# Soundcard configuration

*Stable: 16.10.2019 - 08:37 / Revision: 16.10.2019 - 08:35*

**Contents**



## 1 Overview

This article explains how to configure ST audio peripherals, as well as STM32MP1 boards external audio components, when they are assigned to the **Linux® OS**. In such cases, they are controlled by the ALSA framework.

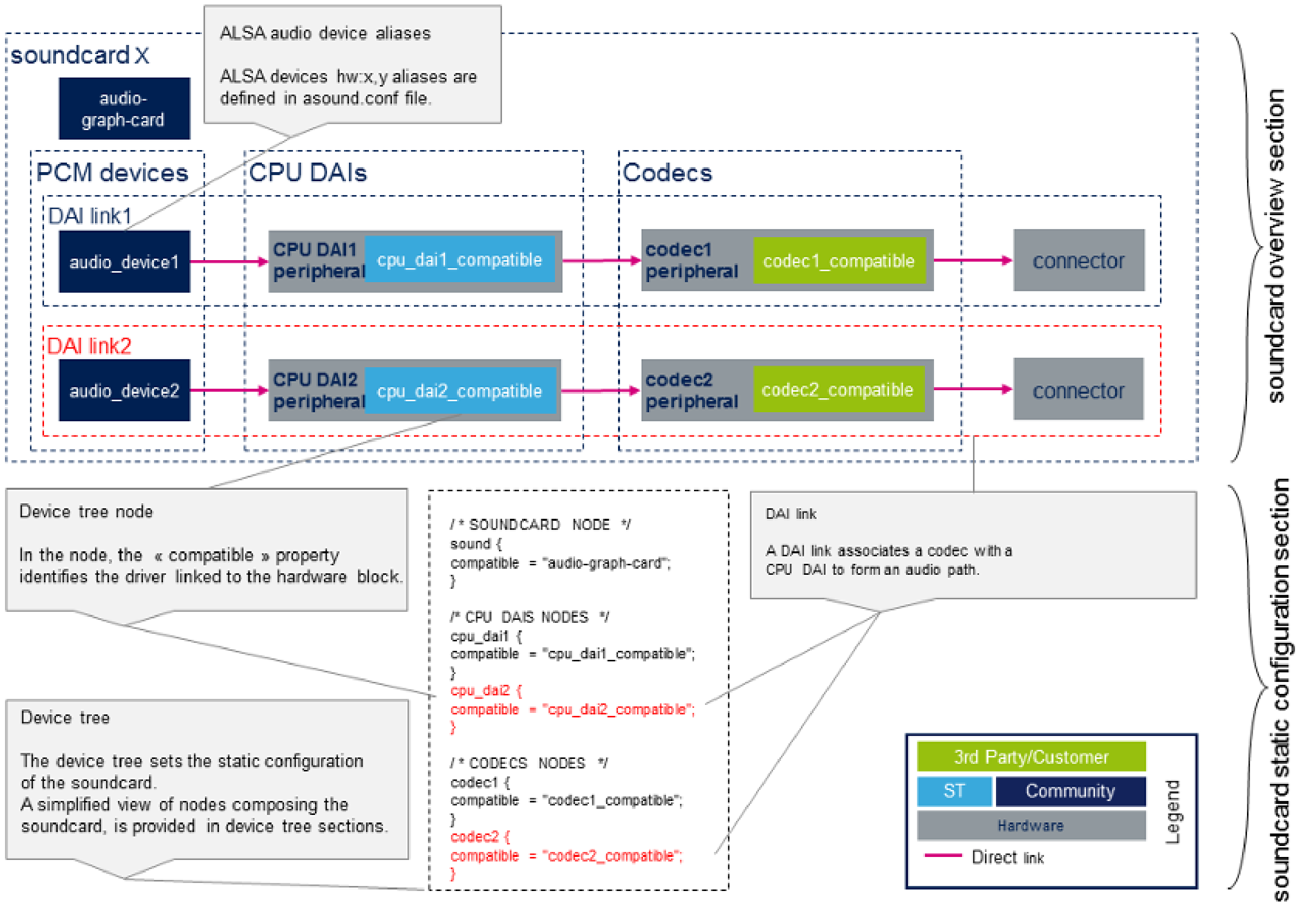
In the ASoC layer of the ALSA framework, audio hardware components are described as CPU DAIs and codec, which are linked together to create DAI links. A sound card is a software component gathering a set of DAI links.

