MODE [t, e, j] :e *# chose the mode of TEJ board, if you don't execute this operation, the mode of TEJ board default E1.*

Note: users also can manually install the driver, please refer to 'Manual Driver Installation'。

3.3.3 FreeSwitch Installation

Step1： Install freeswitch

#cd /usr/src/freeswitch

../bootstrap.sh

＃vim /usr/src/freeswitch/modules.conf

finds the line with "../../libs/openzap/mod_openzap". Then delete the '#' in the beginning of this line.

./configure

#make

#make install

#make samples

3.3.4 Configuration

Note: user can use the method below or 'Manual System Configuration' to complete the configuration based on actual requirements.

#./astcfg_zaptel freeswitch

3.3.5 Startup FreeSwitch

Step1： Modify the configuration file of model

modifies the file modules.conf.xml, which makes the freeswitch loading the openzap model when freeswitch starting.

#cd /usr/local/freeswitch/conf/autoload_configs

＃vim modules.conf.xml

finds the line with mod_openzap, then modifies this line as <load module="mod_openzap"/>.

Step2： Startup FreeSwitch

#cd /usr/local/freeswitch/bin

#./freeswitch

Chapter 4 Manual Driver Installation

Step1： Install the zaptel driver

#cd /opt *# enter into the zaptel source code package directory*

#tar –zxvf zaptel-1.4.8.tar.gz *# decompress the source code*

#cd zaptel-1.4.8 *# enter into the zaptel source code directory*

#./configure

#make *# compile the driver*

#make install *# install the driver*

#modprobe zaptel *# load the driver*

Step2： Install the Synway AST driver

#cd /opt *# enter into the AST source code package directory*

#tar –zxvf SynAST-1.0.0.0.tar.gz *# decompress the source code*

#cp –rf /opt/SynAST-1.0.0.0/src/fxm/fxm8apci /opt/zaptel-1.4.8 *# copy the fxm8apci driver to the zaptel directory. If user uses other board, not FXM-8A/PCI, user should refer to the Table 4-1 to find the corresponding driver file and uses it to substitute "fxm/fxm8apci" in this command based on the actual situation . if the version of zaptel is above 1.4.11, the corresponding driver file should copy to "/opt/zaptel-[version]/kernel"*

#cd /opt/zaptel-1.4.8/fxm8apci *# enter into the AST source code directory which already copy into the zaptel directory in the upper operation. if the version of zaptel is above 1.4.11, please execute this command:" #cd /opt/zaptel-[version]/kernel/fxm8apci ".*

#make *# compile the dirver*

#insmod fxm8apci.ko *# load the driver*

Note: if the user uses the other board, not FXM-8A/PCI, user should refer to the Table 4-1 to find the corresponding driver file and uses it to substitute the "fxm" and "fxm8apci" ,which appeared in the upper example ,based on the actual situation .

Subseries	Boards Model	Source Code	Driver File	Remark
FXM	FXM-8A/PCI	fxm8apci	fxm8apci.ko	The example as upper.
FXM	FXM-16A/PCIe	fxm16apcie	fxm16apcie.ko	Using the other driver file shows in Table 4-1 instead of the "fxm8apci.ko" in the upper example based on the
TEJ	TEJ-1A/PCI TEJ-2A/PCI TEJ-4A/PCI	tejxapci	tejxapci.ko	

CDC	CDC-1522A/PCI CDC-2522A/PCI CDC-3522A/PCI	cdcx522apci	cdcx522apci.ko	specific Synway AST board model.

Table 4-1 corresponding relationship table between the board and driver

Chapter 5 Manual System Configuration

Below taking the FXM-16A/PCIe board and TEJ-4A/PCI board as an example to illuminate how to configure the system.

5.1 Configure zaptel

Step1： Modify the configuration file based on the Table 5-1 below.

board / configuration file	TEJ-4A/PCI				FXM-16A/PCIe (The former 4 models are FXO . the later 4 models are FXS)
	E1		T1/J1		
	ISDN	SS1	ISDN	SS1	
	loadzone=us defaultzone=us				

