Renesas RA Family

DSP Design Workflow with RA8 MCUs

Introduction

This application note describes the use of an ARM® CMSIS-DSP based Audio Equalizer application project that is created with a MATLAB Filter Design workflow. First, the audio equalizer is designed with MATLAB Filter Design Toolboxes and validated in MATLAB by running the included scripts to process audio with the resulting filter implementation. Next, the CMSIS-DSP source code is generated with the MATLAB Embedded Coder tool and is then optimized to run on the Renesas Arm Cortex-M85-based MCUs with Digital Signal Processing (DSP) extension, and Floating Point Unit (FPU) support. This application note explains the steps to design, import, build, and execute the Audio Equalizer project that is included to accompany the application note.

Additionally, the application note concludes by describing how to improve filter performance with Renesas RA8 MCUs using Arm® Cortex-M85 core with Helium™.

The Audio Equalizer is implemented for floating-point filter processing with the CMSIS-DSP filtering function **Biquad Cascade IIR Direct Form II**. All information regarding Arm CMSIS-DSP can be found at the following link:

Arm Software GitHub CMSIS_5 DSP

The details on the Biquad Cascade IIR function can be found at the link:

Biquad Cascade IIR Filters Using a Direct Form II Transposed Structure

Required Resources

Hardware

- EK-RA8D1, Evaluation Kit for RA8D1 MCU Group (renesas.com/ra/ek-ra8d1)
- SparkFun Electronics Audio Codec breakout board with WM8960
- Assorted jumper cables

Development Tools and Software

- The e² studio IDE v2024-10
- Renesas Flexible Software Package (FSP) v5.6.0
- LLVM Embedded Toolchain version 18.1.3
- Segger J-Link® USB driver v7.98g and Serial Terminal App Tera Term v4.99
- Optional - MATLAB Filter Design Tools
- Optional - Segger SystemView v3.58

Target Devices

This application note focuses on the Audio EQ design workflow for the RA8D1 MCU. However, it also applies to any Arm Cortex-M85-based Renesas MCU that supports Serial Sound Interface Enhanced (SSIE), such as the RA8M1, RA8E1, and RA8E2 MCUs.

Contents

1. Application Project Requirements

The following items are required to build and run the Audio Equalizer.

1.1 Required Development Tools

The following tools are required for building the application project. The tools are available for downloading from Renesas.com:

- e² studio version 24.10.0 with FSP version 5.6.0 and LLVM for ARM version 18.1.3
  https://www.renesas.com/en/software-tool/flexible-software-package-fsp
  or by internet web search keyword: RA FSP Download

1.2 Required Hardware

The following items are required to operate the application project:

- EK-RA8D1
  https://www.renesas.com/en/products/microcontrollers-microprocessors/ra-cortex-m-mcus/ek-ra8d1-evaluation-kit-ra8d1-mcu-group
  or by internet web search keyword: EK-RA8D1
- 1x micro USB cable for programming, debugging, and status display on the serial terminal (for example, Tera Term)

The following items are required for the operation of the application project:

- Wolfson WM8960 Codec mounted on SparkFun Electronics Audio Codec Breakout WM8960 (SparkFun part number BOB-21250)
- Stereo Audio source devices, for example, Compact Disc Players and Stereo Audio output devices such as Powered Audio Speakers, or Headphones
- 10 x jumper wires to connect SparkFun Audio Codec Breakout to the EK-RA8D1 digital connections
- 3 x jumper wires to connect audio source device signals, for example, a Compact Disc Player, to Audio Codec audio inputs
- 3 x jumper wires to connect the Audio Codec audio outputs to Powered Speakers or Headphones

1.3 Required Software

The following software item is required for the application project operation

- Tera Term v4.99 for Command Line Interface Tera Term Download

The following software items are optional for the audio filter design and for profiling the operation

- MATLAB with Audio Filter Design Toolboxes – See Section 3. Audio Equalizer Design Workflow, Table 2. MATLAB Toolboxes and Support Add-Ons for more details
- Segger SystemView v3.58 Application

2. How to Build and Run the Audio Equalizer Application

After importing the example project, you can build, download, and run the project. See Figure 1 Build Audio Equalizer Project, which shows the Audio Equalizer project.

Details on how to import example projects can be found in the Renesas e² studio 2023 – 10 or Higher Quick Start Guide.

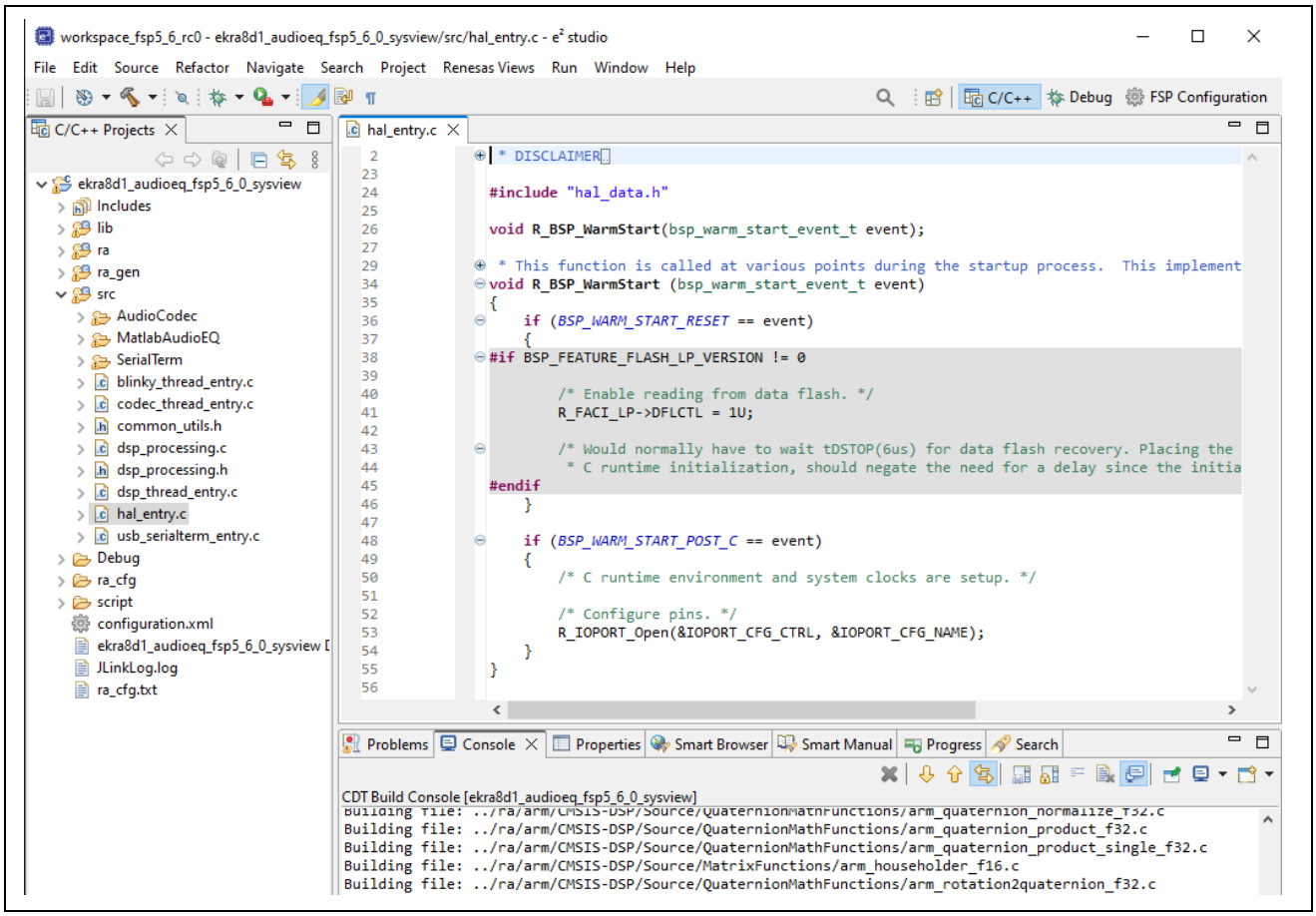


Figure 1. Build Audio Equalizer Project

2.1 Build the Project Source Code

Follow the steps to import and build the project:

- Place the project zip folder into a clean workspace directory and unzip the project there.
- Open e² studio in your workspace directory.
- Navigate to **File -> Import**, which brings an import dialog. In this dialog, select the wizard titled "Existing Projects into Workspace" to finish importing the project (see Figure 1. Build Audio Equalizer Project).
- Build the firmware using the hammer icon on the top toolbar 

The compiler uses the -O3 (optimize most) setting and the LLVM Embedded Toolchain for Arm v18.1.3 in addition to the Arm DSP Library Source, which has optimizations for the Helium extension (see Figure 2. Project Toolchain and Figure 3. Project Summary).

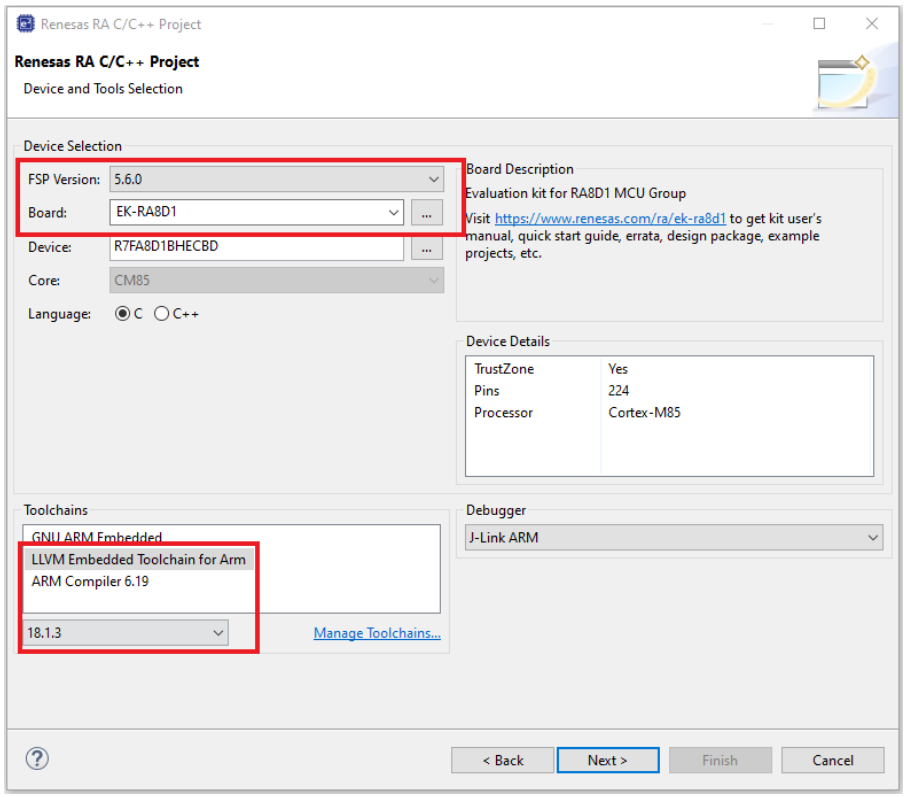


Figure 2. Project Toolchain

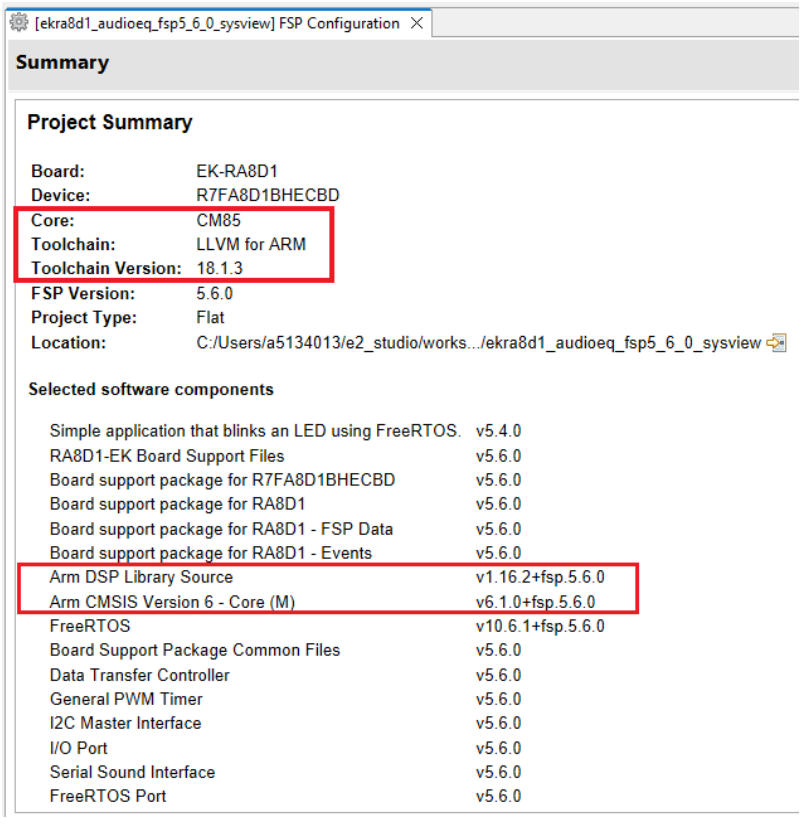


Figure 3. Project Summary

2.2 Connect Audio Codec and Signal Connections

Use the jumper wires to connect the SparkFun Audio Codec board to the EK-RA8D1 according to the connections in Table 1. Audio Codec to EK-RA8D1 Connections.

Table 1. Audio Codec to EK-RA8D1 Connections

Audio Codec	EK-RA8D1	Description
SCL	P512 SCL1	I2C Clock
SDA	P511 SDA1	I2C Data
ADCDAT	P406 SSIRXD0	I2S ADC data to MCU
DACDAT	P405 SSITXD0	I2S DAC data from MCU
DACLRC	P404 SSIWS0	I2S Word Select from MCU
ADCLRC –> connect to DACLRC	-	I2S Word Select from MCU
BCLK	P403 SSISCK0	I2S Bit Clock from MCU
VIN	Power 5V	DC Voltage for Codec Analog
3V3	3V3	DC Voltage for Codec Digital
GND	GND	0V reference for Codec
LINPUT1	-	Audio input Left
RINPUT1	-	Audio input Right
Audio input GND to Codec GND	-	GND for audio inputs
HPL	-	Audio output Left
HPR	-	Audio output Right
Codec GND to Audio output GND	-	GND for audio outputs

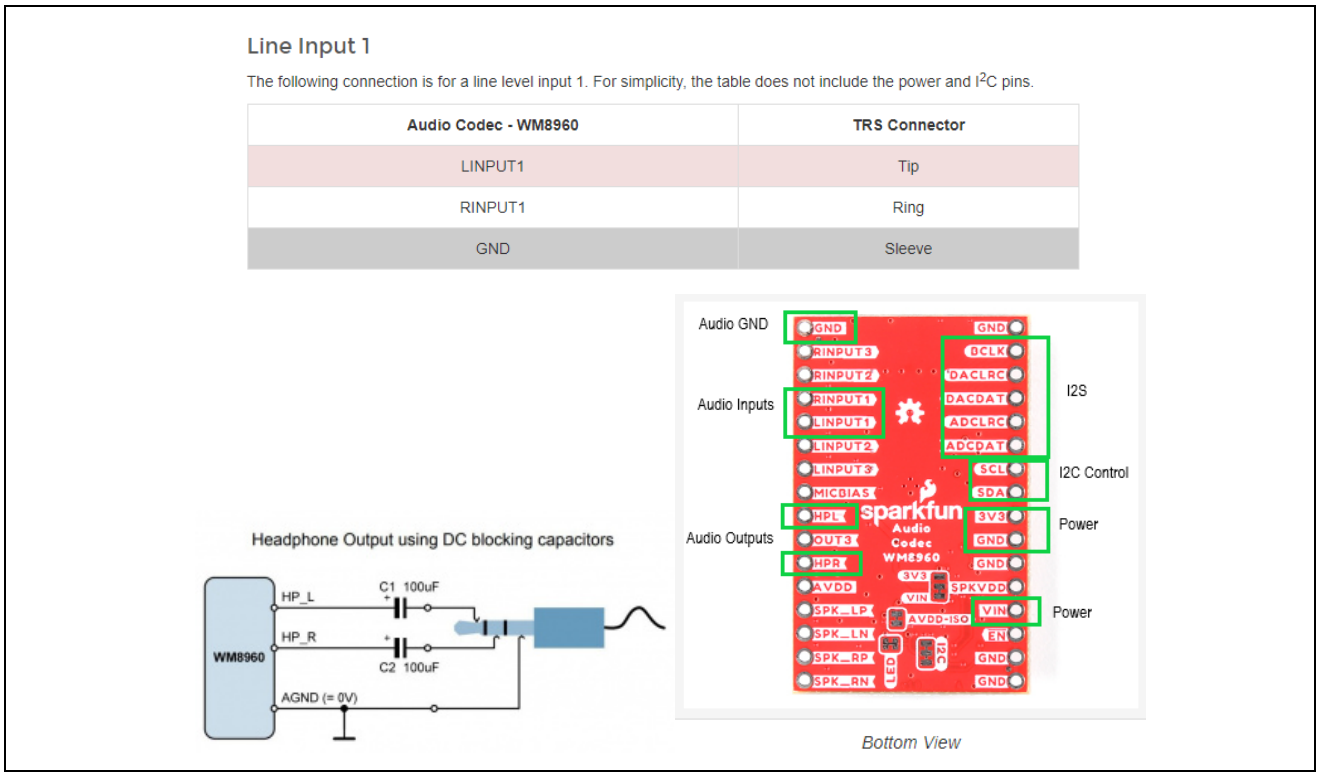


Figure 4. Audio Codec Connections

Use Table 1 to connect the Codec Digital and Power Connections to the EK-RA8D1 first.
In Figure 4. Audio Codec Connections, the SparkFun Codec Board shows the I2S, I2C Control, and Power connections. Use the jumper wires to connect from the Audio Codec board to the EK-RA8D1 Connection Headers, as shown in Figure 5. EK-RA8D1 Connection Headers. For the best signal integrity, solder the wires to the SparkFun board first and then make the connections to the EK-RA8D1.

Next, connect an audio signal source, such as a Compact Disc player, to the SparkFun Codec Audio Inputs LINPUT1, RINPUT1, and GND. Connect the Audio Outputs HPL, HPR, and GND from the SparkFun Audio Codec to the amplified speakers or headphones. For the best signal integrity, solder the connection wires to the SparkFun board.

