

# ADSP-SC5xxx Setup

## Setup

Demo of the AES67/RAVENNA implementation using the sc573-ezkit

## Features

Supported sampling rates are 44.1,48,88.2,96 kHz.

Supported number of channels is 8 in and 8 out at 1 FS and 4 in and 4 out at 2 FS

## Startup

To run the example :

1. Load the ramdisk/sdcard image using u-boot
2. Log in with username is **root** and the password is **adi**.
3. Modify the /home/root/startup.sh with e.g. nano and set the wanted IP address and hostname (a uniq hostname is required for Avahi).
4. Run startup.sh from /home/root/ directory

Default startup.sh Script

```
hostname adsp-sc573-ezkit-xxx
# NIC setup
ifconfig eth0 169.254.20.13 netmask 255.255.0.0
ethtool -C eth0 tx-frames 1
ethtool -C eth0 tx-usecs 0
ethtool -C eth0 rx-usecs 75
# configuring PTP using ptp4l
/usr/bin/sbin/ptp4l -i eth0 -s &
# increase the irq priority
chrt -r -p 99 9
# AES67
insmod /lib/modules/4.19.0-yocto-standard/extra/MergingRavennaALSA.ko
cd Butler/
./Merging_RAVENNA_Daemon &
# only critical message are print in the console
dmesg -n 1
# SHARC firmware load and start
cd ..
mkdir /lib/firmware
cp -- "Firmware/SharcaES67.ldr" "/lib/firmware/SharcaES67.ldr"
echo "Start SHARC core"
cd /lib/firmware
echo SharcaES67.ldr >/sys/class/remoteproc/remoteproc0/firmware
echo start >/sys/class/remoteproc/remoteproc0/state
```


Load a Sharc firmware from Crosscore Embedded Studio :


1. Lines below "# SHARC firmware load and start" need to be commented out
2. Load code on sharc (from Crosscore Embedded Studio)
3. Start sharc code with corecontrol --start 1

## Streams Setup

In a browser (Chrome recommended) go to the advanced pages by typing the board IP address followed by :9090 e.g. 169.254.1.5:9090

The following should be shown:



 Vendor **Merging Technologies**  
 Product **ADSP-SC5xxx**  
 Serial **2751683637736**


Not secure | 169.254.20.13:9090

General settings | PTP | Session sources | Session sinks | Ins/Outs | NIMOS | System

**Device Name**  
 ALSA (on adsp-sc573-ezkit-xxx)  
This is the unique zeroconf device name. Other devices see this device name.


**Audio Configuration**  
 Sample rate: 48 kHz

**Session Sinks Global**  
 Safety Payout Delay (@1FS): 0  
 SSM (requires IGMP v3):

**Network**  
**Interface 1**  
 Link: Up  
 Name: eth0  
 Type: Zeroconf  
 Address: 169.254.20.13  
 Netmask: 255.255.0.0  
 Gateway:   Use as Primary Gateway

Note: changing the network settings require a reboot of the device.

For an exhaustive documentation about the user interface, please refer to the advanced page manual (link available in the Resources section)

Create a Session source with the button  to setup a Transmitter

Anubis\_Premium x | Horus\_80157 x | ALSA (on adsp-sc573-ezkit-xxx) x +

Not secure | 169.254.20.13:9090

RAVENNA  
AES67 *now!* | adsp-sc573-ezkit.local

Vendor **Merging Technologies**  
Product **ADSP-SC5xxx**  
Serial **2751683637736**

MERGING  
AUDIO FOR THE NETWORKING AGE

General settings | PTP | Session sources | **Session sinks** | Ins/Outs | NIMOS | System

1 ALSA (on adsp-sc573-ezkit-xx)

### Configuration

Enabled

IO Audio Device

Name ALSA (on adsp-sc573-ezkit-xxx)\_1

Description

Output Interface(s) Interface 1

Auto-unicast  retrieve unicast address+port from sink (RTSP)

Address 239.1.20.13  user defined

Address sec  user defined

TTL 15

Payload Type 98

Codec L24

Frame size (samples) 48

DSCP 34 (AF41)

RefCik PTP traceable

Channels Channel count 8

ALSA Output 1 - ALSA Output


The URL of the SDP of this session is <http://169.254.20.13:9090/by-id/1>.