/etc/ zaptel.conf	span=1,1,0,ccs,hdb3
	bchan=1-15
	dchan=16
	bchan=17-31
	span=2,1,0,ccs,hdb3
	bchan=32-46
	dchan=47
	bchan=48-62
	span=3,1,0,ccs,hdb3
	bchan=63-77
	dchan=78
	bchan=79-93
	span=4,1,0,ccs,hdb3
	bchan=94-108
	dchan=109
	bchan=110-124
		span=1,1,0,cas,hdb3
		cas=1-15,17-31:1111
		span=2,1,0,cas,hdb3
		cas=32-46,48-62:1111
		span=3,1,0,cas,hdb3
		cas=63-77,79-93:1111
		span=4,1,0,cas,hdb3
		cas=94-108,110-124:1111
			span=1,1,0,esf,b8zs
			bchan=1-23
			dchan=24
			span=2,1,0,esf,b8zs
			bchan=25-47
			dchan=48
			span=3,1,0,esf,b8zs
			bchan=49-71
			dchan=72
			span=4,1,0,esf,b8zs
			bchan=73-95
			dchan=96
				span=1,1,0,esf,b8zs
				e&m=1-23
				span=2,1,0,esf,b8zs
				e&m=25-47
				span=3,1,0,esf,b8zs
				e&m=49-71
				span=4,1,0,esf,b8zs
				e&m=73-95
					#fxo model
					fxsks=1
					fxsks=2
					fxsks=3
					fxsks=4
					fxsks=5
					fxsks=6
					fxsks=7
					fxsks=8
					#fxs model
					fxoks=9
					fxoks=10
					fxoks=11
					fxoks=12
					fxoks=13
					fxoks=14
					fxoks=15
					fxoks=16

Table 5-1 zaptel configuration table

Step2： Query the result of configuration

#ztcfg –vv

Note: If there are more then one AST boards, the sequence of board's channel number was decided by the board's driver loaded order. Which board's driver firstly load, whose channel number is smaller then the other boards. The board's channel number is increase. If the board styles are same, the sequence of board's channel number was decided by the distance between the PCI/PCIe plug and CUP. Which board's PCI/PCIe plug is nearest the CPU, whose channel number is smaller than other boards and The channel number is

increase. The span number can be examined by executing the command:" #cat /proc/zaptel/[span number] ".

5.2 Configurate Asterisk/Trixbox

Modify the configuration file based on the Table 5-2 below

<table>
<tr><th>board
configuration file</th><th colspan="4">TEJ-4A/PCI</th><th>FXM-16A/PCIe (The former 4 models are FXO . the later 4 models are FXS)</th></tr>
<tr><td rowspan="3">/etc/
asterisk/
zapata.conf</td><td colspan="2">E1</td><td colspan="2">T1/J1</td><td rowspan="2"></td></tr>
<tr><td>ISDN</td><td>SS1</td><td>ISDN</td><td>SS1</td></tr>
<tr><td colspan="5">[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_cp
switchtype=euroisdn
channel=>1-15,17-31
channel=>32-46,48-62
channel=>63-77,79-93
channel=>94-108,110-124</td><td></td><td>context=from-pstn
signalling=pri_cpe
switchtype=national
channel=>1-23
channel=>25-47
channel=>49-71
channel=>73-95</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 model
context=from-pstn
signalling=fxs_ks
channel=>1-8
;fxs model
context=from-internal
signalling=fxo_ks
channel=>9-16</td></tr>
<tr><td>/etc/
asterisk/
unicall.conf</td><td></td><td>[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=mx,0,4,7
category=
NATIONAL_SUBSCRIBER
channel=>1-15,17-31
channel=>32-46,48-62</td><td></td><td></td><td></td></tr>
</table>

		channel=>63-77,79-93 channel=>94-108,110-124			

Table 5-2 asterisk/trixbox configuration table

Note:

1) ***if you want to use network-end in ISDN model , you should use pri_net instead of pri_cpe in Table 5-2 .***
2) ***under the model of E1 and SS1, the value of protocolvariant in file unicall.conf is set by different Country and Operation Business. Refer to the Table 5-3 for detail information.***

Country/ Operation Business	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 5-3 protocolvariant corresponding table

3) ***the channel configuration of /etc/asterisk/unicall.conf should not repeat with the channel configuration of /etc/asterisk/zapata.conf, and otherwise it may cause error.***
4) ***if there returns a error "chan_unicall.so is wrong" at the start of Asterisk, you can use the command below to modify it:***
   chcon -t texrel_shlib_t /usr/lib/asterisk/modules/chan_unicall.so

5.3 Configure FreeSwitch

Modify the configuration file based on the Table 5-4 below.

