

AXE2DL + EC64L

## ATCOM® Digital Card AXE2DL

### User Manual

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## Release note

<b>Version</b>	<b>Changed note</b>	<b>Editor</b>	<b>Date</b>
1.0	First Release		
2.0	Release the User manual	Deniel	2013-07-03

## Contact ATCOM

### The Introduction of ATCOM

ATCOM is the leading VoIP hardware manufacturer in global market. We have been keeping innovating with customer's needs oriented, working with partners to establish a total solution for SMB VoIP with IP phone, IP PBX and Asterisk cards.

With over 10 years' experience of R&D, manufacturing and service in network and VoIP field; mission of creating the biggest value for IP terminals, we commit ourselves in supplying the competitive IP phone and other terminals for IP PBX, softswitch, IMS, NGN providers and carriers; supplying the competitive total VoIP solution for SMB market. We keep improving the customer's experience and creating the bigger value with our reliable products. Until now, our VoIP products has been sold to over 60 countries and used by millions of end users.

### Contact sales:

Address	District C, east of 2nd floor, #3, Crown industry buildings, Chegongmiao Industry area, Futian district, Shenzhen, China
Tel	+(86)755-23487618
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Website address: <http://www.atcom.cn/>

Download Center: <http://www.atcom.cn/download.html>

# Chapter 1 the Introduction of AXE2DL

## Overview of the AXE2DL

AXE2DL Asterisk card is the asterisk PCI-E card which supports dual ISDN PRI E1/T1/J1 ports, optional with teleco-grade hardware echo cancellation. Using AXE2DL digital PRI card, open source Asterisk PBX and stand alone PC, users can create their IP PBX telephony solution included all the sophisticated features of traditional PBX, and extended features such as voicemail in IP PBX. With low profile design, the AXE2DL is suitable for 2U server.

## Features

AXE4DL: dual ISDN PRI E1/T1/J1 ports  
Support original Dahdi, Zaptel(No patch)  
Support Asterisk, Freeswitch, Yate  
Support Elastix, Trixbox, AsteriskNOW, PBX in a Flash  
100% compatible with all features of Asterisk PBX  
Supports standard telephony and data protocols(Including PRI, CAS, CCS for voice and PPP, HDLC, Cisco HDLC and Frame Relay for data modes)  
Supports chan\_ss7 and Openr2  
Optional hardware echo cancellation  
Select the hardware echo cancellation direction  
With low profile design, the AXE2DL is suitable for 2U server.

## Applications

ISDN PRI IP PBX  
ISDN least cost router  
Calling Card Platforms  
IVR system  
Call Center  
Traditional Calls/VoIP Calls Conference  
VoIP Gateway  
Callback Service

## Optional DSP Hardware Echo Cancellation Module

G.168–2002 echo cancellation in hardware  
1024 taps/128 ms tail per channel on all channel densities  
DTMF decoding and tone recognition  
Voice quality enhancement: music protection, acoustic echo control, and adaptive noise reduction  
Does not increase the physical size of the card, and no additional slot is required

## **Voice Modes**

PRI CPE and PRI NET:

EuroISDN

4ESS(AT&T)

5ESS(Lucent)