For additional information, refer to the connections information in Figure 4. Audio Codec Connections, Figure 5. EK-RA8D1 Connection Headers and the EK-RA8D1 User Manual EK-RA8D1 v1 – User Manual.

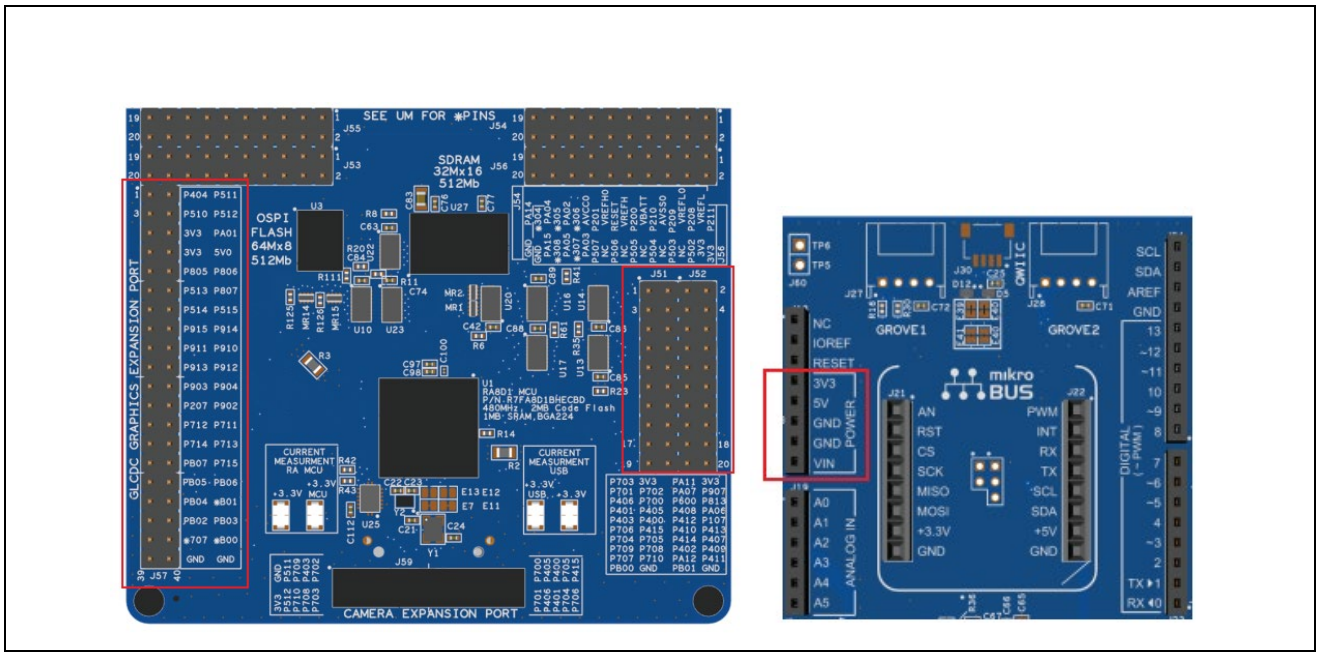


Figure 5. EK-RA8D1 Connection Headers

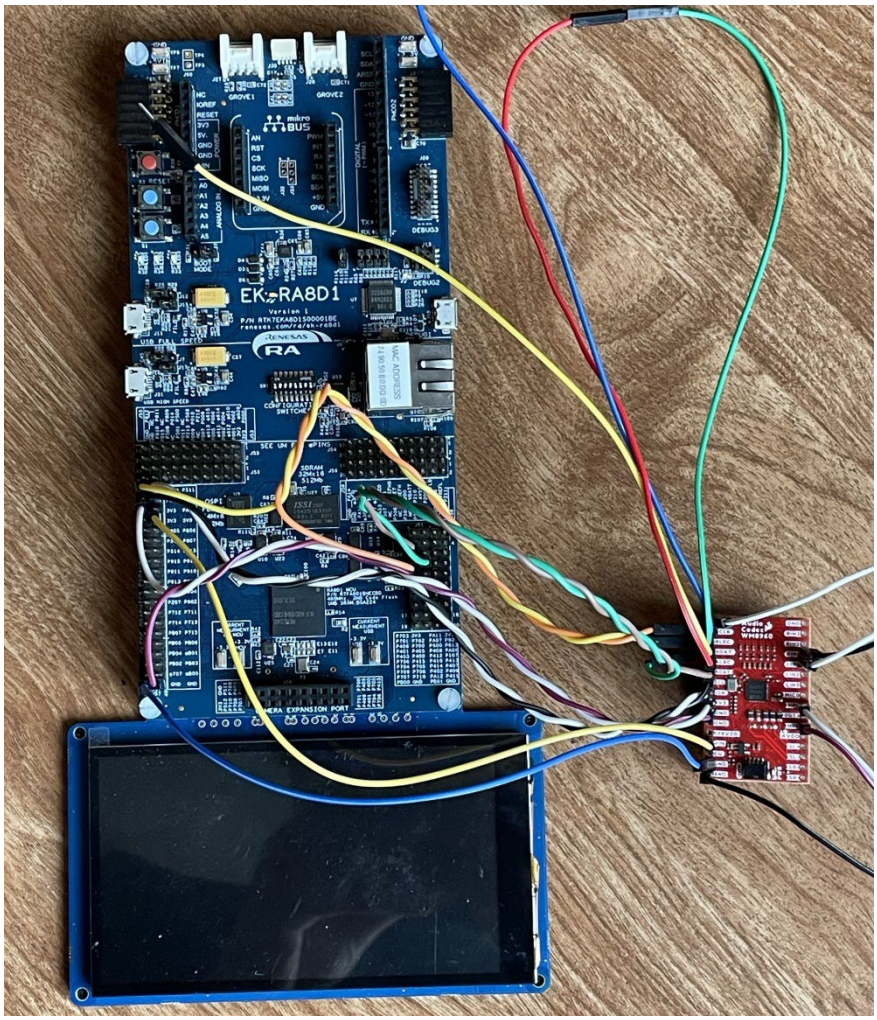


Figure 6. EK-RA8D1 and Audio Codec Wiring Example

See Figure 6. EK-RA8D1 and Audio Codec Wiring Example for an example of how to wire the Audio Codec to the EK-RA8D1.

Twist the analog audio input wires and audio output wires into pairs to minimize conducted noise due to proximity to the digital connections between the Audio Codec and EK-RA8D1 board.

Where possible, twist the wire pairs for signal wires carrying I2S ADC and DAC digital data/clock connections. It was determined that digital noise conduction was minimized for the I2S ADCDAT and DACLRC by tapping the twisted wire pair close to the board near the camera expansion port.

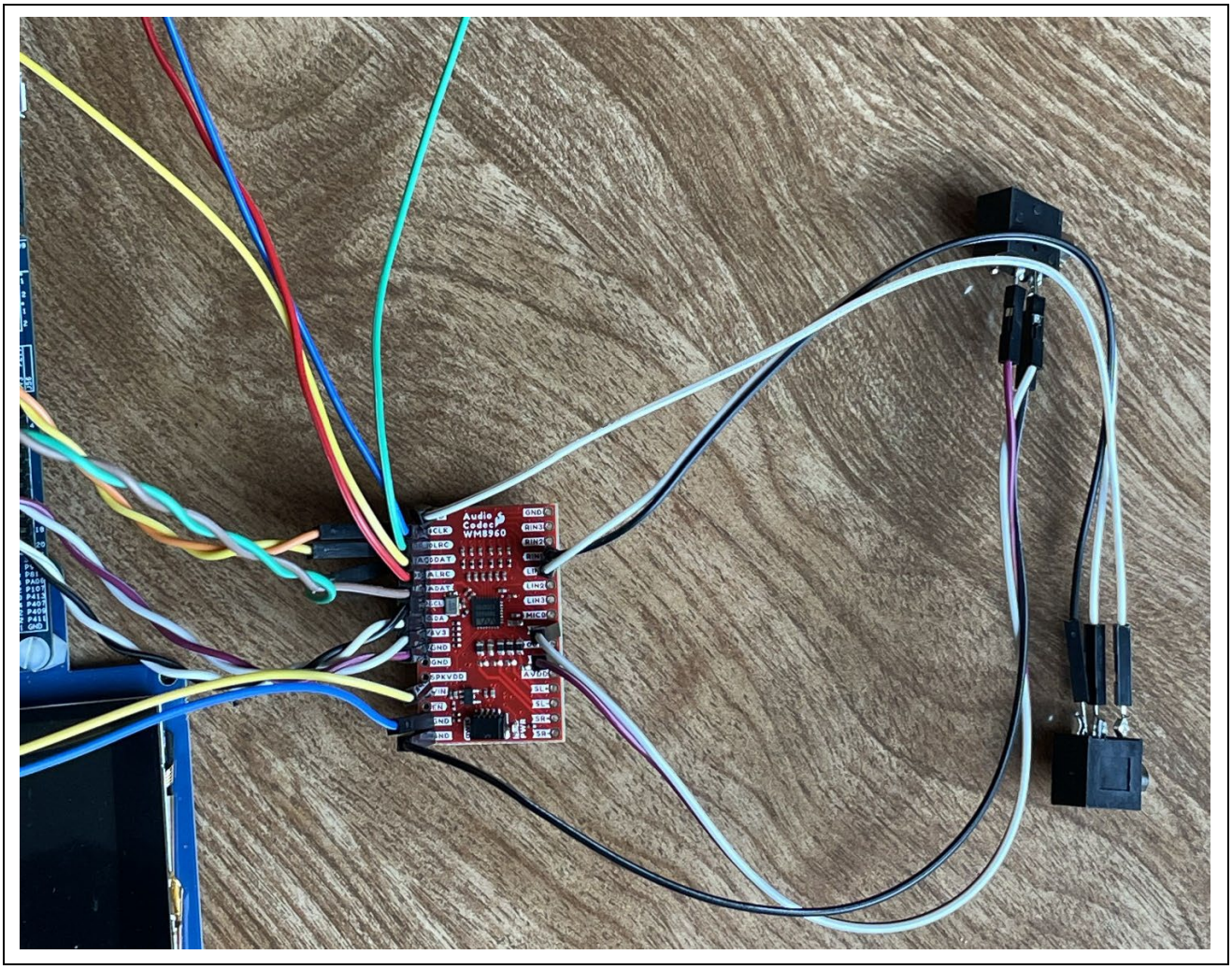


Figure 7. Audio Codec Wiring

See Figure 7. Audio Codec Wiring is an example of how the SparkFun Audio Codec can be connected. Notice the two female 3.5mm stereo audio jacks that are used to connect the audio source (LINPUT1, RINPUT1, and GND) and audio outputs (HPL, HPR, and GND) to the Audio Codec board.

After the Audio Equalizer (EQ) project is downloaded and running, be careful to set the audio signal source with a low-level (amplitude), feeding the Audio Codec. Overdriving the Audio Codec results in audible distortion at the Codec audio output. Slowly raise the audio signal source level until the output signal can be heard clearly but without distortion.

2.3 Setup Command Line Interface (CLI)

The Audio Equalizer command line interface with menu, operation mode, and processing latency are sent to the serial terminal emulator over UART. Connect to Tera Term using the following steps.

Launch Tera Term from the Windows Start Menu. As seen in Figure 8. Launch Serial Terminal Application, open a new connection by selecting **Serial** option and click on **Port** to choose “COMxx: JLink CDC UART Port (COMxx)” and then click **OK**. The Com port number is different from what is shown. This depends on the number of serial devices connected to your PC.

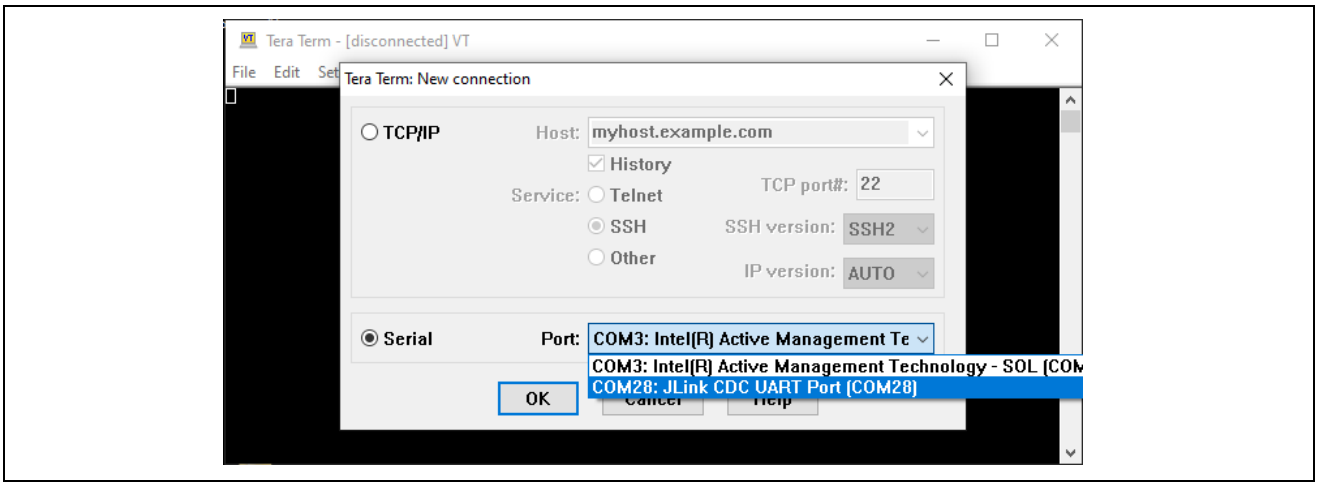


Figure 8. Launch Serial Terminal Application

Configure Tera Term by selecting **Setup -> Serial port**. Set **Speed** to **115200**. All other settings should match those as shown in Figure 9. Then click **New setting**.

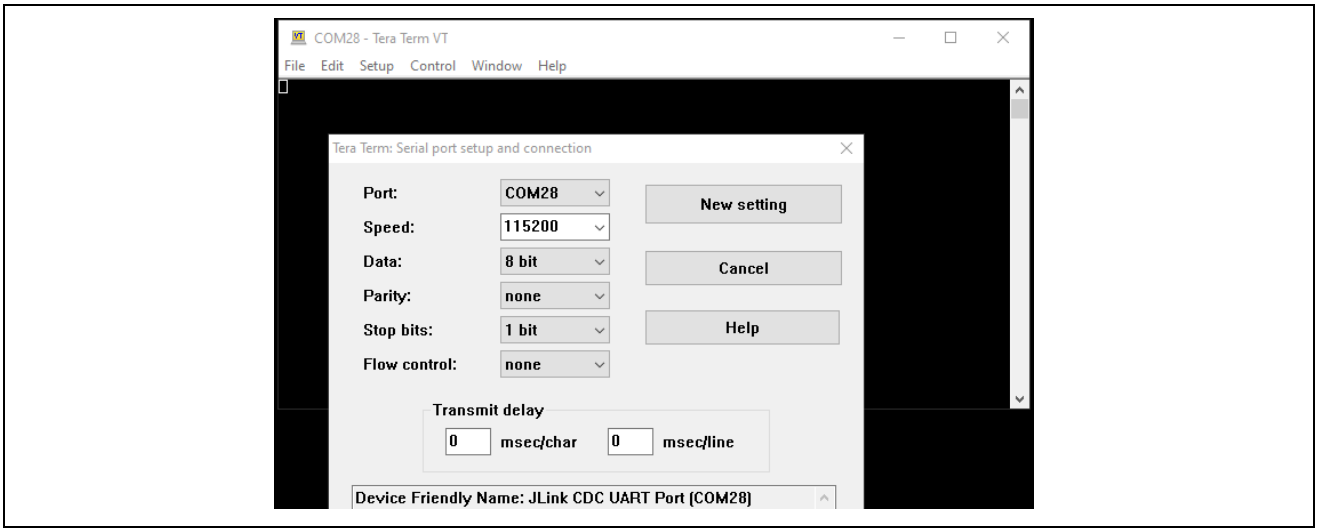


Figure 9. Tera Term Settings

Proceed with the steps to download and start the Audio EQ project in Section 2.4 Download and Operate Project. When the Audio EQ project is started, the Command Line Interface menu is displayed on Tera Term. There are four menu options to select with the Audio EQ project. See Figure 10. Audio EQ Command Line Interface for more information on working with the Audio EQ application.

2.4 Download and Operate Project

Once the build process is finished, then run the project by clicking the Debug icon  to download to the target board and click Run  .

The Audio EQ starts with DSP processing active. Figure 10 Audio EQ Command Line Interface shows the CLI menu.

The Audio EQ starts with DSP Processing active, and you will hear several seconds of silence before the audio output plays once the audio input source is started. This is due to the circular buffers in the system.

To hear the unprocessed audio input, press 1 to bypass DSP Processing.

After listening to the audio for a few seconds, press 2 to Run DSP Processing.

You will hear a difference in the frequency response with the processed audio signal. Compared to the unprocessed input signal there will be more Bass (Low) and Treble (High) frequency content present.