board / configuration file	TEJ-4A/PCI		FXM-16A/PCIe (The former 4 models are FXO . the later 4 models are FXS)
	E1	T1/J1	
	ISDN	ISDN	
/usr/ local/ freeswitch/ conf/ openzap.conf	[span zt] name =>OpenZAP1 number=>1 trunk_type=>e1 b-channel=>1->15 d-channel=>16 b-channel=>17->31 [span zt] name =>OpenZAP2 number=>2	[span zt] name =>OpenZAP1 number=>1 trunk_type=>t1 b-channel=>1->23 d-channel=>24 [span zt] name =>OpenZAP2 number=>2 trunk_type=>t1	#fxo port [span zt] name => OpenZAP-FX01 number => 1 fxo-channel => 1 [span zt] name => OpenZAP-FX02 number => 2 fxo-channel => 2 [span zt] name => OpenZAP-FX03

	trunk_type=>e1 b-channel=>32->46 d-channel=>47 b-channel=>48->62 [span zt] name =>OpenZAP3 number=>3 trunk_type=>e1 b-channel=>63->77 d-channel=>78 b-channel=>79->93 [span zt] name =>OpenZAP4 number=>4 trunk_type=>e1 b-channel=>94->108 d-channel=>109 b-channel=>110->124	b-channel=>25->47 d-channel=>48 [span zt] name =>OpenZAP3 number=>3 trunk_type=>t1 b-channel=>49->71 d-channel=>72 [span zt] name =>OpenZAP4 number=>4 trunk_type=>t1 b-channel=>73->95 d-channel=>96	number => 3 fxo-channel => 3 [span zt] name => OpenZAP-FX04 number => 5 fxo-channel => 4 [span zt] name => OpenZAP-FX05 number => 5 fxo-channel => 5 [span zt] name => OpenZAP-FX06 number => 6 fxo-channel => 6 [span zt] name => OpenZAP-FX07 number => 7 fxo-channel => 7 [span zt] name => OpenZAP-FX08 number => 8 fxo-channel => 8 #fxs port [span zt] name => OpenZAP-FXS9 number => 9 fxs-channel => 9 [span zt] name => OpenZAP-FXS10 number => 10 fxs-channel => 10 [span zt] name => OpenZAP-FXS11 number => 11 fxs-channel =>11 [span zt] name => OpenZAP-FXS12 number => 12 fxs-channel =>12 [span zt] name => OpenZAP-FXS13 number => 13 fxs-channel =>13 [span zt] name => OpenZAP-FXS14 number => 14 fxs-channel =>14 [span zt] name => OpenZAP-FXS15 number => 15 fxs-channel =>15 [span zt] name => OpenZAP-FXS16 number => 16 fxs-channel =>16
/usr/ local/ freeswitch/	***added the following content under the </settings>*** <pri_spans> 		***added the following content under the </settings>*** <analog_spans>

conf/ autoload_configs/ openzap.conf.xml	<param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> </pri_spans>	<param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/>

```
        <param name="digit-timeout" value="2000"/>
        <param name="max-digits" value="11"/>
        <param name="dialplan" value="XML"/>
         <param name="context" value="default"/>
  <param name="enable-callerid" value="true"/>
 </span>
<span id="9">
        <param name="tonegroup" value="us"/>
        <param name="digit-timeout" value="2000"/>
        <param name="max-digits" value="11"/>
        <param name="dialplan" value="XML"/>
         <param name="context" value="default"/>
  <param name="enable-callerid" value="true"/>
 </span>
<span id="10">
        <param name="tonegroup" value="us"/>
        <param name="digit-timeout" value="2000"/>
        <param name="max-digits" value="11"/>
        <param name="dialplan" value="XML"/>
         <param name="context" value="default"/>
  <param name="enable-callerid" value="true"/>
 </span>
<span id="11">
        <param name="tonegroup" value="us"/>
        <param name="digit-timeout" value="2000"/>
        <param name="max-digits" value="11"/>
        <param name="dialplan" value="XML"/>
         <param name="context" value="default"/>
  <param name="enable-callerid" value="true"/>
 </span>
<span id="12">
        <param name="tonegroup" value="us"/>
        <param name="digit-timeout" value="2000"/>
        <param name="max-digits" value="11"/>
        <param name="dialplan" value="XML"/>
         <param name="context" value="default"/>
  <param name="enable-callerid" value="true"/>
 </span>
<span id="13">
        <param name="tonegroup" value="us"/>
        <param name="digit-timeout" value="2000"/>
        <param name="max-digits" value="11"/>
        <param name="dialplan" value="XML"/>
         <param name="context" value="default"/>
  <param name="enable-callerid" value="true"/>
 </span>
<span id="14">
        <param name="tonegroup" value="us"/>
        <param name="digit-timeout" value="2000"/>
        <param name="max-digits" value="11"/>
        <param name="dialplan" value="XML"/>
         <param name="context" value="default"/>
  <param name="enable-callerid" value="true"/>
 </span>
<span id="15">
        <param name="tonegroup" value="us"/>
        <param name="digit-timeout" value="2000"/>
        <param name="max-digits" value="11"/>
```