DMS100

## **Hardware Requirement**

1.6-Ghz Pentium IV

512 MB RAM

PCI-E slot

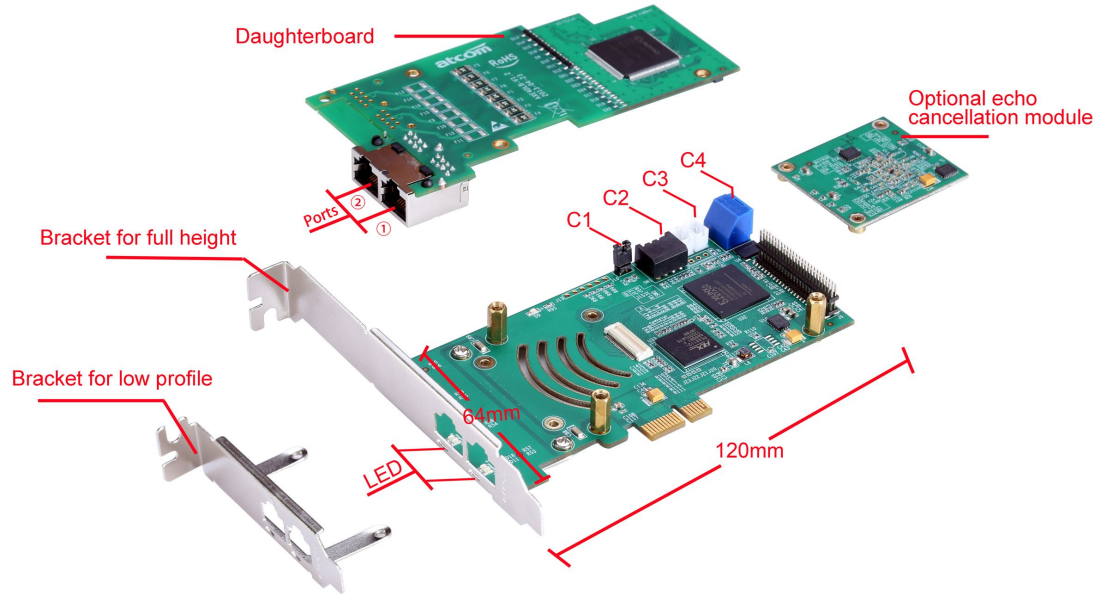
## **AXE4DL Dimension**

120mm (Length)\*64mm (height)

## **Operating System**

Linux (all versions, releases and distributions from 1.0 up)

# Chapter 2 Hardware Introduction



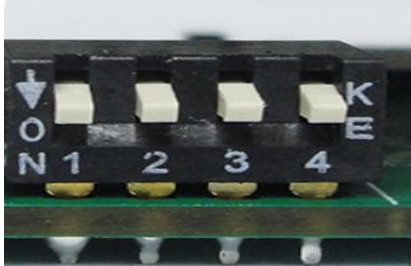
AXE2DL



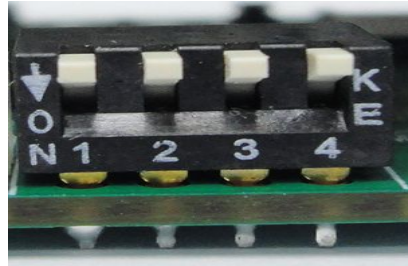
The Front View of EC64L

LED: The LED of the four ports will be red clearly, when the driver of the card is loaded correctly and the “/etc/dahdi/system.conf” file is configured correctly, ;  
 the LED will be green clearly, when the PRI line is connected correctly, and it synchronizes to the other equipment successfully, .

C2: It is used for selecting E1, T1, J1 mode. In default, it will be set it up according to users requirement before shipping out. Users can set up E1, T1/J1 like the following:

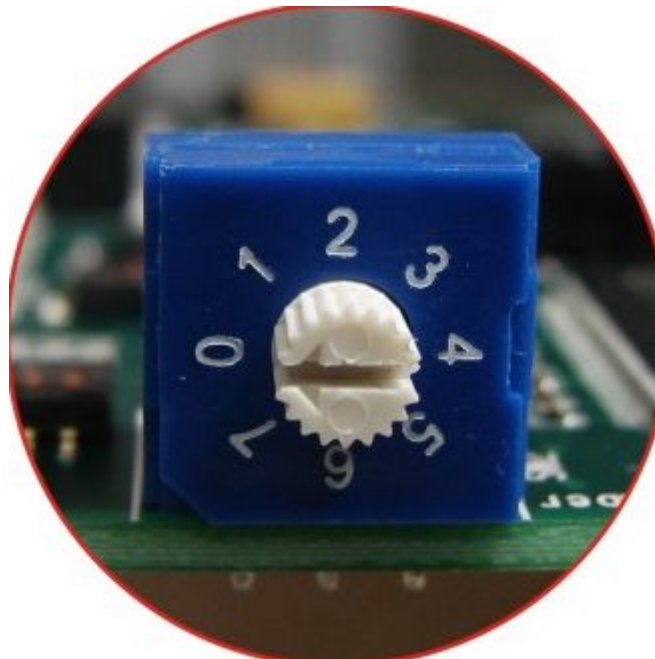


For E1 Mode



For T1 and J1 Mode

C4: It is used when there are more than one PRI card in the one server;  
 For example: if users have two cards in the server, then turn it to 0 in the one card, and turn it to 1 in the other card; If users have four cards in the server simultaneously, then turn it to 0,1,2,3 respectively.



