Odin TeleSystems Inc.

Asterisk on Alvis-CSI

Installation Guide

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1 Abstract

This document provides a HOW-TO description of an Asterisk installation and setup on Odin TeleSystems' Alvis-CSI family products. This presentation is targeted to system integrators and application developers who are developing Asterisk-enabled telecommunications systems and/or software applications for Alvis-CSI utilizing the Odin Telecom frameworX (OTX) for Asterisk Softswitch/PBX.

2 Installation

This section describes the Asterisk installation on Odin's Alvis-CSI boards on example of 2 SIP phones connection. You can choose other types of connection if needed.
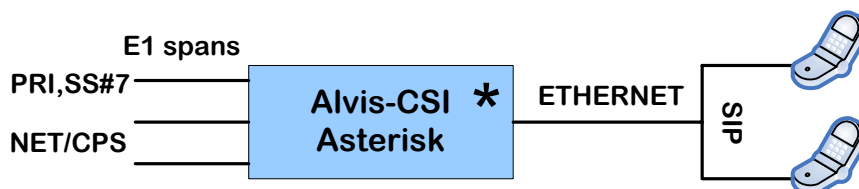
2.1 Installation

First make sure that Alvis-CSI works, configured and connected properly. Try to ping and access the CSI via the telnet/ssh. It is recommended to upgrade the firmware to the recent released version (at least 1.8.2). Please refer to Odin document "Alvis-CSI Firmware Upgrade HOWTO".

Connect the usb-flash and set "usbboot=1" in the u-boot. Next step mount the USB Flash, format it as an ext3 partition and unpack the Asterisk to the USB flash's root. Note that currently you can work with Asterisk only using usb-flash.

An alternative installation procedure includes the installation of the Asterisk add-on tar.gz achieve in the "/opt" folder. Then run Asterisk and do moving of the Root FileSystem to the USB-Flash. It is recommended only for experienced users!

Connect the Serial cable, reboot the CSI and confirm that CSI boots from the USB flash. (Login to the console and type: "mount". It should not say that rootfs is mounted on the mtdblockX).

Also to access the Asterisk CLI, type: "asterisk –rvvvvv".
System should show: "CLI>".
Then type: "quit".

Limitations of the current version are:

- Native bridging mode is not supported yet;
- Voice Codecs transcoding is not implemented yet;
- Hardware Line/Network Echo Cancelling is not integrated yet;
- Asterisk GUI has not been installed yet.

These limitations will be removed in the next version of the Asterisk-CSI.

2.2 Simple PRI-Test Alvis-CSI – to-Alvis-CSI Configuration Setup

You will need some 2 SIP phones, no matter SoftPhones (Like X-lite) or Hardware IP-Phones (LinkSys SPA94x are recommended). In case of SoftPhones they should be installed on 2 different PC's to avoid the SoundCard conflicts.
Also you will need 2 PC's with Alvis-CSI connected each other via the E1 (Li0 to Li1).
Also both units should be connected via the Ethernet.
For SIP phones connection with Asterisk on Alvis-CSI boards you need:

- 2 Alvis-CSI devices – named as "host1" and "host2";
- 2 SIP subscribers called "001" and "002";
- default configuration files generated during Asterisk assembly.

Please configure the "001" subscriber for "host1". To do this, please add the following code in the end of "/etc/asterisk/sip.conf" file on the "host1":

```
[001]
username=001    ; login
secret=001      ; password
type=friend     ; the subscriber can receive and make calls
host=dynamic    ; TODO to clarify
context=default ; user's context (uses in dialplan)
;port=5060      ; reserve port
qualify=yes     ; more detailed output for "sip show peers" command
```

In this file the character ";" is used as a comment. Repeat the similar procedure on the "host2" for the subscriber "002":

```
[002]
username=002    ; login
secret=002      ; password
type=friend     ; the subscriber can receive and make calls
host=dynamic    ; TODO to clarify
context=default ; user's context (uses in dialplan)
qualify=yes     ; more detailed output for "sip show peers" command
```

Assume that the Asterisk console is opened. Then you can load new settings using the command:

```
> sip reload
```

Please run this command on both Asterisks. To connect to the Asterisk console (when program is running) use the following command:

```
# asterisk –rvvvvv
```

Now it is needed to configure the SIP phones. Details vary from program to program, and even depend on the program's version. If phones configuration is successful you will get the appropriate message on the Asterisk console. To browse the registered subscribers, use the command:

```
> sip show peers
```