Each of the following STM32 MPU board sections describes one or more sound cards. A schematic for each sound card is provided, as well as its means of static and dynamic configuration.

### 1.1 Sound card schematic

The sound card schematic gives an overview of the hardware and software components forming the sound card, and their relationships.

The example sound card schematic given below emphasizes the links between the sound card and the device-tree section.




## 1.2 Static configuration

- Device tree

The device tree allows the description, configuration and connection of the audio hardware components to define the sound card. The user has to follow the audio graph card bindings[1] to configure the sound card and device graph bindings[2] to connect audio components. The user must also refer to the audio component (codec and CPU DAI) bindings to configure these components properly. The bindings of the audio components can be found in device-tree samples in following sections and in the References chapter.

The STMicroelectronics configuration tool STM32CubeMX, allows the generation of CPU DAI device tree nodes.

| | |
|---|---|
| ⚠ | **STM32CubeMX does not allow configuration of sound card and codec nodes, which are board dependent. The sound card node and the codec nodes have to be filled manually through user sections.** |

- asound.conf [3]

The optional asound.conf[3] system-global custom settings file, provides extra functionalities, such as routing and audio sample conversion. It can be found in the /etc directory.

- Sound card configuration files [3]

The alsa-lib layer provides card configuration files in /usr/share/alsa/cards directory. These files allow to map ALSA hardware devices on standard devices, such as "front", "hdmi" or "iec958" devices. The label defined in sound card device tree node defines the name of the card. The card configuration is retrieved from this card name, according to /usr/share/alsa/cards/aliases.conf mapping.