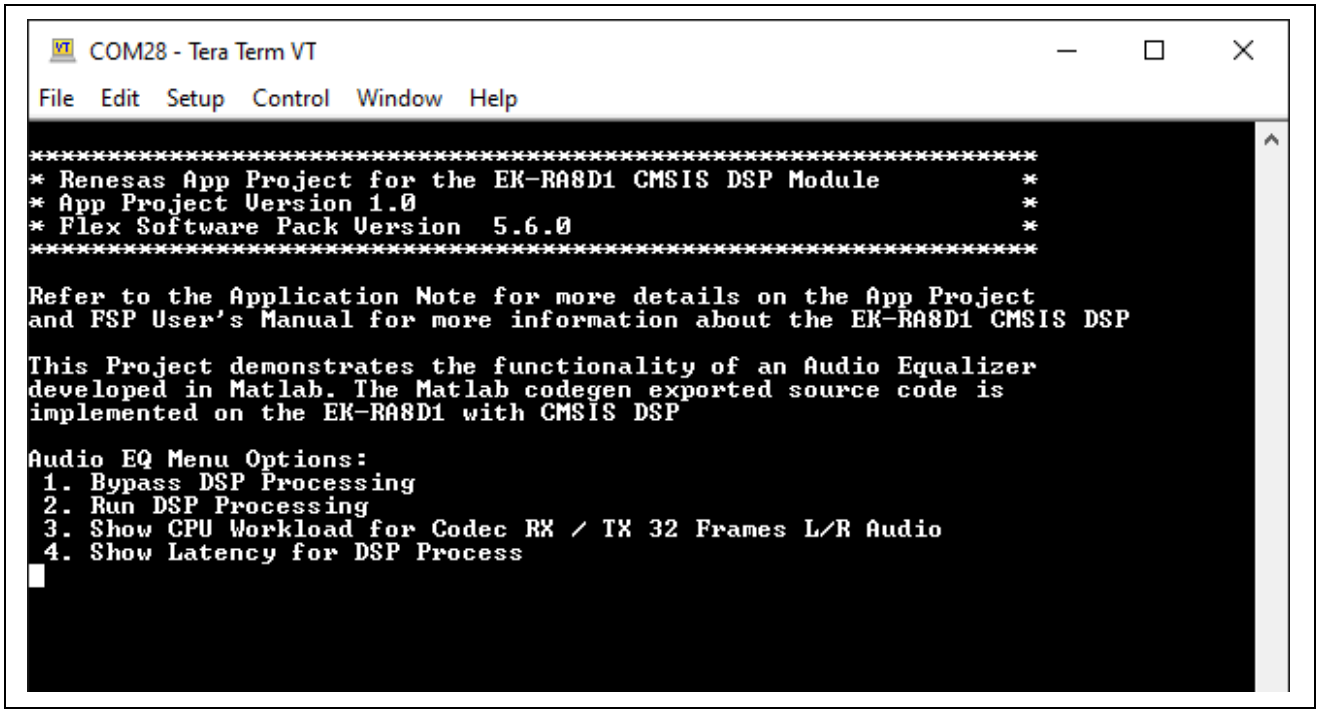


Figure 10. Audio EQ Command Line Interface

The following list details each of the CLI menu options and how it affects the application:

1. **Bypass DSP Processing**
   Used to stop DSP processing. Audio input received is passed (with no processing) to Audio output. Use this menu option to switch between DSP processing mode and Bypass mode to hear the result of the Audio EQ processing. Also, Bypass mode can be used to set the audio input signal. Set the Bypass DSP mode and raise the level of the audio input signal so that no distortion is present while monitoring the audio output. If you hear audio distortion in the output signal in Bypass or Run DSP mode, then lower the input signal level.
2. **Run DSP Processing**
   Activates the DSP Processing, which is the Audio Equalizer (5 bands with frequency centers at [140 Hz, 1000 Hz, 2400 Hz, 4800 Hz, 10000 Hz]. Gains values in Decibels (dB) are set for each of the frequency centers [6.0, 0.0, 1.0, 1.0, 6.0].
3. **Show CPU Workload for Codec RX / TX 32 Frames L/R Audio**
   Shows the amount of time required to receive 32 frames of interleaved channel data from the Audio Codec ADC and to transmit 32 frames of interleaved left/right audio data to the Audio Codec DAC over I2S interface. I2S Sample rate is 44.1 kHz with 16-bit fixed point (integer) data. One frame is one sample of data for the left and right audio channels.
4. **Show Latency for DSP Process**
   Shows the amount of time required to process 32 samples of left/right audio data. The fixed data is copied from the input circular buffer and de-interleaved, converted to floating point and processed, and then converted to fixed point interleaved data before storing it in the output circular buffer.

An example frequency response is shown in Figure 11. Audio EQ Frequency Response for the Audio EQ in Bypass DSP Processing mode (yellow graph) and Run DSP Processing mode (blue graph). The graphs show the frequency vs. amplitude for a Rock music input signal that contains many instruments and vocals.

For the yellow graph, notice that the bass frequencies (20 Hz to 1000 Hz) are lower in amplitude (dBm) than the DSP-processed signal (blue graph), and high frequencies (10 kHz to 20 kHz) are lower in amplitude than the DSP-processed signal. In Run DSP Processing mode, the filter response is set to boost frequencies for the band centered at 140 Hz by 6 dB and the band 10 kHz by 6 dB. With the DSP Processing mode active, the higher amplitude response of the frequency bands can be observed, as indicated by the blue waveform.

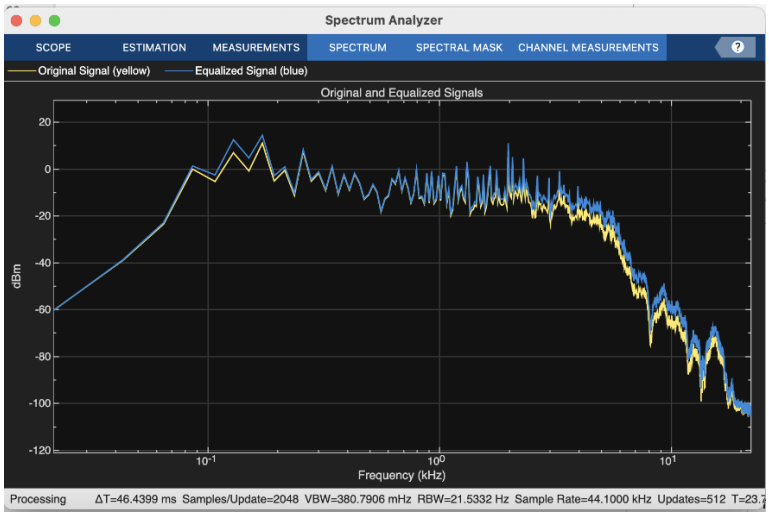


Figure 11. Audio EQ Frequency Response

3. Audio Equalizer Design Workflow

The Arm CMSIS-DSP examples Arm Software GitHub CMSIS_5 DSP provide an introduction and reference on how to implement fundamental processing with CMSIS-DSP functions. However, in the field of Digital Signal Processing, de facto standard design tools such as MathWorks MATLAB are often used to iteratively design and validate filters before implementing and optimizing the source code for an embedded system target.

This audio equalizer (EQ) application note goes into detail on using MATLAB as an audio filter design workflow, generating the CMSIS-DSP supported source code for Cortex-M processors, and optimizing the implementation on Renesas RA8 MCUs for the lowest processing latency.

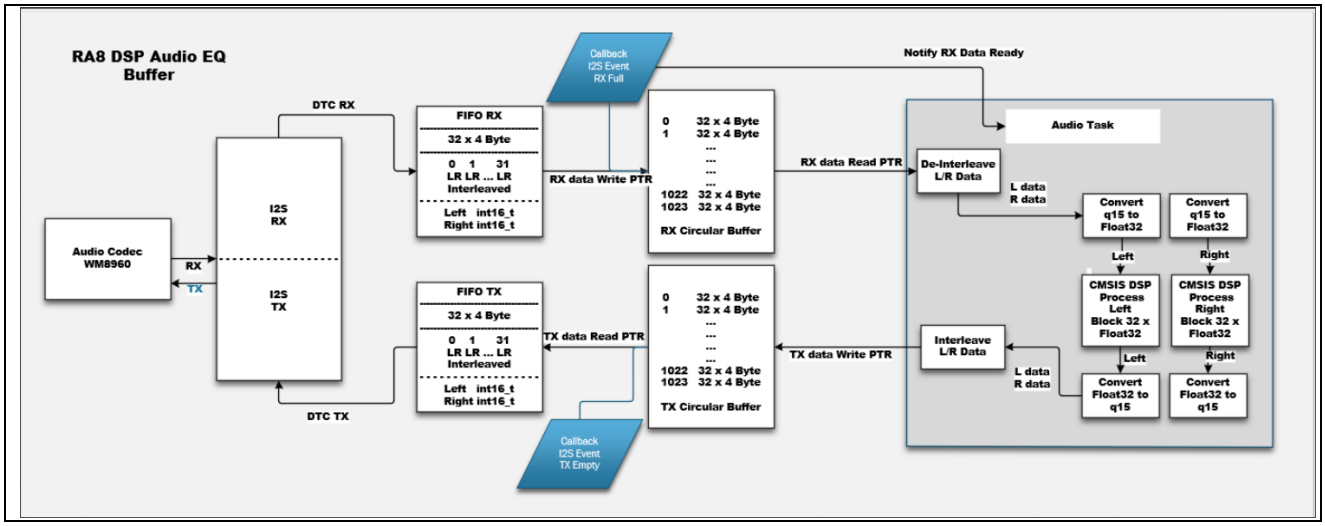


Figure 12. Audio EQ Block Diagram

Figure 12. Audio EQ Block Diagram shows the system block diagram of the audio EQ. The basic operation steps are:

- Input audio data is received from the Audio Codec by FIFO RX and stored in the RX Circular Buffer

- RX Circular Buffer fixed-point data is de-interleaved into left and right channel data and converted to floating point data.
- Process L & R channel data with the Audio EQ CMSIS DSP floating point functions on a block of 32 samples.
- Processed Left and Right channel data is converted back to fixed-point data and interleaved.
- Interleaved data is stored in the TX Circular Buffer, loaded into FIFO TX, and sent to the Audio Codec as output audio for the listener.

3.1 MATLAB Software Requirements

For this project, MATLAB version R2023b MATLAB (mathworks.com) was used along with the Audio Toolbox Audio Toolbox and DSP System Toolbox DSP System Toolbox. In addition, the Signal Processing Toolbox Signal Processing Toolbox was installed since it is a dependency for the DSP System Toolbox and Audio Toolbox.

In order to export the audio filter designed in MATLAB and generate source code targeted for Arm Cortex-M processors, it is necessary to install MATLAB Coder MATLAB Coder, Embedded Coder Embedded Coder, and the support package Embedded Coder Support Package for ARM Cortex-M Processors.

For instructions on how to install the toolboxes and support add-ons, consult the installation instructions at the links provided above. Trial versions for evaluation of the toolboxes and add-ons are available by contacting MATLAB customer support. If you have any installation issues or questions about the software components, contact MATLAB customer support at Support - MATLAB.

See Table 2. MATLAB Toolboxes and Support Add-Ons for an explanation of the MATLAB toolboxes and support add-ons used in the project.

Table 2. MATLAB Toolboxes and Support Add-Ons

Software Component	Component Type	Description
MATLAB R2023b	Application	Main application
Audio Toolbox v23.2	Toolbox (Add-On)	Used to add audio file streaming support to test filter design on PC in real-time
DSP System Toolbox v23.2	Toolbox (Add-On)	Required by Audio Toolbox and used for Audio Filter Design
Signal Processing Toolbox v23.2	Toolbox (Add-On)	Required by DSP System Toolbox and Audio Toolbox
MATLAB Coder v23.2	Add-On	Generates C/C++ code from MATLAB code
Embedded Coder v23.2	Add-On	Extends MATLAB coder to generate C/C++ code optimized for embedded systems
Embedded Coder Support Package for ARM Cortex-M Processors v23.2.1	Add-On	Extends Embedded Coder to generate code optimized for Cortex-M processors

For this project, the DSP System Toolbox was chosen to design the Audio EQ filter because it provides useful System objects that include embedded coder support for ARM Cortex-M processors. Specifically, the Audio EQ was implemented as a Band Pass Filter (BPF) using cascaded Second Order Sections (SOS) with the dsp.BiquadFilter System object Biquad Filter.

A Biquad is a type of linear filter that implements a transfer function that is the ratio of two quadratic functions (the numerator and denominator). What is unique about a Biquad filter is the ability of the user parameter settings (the coefficient design) to control the response characteristics of the filter. The parameters are Gain, Band Center Frequency, and Bandwidth (Selectivity/Quality-Q). The user sets the desired parameters, and the numerator and denominator coefficients of the transfer function are calculated to design the filter.

It is beyond the scope of this application note to cover the filter design specifics and details of the MATLAB filter toolboxes. Refer to MATLAB documentation for more information. For other filter System objects that support Cortex-M code generation, see the documentation for DSP System Toolbox.

If you are developing a filter with a different System object or Toolbox, be sure to consult the MATLAB documentation to ensure that the filter provides the required embedded coder support for ARM Cortex-M cores. Not all MATLAB signal processing and filter design Toolboxes provide embedded coder support for ARM Cortex-M cores. Before creating the design, review the documentation first for the filter or filter design tool under consideration. If the documentation for the Toolbox or System object is not clear, then Contact MATLAB customer support at Support - MATLAB.

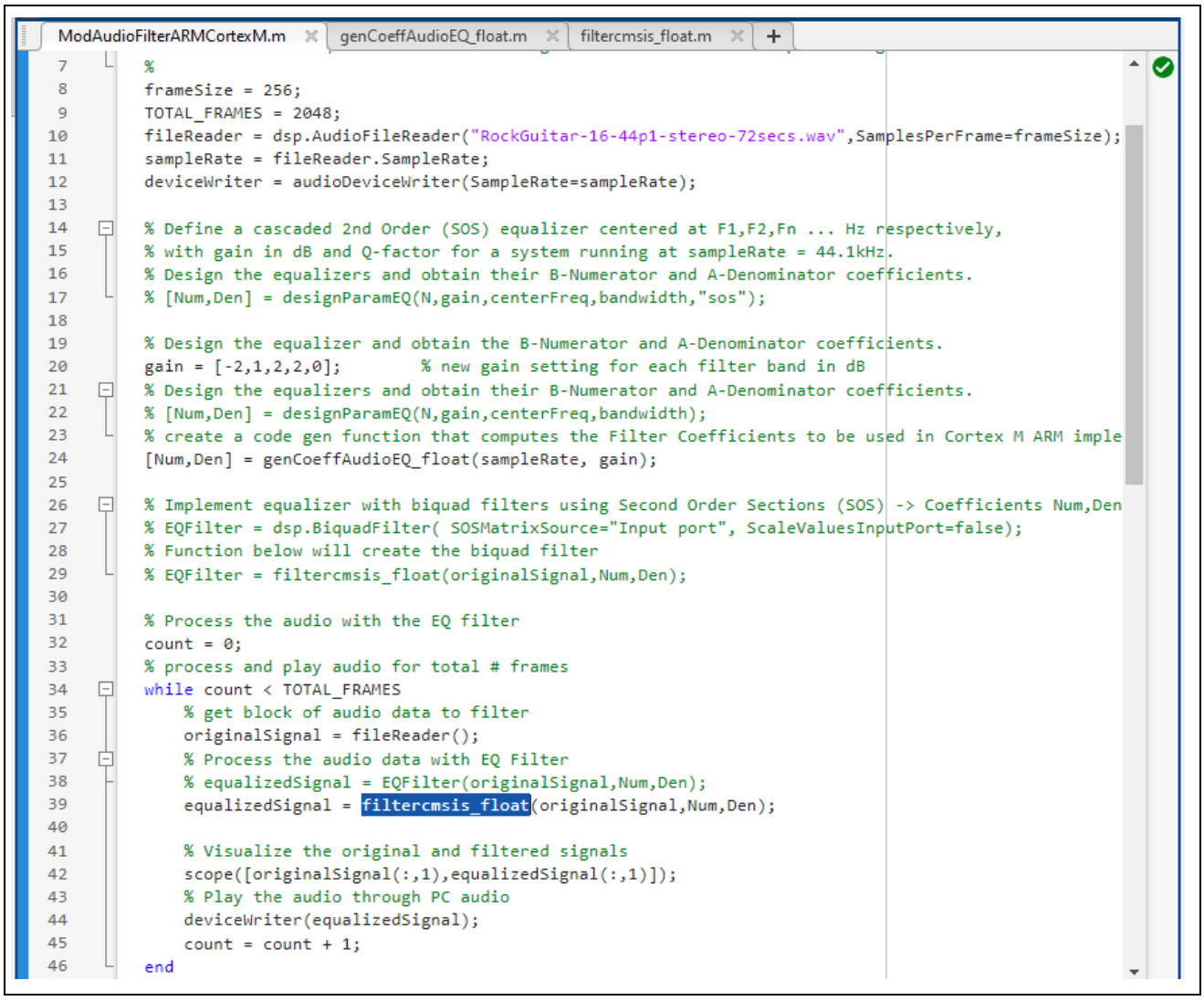


Figure 13. MATLAB Audio Equalizer Design

See Figure 13. MATLAB Audio Equalizer Design, Figure 14. Audio EQ Filter Coefficient Design, and Figure 15. Audio EQ Biquad Filter Function, which details the MATLAB script (ModAudioFilterARMCortex.m) for the workflow used to develop and validate the Audio EQ filter. The steps are summarized as follows:

1. Set up Audio Toolbox resource PC Audio File Reader to acquire input signal to process with DSP Filter and set up resource File Writer to output the processed audio signal.
2. Design the Filter Coefficients used to implement the five Frequency Bands at the specified Gain Setting for each band. This determines the numerator and denominator (coefficients) of the Filter Transfer function.
3. Implement and run the Filter processing on the audio input signal.
4. Validate the Frequency Response of the input and output (processed) audio signals with the MATLAB Scope Tool.
5. Listen to processed audio to evaluate the Filter Design with the Audio File Writer on PC.
6. Repeat Steps 2,3,4 and 5 for Filter Design and Validation until the desired result is obtained.
7. Configure Embedded Coder for Cortex-M Source Code Generation.
8. Generate and Export Floating-Point or Fixed-Point Source Code for Cortex-M Processor.
9. Implement and optimize the Cortex-M processor.

The workflow is covered in detail in the next sections.

3.2 Create Filter Coefficient and Processing Functions

For the Embedded Coder tool to export CMSIS DSP C/C++ source code, the top-level MATLAB script (see Figure 13. MATLAB Audio Equalizer Design) must call modular functions that implement filter coefficient generation and filter processing. The functions must be put into their own module script file with a callable and exportable function format.

For more information on Embedded Coder Code Generation from MATLAB Code and specifics on fine-grain control of the generated functions, see Embedded Coder Capabilities for Code Generation from MATLAB Code. It is recommended to take the time to review the MATLAB documentation on Embedded Coder. It can be difficult to get the code generation to work successfully when the codegen step is performed in Section 3.3 Export Filter Functions with Embedded Coder Tool. Some example tutorials can be found at codegen - Generate C/C++ code.

For this project, the Filter Coefficient design function prototype has a specific form, as shown in Figure 14. Audio EQ Filter Coefficient Design.

function [Num,Den] = genCoeffAudioEQ_float(sampleRate, gain)

Input arguments are sampleRate, and gain

With **return argument list** at the end of the function:

[**Num,Den**] = designParamEQ(N, gain, centerFreq, bandwidth,”sos”);

The function, designParamEQ(…) is the DSP System Toolbox filter design tool that creates the transfer function coefficients for the Second Order Sections (“sos”) used with the dsp.BiquadFilter system object.

Audio EQ is implemented as a series of five cascaded Biquads with Second Order Sections. It is similar to a Band pass filter but with more direct control over the filter response characteristics set by user parameters that calculate the transfer function coefficients. The filter parameters are Gain, Band Center Frequency, and Bandwidth (the inverse is Quality – Q). A high bandwidth is a low Q value, such as 0.707. A low bandwidth is a high Q value, such as 2.0.