Create a Session sink with the button  to setup a Receiver and select a Source to listen to

Anubis\_Premium x | Horus\_80157 x | ALSA (on adsp-sc573-ezkit-xxx) x +

Not secure | 169.254.20.13:9090

**RAVENNA**  
**AES67** *now!*

Vendor **Merging Technologies**  
 Product **ADSP-SC5xxx**  
 Serial **2751683637736**


adsp-sc573-ezkit.local

General settings | PTP | Session sources | Session sinks | Ins/Outs | NIMOS | System

### Configuration

IO: Audio Device

Label:

Description:

Source:   Manual (Auto)

Delay (samples):

Ignore refclk GMID:

Ignore refclk Domain:

Relaxed check:

Channels:

- NUC\_HAPI
- sap://Anubis\_Premium\_2081
- sap://Anubis\_Premium\_Combo 1/2
- sap://Anubis\_Premium\_Jack 3/4
- Loop
- NUC\_HAPI

### Session Info

Session status: Initializing

RTP status:

Session name:

Playout delay:

RTSP Host:

### Interface 1

RTP status:

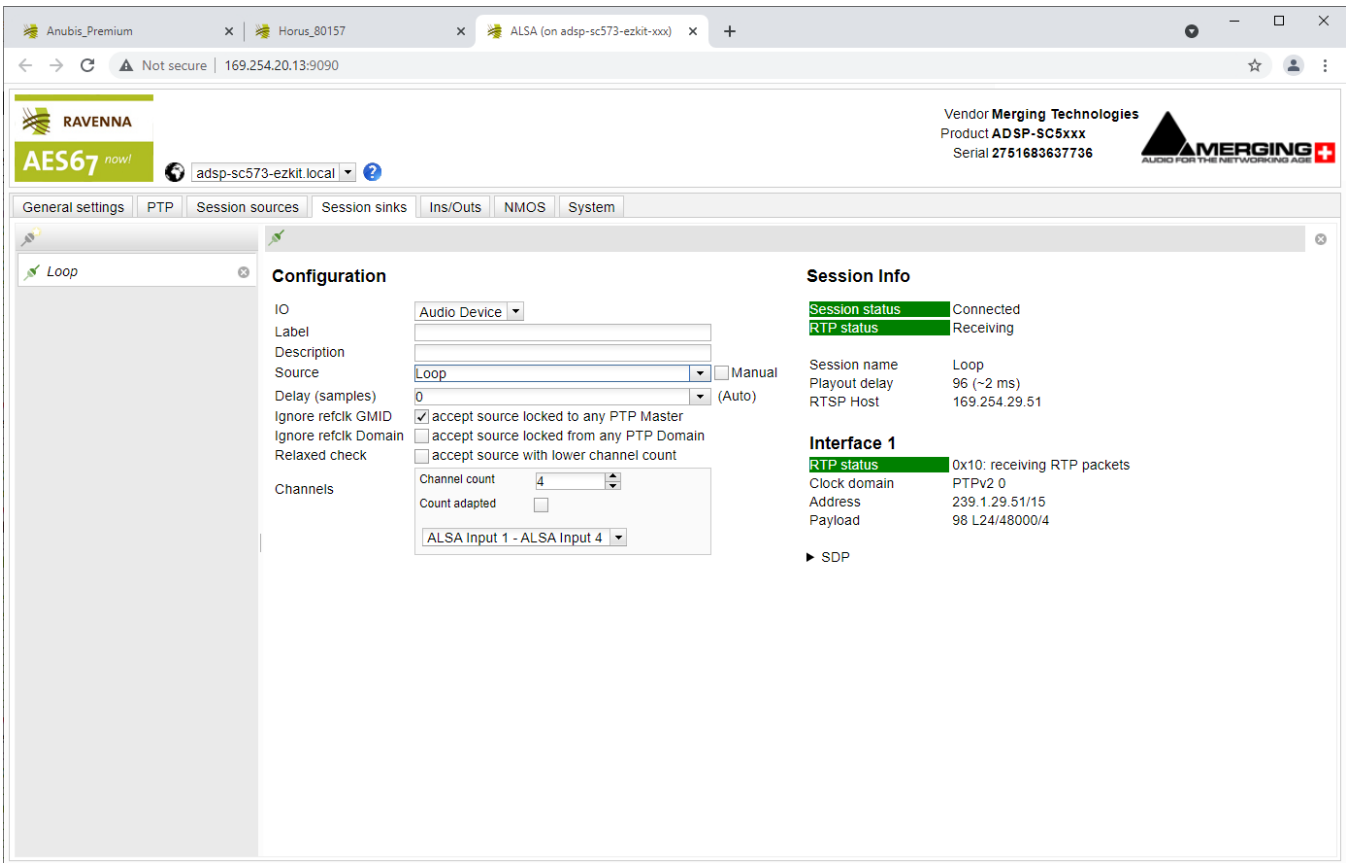
Clock domain:

Address:

Payload:

► SDP

A stream will be initialized and its status will be reported. If everything is green, the audio is working.



## Options

### Butler configuration options

Next to the Butler (Merging\_RAVENNA\_Daemon) there is configuration file : merging\_ravenna\_daemon.conf

The following options can be set :

- interface\_name : Network interface name used by RAVENNA/AES67 network. e.g eth0, eth2, enc0, br1...
- device\_name : By default the name is "Merging ALSA-AES67 (on <hostname>". This can be changed but the name has to be unique on the network (used by Zeroconf) and white spaces are not supported
- web\_app\_port : Port number on which the RAVENNA/AES67 webserver will listen to
- web\_app\_path : Path of the webapp folder provided in the package. Should terminate by webapp/advanced
- tic\_frame\_size\_at\_1fs : Frame size in sample at 1Fs (44.1 / 48 kHz). e.g 48 for AES67
- config\_pathname : Path where the config file will be saved e.g streamer and receiver
- max\_tic\_frame\_size : In case of a high value of tic\_frame\_size\_at\_1fs, this have to be set to 8192
- source\_name\_prefix : The name of the source that will be automatically used. Useful in the high-end world
- default\_sample\_rate : Samplerate (44100, 48000, 96000, ...) set at daemon launch. Useful to start at the nominal system samplerate

For the ADI project the following options are set :

```
config_pathname=./ALSA_configuration.cfg
web_app_path=./webapp/advanced
interface_name=eth0
web_app_port=9090
tic_frame_size_at_1fs=48
default_sample_rate=48000
```

### SHARC working modes

The SHARC has 3 working modes :

1. Loopback the audio data in the L2
2. Send the audio data from the L2 SRAM to the DAC

3. Send the audio data from the ADC to the L2 SRAM

The working mode is defined at the beginning of the src/main.c file.  
Working mode 1) and 3) cannot both be set.

The ADC/DAC implementation only supports 4 channels at the moment.  
The sampling rate of the ADC/DAC is defined in inc/common.h "#define ADA\_FS 1"

## LKM loopback modes

2 loopback mode are possibles :

1. Loop in Sharc : MT\_SHARC\_LOOPBACK
2. Loop in the LKM : MT\_LOOPBACK

Loopback #define option in

workspace/sources/meta-merging-ravenna/recipes-merging-ravenna/merging-ravenna-alsa-mod/files/ravenna-alsa-lkm/common/MergingRAVENNACommon.h

The provided package is set to mode 1 (loopback in SHARC)

## Misc

### Known issues

- PTP Domain is limited to 0, DSCP not working, Master Slave status not reported, Statistics not working [RAV-1537](#)
- Max Sink and Sources size should be limited to 4 @ 96kHz and 88.2 kHz [RAV-1527](#)
- When using ADC or DAC one cannot change sampling rate and number of channels during runtime.

## Ressources

Sources

Ask by email to Merging

ADI Merging AES67LKM repository

<https://bitbucket.org/MergingTechnologies/ravenna-alsa-lkm/src/yocto/>

Web user interface documentation

[Merging RAVENNA Advanced Pages User Guide.](#)