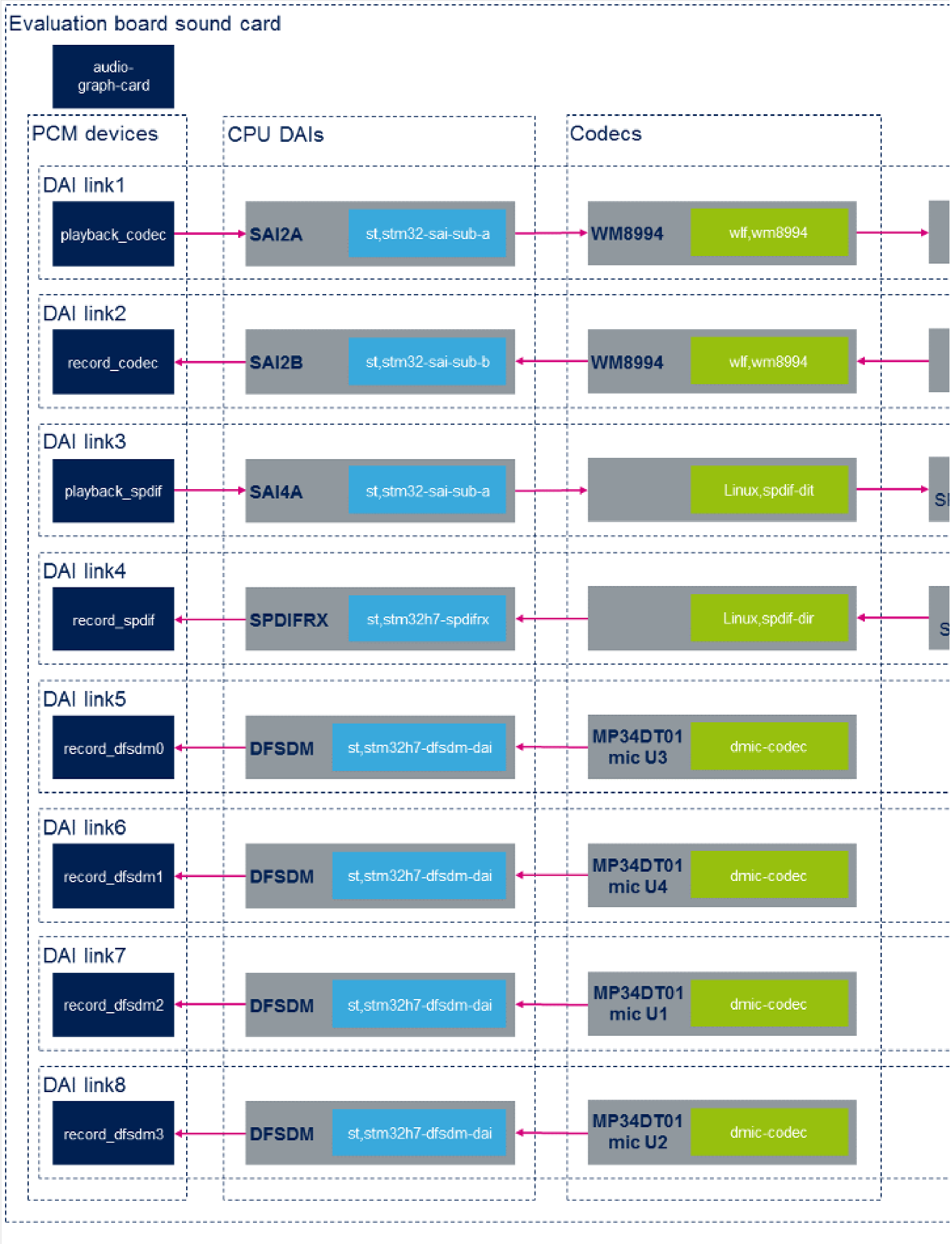
## 1.3 Dynamic configuration

The codecs and CPU DAI drivers also provide ALSA controls, allowing dynamic configuration of the sound card. The controls can be changed at runtime through the amixer utility to modify certain settings in the audio path. For instance, such controls can be used to modify the audio volume or the mute state of a block in the codec.

A customized configuration of these controls can be saved in the asound.state configuration file using the alsactl utility. This configuration can be restored at boot time through alsactl. STM32MPU sound cards come with a dedicated asound.state configuration file providing relevant control settings.

