Odin TeleSystems Inc.

Asterisk® on Alvis-PBX

Getting Started

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1   Abstract

This document provides information of how to get started with Asterisk on Odin TeleSystems' Alvis-PBX family of products. This presentation is targeted to system integrators and application developers who are developing Asterisk-enabled telecommunications systems and/or software applications for Alvis-PBX.

2   Installation

2.1   Unpacking the unit

The Alvis-PBX is supplied with a power adapter and a console cable (in addition to the main unit). Figure 1 shows the main unit of Alvis-4-PBX with all external interfaces connected.

Figure 1

The power adapter has different inserts that will match the wall-power plug in most regions of the world. See Figure 2 below.

Figure 2

To prepare the Alvis-PBX for use follow these steps:

1. Remove the protective insert from the power adapter.
2. Insert the proper wall-power insert into the power adapter.
3. Plug the power adapter into a wall socket (100V-240V AC, 50Hz-60Hz).

2.2   Connecting the Console

During the initial configuration of Alvis-PBX it is useful to connect the console to the unit. The console is accessed via a serial port labeled 'CONSOLE' on the front panel of Alvis-PBX. Plug in the supplied console cable into the CONSOLE port and connect the other end of the cable (DB-9) to a COM port of a PC.

Use a terminal emulator program (like HyperTerminal, or putty) and configure the following parameters:

- COM-port number – typically COM1 or COM2 if you connected it to a "native" COM port of your PC
- Baudrate – 115200
- Data bits – 8
- Stop bits – 1
- Parity – None

Please see example of the configuration panel from 'putty' below:

Figure 3

Once the Alvis-PBX is booted (see chapter 2.4), the Alvis-PBX can also be controlled from a telnet session. See example in chapter 2.6.

2.3   USB Flash Drive

Insert the supplied USB Flash Drive into the connector labeled USB. The USB Flash drive contains the file system that the unit will boot from.

Please note that the upcoming production version of Alvis-PBX will boot and run Asterisk from an internal device within the unit itself, hence the USB port is freed up for other usage (if needed).

An alternative procedure involves running Asterisk from an NFS-mounted file system (which is out of the scope of this document).

2.4   Power Up

The Alvis-PBX either boots directly upon power-up or by pressing the Reset push-button (accessible via a small hole in the front panel marked 'RESET'). The console (if attached) will display the boot messages.

Plug the power cable into the connector labeled "POWER" on the main unit. You should see boot messages on the console. See Figure 4.

```
COM1 - PuTTY
TI UBL Version: 1.12, Flash type: NAND
Booting PSP Boot Loader
PSPBootMode = NAND
Starting NAND Copy...
Initializing NAND flash...
Valid MagicNum found.
NAND Boot success.
   DONE
```

Figure 4

The initial messages in the console will be from u-boot. U-boot is one of the initial boot loaders. U-boot will give the user a chance (5 seconds) to press any key and break the normal boot process. If no key is pressed within 5 seconds u-boot will run its default boot command.

After the boot process completes the login prompt is displayed. Log in as 'root'. The default password is blank (no password). See Figure 5.

```
COM1 - PuTTY
MontaVista(R) Linux(R) Professional Edition 4.0 (0501140)

10.0.1.211 login: root
Linux (none) 2.6.10_mvl401-davinci_evm #71 Tue Sep 23 10:19:1
1 GNU/Linux

Welcome to MontaVista(R) Linux(R) Professional Edition 4.0 (0

   Welcome to Alvis-CSI.
   DMP serial number is: DM1R151001011
   IP address is: 10.0.1.211

root@10.0.1.211:~#
root@10.0.1.211:~#
```

Figure 5

The IP address of the unit is displayed as part of the prompt.

2.5   Setting the IP address of Alvis-PBX

The default IP address is 10.0.1.2 (static address). The IP-address of the Alvis-PBX is set by environment variable in u-boot. The Alvis-PBX can operate with either static or dynamic IP addresses. If you choose a dynamic IP address you must have a DHCP server configured on the same subnet as the Alvis-PBX.

The IP address configuration can be modified via the Alvis-PBX web interface, by using the 'uboot' command in Linux, or by editing the uboot variable directly in uboot mode

(which is the mode the DMP is in right after a reboot; before Linux boots).

2.5.1 Setting IP Configuration using the Web interface

To set the IP address using the Alvis Web interface first connect the Alvis-DMP unit to an Ethernet subnet with address 10.0.1.x, then start a web browser and enter the IP address (10.0.1.2) of the Alvis-DMP in the browser's address (URL) field.
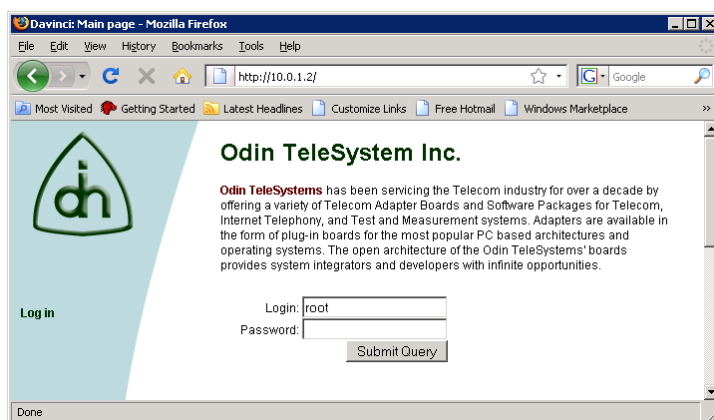
Figure 6

Enter the root password (which is blank [no password] by default).