The band center frequencies and bandwidths have been chosen so the user can easily notice the changes to the input signal when DSP processing is active. The low frequency in band 1 (140 Hz and lower) has a low bandwidth (high Q), so gain changes are focused on the bass guitar and bass drum instruments of an input signal. The high-frequency response (10 kHz and higher) in band 5 has a high bandwidth (low Q) so gain changes are noticeable in the upper vocal harmonics, cymbals, drums, and percussion instruments.

MATLAB embedded coder produces the best results (for example, exports source code with no errors) when the top-level MATLAB script is made up of callable modular functions that perform minimal functionality.

genCoeffAudioEQ_float.m

```
function [Num,Den] = genCoeffAudioEQ_float(sampleRate, gain)
% Calculate the 5 Band Audio EQ filter coefficients with DSP Toolbox using float32 data type
%
%#codegen
% Design a cascaded 2nd Order (SOS) equalizer centered at F1,F2,Fn ... Hz respectively,
% with gain in dB and Q-factor for a system running at sampleRate = 44.1kHz.
N    = [2,2,2,2,2];      % order for each filter band -> 2nd Order filter sections cascaded
F1 = 140;                % center freq in Hz - Band 1
W1 = F1/(sampleRate/2);  % center freq (as ratio of freq / nyquist)
Q1 = 2.0;                % Selectivity Q (Quality) - Band 1
BW1 = W1/Q1;             % Bandwidth (inverse of Q) - Band 1

F2 = 1000;               % center freq in Hz - Band 2
W2 = F2/(sampleRate/2);  % center freq (as ratio of freq / nyquist)
Q2 = 1.0;                % Selectivity Q (Quality) - Band 2
BW2 = W2/Q2;             % Bandwidth (inverse of Q) - Band 2

F3 = 2400;               % center freq in Hz - Band 3
W3 = F3/(sampleRate/2);  % center freq (as ratio of freq / nyquist)
Q3 = 1.0;                % Selectivity Q (Quality) - Band 3
BW3 = W3/Q3;             % Bandwidth (inverse of Q) - Band 3

F4 = 4800;               % center freq in Hz - Band 4
W4 = F4/(sampleRate/2);  % center freq (as ratio of freq / nyquist)
Q4 = 0.707;              % Selectivity Q (Quality) - Band 4
BW4 = W4/Q4;             % Bandwidth (inverse of Q) - Band 4

F5 = 10000;              % center freq in Hz - Band 5
W5 = F5/(sampleRate/2);  % center freq (as ratio of freq / nyquist)
Q5 = 0.707;              % Selectivity Q (Quality) - Band 5
BW5 = W5/Q5;             % Bandwidth (inverse of Q) - Band 5
%
centerFreq = [W1,W2,W3,W4,W5];      % center freq for each filter band (radians)
bandwidth  = [BW1,BW2,BW3,BW4,BW5]; % bandwidth of each band in (radians)
% Design the equalizers and obtain their B-Numerator and A-Denominator coefficients.
[Num,Den] = designParamEQ(N,gain,centerFreq,bandwidth,"sos");
end
```

Figure 14. Audio EQ Filter Coefficient Design

The Filter Processing function prototype has the form shown in Figure 15. Audio EQ Biquad Filter Function

function output = filtercmsis_float(input, num, den)

Input arguments are input (samples), and filter transfer function numerator and denominator

With **return argument output (samples)** at the end of the function:

output = filter(single(input), single(num), single(den));

The type-cast single(arg) forces the CMSIS DSP implementation to use single precision float (float32).

The Audio EQ filter processing function is implemented with the transfer function specified by input arguments num and den, which are the filter coefficients designed by the Filter Coefficient design function in Figure 14. Audio EQ Filter Coefficient Design.

filter = dsp.BiquadFilter(…) is the DSP System Toolbox filter that creates the Audio EQ as five bands using cascaded Second Order Sections (“sos”).

ModAudioFilterARMCortexM.m | genCoeffAudioEQ_float.m | filtercmsis_float.m

```
function output = filtercmsis_float(input,num,den)
% implement the function with DSP Toolbox filter with float32 data type
% and use Cortex M CMSIS DSP functions
%#codegen
persistent filter;
if isempty(filter)
    filter = dsp.BiquadFilter( SOSMatrixSource="Input port", ScaleValuesInputPort=false);
end
% typecast filter input args as single precision float to use CMSIS DSP
output = filter(single(input),single(num),single(den));
end
```

Figure 15. Audio EQ Biquad Filter Function

3.3 Export Filter Functions with Embedded Coder Tool

The MATLAB script is in Figure 16. MATLAB Code Gen - Floating Point for Audio Equalizer shows how to create an Embedded Coder configuration to generate source code for the Cortex-M processor using single precision float implementation. The **cfg.dialog** step opens a Configuration Dialog at run-time to review the settings for code generation. Review the information and click OK to proceed with code generation.

The **codegen** commands create the Cortex-M CMSIS DSP-based C language source code implementations in single precision float for the following functions:

For the Filter Coefficient design, the implementation is created with function **genCoeffAudioEQ_float(sampleRate, gain)** with single float args **sampleRate**, and **gain** for each frequency band and returns the **num** and **den** of the filter transfer function coefficients.

For the Filter processing, the implementation is created with the function

filtercmsis_float(input, num, den) with single float args **input** samples, **num** and **den** coefficients, and returns **output** samples.

ModAudioFilterARMCortexM.m | genCoeffAudioEQ_float.m

```
47
48  %% Configure for ARM Cortex M CMSIS DSP Source Code Generation
49  % Create a code generation configuration object for use with codegen when generating a C/C++ static library.
50  %
51  % I used EmbeddedCodeConfig to generate Script which is pasted here after setting Hardware->Devices to use
52  % HW ARM Cortex M and GNU GCC compiler tool chain.
53  cfg = coder.config( "lib", "ecoder", true );
54  cfg.CodeReplacementLibrary = "ARM Cortex-M";
55  cfg.Toolchain = "GNU Tools for ARM Embedded Processors";
56  cfg.HardwareImplementation.ProdHWDeviceType = "ARM Compatible->ARM Cortex-M";
57  cfg.HardwareImplementation.TargetHWDeviceType = "Generic->Custom";
58  cfg.GenCodeOnly = true;
59
60  % Open the Custom code panel of the configuration dialog and verify the settings.
61  cfg.dialog
62
63  %%  Generate Code for Cortex ARM M Floating Point versions of Filter and EQ Coefficient Generation
64  % Generate C code for the MATLAB function
65  % filtercmsis_float.m
66  %
67  % Typecast the input args to single (precision float) to get CMSIS DSP
68  % source code generated.
69  %
70  % Case the input datatype as single precision float
71  codegen filtercmsis_float -args {single(u1),single(Num),single(Den)} -config cfg -report
72  % When code generation finishes successfully, click View report to display the code generation report.
73  %
74  % See genCoeffAudioEQ_float.c file
75  codegen genCoeffAudioEQ_float -args {single(sampleRate),single(gain)} -config cfg -report
```

Figure 16. MATLAB Code Gen - Floating Point for Audio Equalizer

```
77    %% Generate Code for Cortex ARM M Fixed-Point versions of Filter and EQ Coefficient Generation
78    % Generate fixed-point C code for the MATLAB function filtercmsis_fixedpt_q15.m
79    % cast the Input data and Numerator and Denominator coeffs as
80    % Fixed-Point Numeric objects -> fi(v,s,w,f)
81    % v = data to convert
82    % s = Signedness : true
83    % w = Word length in bits : 16     => 16-bit
84    % f = Fraction length in bits : 15 => q15
85    codegen filtercmsis_fixedpt_q15 -args {fi(u1, true, 16, 15),fi(Num),fi(Den)} -config cfg -report
86
87    % See genCoeffAudioEQ_fixedpt.c file
88    % Design and Generate Audio EQ filter coefficients in Fixed-Point datatype
89    % [Num,Den] = designParamEQ(N,gain,centerFreq,bandwidth,"sos") function can not be converted to
90    % Fixed-Point
91    % A work around is to call with single precision args and convert
92    % [Num,Den] coefficients to fixed-point q14. Then pass to filter
93    % filtercmsis_fixedpt_q15() source code on ARM Cortex M
94    codegen genCoeffAudioEQ_fixedpt -args {single(sampleRate),single(gain)} -config cfg -report
95
96    %% Release Audio Toolbox Resources
97    release(deviceWriter)
98    release(fileReader)
99    release(scope)
100
```

Figure 17. MATLAB Code Gen - Fixed-Point for Audio Equalizer

Although we will focus on the floating-point implementation in the Audio EQ project, the fixed-point codegen command is shown here for those who wish to investigate fixed-point filters.

The MATLAB script is in Figure 17. MATLAB Code Gen - Fixed-Point for Audio Equalizer shows the codegen commands to create the Cortex-M CMSIS DSP-based C language source code implementations in fixed-point (q15, 16-bit integer) for the following functions:

For the Filter Coefficient design, **genCoeffAudioEQ_fixedpt**(**sampleRate, gain**) calls the filter design function **designParamEQ**(..), but it does not have an equivalent in the MATLAB toolbox that can generate a set of fixed-point coefficients.

So, as a workaround, the floating-point version must be called to generate the **num** and **den** coefficients, and then it is required to convert to fixed-point values to call filter processing function **filtercmsis_fixedpt_q15(input, fi(Num), fi(Den))** where the input arguments are fixed-point values.

This brings up an important point: not all MATLAB signal processing and filter design Toolboxes provide Cortex-M embedded coder support for both floating-point and fixed-point source code implementations. Consult the MATLAB documentation first to determine the coder support for the Toolbox and Filter System objects you wish to use.

3.4 Understanding the Generated Source Code

The MATLAB code generation script is in Figure 16. MATLAB Code Gen - Floating Point for Audio Equalizer and Figure 17. MATLAB Code Gen - Fixed-Point for Audio Equalizer produces the source code output with folder structure shown in Figure 18. MATLAB Code Generation Output.

The folder contains the MATLAB *.m files, which include the main MATLAB script ModAudioFilterARMCortexM.m that calls the exportable functions defined in **genCoeffAudioEQ_float.m** and **filtercmsis_float.m**. We discussed these functions previously, as shown in Figure 14. Audio EQ Filter Coefficient Design and Figure 15. Audio EQ Biquad Filter Function.

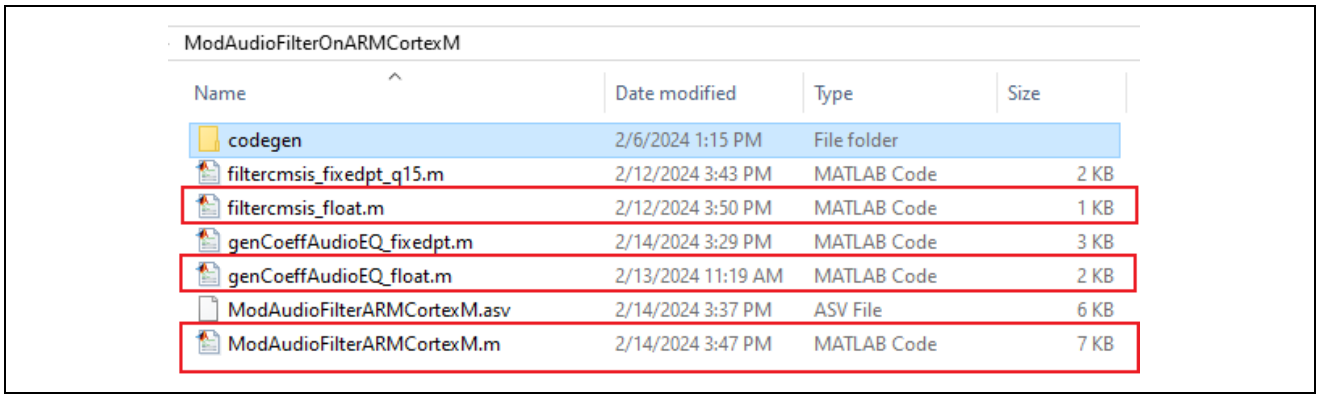


Figure 18. MATLAB Code Generation Output

In Figure 19. MATLAB Code Generation – Codegen Library, the codegen library folder contains the CMSIS DSP C language source code for the exportable functions. Notice that each **codegen** command called in the MATLAB script from Figure 16. MATLAB Code Gen - Floating Point for Audio Equalizer and Figure 17. MATLAB Code Gen - Fixed-Point for Audio Equalizer produces a separate folder with the source code for the exported function.

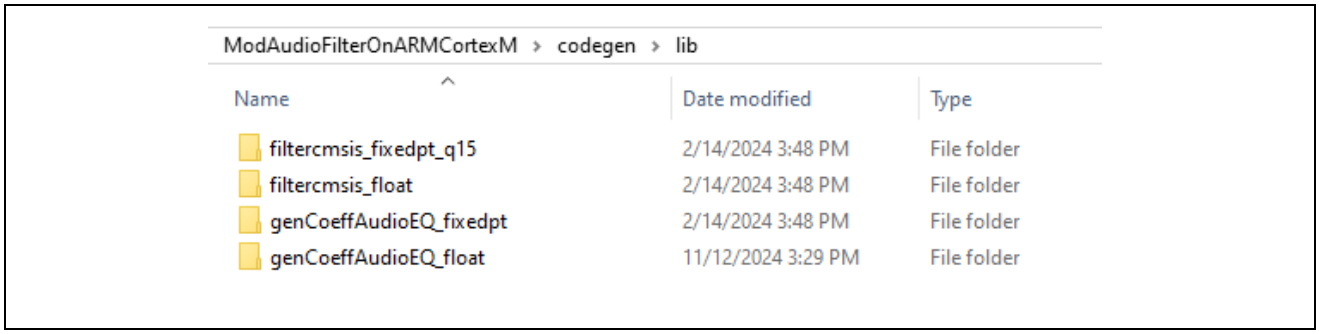


Figure 19. MATLAB Code Generation – Codegen Library

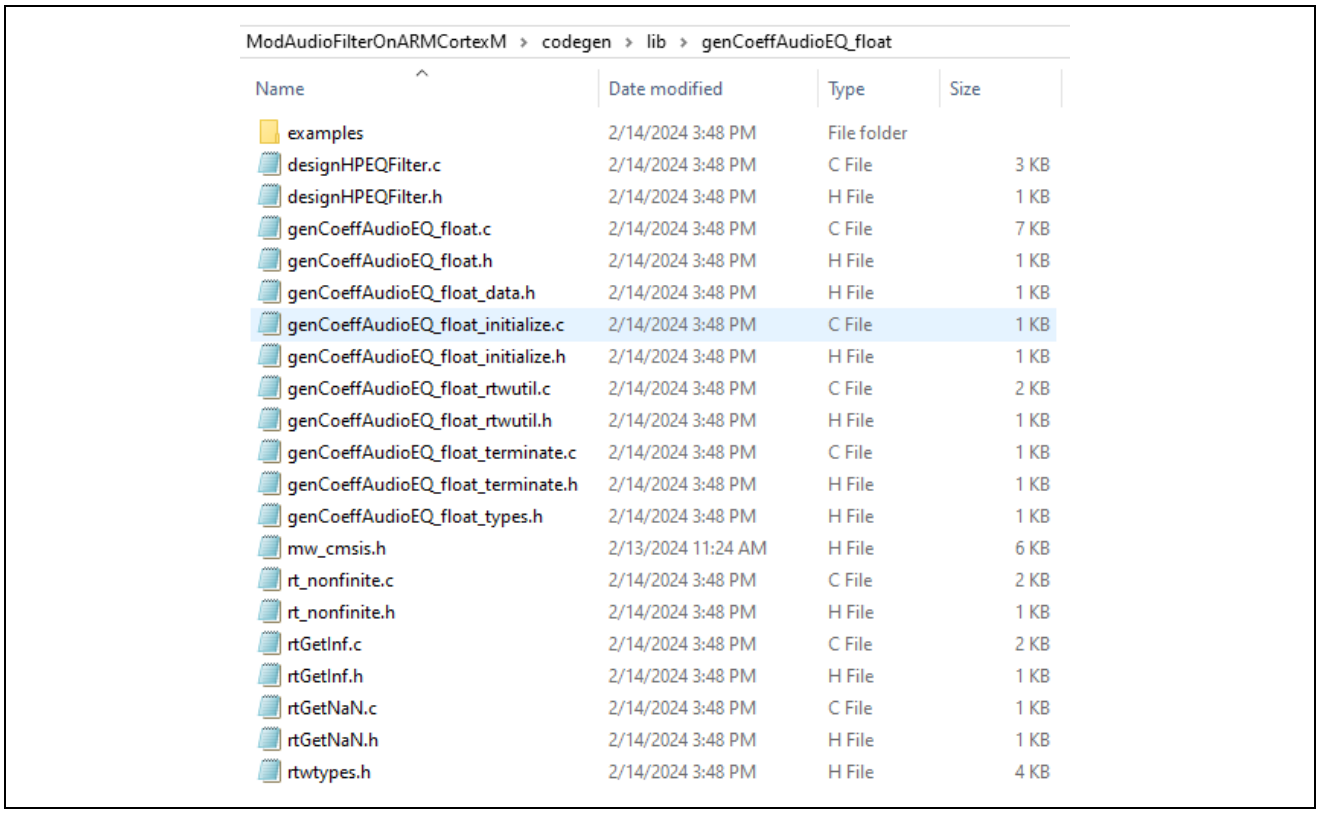


Figure 20. MATLAB Code Generation – Generate Filter Coefficients

We will focus on the floating-point implementations in the Audio EQ project. Figure 20. MATLAB Code Generation – Generate Filter Coefficients shows, the source code files to generate the Filter Coefficients and Figure 21. MATLAB Code Gen – Filter shows the files to implement the Filter Processing.

Each source code directory contains an **examples folder** with main.c and main.h files that show a recommendation of how the source code should be implemented by your target application. The main.c is a starting point for porting this project to the RA8D1 version of the Audio EQ.