## 2 STM32MP15 evaluation board sound card configuration

### 2.1 Sound card overview




STM32MP15 Evaluation board sound card

Template:WarningImageMapOverlay

## 2.2 Static configuration

The extract below is from the STM32MP15 evaluation board device tree. Only the nodes associated to the sound card, and the most relevant properties are shown here. For example, the properties linking nodes to form the first DAI link are emphasized with green font.

```
/ * SOUNDCARD */
        sound {
                compatible = "audio-graph-card[1]";
                label = "STM32MP1-EV";                /* Sound card identified as STM32M
                routing =
                        "AIF1CLK" , "MCLK1",
                        "AIF2CLK" , "MCLK1",
                        "IN1LN" , "MICBIAS2",
                        "DMIC2DAT" , "MICBIAS1",
                        "DMIC1DAT" , "MICBIAS1";
                dais = <&sai2a_port &sai2b_port &sai4a_port &spdifrx_port
                        &dfsdm0_port &dfsdm1_port &dfsdm2_port &dfsdm3_port>;
        };

/ * CODECS */
        spdif_out: spdif-out {
                compatible = "linux,spdif-dit[4]";

                spdif_out_port: port@0 {
                        spdif_out_endpoint: endpoint {
                                remote-endpoint = <&sai4a_endpoint>;
                        };
                };
        };

        spdif_in: spdif-in {
                compatible = "linux,spdif-dir[5]";
                spdif_in_port: port@0 {
                        spdif_in_endpoint: endpoint {
                                remote-endpoint = <&spdifrx_endpoint>;
                        };
                };
        };

        dmic0: dmic@0 {
                compatible = "dmic-codec";
                port {
                        dmic0_endpoint: endpoint {
                                remote-endpoint = <&dfsdm_endpoint0>;
                        };
                };
        };

        dmic1: dmic@1 {
                compatible = "dmic-codec";
                port {
                        dmic1_endpoint: endpoint {
                                remote-endpoint = <&dfsdm_endpoint1>;
                        };
                };
        };

        dmic2: dmic@2 {
                compatible = "dmic-codec";
                port {
                        dmic2_endpoint: endpoint {
```

```
                                remote-endpoint = <&dfsdm_endpoint2>;
                        };
                };
        };

        dmic3: dmic@3 {
                compatible = "dmic-codec";
                port {
                        dmic3_endpoint: endpoint {
                                remote-endpoint = <&dfsdm_endpoint3>;
                        };
                };
        };
};

&i2c2 {
        wm8994: wm8994@1b {
                compatible = "wlf,wm8994";
                ...
                clocks = <&sai2a>;
                clock-names = "MCLK1";

                ports {
                        #address-cells = <1>;
                        #size-cells = <0>;

                        wm8994_tx_port: port@0 {
                                wm8994_tx_endpoint: endpoint {
                                        remote-endpoint = <&sai2a_endpoint>;
                                };
                        };

                        wm8994_rx_port: port@1 {
                                wm8994_rx_endpoint: endpoint {
                                        remote-endpoint = <&sai2b_endpoint>;
                                };
                        };
                };
        };
};

/* CPU DAIS */
&sai2 {
        clocks = <&rcc SAI2>, <&rcc PLL3_Q>, <&rcc PLL4_Q>;
        pinctrl-names = "default", "sleep";
        pinctrl-0 = <&sai2a_pins_a>, <&sai2b_pins_a>;
        pinctrl-1 = <&sai2a_sleep_pins_a>, <&sai2b_sleep_pins_a>;
        clock-names = "pclk", "x8k", "x11k";

        sai2a: audio-controller@4400b004 {
                compatible = "st,stm32-sai-sub-a";
                dma-names = "tx";                               /* SAI set as transmitter
                clocks = <&rcc SAI2_K>;
                clock-names = "sai_ck";

                sai2a_port: port@0 {
                        sai2a_endpoint: endpoint {
                                remote-endpoint = <&wm8994_tx_endpoint>;
                                format = "i2s";
                                mclk-fs = <256>;          /* SAI is master clock provider
                        };
                };
        };

        sai2b: audio-controller@4400b024 {
                compatible = "st,stm32-sai-sub-b";
                dma-names = "rx";                               /* SAI set as receiver */
```

```
                clocks = <&rcc SAI2_K>, <&sai2a>;
                clock-names = "sai_ck", "MCLK";

                sai2b_port: port@0 {
                        sai2b_endpoint: endpoint {
                                remote-endpoint = <&wm8994_rx_endpoint>;
                                format = "i2s";
                                mclk-fs = <256>;          /* SAI is master clock provider