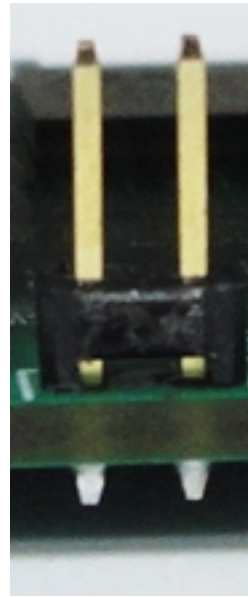


C1: It is used to select the hardware echo cancellation direction.

In default, It will not use the jumper, and it will delete the echo for the far end side; if users need delete the echo for the local side, please use the jumper like the following illustration.



Delete the echo for the local side

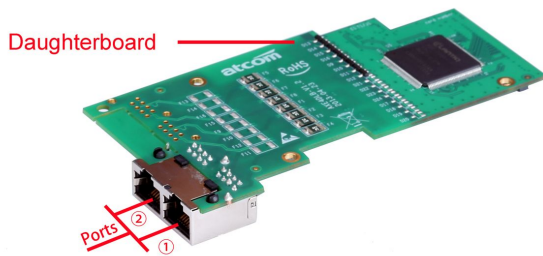


Delete the echo for the far end side

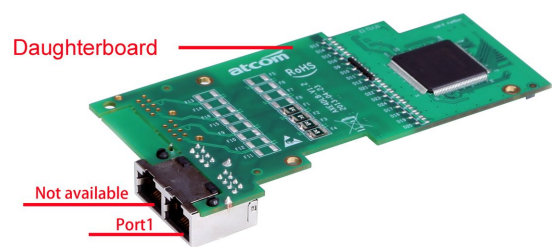
C3: It is not available.

Notice:

The daughterboard is very similar between the AXE2DL and AXE1DL card. Users can get the difference from the following illustration.



AXE2DL daughterboard



AXE1DL daughterboard

## Chapter 2 Test Environment

Test Environment:

Libpri-1.4.7

dahdi-linux-complete-2.6.2+2.6.2

asterisk-1.8.7.0

Centos6.0

(kernel version: 2.6.32-279.22.1.el6.i686)

AXE2DL+EC64L

## Chapter 3 Software Installation

After inserting the card into the PCI slot and boot the server, please use the “lspci” command to check the PCI bus compatibility. From the correct output, users can see the following line:

```
-----
03:01.0 Communication controller: Digium, Inc. Wildcard TE410P quad-span T1/E1/J1 card 3.3V
(rev 02)
-----
```

The TE210P will be found. if users can not see one line like the line above, please power off the server and try to use another PCI slot;

If it does not help still, users have to check the compatibility issue between the card and the PCI-E bus. Please try to use another PCI-E slot or other server and try it again.

1. To install asterisk and dahdi, users have to use “yum” command to install the following prerequisite packages:

```
yum install -y bison bison-devel zlib zlib-devel openssl openssl-devel gnutls-devel gcc
gcc-c++ ncurses ncurses-devel
```

2. Download libpri, dahdi, and asterisk packages

Notice: If users need use the latest version, please check them on asterisk download center.

```
[root@localhost src]#
```

```
wget http://downloads.asterisk.org/pub/telephony/libpri/libpri-1.4.14.tar.gz
```

```
[root@localhost src]#
```

```
wget
```

```
http://downloads.asterisk.org/pub/telephony/dahdi-linux-complete/dahdi-linux-complete-2.6.2+2.6.2.tar.gz
```

```
[root@localhost src]#
```

```
wget http://downloads.asterisk.org/pub/telephony/asterisk/releases/asterisk-1.8.7.0.tar.gz
```

### 3. Installation the packages

#### Install libpri

- 1) [root@localhost src]# tar -xvzf libpri-1.4.14.tar.gz
- 2) [root@localhost libpri-1.4.14]# make
- 3) [root@localhost libpri-1.4.14]# make install