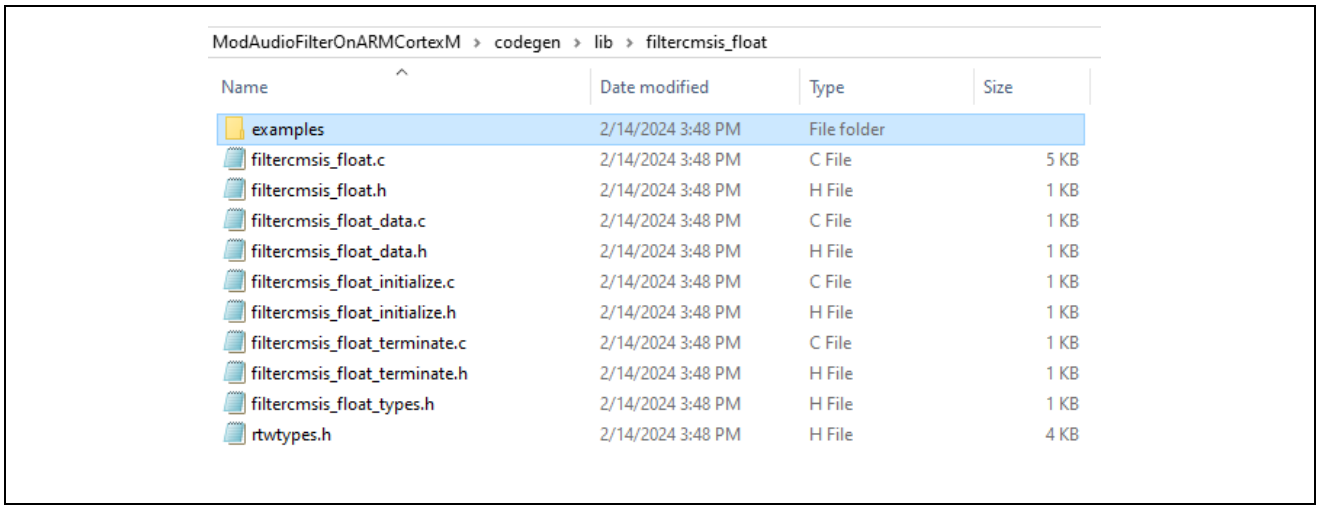


Figure 21. MATLAB Code Gen – Filter

See Figure 22. Examples - Filter Coefficient Call and Figure 23. Examples - Filter Processing Call for the Coefficient Generation and Filter Processing examples.

```
int main(int argc, char **argv)
{
  (void)argc;
  (void)argv;
  /* The initialize function is being called automatically from your entry-point
   * function. So, a call to initialize is not included here. */
  /* Invoke the entry-point functions.
You can call entry-point functions multiple times. */
  main_genCoeffAudioEQ_float();
  /* Terminate the application.
You do not need to do this more than one time. */
  genCoeffAudioEQ_float_terminate();
  return 0;
}

/*
 * Arguments    : void
 * Return Type  : void
 */
void main_genCoeffAudioEQ_float(void)
{
  float Num[15];
  float Den[10];
  float fv[5];
  /* Initialize function 'genCoeffAudioEQ_float' input arguments. */
  /* Initialize function input argument 'gain'. */
  /* Call the entry-point 'genCoeffAudioEQ_float'. */
  argInit_1x5_real32_T(fv);
  genCoeffAudioEQ_float(argInit_real32_T(), fv, Num, Den);
}
```

genCoeffAudioEQ_float.c

```
 2  * File: genCoeffAudioEQ_float.c
 7
 8  /* Include Files */
 9  #include <genCoeffAudioEQ_float.h>
10  #include <math.h>
11  #include <designHPEQFilter.h>
12  #include <genCoeffAudioEQ_float_rtwutil.h>
13  #include <rt_nonfinite.h>
14
15
16  /* Function Definitions */
17  /*
18   * Calculate the 5 Band Audio EQ filter coefficients with DSP Toolbox using
19   * float32 data type
20   *
21   *
22   * Arguments    : float sampleRate
23   *                const float gain[5]
24   *                float Num[15]
25   *                float Den[10]
26   * Return Type  : void
27   */
28  void genCoeffAudioEQ_float(float sampleRate, const float gain[5], float Num[15],
29                             float Den[10])
30  {
```

Figure 22. Examples - Filter Coefficient Call

In Figure 22. Examples – Filter Coefficient Call**,** the array of gain values for each frequency band is named **float fv[5]**. The variable was renamed to float gain[5] in the EK-RA8D1 ported version to improve readability in the source code. One point to remember is that the exported source code is a starting point for the target application and may require additional optimizations in the final application.

Details on optimizing and improving readability are covered in Section 3.5 Modify Source Code for Best Performance on RA8 MCUs

In Figure 23. Examples - Filter Processing Call**,** numerator array **float fv1[15],** and denominator array **float fv2[10]** were renamed to num[15] and den[10]. Later, it was determined to improve processing latency that all filter coefficient initialization should be performed once in a custom function **filtercmsis_biquad_init**(const float num[15], const float den[10]).

So, the call to **filtercmsis_float**(…) was changed to remove num[15] and den[10] in the function argument list. Since this is a stereo EQ, there are two instances required to maintain the filter state: **filtercmsis_float_L**(input[32], output[32]) and **filtercmsis_float_R(**input[32], output[32) are called in the EK-RA8D1 ported version.

Note: The input audio samples array **float fv[75]** has size = 75 samples. This value was changed to 32 samples in the EK-RA8D1 ported version to better align with the maximum number of samples in a I2S Receive (DTC) transfer payload.

```
int main(int argc, char **argv)
{
  (void)argc;
  (void)argv;
  /* The initialize function is being called automatically from your entry-point
   * function. So, a call to initialize is not included here. */
  /* Invoke the entry-point functions.
You can call entry-point functions multiple times. */
  main_filtercmsis_float();
  /* Terminate the application.
You do not need to do this more than one time. */
  filtercmsis_float_terminate();
  return 0;
}

/*
 * Arguments    : void
 * Return Type  : void
 */
void main_filtercmsis_float(void)
{
  float fv[75];
  float output[75];
  float fv1[15];
  float fv2[10];
  /* Initialize function 'filtercmsis_float' input arguments. */
  /* Initialize function input argument 'input'. */
  /* Initialize function input argument 'num'. */
  /* Initialize function input argument 'den'. */
  /* Call the entry-point 'filtercmsis_float'. */
  argInit_75x1_real32_T(fv);
  argInit_3x5_real32_T(fv1);
  argInit_2x5_real32_T(fv2);
  filtercmsis_float(fv, fv1, fv2, output);
}
```

filtercmsis_float.c

```
56
57  // I removed num,den since filtercmsis_biquad_init(num,den) called once to init biquad filter
58  void filtercmsis_float_L(const float input[32], float output[32])
59  {
60      /*
61      * I Modified the original Matlab codegen source code by moving biquad filter init to
62      * separate function filtercmsis_biquad_init(const float num[15],const float den[10])
63      */
64
```

Figure 23. Examples - Filter Processing Call

3.5 Modify Source Code for Best Performance on RA8 MCUs

After analyzing the MATLAB source code generated with the provided example functions in Figure 22. Examples - Filter Coefficient Call and Figure 23. Examples - Filter Processing Call, it was determined that modifications would be required for the best performance on the RA8 MCU. A multi-threaded FreeRTOS project has been created in e^2 studio to use the generated MATLAB source code. A thread called DSP Thread is used to implement the Audio EQ processing with the ARM CMSIS DSP Library. The entry function **dsp_thread_entry**(void *pvParameters) is implemented to handle DSP filter coefficient and memory initialization as well as processing with **DSP_Process_Data**(…) which runs continuously to process audio data arriving from the I2S connected Audio Codec. See Figure 24. DSP Thread.

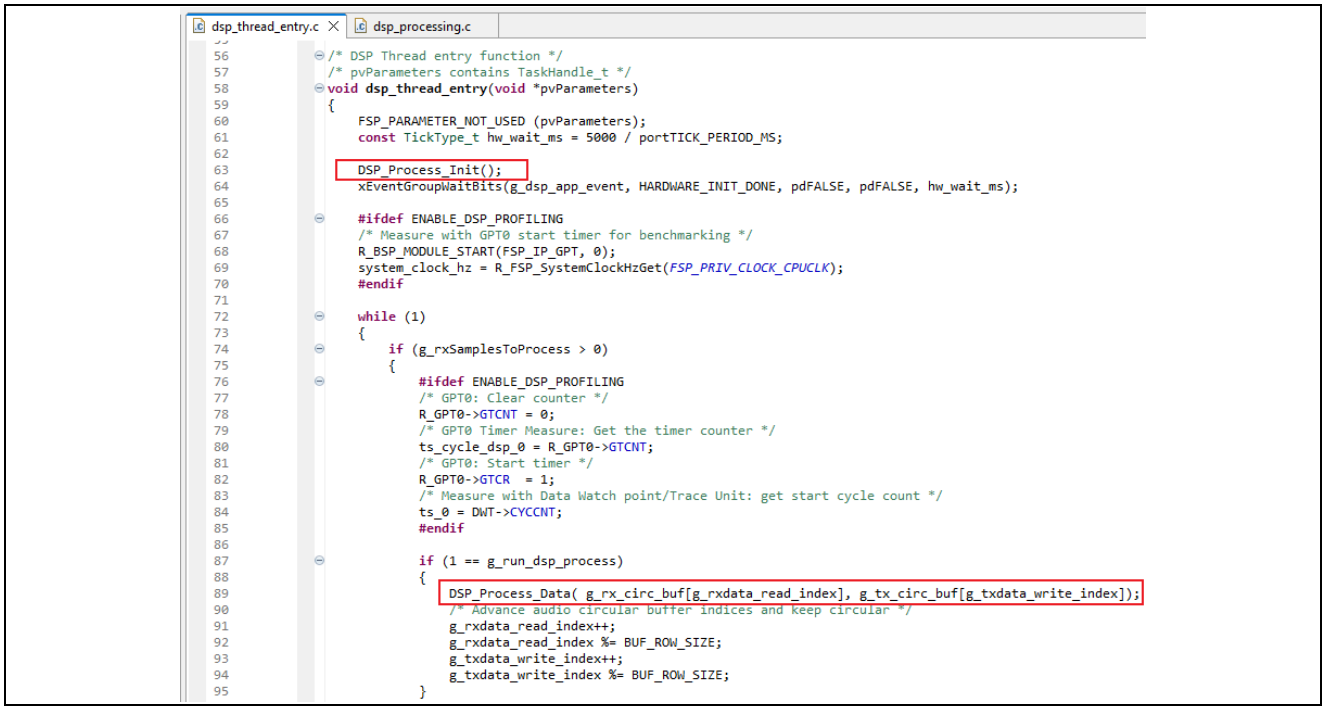


Figure 24. DSP Thread

In the MATLAB script, filter coefficient design and generation are called every time the filter processing is run on a group of audio samples. However, it was determined that the EK-RA8D1 embedded system processing latency could be improved by running the filter coefficient generation and Biquad filter initialization once with **DSP_Process_Init**() before calling the filter processing **DSP_Process_Data**(…), which runs continuously on the incoming audio. DSP_Process_Init() should be called again when any of the filter parameters change such as Gain. For example, in the case of a user parameter control with LCD is added to the project.

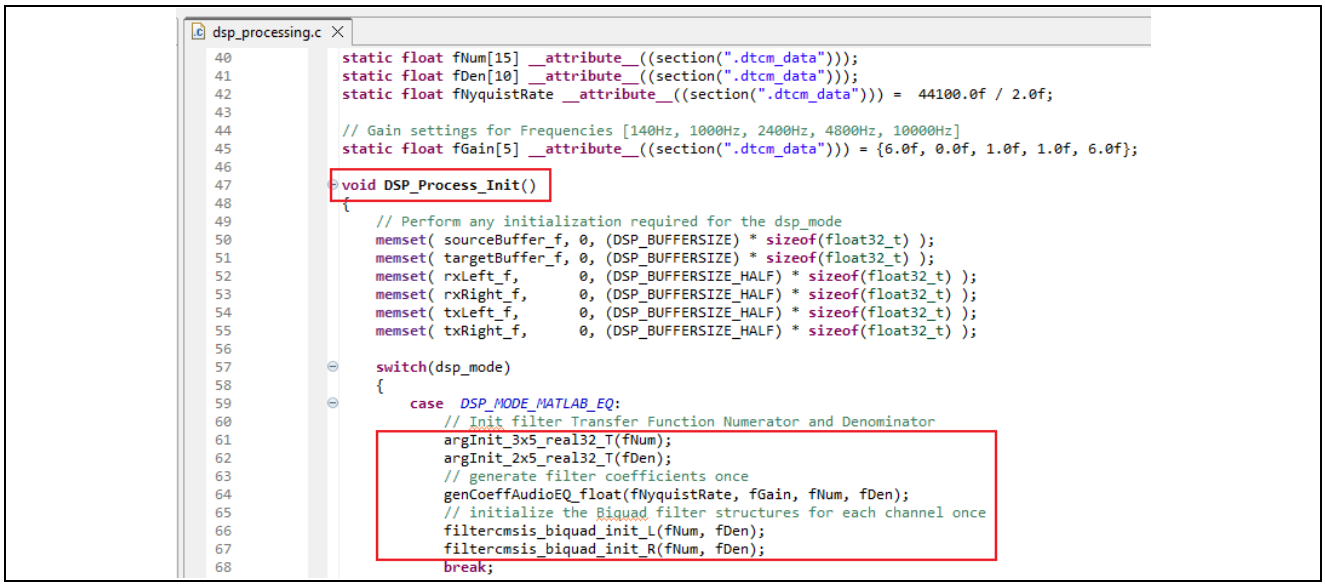


Figure 25. DSP Process Init

As seen in Figure 25. DSP Process Init, **DSP_Process_Init**() is implemented by calling the source code functions generated by MATLAB.

The functions **filtercmsis_biquad_init_L**(fNum, fDen) and **filtercmsis_biquad_init_R**(fNum, fDen) are custom functions added to initialize the CMSIS Biquad filter with **arm_biquad_cascade_df2T_init_f32**(…).

The next section covers how to enable the DSP process latency measurement.

3.6 DSP Process Latency Measurement

3.6.1 How to Enable Latency Measurement

Support has been added to the Audio EQ project to enable execution time (latency) measurement on the Audio Codec data handler and DSP process. In this section, the term latency refers to the time it takes to process a group of samples with a filter or send and receive a group of samples with the Audio Codec data handler. This allows profiling before and after changes are made to the Compiler optimization settings and DSP processing implementation.

As observed in Figure 26. Enable Latency Measurement, to enable latency measurement use #define **ENABLE_CODEC_PROFILING** and

#define **ENABLE_DSP_PROFILING** in files codec_thread_entry.c and dsp_thread_entry.c.

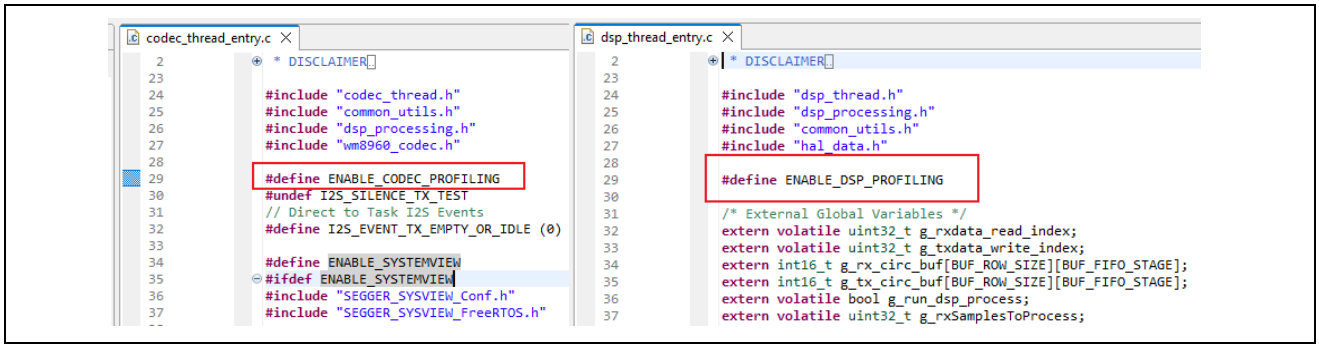


Figure 26. Enable Latency Measurement

3.6.2 Adding Support Code to Measure Latency

As shown in Figure 27. Store DWT and GPT Before and After Calling the DSP Process Function, the DWT counter (DWT->CYCCNT) and GPT timer count are saved before and after calling the Audio Codec handler **R_SSI_WriteRead**(…) or DSP processing function.

When running a debug session with e^2 studio IDE, the DWT measurement is valid but is not for operating the board in standalone mode. When the Command Line Interface is used, the GPT timer measurement is

always valid for debug or standalone operation. The latency values are displayed in the Command Line Interface (CLI) with menu options 3 and 4. See Section 2.4 Download and Operate Project for more details.