                        };
                };
        };
};

&sai4 {
        clocks = <&rcc SAI4>, <&rcc PLL3_Q>, <&rcc PLL4_Q>;
        clock-names = "pclk", "x8k", "x11k";

        sai4a: audio-controller@50027004 {
                compatible = "st,stm32-sai-sub-a";
                dma-names = "tx";
                st,iec60958;                                    /* SAI configured for S/PD
                pinctrl-names = "default", "sleep";
                pinctrl-0 = <&sai4a_pins_a>;
                pinctrl-1 = <&sai4a_sleep_pins_a>;
                clocks = <&rcc SAI4_K>;
                clock-names = "sai_ck";

                sai4a_port: port@0 {
                        sai4a_endpoint: endpoint {
                                remote-endpoint = <&spdif_out_endpoint>;
                        };
                };
        };
};

&spdifrx {
        compatible = "st,stm32h7-spdifrx";
        pinctrl-names = "default", "sleep";
        pinctrl-0 = <&spdifrx_pins_a>;
        pinctrl-1 = <&spdifrx_sleep_pins_a>;

        spdifrx_port: port@0 {
                spdifrx_endpoint: endpoint {
                        remote-endpoint = <&spdif_in_endpoint>;
                };
        };

&dfsdm {
        compatible = "st,stm32mp1-dfsdm";
        pinctrl-names = "default", "sleep";
        pinctrl-0 = <&dfsdm_clkout_pins_a
                     &dfsdm_data1_pins_a &dfsdm_data3_pins_a>;
        pinctrl-1 = <&dfsdm_clkout_sleep_pins_a
                     &dfsdm_data1_sleep_pins_a &dfsdm_data3_sleep_pins_a>;
        spi-max-frequency = <2048000>;

        clocks = <&rcc DFSDM_K>, <&rcc ADFSDM_K>;
        clock-names = "dfsdm", "audio";

        dfsdm0: filter@0 {
                compatible = "st,stm32-dfsdm-dmic";
                st,adc-channels = <3>;                  /* Use channel 3, shared with mi
                st,adc-channel-names = "dmic_u1";     /* Left mic U1 associated with Rig
                st,adc-channel-types = "SPI_R";      /* Rising edge for left channel */
                st,adc-channel-clk-src = "CLKOUT";    /* CKOUT clocks the microphones */
                st,filter-order = <3>;
```

```
                asoc_pdm0: dfsdm-dai {
                        compatible = "st,stm32h7-dfsdm-dai";
                        #sound-dai-cells = <0>;
                        io-channels = <&dfsdm0 0>;
                        cpu_port0: port {
                                dfsdm_endpoint0: endpoint {
                                        remote-endpoint = <&dmic0_endpoint>;
                                };
                        };
                };
        };

        dfsdm1: filter@1 {
                compatible = "st,stm32-dfsdm-dmic";
                st,adc-channels = <1>;
                st,adc-channel-names = "dmic_u2";
                st,adc-channel-types = "SPI_F";
                st,adc-channel-clk-src = "CLKOUT";
                st,filter-order = <3>;

                asoc_pdm1: dfsdm-dai {
                        compatible = "st,stm32h7-dfsdm-dai";
                        #sound-dai-cells = <0>;
                        io-channels = <&dfsdm1 0>;
                        cpu_port1: port {
                                dfsdm_endpoint1: endpoint {
                                        remote-endpoint = <&dmic1_endpoint>;
                                };
                        };
                };
        };

        dfsdm2: filter@2 {
                compatible = "st,stm32-dfsdm-dmic";
                st,adc-channels = <3>;                  /* Use channel 3, shared with mi
                st,adc-channel-names = "dmic_u3";      /* Right mic U3 associated with Le
                st,adc-channel-types = "SPI_F";       /* Falling edge for Right channel *
                st,adc-channel-clk-src = "CLKOUT";
                st,filter-order = <3>;

                asoc_pdm2: dfsdm-dai {
                        compatible = "st,stm32h7-dfsdm-dai";
                        #sound-dai-cells = <0>;
                        io-channels = <&dfsdm2 0>;
                        cpu_port2: port {
                                dfsdm_endpoint2: endpoint {
                                        remote-endpoint = <&dmic2_endpoint>;
                                };
                        };
                };
        };

        dfsdm3: filter@3 {
                compatible = "st,stm32-dfsdm-dmic";
                st,adc-channels = <1>;
                st,adc-channel-names = "dmic_u4";
                st,adc-channel-types = "SPI_R";
                st,adc-channel-clk-src = "CLKOUT";
                st,filter-order = <3>;

                asoc_pdm3: dfsdm-dai {
                        compatible = "st,stm32h7-dfsdm-dai";
                        #sound-dai-cells = <0>;
                        io-channels = <&dfsdm3 0>;
                        cpu_port3: port {
                                dfsdm_endpoint3: endpoint {
                                        remote-endpoint = <&dmic3_endpoint>;
```

```
                                        };
                                };
                        };
                };
        };
```