For further details for the web interface please refer to the Alvis-CSI Technical Description document.

2.5.2 Setting IP Configuration using the 'uboot' command in Linux

The Alvis-PBX file system contains a special utility called 'uboot' which can be used to set and retrieve uboot variables.

Type 'uboot' without any parameters to see a listing of available parameters.

```
root@10.0.1.2:~# uboot
Usage: uboot [opts] cmd [cmd_args]
Where opts can be:
  --file, -f file_name  -- Load uboot vars from file, default /dev/mtd0
  --yes, -y             -- Store anyway even if list is empty
Where cmd can be:
  set [{name} {value}]  -- Set var {name} to {value}
  get [{name}]          -- Get var named {name}
  list                  -- Get list of all vars
```

To get the current IP address type:

```
root@10.0.1.2:~# uboot get ipaddr
10.0.1.2
```

To set the IP address to 192.168.1.10 type:

> root@10.0.1.2:~# uboot set ipaddr 192.168.1.10
> root@10.0.1.2:~#

A reboot is needed when the IP address is changed.

> root@10.0.1.2:~# reboot

2.5.3 Setting IP Configuration using in u-boot mode

To change the IP-address in u-boot mode first connect the Console (see chapter 2.2) and then boot the processor (see chapter **Error! Reference source not found.**). Press any key during the first 5 seconds of the u-boot boot process. The u-boot prompt should then be displayed:

> A-Boot >

Below is a list of useful u-boot commands:

- Help – provides a list of available u-boot commands
- Printenv – displays all u-boot environment variables
- Setenv – sets (or changes) a variable in the environment
- Saveenv – saves the environment in the NAND flash memory
- Boot – boots the processor using the default boot command

The IP-address of the DMP is stored in the environment variable called 'ipaddr'. To set the IP-address to 10.0.1.150 use the following commands:

> Setenv ipaddr 10.0.1.150
> Saveenv

To boot the processor run the command 'boot'.

Other environments variables that can be configured are:

- netmask – Ethernet subnet mask
- usboot – If set to '1' then the Alvis-PBX will boot from a USB Flash Drive. If set to '0' it will boot from a local device.
- spans – If set to '2' the unit will use Li0 and Li1 for its ISDN PRI spans. If set to '4' the unit will use Li0, Li1, Li2, and Li3 for its ISDN spans.
- ntpserver – Network Time Protocol server

2.6 Accessing Alvis-PBX via Telnet or SSH

Alvis-PBX can be accessed via telnet or SSH. However, in order to use SSH you must first set a password for 'root' via a telnet session or via the console.

3   Configuration of Asterisk

This section provides an example of an Asterisk configuration using two Alvis-PBX units and two SIP phones. The two Alvis-PBX units are configured in E1 mode. One unit is NET, and the other unit is CPE.

3.1   Asterisk CLI

First make sure that Alvis-PBX configured and connected properly to the Ethernet. Try to ping and access the Alvis-PBX using telnet or ssh.

Make sure that the USB Flash Drive containing the file system with the Asterisk package is inserted into the USB connector of Alvis-PBX.

Connect the Serial cable (if not already connected), reboot the Alvis-PBX and confirm that Alvis-PBX boots from the USB Flash drive.

Log in to the unit and type the following:

    mount

Make sure that rootfs is not mounted on mtdblockX.

To access the Asterisk CLI, type the following:

    asterisk –rvvvvv

The prompt should show: "CLI>".

Then type: "quit" to exit from CLI for now.

3.2   Alvis-PBX to Alvis-PBX Configuration Setup

You will need two SIP soft phones (e.g. X-lite) or Hardware IP-Phones (LinkSys SPA94x or similar). In case you use soft phones they should be installed on 2 different PCs to avoid any sound card conflicts.
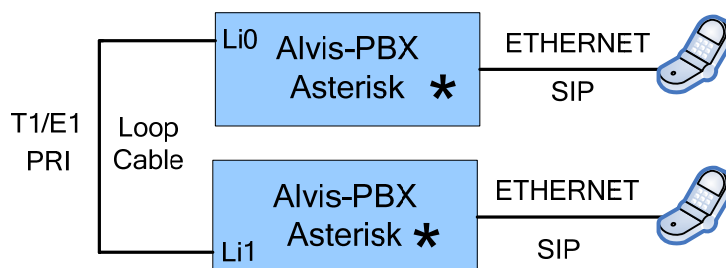

Figure 7

You will also need 2 Alvis-PBX units connected to each other via the T1/E1 interface (loop cable from Li0 of one unit to Li1 of the other unit). Both units should also be connected to the Ethernet.