#### Install dahdi-linux

- 4) [root@localhost src]# tar -xvzf dahdi-linux-complete-2.6.2+2.6.2.tar.gz
- 5) [root@localhost src]# cd dahdi-linux-complete-2.6.2+2.6.2
- 6) [root@localhost dahdi-linux-2.6.2]# make
- 7) [root@localhost dahdi-linux-2.6.2]# make install
- 8) [root@localhost dahdi-linux-2.6.2]# make config

#### Install asterisk

- 1) [root@localhost src]# tar -xvzf asterisk-1.8.7.0.tar.gz
- 2) [root@localhost asterisk-1.8.7.0]# ./configure
- 3) [root@localhost asterisk-1.8.7.0]# make
- 4) [root@localhost asterisk-1.8.7.0]# make install
- 5) [root@localhost asterisk-1.8.7.0]# make samples

Notice: if uses run the command: “make samples”, all the asterisk configuration files will be set as the samples.

## Chapter 4 Software Configuration

1. Please run the “dahdi\_genconf” command to configure the “/etc/dahdi/system.conf” file and generate “/etc/asterisk/dahdi-channels.conf” file.

```
[root@localhost ~]# dahdi_genconf
```

It will not show any output when the command run successfully.

After do that successfully, it will get the following configuration in the “system.conf” file with E1 mode.

```
[root@localhost ~]# cat /etc/dahdi/system.conf
# Autogenerated by /usr/sbin/dahdi_genconf  do not hand edit
# This file is parsed by the Dahdi Configurator, dahdi_cfg
# Span 1: TE2/0/1 "T2XXP (PCI) Card 0 Span 1" (MASTER)
span=1,1,0,ccs,hdb3,crc4
# termtype: te
bchan=1-15,17-31
dchan=16
echocanceller=mg2,1-15,17-31

# Span 2: TE2/0/2 "T2XXP (PCI) Card 0 Span 2"
span=2,2,0,ccs,hdb3,crc4
# termtype: te
bchan=32-46,48-62
dchan=47
echocanceller=mg2,32-46,48-62
```

After running dahdi\_genconf successfully, the dahdi-channels.conf file will get the following configuration.

```
[root@localhost ~]# cat /etc/asterisk/dahdi-channels.conf
; Autogenerated by /usr/sbin/dahdi_genconf on Mon Aug  5 15:02:51 2013
; If you edit this file and execute /usr/sbin/dahdi_genconf again,
; your manual changes will be LOST.
; Dahdi Channels Configurations (chan_dahdi.conf)
;
; This is not intended to be a complete chan_dahdi.conf. Rather, it is intended
; to be #include-d by /etc/chan_dahdi.conf that will include the global settings
;
; Span 1: TE2/0/1 "T2XXP (PCI) Card 0 Span 1" (MASTER)
group=0,11
context=from-pstn
switchtype = euroisdn
signalling = pri_cpe
channel => 1-15,17-31
context = default
group = 63

; Span 2: TE4/0/2 "T4XXP (PCI) Card 0 Span 2"
group=0,12
context=from-pstn
switchtype = euroisdn
signalling = pri_cpe
channel => 32-46,48-62
context = default
group = 63
```

2. Please add the line: “#include dahdi-channels.conf” into the end of the “chan\_dahdi.conf” file. And users can run the following command to add it to the file.

```
[root@localhost ~]# echo #include dahdi-channels.conf >> /etc/asterisk/chan_dahdi.conf
```

3. Please run asterisk with the following command:

```
[root@localhost ~]# asterisk
[root@localhost ~]# asterisk -vvgr
[root@localhost ~]# reload
```

4. Please run the “dahdi show channels” command in the CLI. Users will get 64 channels.

5. How to check if the hardware echo cancellation works or not
  - 1) After booting the server with the AXE2DL and the echo cancellation module, please run the “dmesg” command to check if the hardware is detected or not. Users will get the following information.

```
VPM450: echo cancellation for 64 channels
wct4xxp 0000:05:04.0: VPM450: hardware DTMF disabled.
wct4xxp 0000:05:04.0: VPM450: Present and operational servicing 4 span(s)
```

- 2) When users are using the hardware echo cancellation module, please disable the software echo cancellation in “/etc/dahdi/system.conf”, and enable the parameter: “echocancel=yes” line in the “/etc/asterisk/chan\_dahdi.conf” file. After do that, please restart dahdi and asterisk.