The card-specific alsa-lib configuration file for STMP32MP15 Evaluation board is /usr/share/alsa/cards /STM32MP1EV.conf.

## 2.3 Dynamic configuration

The table below gives an overview of the controls allowing configuration of the STM32MPU evaluation board "sound" sound card.

| audio device | CPU DAI | codec |
|---|---|---|
| playback_codec | no controls available | configure codec output path |
| record_codec | no controls available | configure codec input path |
| playback_spdif | configure iec958 | no controls available |
| record_spdif | configure SPDFIRX input path | no controls available |

### 2.3.1 Wolfson wm8994 output configuration

Control commands to configure aif1 interface to headset output (HPOUT1L/R) path, on wm8994 codec:

```
amixer -c STM32MP1EV cset name='AIF1DAC1 Volume' '96' '96'
amixer -c STM32MP1EV cset name='Headphone Volume' '63' '63'
amixer -c STM32MP1EV cset name='DAC1 Volume' '50' '50'
amixer -c STM32MP1EV cset name='DAC1L Mixer AIF1.1 Switch' 'on'
amixer -c STM32MP1EV cset name='DAC1R Mixer AIF1.1 Switch' 'on'
amixer -c STM32MP1EV cset name='DAC1 Switch' 'on' 'on'
amixer -c STM32MP1EV cset name='Left Output Mixer DAC Switch' 'on'
amixer -c STM32MP1EV cset name='Right Output Mixer DAC Switch' 'on'
amixer -c STM32MP1EV cset name='Headphone Switch' 'on' 'on'
```

### 2.3.2 Wolfson wm8994 input configuration

Control commands to configure headset microphone input (IN1LN) to aif2 interface, on wm8994 codec:

```
amixer -c STM32MP1EV cset name='IN1L PGA IN1LN Switch' 'on'
amixer -c STM32MP1EV cset name='IN1L PGA IN1LP Switch' 'off'
amixer -c STM32MP1EV cset name='IN1L Volume' '25'
amixer -c STM32MP1EV cset name='IN1L Switch' 'on'
amixer -c STM32MP1EV cset name='MIXINL IN1L Switch' 'on'
amixer -c STM32MP1EV cset name='MIXINL IN1L Volume' '1'
amixer -c STM32MP1EV cset name='MIXINL IN1LP Volume' '0'
amixer -c STM32MP1EV cset name='AIF1ADCL Source' 'Left'
amixer -c STM32MP1EV cset name='ADCL Mux' 'ADC'
amixer -c STM32MP1EV cset name='DAC2 Left Sidetone Volume' '12'
amixer -c STM32MP1EV cset name='DAC2 Right Sidetone Volume' '12'
amixer -c STM32MP1EV cset name='AIF2DAC2L Mixer Left Sidetone Switch' 'on'
amixer -c STM32MP1EV cset name='AIF2DAC2R Mixer Right Sidetone Switch' 'on'
amixer -c STM32MP1EV cset name='DAC2 Volume' '96' '96'
```

```
amixer -c STM32MP1EV cset name='DAC2 Switch' 'on' 'on'
amixer -c STM32MP1EV cset name='AIF2ADC Volume' '96' '96'
amixer -c STM32MP1EV cset name='AIF2ADC Mux' 'AIF2ADCDAT'
amixer -c STM32MP1EV cset name='AIF2 Boost Volume' '1'
amixer -c STM32MP1EV cset name='ADC OSR' 'Low Power'
```