Suppose that each subscriber is registered at its telephone exchange. Edit the "/etc/asterisk/extensions.conf" file, for example, on the "host1". Add the following code at the [default] section:

```
exten => _7XX, 1, Answer
exten => _7XX, n, SayDigits(007)
exten => _7XX, n, HangUp
```

Load the new dialplan from the Asterisk console:

```
> dialplan reload
```

Now you can check the connection "001" with the "host1". If call from SIP phone to any three-digit number beginning from 7, you can hear the Alice voice calling the "007" digits, after what the connection will be closed.
Half of the way is passed! It is remained to set the connection between dial exchanges. Make a physical connection: link the 1st span on the "host1" (upper left), for example, with the 2nd span on the "host2" (lower left). In this connection on the "host1" edit "/etc/dahdi/system.conf" file follows:

```
span=1,1,0,ccs,hdb3,crc4
bchan=1-15
dchan=16
bchan=17-31
```

On the "host2" in "/etc/dahdi/system.conf" file type the code:

```
span=2,1,0,ccs,hdb3,crc4
bchan=32-46
dchan=47
bchan=48-62
```

Related changes at other connection are obvious. You need only to compare the ranges of logical and physical channels.

Next edit "/etc/asterisk/chan_dahdi.conf" file on both devices. For the "host1" you should add:

```
signalling=pri_cpe
group=1
channel=>1-15
channel=>17-31
```

Respectively for "host2":

```
signalling=pri_net
group=2
channel=>32-46
channel=>48-62
```

After this you need to run the following command on Asterisk console:

```
>dahdi restart
```

Now it is turn to edit dialplans. For "001" in "/etc/asterisk/extensions.conf" file on the "host1" add the code in [default] section:

```
exten => _X.,1,Dial(DAHDI/G1/${EXTEN},,t)
exten => _X.,n,Hangup()

exten => 001,1,Dial(SIP/001);
exten => 001,n,HangUp();
```

Respectively for "002" on the "host2":

```
exten => _X.,1,Dial(DAHDI/G2/${EXTEN},,t)
exten => _X.,n,Hangup()

exten => 002,1,Dial(SIP/002);
exten => 002,n,HangUp();
```

Load new dialplan on both Asterisks:

```
> dialplan reload
```

2.2.1   Checking the Configuration

Make sure that configuration works. First check if PRI is loaded by following command:

```
> PRI show spans
```

This command shows the link state. It can be "down" or "up". You will get the "up" state if PRI loaded correctly. Then check the dahdi loading and lines connection using the command:

```
> dahdi show status
```

You will see the "OK" Alarm if succeed and "RED" Alarm otherwise.

And now you can call!

Try to call from host1 to the 002 and back.

2.3   Real PRI connection basics

In the previous chapter we have assumed the CSI units are connected with each other. In the real installations on 99% the CSI is connected to some PSTN networks via the E1 PRI

connections.

In that case the CSI is acting as a Layer1 Slave. To check this, please set the "E1Master=0" in the u-boot variable. Also check the Layer1 CRC4 setting from your provider.
To check if the Layer1 is ok, please examine the state of the E1 LEDs on the CSI's Front Panel. Also check the Layer1 events coming to the syslog. If all is ok, the link should be "up" and there will be no flooding Layer1 messages arrived.

Note: there are 2 PRI roles - CPE and NET. Li0 is configured as CPE and Li1 as NET by default. In the particular link both parties should not be the same type. Otherwise you will see some error messages in the Asterisk CLI.

2.4 Troubleshooting

First of all, please check if the Layer1 is ok. …
Then check that you have dahdi installed and running. …
Check the sip subscriptions: "sip show peers".
TBD

3 Reference documents

The following documents provide further detailed information related to the Alvis-CSI board:

- Alvis-CSI Technical Description (Odin document # 1111-1-HCA-1020-1)
- Alvis-CSI Product Brief (Odin document # 2020-1-HCA-1020-1)
- Asterisk Press Release
- Alvis-CSI Firmware Upgrade HOWTO (Odin document number 1412-1-HCA-1020-1)

4 Glossary

- OTX – Odin Telecom FrameworX
- Alvis – From the Norse mythology - The dwarf Alvis wanted to marry Thrud (daughter of Thor) but Thor tricked him into being above ground when the sun came up, turning him into stone. Alvis is also a product family based on Texas Instruments DaVinci processor.
- SIP – Session Initiation Protocol
- PRI - Primary Rate Interface