- 3) Run asterisk, and make a call by channel 1, users can get the following information marked with red line, if the the echo cancellation module is working.

Notice: If users enable the software echo cancellation, it can get the following red lines also. So please disable the software echo cancellation when users use the hardware cancellation.

```
localhost*CLI> dahdi show channel 1
Channel: 1
File Descriptor: 17
Span: 1
Extension: 300
Dialing: no
Context: from-pstn
Caller ID: 900
Calling TON: 33
Caller ID name: 900
Mailbox: none
Destroy: 0
InAlarm: 0
Signalling Type: ISDN PRI
Radio: 0
Owner: DAHDI/1-1
Real: DAHDI/1-1
Callwait: <None>
Threeway: <None>
Confno: -1
Propagated Conference: -1
Real in conference: 0
DSP: yes
Busy Detection: no
```

TDD: no

Relax DTMF: no

Dialing/CallwaitCAS: 0/0

Default law: alaw

Fax Handled: no

Pulse phone: no

DND: no

**Echo Cancellation:**

**128 taps**

**currently ON**

Wait for dialtone: 0ms

PRI Flags: Call

PRI Logical Span: Implicit



## Chapter 5 Test

### 1 LED state

LED: If the driver of the card is loaded correctly and the “/etc/dahdi/system.conf” file is configured correctly, the four ports of LED will be red clearly.

If the PRI line is connected correctly, and it synchronizes to the other equipment successfully, then the LED will be green clearly.

### 2 Use the E1/T1/J1 cables to connect to the ports. And Check the PRI state

#### 1) Load the asterisk

```
[root@localhost asterisk]# asterisk -vvgr
```

```
*CLI> reload
```

#### 2) Check the PRI state

users will get the following port information, all the four ports are in up status.

```
localhost*CLI> pri show spans
```

```
PRI span 1/0: Up, Active
```

```
PRI span 2/0: Up, Active
```

### 3 Add the dial plan for the PBX

Please edit the dial-plan in the “extensions.conf” file, users can refer to the following example.

```
[from-internal]
```

```
exten=>_1.,1,Dial(dahdi/i1/${EXTEN:1})
```

```
exten=>_1.,2,Hangup()
```

```
exten=>_2.,1,Dial(dahdi/i2/${EXTEN:1})
```

```
exten=>_2.,2,Hangup()
```

```
[from-pstn]
```

```
exten=>s,1,Answer()
```

```
exten=>s,2,Dial(sip/500)
```

```
exten=>s,3,hangup()
```

- 4 Use a registered sip phone to make an outgoing call.

According to the dial-plan above, if users want to make the outgoing call via the first port, users have to add a prefix “1”;

If users want to make a calling via the second port, users have to add a prefix “2”.

For example, the destination number is 0755-83018618, so dialing 0755-83018618, And then users can get the following output from asterisk CLI:

```
-----
*CLI> == Using SIP RTP CoS mark 5
--Executing [83018618@from-internal:1] Dial("SIP/500-00000003",
    "dahdi/1/83018618") in new stack
-- Called 1/83018618
-- DAHDI/1-1 answered SIP/500-00000003
-----
```

The called party ringing, and pick up the phone, so the call is successful.

- 5 Please use a phone to make an incoming call.

For example, the number of the PRI line is 0755-23485319, using a phone to dial 0755-23485319. The sip 500 rings, and users can get the following output from asterisk CLI:

```
-----
*CLI> -- Starting simple switch on 'DAHDI/1-1'
[Jul 1 11:42:53] NOTICE[20678]: chan_dahdi.c:8734 ss_thread: Got event 18 (Ring Begin)...
-- Executing [s@from-pstn:1] Answer("DAHDI/1-1", "") in new stack
-- Executing [s@from-pstn:2] Dial("DAHDI/1-1", "sip/500") in new stack
== Using SIP RTP CoS mark 5
-- Called 500
-- SIP/500-00000004 is ringing
-- SIP/500-00000004 answered DAHDI/1-1
-----
```

Users can test other three ports in the same way.

## Chapter 6 Reference

<http://www.asteriskguru.com/>

<http://www.asterisk.org/downloads>

<http://www.atcom.cn/>