		<param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> <param name="tonegroup" value="us"/> <param name="digit-timeout" value="2000"/> <param name="max-digits" value="11"/> <param name="dialplan" value="XML"/> <param name="context" value="default"/> <param name="enable-callerid" value="true"/> </analog_spans>

Table 5-4 freeswitch configuration table

Chapter 6 Test

6.1 Preparation

Use FXM-16A/PCIe board and TEJ-4A/PCI board for test. The former 4 models of FXM-16A/PCIe board are FXO and the later 4 models are FXS. Meanwhile, the configuration of TEJ-4A/PCI is E1 model and ISDN signal.

Examine the configuration of zaptel:

#ztcfg –vv

6.2 Test Example

6.2.1 Asterisk Environment

Step1： Examine the configuration of Asterisk

#asterisk –vvvc *# startup asterisk*

*CLI>zap show channels *# examine the channel state*

Step2： Test Case 1 (FXM-16A/PCIe)

1) Added dialing rules to '/etc/asterisk/extensions.conf':

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

2) Using the station channel 15 to dial the phone number 3000 of the station channel 13 for test.

Step3： Test Case 2 (TEJ-4A/PCI)

1) Register a sip UA in asterisk.

2) Added dialing rules to '/etc/asterisk/extensions.conf':

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

3) Using the sip to dial the phone number 3000 of the station channel 13 for test.

6.2.2 Trixbox environment

Step1： Examine the configuration of Trixbox

#asterisk –vvr

*CLI>zap show channels

Step2： Test Case 1 (FXM-16A/PCIe)

1) Open the window of Trixbox WEB management and add extention for channel 13 and channel 15 in pbx ->system seting->extention.

2) Using channel 13 and channel 15 to dial out for test.

Step3： Test Case 2 (TEJ-4A/PCI)

1) open the window of Trixbox WEB management and Register a sip channel in the pbx ->system seting->extention.

2) using the eyebeam software terminal to register a sip terminal.

3) in pbx->system seting->outbound route, add hall-call rules which through TEJ channel dial out.

4) using the sip to dial out for test.

6.2.3 FreeSwitch Environment

Step1： Examine the configuration of FreeSwitch

#cd /usr/local/freeswitch/bin
#./freeswitch
freeswitch@> oz list

Step2： Test Case 1 (FXM-16A/PCIe)

1) ＃vim /usr/local/freeswitch/conf/dialplan/default.xml

2) modify the beginning of file as following:

```
<?xml version="1.0" encoding="utf-8"?>
    <!-- http://wiki.freeswitch.org/wiki/Dialplan_XML -->
     <include>
       <context name="default">
         <extension name="fxstofxo">
             <condition field="destination_number" expression="^(.*)$">
                  <action application="set" data="dialed_number=$1"/>
                  <action application="bridge" data="openzap/1/1/${dialed_number}"/>
             </condition>
        </extension>
```

3) restarting the freeswitch or inputting the command reloadxml under the command freeswitch to reload the dialing rule.

4) Using the phone to dial out for test.

Appendix A FAQ

Question 1: Why in the process of using FXM-8A/PCI and FXM-16A/PCIe board result in noise?

This problem may be caused by not completely download the firmware of zaptel. So, please examine the network to confirm it was already connected when you execute the command ./Setup install.

Question 2: Why the LED of board always flash with the circuitry already connected when using the TEJ board?

This problem may be caused by error configure the impedance of board. User can pull out or plug into the corresponding port's jumper-cap for test.

Question 3: How to determine the sequence of boards when there have more then one AST board?

If the configuration of zaptel is automatic, the sequence of boards can find in file /etc/zaptel.conf.

If the configuration of zaptel is manual, the sequence of boards can be find by executing the command: " #cat /var/log/messages | grep "Launching card" "

Please note the command executed time when you judge the sequence of boards according to the boards' serial number.

Question 4: How to deal with the error which occurs in the switch_odbc model while compiling the freeswitch?

This problem may be caused by lacking of unixodbc package. So the user must completely download the unixodbc package and install it, then re-install the freeswich.