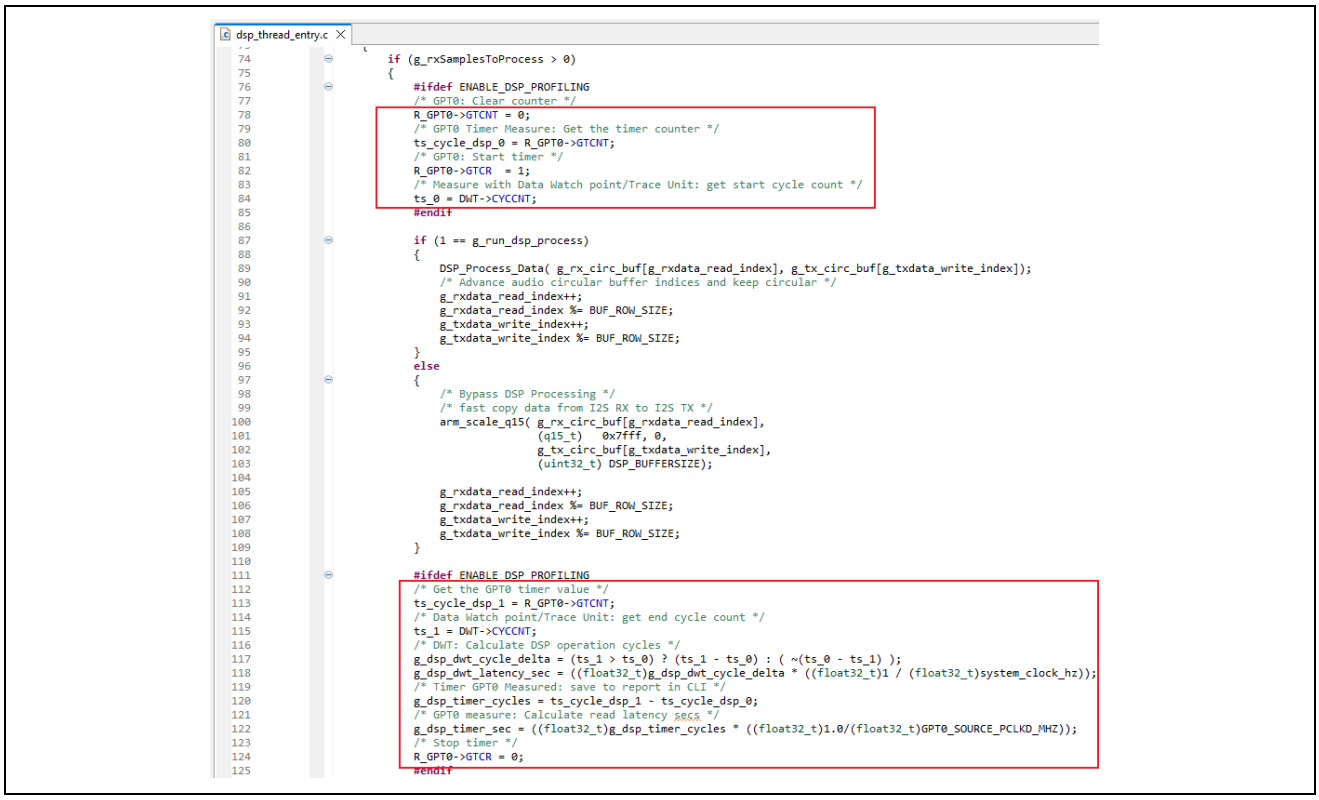


Figure 27. Store DWT and GPT Before and After Calling the DSP Process Function

As observed in the Figure 27. Store DWT and GPT Before and After Calling the DSP Process Function, the execution time of the DSP processing function call is calculated based on DWT or GPT timer cycles and the MCU system or timer clock frequency using the following formula:

Time = DWT cycles x (1 / System Clock)
Time = GPT0 Timer cycles x (1 / Timer Clock)

Note:

- DWT cycles are the count of MCU execution cycles.
- R_GPT0 -> GTCNT contains the count of GPT0 timer cycles
- For the EK-RA8D1, the system clock is CPUCLK set to 480 MHz in this application project.
- For the EK-RA8D1, the GPT0 timer clock source is PCLKD set to 120 MHz

3.7 Changing Segger SystemView Support

A system level perspective of the Audio EQ project can be obtained by using support for Segger SystemView. The tool is integrated into the project and enabled by default to show the process thread priorities, timing, and resource utilization.

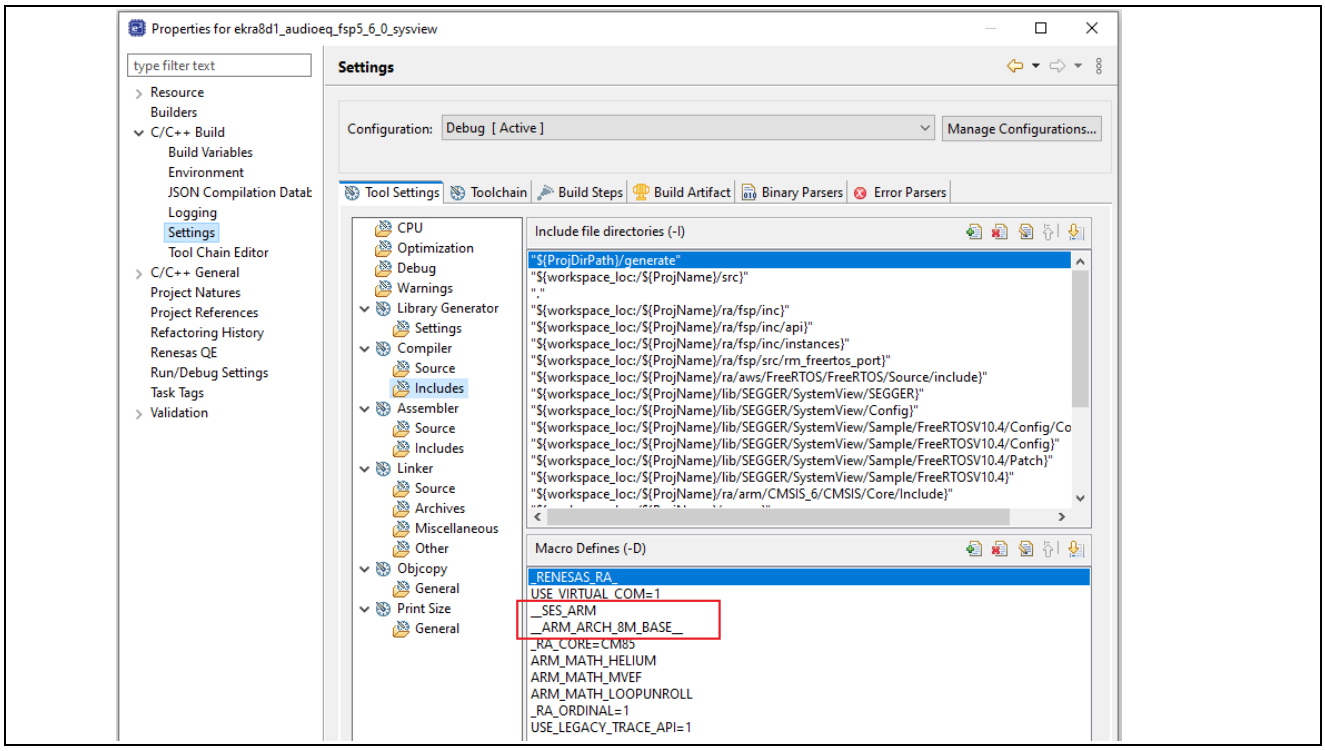


Figure 28. Compiler Macro Defines for SystemView

As observed in Figure 28. Compiler Macro Defines for SystemView, Figure 29. Codec Thread – Defines for SystemView and Figure 30. Codec Thread - Enable Trace for SystemView, Segger SystemView support is enabled to profile the multi-threaded processes in the Audio EQ project. The Macro Defines (-D) __SES_ARM and __ARM_ARCH_8M_BASE__ are used to set the SystemView configuration options required for the RA8 MCU (CM85 core).

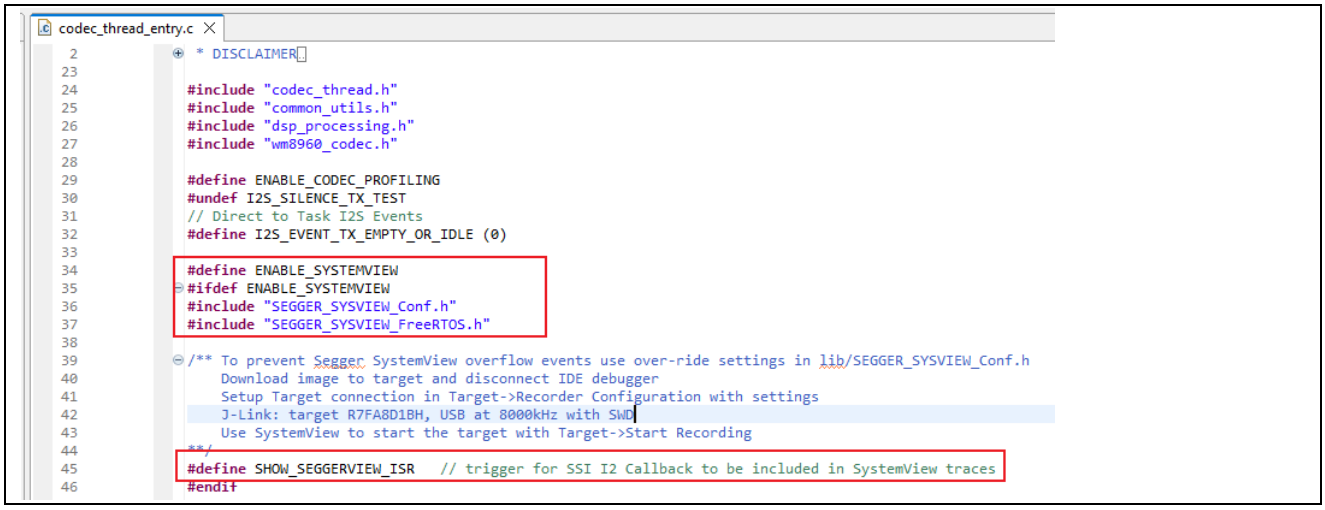


Figure 29. Codec Thread – Defines for SystemView

See Figure 29. Codec Thread – Defines for SystemView and Figure 30. Codec Thread - Enable Trace for SystemView, where SystemView support is enabled in the source code. All SystemView support can be disabled with **#undef ENABLE_SYSTEMVIEW**.

Since it can generate a large amount of trace buffer data, the Audio Codec handler ISR trace can be disabled with **#undef SHOW_SEGGERVIEW_ISR** as required.

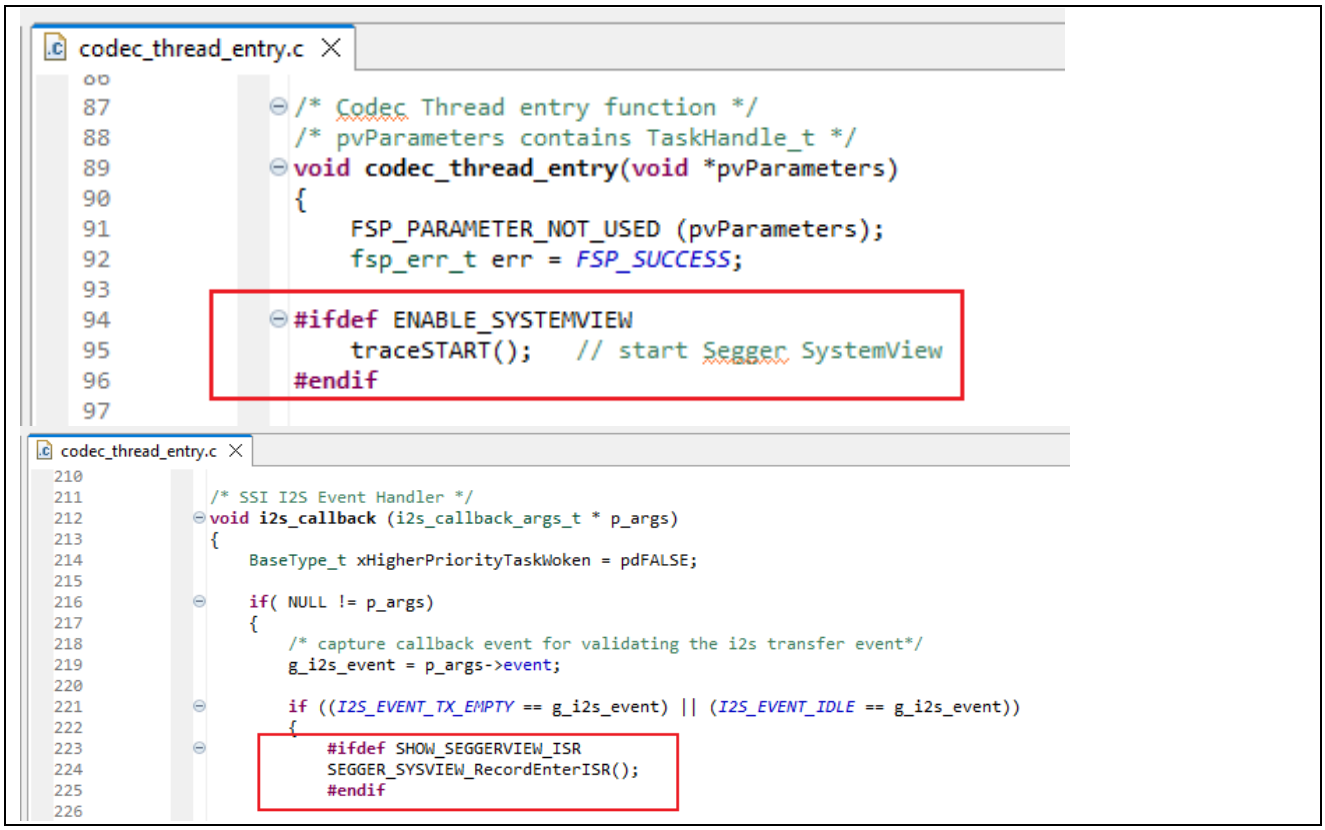


Figure 30. Codec Thread - Enable Trace for SystemView

3.7.1 Integration of Segger SystemView into Example Project

Segger SystemView tool was integrated into the Audio EQ example project running on FSP v5.6 with FreeRTOS v10.6.1

Use the following steps to integrate the SystemView source code:

- Follow the instructions to integrate the SystemView source code with the project from FreeRTOS with SystemView - SEGGER Wiki.
- Use the source code from SEGGER - The Embedded Experts - Downloads - SystemView
- Choose SystemView Target Sources v3.58 and use the Sample -> FreeRTOSV10.4 even though the FreeRTOS version in FSP v5.6.0 is v10.6.1.
- Segger Tech Support recommends V10.4 example for use with FSP v5.6 and FreeRTOS V10.6.1 with EK-RA8D1 Core CM85.
- Add #include "SEGGER_SYSVIEW_FreeRTOS.h" to all FreeRTOS/Source files that have occurrences of #include "FreeRTOS.h".
- Be sure to place #include "SEGGER_SYSVIEW_FreeRTOS.h" before #include "FreeRTOS.h" in all the affected files. See Figure 31. SystemView Integration

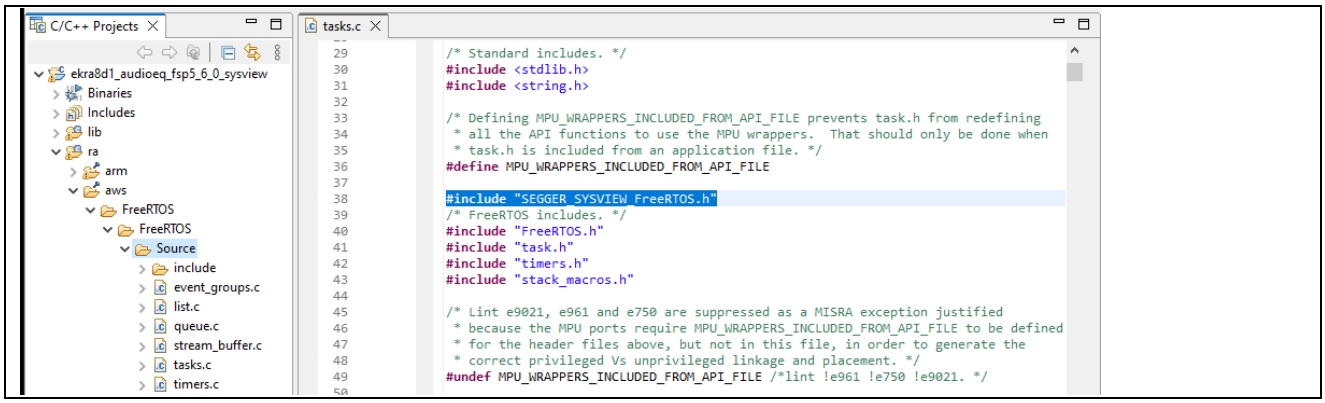


Figure 31. SystemView Integration

To build Segger SystemView with LLVM 18.1.3, the directories under /lib/SEGGER/SystemView/... must be excluded from the build since the files are not required in the build:

- project/lib/SEGGER/SystemView/Sample/RTT
- project/lib/SEGGER/SystemView/SEGGER/Syscalls

If the Audio EQ example project is ever updated to use a newer version of FSP with FreeRTOS, the SystemView source code must be reintegrated into the example project by following the steps covered in Section 3.7 Codec Thread – Defines for SystemView.

Additional coverage of SystemView is provided in Section 4.6 Using Segger SystemView with instructions on how to measure CPU Utilization for the Audio EQ project.

4. Audio Equalizer Software Architecture

The next sections detail the software architecture for the Audio EQ project. There are several sub-systems that handle moving the audio data from the Audio Codec, storing the data, processing the data with the DSP filters, and providing the Command Line Interface to control the Audio EQ and measure latency.

Central to the architecture of the system is the Audio Codec. Since the Audio Codec depends on the buffering sub-system, the Audio Circular Buffer is introduced first.

4.1 Audio Circular Buffers

As seen in Figure 32. Audio Circular Buffers, the Audio EQ, requires a data buffering sub-system that receives and transmits data with the Audio Codec connected to the SSI I2S Master Interface.
The Circular Buffer scheme buffers about 743 milliseconds of audio data. This permits the CMSIS DSP Filter to obtain data for processing and prevents I2S RX and TX FIFO underflow and overflow conditions by decoupling the audio codec interface from the audio DSP processing task.
The I2S sample rate is 44.1 kHz with stereo data @ 16-bit sample data per channel. So, the RX and TX Circular Buffer provides 32x1024 of 16-bit data = 32,768 samples of 16-bit data per channel. At 44100 samples/sec, this provides (32768 samples) / (44100 samples/sec) = 0.743 secs of buffered audio data per channel.