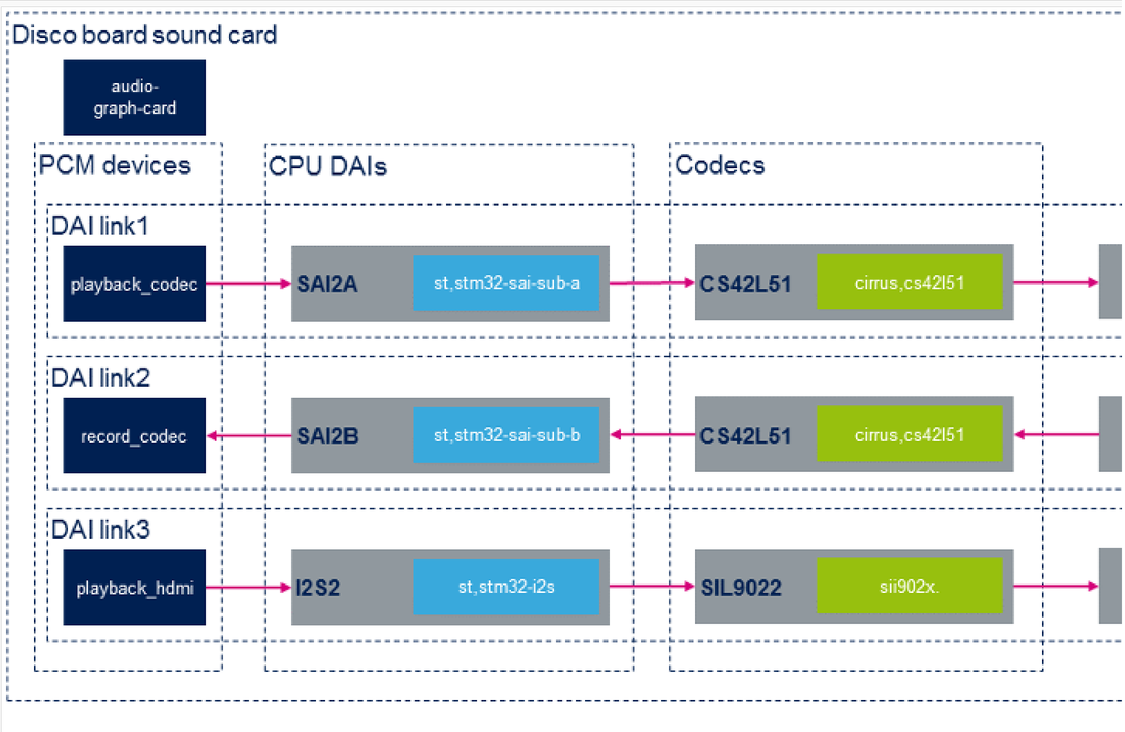
### 2.3.3 SPDFIRX input configuration

Control commands to configure rx1 input path on SPDFIRX:

```
amixer -c STM32MP1EV cset name='SPDIFRX input' 1
```

# 3 STM32MP15 disco board sound card configuration

## 3.1 Sound card overview




STM32MP15 Disco board sound card

Template:WarningImageMapOverlay

## 3.2 Static configuration

The extract below is from the STM32MP15 disco board device tree. Only the nodes associated to the sound card, and the most relevant properties are shown here. As an example, the properties linking nodes to form the first DAI link are emphasized with green font.

```
/ {
/ * SOUNDCARD */
        sound {
                compatible = "audio-graph-card";
                label = "STM32MP1-DK";                  /* Sound card identified as STM32M
                routing =
                        "Playback" , "MCLK",
                        "Capture" , "MCLK",
                        "MICL" , "Mic Bias";
                dais = <&sai2a_port &sai2b_port &i2s2_port>;
                status = "okay";
        };
};

 / * CODECS */
&i2c1 {
        cs42l51: cs42l51@4a {
                compatible = "cirrus,cs42l51";
                ...

                clocks = <&sai2a>;
                clock-names = "MCLK";

                cs42l51_port: port {
                        #address-cells = <1>;
                        #size-cells = <0>;

                        cs42l51_tx_endpoint: endpoint@0 {
                                reg = <0>;
                                remote-endpoint = <&sai2a_endpoint;
                                frame-master;                   /* codec is master
                                bitclock-master;
                        };

                        cs42l51_rx_endpoint: endpoint@1 {
                                reg = <1>;
                                remote-endpoint = <&sai2b_endpoint>;
                                frame-master;                   /* codec is master
                                bitclock-master;
                        };
                };
        };

        hdmi-transmitter@39 {
                compatible = "sil,sii9022";
                ...

                ports {
                        #address-cells = <1>;
                        #size-cells = <0>;

                        port@0 {
                                reg = <0>;
                                sii9022_in: endpoint {
                                        remote-endpoint = <&ltdc_ep0_out>;
                                };
                        };

                        port@1 {
                                reg = <1>;
                                sii9022_tx_endpoint: endpoint {
                                        remote-endpoint = <&i2s2_endpoint>;
                                };
                        };
                };
```

```
        };
};


/* CPU DAIS */
&sai2 {
        clocks = <&rcc SAI2>, <&rcc PLL3_Q>, <&rcc PLL3_Q>;
        clock-names = "pclk", "x8k", "x11k";
        pinctrl-names = "default", "sleep";
        pinctrl-0 = <&sai2a_pins_a>, <&sai2b_pins_b>;
        pinctrl-1 = <&sai2a_sleep_pins_a>, <&sai2b_sleep_pins_b>;
        status = "okay";

        sai2a: audio-controller@4400b004 {
                compatible = "st,stm32-sai-sub-a";
                #clock-cells = <0>;
                dma-names = "tx";                               /* SAI set as transmitter
                clocks = <&rcc SAI2_K>;
                clock-names = "sai_ck";

                sai2a_port: port {
                        sai2a_endpoint: endpoint {
                                remote-endpoint = <&cs42l51_tx_endpoint>;
                                format = "i2s";
                                mclk-fs = <256>;
                                dai-tdm-slot-num = <2>;
                                dai-tdm-slot-width = <32>;
                        };
                };
        };

        sai2b: audio-controller@4400b024 {
                dma-names = "rx";                               /* SAI set as receiver */
                st,sync = <&sai2a 2>;                   /* SAI2B is slave of SAI2A */
                clocks = <&rcc SAI2_K>, <&sai2a>;
                clock-names = "sai_ck", "MCLK";

                sai2b_port: port {
                        sai2b_endpoint: endpoint {
                                remote-endpoint = <&cs42l51_rx_endpoint>;
                                format = "i2s";
                                mclk-fs = <256>;
                                dai-tdm-slot-num = <2>;
                                dai-tdm-slot-width = <32>;
                        };
                };
        };
};

&i2s2 {
        clocks = <&rcc SPI2>, <&rcc SPI2_K>, <&rcc PLL3_Q>, <&rcc PLL4_Q>;
        clock-names = "pclk", "i2sclk", "x8k", "x11k";
        pinctrl-names = "default", "sleep";
        pinctrl-0 = <&i2s2_pins_a>;
        pinctrl-1 = <&i2s2_pins_sleep_a>;
        status = "okay";

        i2s2_port: port {
                i2s2_endpoint: endpoint {
                        remote-endpoint = <&sii9022_tx_endpoint>;
                        format = "i2s";
                        mclk-fs = <256>;
                };
        };
};
```