- Two Alvis-PBX devices – named as "host1" and "host2";
- Two SIP subscribers called "001" and "002";
- Default configuration files supplied with the default Asterisk package for Alvis-PBX.

Please configure the "001" subscriber for "host1". To do this, please add the following code at 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 procedure on "host2" for 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
```

Start the Asterisk console (CLI) with the the following command:

```
# asterisk –rvvvvv
```

Then load new settings using the following command:

```
> sip reload
```

Run this command on both Alvis-PBX units.

Now you need to configure the SIP phones. Details vary from program to program (or phone to phone). If the phone 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 now registered at its Alvis-PBX. 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 you call from a SIP phone to any three-digit number beginning from 7 you can hear Alice's voice calling the "007" digits. After that the connection will be closed.

We are now half-way there. What remains is to set up the connection between the two Alvis-PBX units. Make sure you have connected the loop cable between the two units (connect a T1/E1 loop cable between the 1st span (Li0) of "host1" to the 2nd span (Li1) of "host2").

Then edit the "/etc/dahdi/system.conf" file on "host1" as follows:

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

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

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

Next edit "/etc/asterisk/chan_dahdi.conf" file on both devices. For "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 time to edit dialplans. For subscriber "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 subscriber "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 Alvis-PBX units:

```
> dialplan reload
```

3.3   Checking the Configuration

Make sure that configuration works. First check the status of Layer 1 of the ISDN PRI link by using the following command:

```
> PRI show spans
```

This command shows the link state. It can be "down" or "up". You will get the "up" state if the T1/E1 interface is connected properly from a physical perspective.

Then check the status of dahdi and T1/E1 spans using the following command:

```
> dahdi show status
```

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

At this point you can attempt to make a call.

Try to call from subscriber 001 to subscriber 002 (and vice versa).

3.4   ISDN PRI basics

In the previous chapter we have assumed that the two Alvis-PBX units are connected to each other. In real-life installations the E1/T1 ports of the Alvis-PBX are connected to a PSTN network running the ISDN PRI protocol.
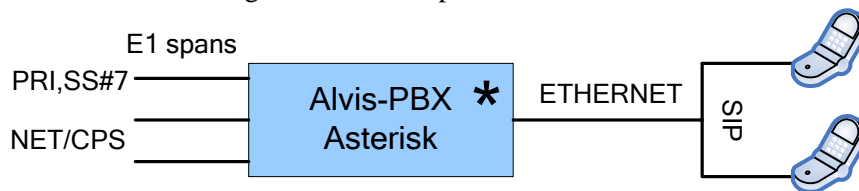
Figure 8

In real-life installations, the Alvis-PBX is acting as a Layer1 Slave (Alvis-PBX will recover the clock from the incoming T1/E1 span). To configure Alvis-PBX is Layer 1 Slave mode, set the "E1Master" u-boot variable to "0" in the u-boot variable

environment.

To check if the Layer1 configuration is ok, please examine the state of the T1/E1 LEDs on the front panel of Alvis-PBX. They should be green. Also check for possible Layer1 events in the syslog of the system. If everything is configured correctly, the link should be "up" and there should not be a continuous stream of Layer1 events (messages).

Note that there are two PRI modes: CPE and NET. In our example Li0 is configured as CPE and Li1 as NET. One end of the T1/E1 span should be in one mode, and the other end must be in the other mode. You can not successfully run an ISDN PRI link when both ends are configured as CPE, or if both ends are configured as NET.

3.5   Troubleshooting

Check if the Layer1 is ok.
Check that the dahdi driver is installed and running.
Check the sip subscriptions: "sip show peers".
Check your cables and their pinout. Ethernet and T1/E1 both use RJ-45 connectors but they use a different pinout.

4   Limitations

Limitations of the current version are:

- G.723 and G.729 Voice Codecs transcoding is not implemented yet;
- Hardware Line/Network Echo Cancelling is not integrated yet;
- Asterisk FreePBX GUI has not been installed yet.

These limitations will be removed in a future release of the Asterisk package for Alvis-PBX.

5   Reference documents

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

- Alvis-CSI Technical Description (Odin document # 1111-1-HCA-1020-1)
- Alvis-PBX Product Brief (Odin document # 2020-1-HCA-1024-1)
- Alvis-CSI Firmware Upgrade HOWTO (Odin document number 1412-1-HCA-1020-1)

6   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.

- PBX - Private Branch Exchange is a telephone exchange that serves a particular business or office
- Alvis-PBX – A general purpose version of Alvis-PBX with additional features for using the unit as a T1/E1 monitoring probe
- CSI – Complete System Integration
- SIP – Session Initiation Protocol
- PRI - Primary Rate Interface
- ISDN – Integrated Services Digital Network