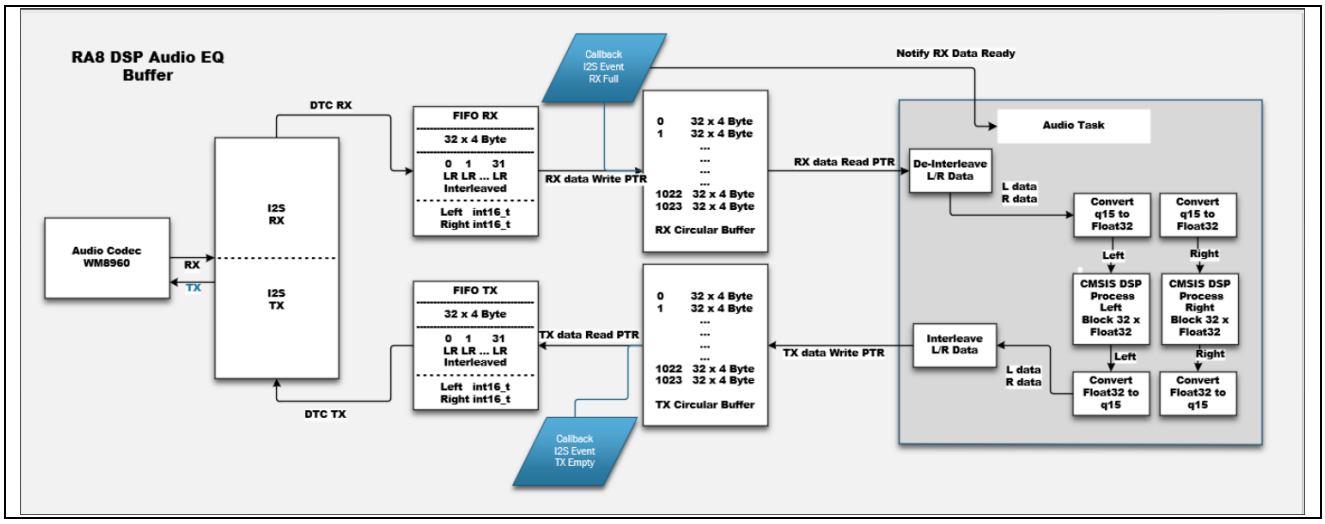


Figure 32. Audio Circular Buffers

The maximum length of the RX and TX Circular Buffers can be scaled as required. However, consult the total amount of internal or external memory available for the RA8 MCU.

dsp_processing.h

```
2   * DISCLAIMER
23
24  #ifndef DSP_PROCESSING_H_
25  #define DSP_PROCESSING_H_
26
27  #include "hal_data.h"
28
29  /* Defines */
30  #define BUF_ROW_SIZE        1024
31  #define BUF_FIFO_STAGE      64
32  #define DSP_BUFFERSIZE      64
33  #define DSP_BUFFERSIZE_HALF 32
```

codec_thread_entry.c

```
71
72  int16_t g_rx_circ_buf[BUF_ROW_SIZE][BUF_FIFO_STAGE] BSP_ALIGN_VARIABLE(32) BSP_PLACE_IN_SECTION(".nocache");
73  int16_t g_tx_circ_buf[BUF_ROW_SIZE][BUF_FIFO_STAGE] BSP_ALIGN_VARIABLE(32) BSP_PLACE_IN_SECTION(".nocache");
74
75
76  /* Local Variables */
77  static uint32_t rxdata_write_index = 0;
78  static uint32_t txdata_read_index  = 0;
```

Figure 33. Audio Circular Buffers – Source Code

If the total audio input-to-output latency of the system must be reduced, then the buffer size can be decreased. Refer to Figure 33. Audio Circular Buffers – Source Code in the source code for #defines **BUF_ROW_SIZE** and **BUF_FIFO_STAGE** that set the Circular Buffer size.

If the block size of data (a set of samples) processed by the CMSIS Filter in one function call must be increased, then the max length of the buffer may be increased. The system input-to-output latency and CMSIS Filter processing time can be balanced by adjusting the Circular Buffer maximum length, and the CMSIS Filter processing block size. In Audio EQ, the DSP processing is performed on a block of 32 samples each for left-channel and right-channel audio every time the DSP filter function is called.

The Audio Codec I2S Callback Event handles filling the RX Circular Buffer with data received from the DTC FIFO RX. Both read/write pointers wrap back to offset 0 when the end of the buffer is reached.
The audio task is notified when RX data is present in the buffer and ready for processing.
The I2S Callback Event TX FIFO Empty handles the dequeuing of the TX Circular Buffer and filling the DTC FIFO TX.
In Figure 33. Audio Circular Buffers – Source Code, the #define **BUF_FIFO_STAGE 64** was set to match the maximum number of bytes that can be transferred by the DTC FIFO in one transaction**.** The DTC FIFO left- and right-channel data is interleaved, giving a total of 64 x 2 bytes = (left: 32 x 2 bytes) + (right: 32 x 2 bytes).

In Figure 32. Audio Circular Buffers, the q15 to Float32 data conversion blocks use the fast CMSIS DSP Library functions.
The De-Interleave and Interleave blocks handle converting the serialized I2S format (left channel then right channel) data to two parallel streams of channel processing with the CMSIS DSP functions.
In this project, CMSIS DSP functions process one channel at a time, so two function calls are required to handle stereo audio.

4.2 Audio Codec Thread

In conjunction with the audio circular buffers, the Audio Codec thread is responsible for receiving and transmitting the audio data from the I2S-connected Audio Codec. The DSP processing thread picks up the data stored in the receive circular buffer, processes one block of data (left channel: 32 samples of 16-bit data, right channel: 32 samples of 16-bit data), and stores the results back to the transmit circular buffer.

4.2.1 I2S Stack Settings

As observed in Figure 34. FSP Configuration - I2S Stack Settings, the FSP configuration for the I2S Master stack settings is shown. The callback i2s_callback implemented in codec_thread_entry.c is used to receive and transmit the audio data that is stored in the Audio Circular Buffers. See Section 4.1 Audio Circular Buffers and Section 4.2.3 I2S Callback for more information.

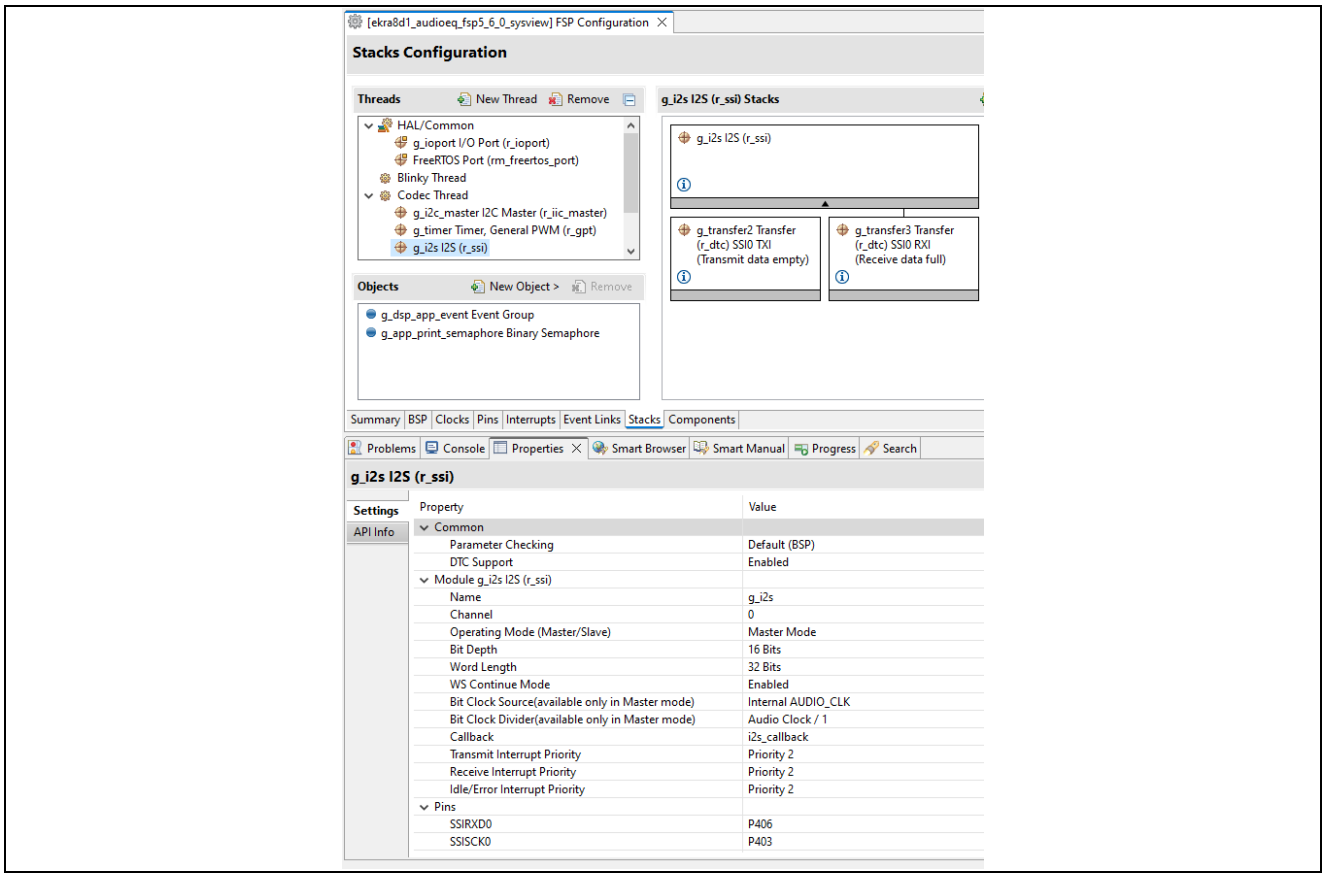


Figure 34. FSP Configuration - I2S Stack Settings

The I2S Master clock source for the Audio Codec is the **Internal AUDIO_CLK** which is provided by the General PWM Timer running on channel 2.

Note in the RA8 MCU series, only the GTIOC2A output or AUDIO_CLK input pin can be used as AUDIO_CLK. Refer to the Hardware User's Manual, as it may vary depending on the MCU series. The details for the I2S Audio Clock are covered in the next section.

The I2S Master bit depth is 16-bit per sample, and the Word Length is 32-bit since one frame of data consists of interleaved left and right samples. 1 x Left sample (16) + 1 x Right sample (16) = 32-bits.

In the example project, I2S DTC-supported transfers are enabled, as observed in Figure 34. FSP Configuration - I2S Stack Settings. DTC support is optional and can be disabled. With DTC support enabled, the DSP processing latency is improved. For example, with DTC disabled, the Audio EQ takes 9.8 µs to

process 32 samples of stereo data. With DTC enabled, the latency is 9.64 µs to process 32 samples of stereo data. A percentage change of -1.63% is not a significant decrease, but for large amounts of data processing, and percent decrease helps with improving processing latency.

Use Segger SystemView to see the effect on the system throughput and MCU with DTC transfers enabled and disabled. To disable DTC support, see Figure 35. FSP Configuration - I2S DTC Support Disabled.

For more information, see Section 4.6 Using Segger SystemView.

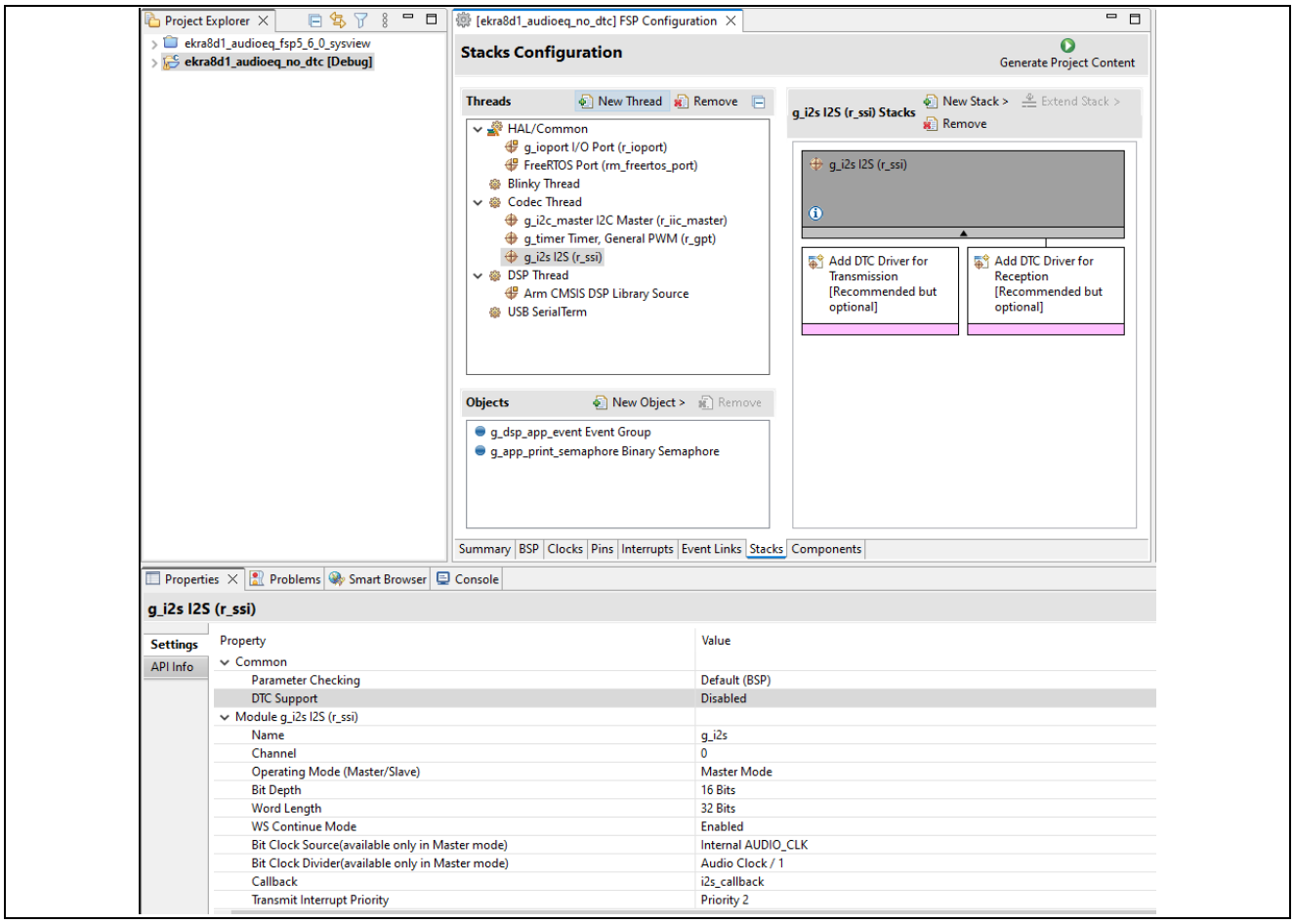


Figure 35 FSP Configuration - I2S DTC Support Disabled

4.2.2 I2S Audio Clock

In Figure 36. FSP Configuration GPT Timer Settings - I2S Audio Clock, the FSP configuration is shown for the General PWM Timer, which is being used as the internal I2S Audio Clock source.

The bit rate of the Timer is used to clock the interleaved stereo channel data of 16-bit per channel sample at 44100 samples/sec.

Timer Bit Rate: 1411200 bits/sec = 44100 samples/sec x 16-bits/channel-sample x 2 channels

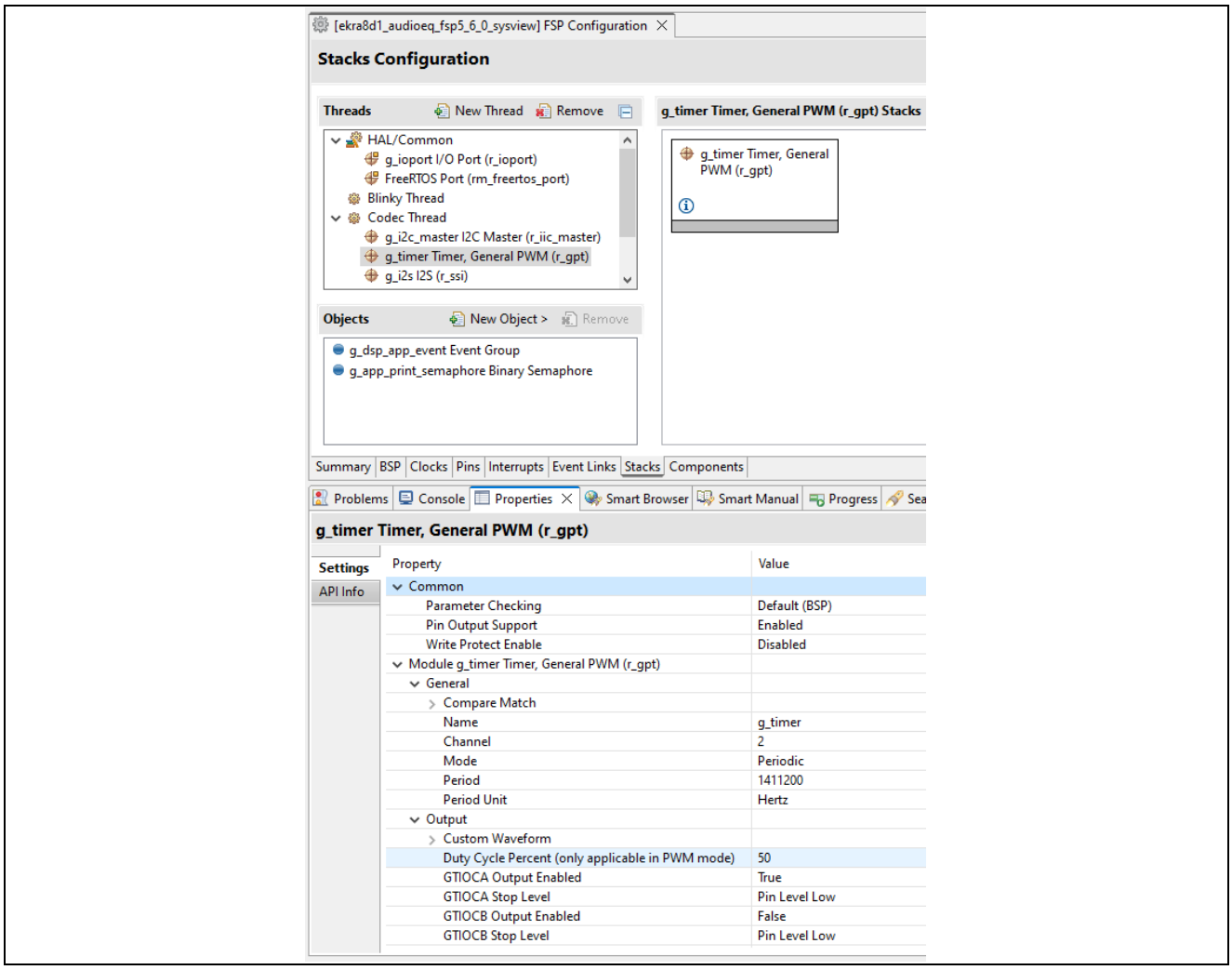


Figure 36. FSP Configuration GPT Timer Settings - I2S Audio Clock

4.2.3 I2S Callback

In Figure 37. Codec Thread - I2S Callback, the I2S Master callback function **i2s_callback**(…) is implemented in codec_thread_entry.c. It is used to receive and transmit the audio data stored in the Audio Circular Buffers. For I2S Events **I2S_EVENT_TX_EMPTY** and **I2S_EVENT_IDLE**, the DTC RX FIFO is read and written to the Receive Audio Circular Buffer, and the DTC TX FIFO is filled with the Transmit Audio Circular Buffer data.