The card-specific alsa-lib configuration file for STMP32MP15 Disco board is /usr/share/alsa/cards /STM32MP1DK.conf.

## 3.3 Dynamic configuration

The table below gives an overview of the controls allowing the configuration of the STM32MPU disco board sound card.

| audio device | CPU DAI | codec |
|---|---|---|
| playback_codec | no controls available | configure codec output path |
| record_codec | no controls available | configure codec input path |
| playback_hdmi | no controls available | no controls available |

### 3.3.1 Cirrus cs42l51 output configuration

Control commands to configure the aif interface to headset output (AOUTA/B) path, on the cs42l51 codec:

```
amixer -c STM32MP1DK cset name='PCM Playback Switch' 'on','on'
amixer -c STM32MP1DK cset name='PCM Playback Volume' '63','63'
amixer -c STM32MP1DK cset name='Analog Playback Volume' '204','204'
amixer -c STM32MP1DK cset name='PCM channel mixer' 'L R'
```

### 3.3.2 Cirrus cs42l51 input configuration

Control commands to configure headset microphone input (MICIN1/AIN3A) to the aif interface, on the cs42l51 codec:

```
amixer -c STM32MP1DK cset name='PGA-ADC Mux Left' '3'
amixer -c STM32MP1DK cset name='Mic Boost Volume' '1','1'
```

## 4 References

1. ↑ [1.0] [1.1] Documentation/devicetree/bindings/sound/audio-graph-card.txt
2. ↑ Documentation/devicetree/bindings/graph.txt
3. ↑ [3.0] [3.1] [3.2] asound.conf
4. ↑ Documentation/devicetree/bindings/sound/spdif-transmitter.txt
5. ↑ Documentation/devicetree/bindings/sound/spdif-receiver.txt

Operating System

Advanced Linux sound architecture

ALSA System on Chip

Digital Audio Interface

Microprocessor Unit

Central processing unit

Serial Audio Interface

Sony/Philips Digital Interface Format

Digital Filter for Sigma-Delta Modulator

Serial Peripheral Interface

Digital-to-analog converter

Analog-to-digital converter. The process of converting a sampled analog signal to a digital code that represents the amplitude of the original signal sample.