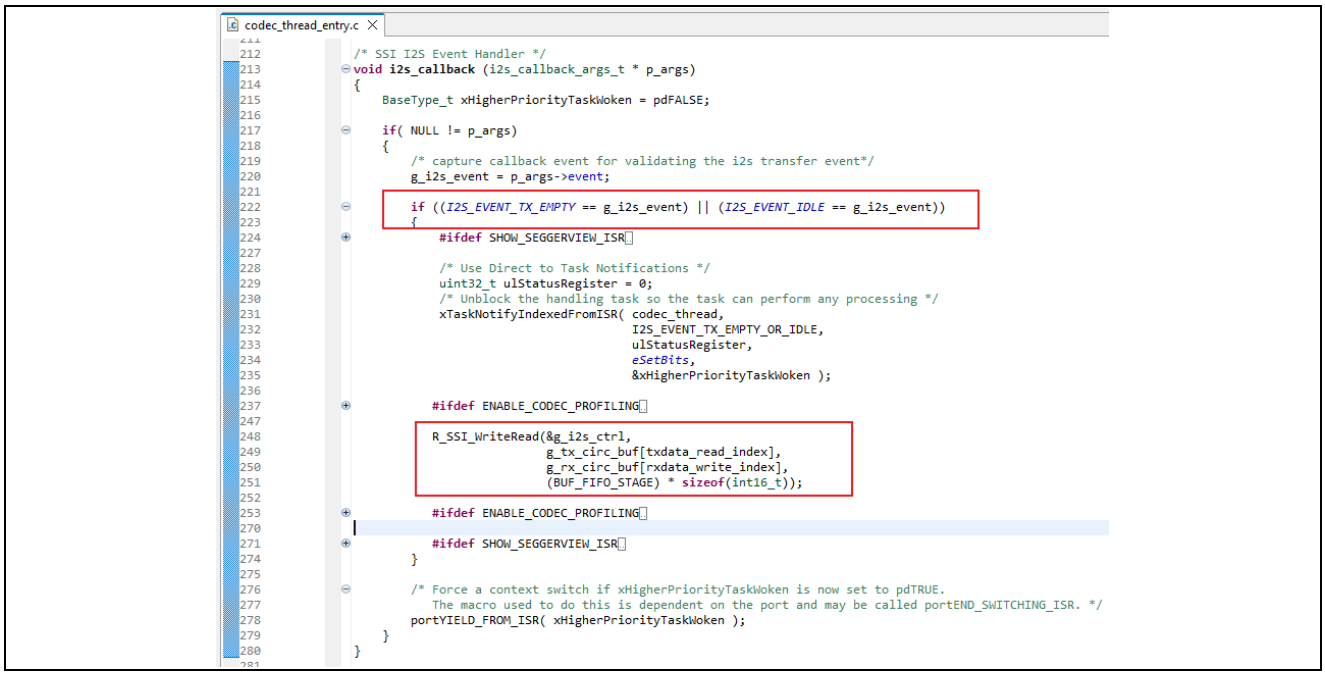


Figure 37. Codec Thread - I2S Callback

The **R_SSI_WriteRead**(…) performs this operation, and both RX and TX operations are handled simultaneously to keep the I2S interface data flow from stalling. If the data flow stalls, then a click or pop in the audio input or output can be heard.

Each call to **R_SSI_WriteRead**(…) reads and writes 32 samples of left-and right-channel data (16-bit integer) interleaved to the Audio Codec over the I2S Master interface.

The initialization of the Audio Codec is covered next in Section 4.2.4 Codec Initialization.

4.2.4 Codec Initialization

The Wolfson WM8960 is the Audio Codec used for the Audio EQ project. The I2C interface is used to initialize and control the codec, and the digital data of the audio is transmitted and received by interfacing with the MCU I2S Master interface. In Figure 38. Codec Initialization, the Codec thread initializes the device with the function **init_codec()**.

In file codec_thread_entry.c, the #defines **CMSIS_PROCESS_I2S_MODE**, **CODEC_ANALOG_PASSTHRU**, and **CODEC_ADC_DAC_PASSTHRU** set the operating mode for the Audio Codec. For Audio EQ normal operation, the mode **CMSIS_PROCESS_I2S_MODE** is enabled.

codec_thread_entry.c

```
90  void codec_thread_entry(void *pvParameters)
91  {
92      FSP_PARAMETER_NOT_USED (pvParameters);
93      fsp_err_t err = FSP_SUCCESS;
94
95  #ifdef ENABLE_SYSTEMVIEW
96      traceSTART();   // start Segger SystemView
97  #endif
98
99      /*
100      * Audio Codec Op Modes
101      * Pick one mode to define
102      * CMSIS_PROCESS_I2S_MODE : used for DSP Filter Operation
103      * CODEC_ANALOG_PASSTHRU  : test mode that patches analog input
104      * to analog output - Digital ADC/DAC path not used
105      * CODEC_ADC_DAC_PASSTHRU : test mode that patches Digital
106      * input to Digital output, relies on MCU for I2S clock gen
107      */
108 #define CMSIS_PROCESS_I2S_MODE
109 #undef CODEC_ANALOG_PASSTHRU
110 #undef CODEC_ADC_DAC_PASSTHRU
111
112 #ifdef CMSIS_PROCESS_I2S_MODE
113     err = init_codec();
114     if(FSP_SUCCESS != err)
115     {
116         /* Audio Codec init failed, so cleanup*/
117         deinit_codec();
118         APP_ERR_TRAP(err);
119     }
120
121 #elif defined CODEC_ANALOG_PASSTHRU
122     init_codec_analog_passthru();
123 #elif defined CODEC_ADC_DAC_PASSTHRU
124     init_codec_loopback();
125 #endif
126
```

wm8960_codec.c

```
183 /**********************************************************************************************************************
184  *  @brief       initialize IIC master module and set up Audio Codec
185  *  @param[IN]   None
186  *  @retval      FSP_SUCCESS                Upon successful open and start of timer
187  *  @retval      Any Other Error code apart from FSP_SUCCESS is  Unsuccessful open or start
188  *********************************************************************************************************************/
189 fsp_err_t init_codec(void)
190 {
191     fsp_err_t err  = FSP_SUCCESS;
192     fsp_err_t err2 = FSP_SUCCESS;
193
194     /* opening IIC master module */
195     err = R_IIC_MASTER_Open(&g_i2c_master_ctrl, &g_i2c_master_cfg);
196     /* handle error */
197     if (FSP_SUCCESS != err)
198     {
199         return err;
200     }
201
202     /* Reset the WM8960 codec and local Control Registers*/
203     err = codec_reset();
204     /* see i2c_codec.h - Set to select the audio codec operation mode */
205     err2 = codec_i2s_operation_setup();
206     return (err || err2);
207 }
208
```

Figure 38. Codec Initialization

The other two test modes are available for troubleshooting I2C and I2S connections with the Audio Codec. See the file /AudioCodec/wm8960.c in the project for more information on all the operating modes.

The test mode set by **CODEC_ANALOG_PASSTHRU** can be used to check that the Audio Codec is wired correctly to the MCU I2C control interface and analog audio input/output connections. The test mode set by **CODEC_ADC_DAC_PASSTHRU** can be used to check that the I2S interface is connected properly and the I2S Master is clocking the codec correctly.

As observed in Figure 38. Codec Initialization in file wm8960_codec.c, the function **init_codec()** opens the I2C Master to control the Audio Codec. Next, the Audio Codec is reset with **codec_reset()**, and the Audio Codec is put into I2S operation mode.

See Figure 39. FSP Configuration - I2C Master Stack Settings for the FSP Configuration of the I2C Master stack settings. The I2C connections to the Audio Codec board are covered in detail in Section 2.2 Connect Audio Codec and Signal Connections.

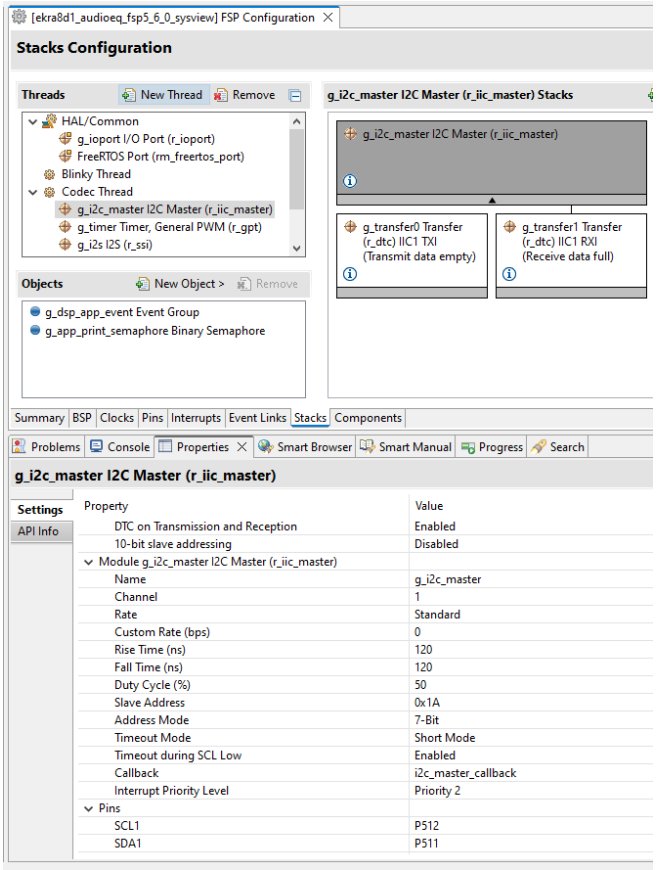
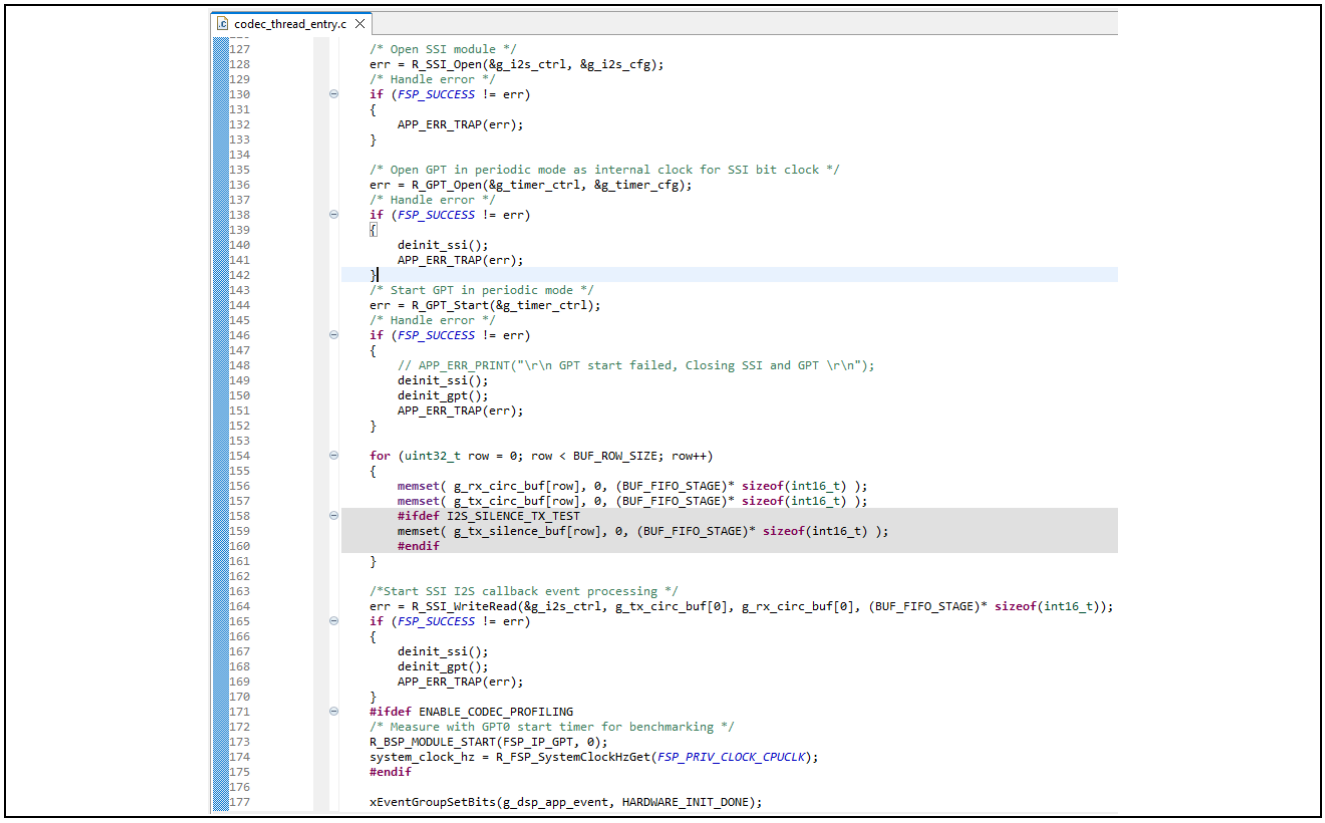


Figure 39. FSP Configuration - I2C Master Stack Settings

4.2.5 I2S Initialization

In the Codec thread, the SSI module is initialized in Master mode for use with the Audio Codec and uses the General Purpose Timer in Periodic mode as the Audio SSI bit clock source. As observed in Figure 40. I2S Initialization, the Codec thread initializes the SSI and Timer modules, starts the SSI I2S callback event processing, and notifies the DSP processing thread that the audio hardware initialization has been completed.

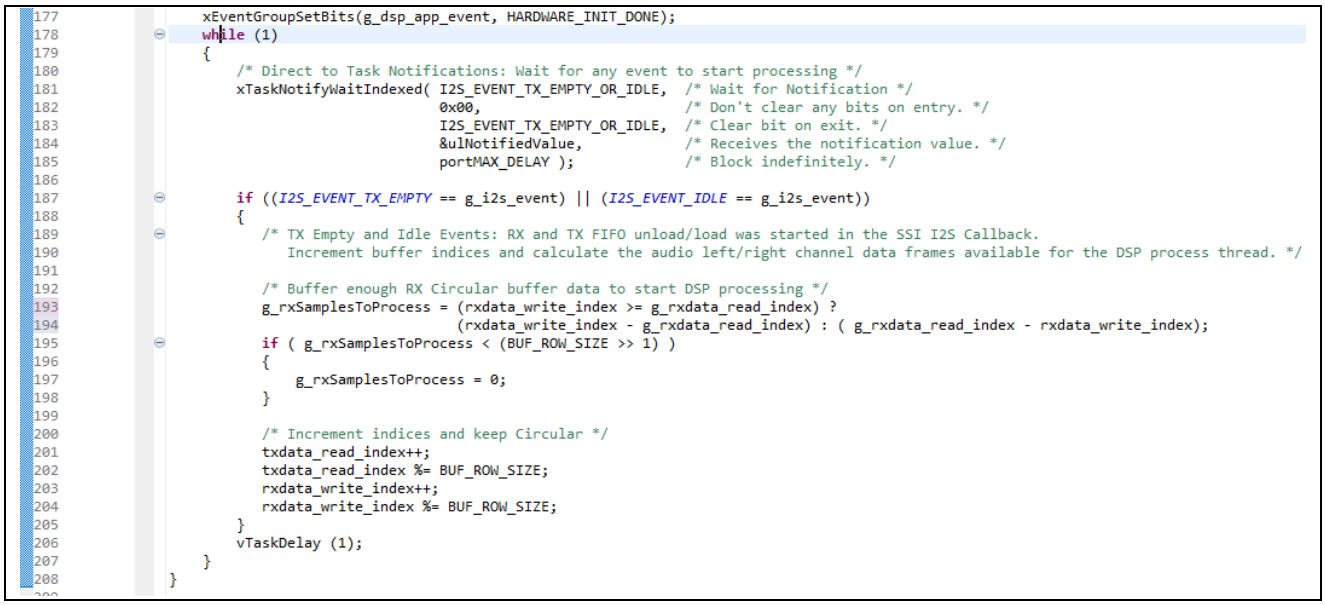


```
codec_thread_entry.c
127     /* Open SSI module */
128     err = R_SSI_Open(&g_i2s_ctrl, &g_i2s_cfg);
129     /* Handle error */
130     if (FSP_SUCCESS != err)
131     {
132         APP_ERR_TRAP(err);
133     }
134
135     /* Open GPT in periodic mode as internal clock for SSI bit clock */
136     err = R_GPT_Open(&g_timer_ctrl, &g_timer_cfg);
137     /* Handle error */
138     if (FSP_SUCCESS != err)
139     {
140         deinit_ssi();
141         APP_ERR_TRAP(err);
142     }
143     /* Start GPT in periodic mode */
144     err = R_GPT_Start(&g_timer_ctrl);
145     /* Handle error */
146     if (FSP_SUCCESS != err)
147     {
148         // APP_ERR_PRINT("\r\n GPT start failed, Closing SSI and GPT \r\n");
149         deinit_ssi();
150         deinit_gpt();
151         APP_ERR_TRAP(err);
152     }
153
154     for (uint32_t row = 0; row < BUF_ROW_SIZE; row++)
155     {
156         memset( g_rx_circ_buf[row], 0, (BUF_FIFO_STAGE)* sizeof(int16_t) );
157         memset( g_tx_circ_buf[row], 0, (BUF_FIFO_STAGE)* sizeof(int16_t) );
158         #ifdef I2S_SILENCE_TX_TEST
159         memset( g_tx_silence_buf[row], 0, (BUF_FIFO_STAGE)* sizeof(int16_t) );
160         #endif
161     }
162
163     /*Start SSI I2S callback event processing */
164     err = R_SSI_WriteRead(&g_i2s_ctrl, g_tx_circ_buf[0], g_rx_circ_buf[0], (BUF_FIFO_STAGE)* sizeof(int16_t));
165     if (FSP_SUCCESS != err)
166     {
167         deinit_ssi();
168         deinit_gpt();
169         APP_ERR_TRAP(err);
170     }
171     #ifdef ENABLE_CODEC_PROFILING
172     /* Measure with GPT0 start timer for benchmarking */
173     R_BSP_MODULE_START(FSP_IP_GPT, 0);
174     system_clock_hz = R_FSP_SystemClockHzGet(FSP_PRIV_CLOCK_CPUCLK);
175     #endif
176
177     xEventGroupSetBits(g_dsp_app_event, HARDWARE_INIT_DONE);
```

Figure 40. I2S Initialization

4.2.6 Codec Thread Processing

In Figure 41. Codec Thread Processing, the Codec thread processing waits for I2S events, maintains the Audio Circular Buffer indices, and determines the number of received samples available to process.



```
177     xEventGroupSetBits(g_dsp_app_event, HARDWARE_INIT_DONE);
178     while (1)
179     {
180         /* Direct to Task Notifications: Wait for any event to start processing */
181         xTaskNotifyWaitIndexed( I2S_EVENT_TX_EMPTY_OR_IDLE,  /* Wait for Notification */
182                                 0x00,                        /* Don't clear any bits on entry. */
183                                 I2S_EVENT_TX_EMPTY_OR_IDLE,  /* Clear bit on exit. */
184                                 &ulNotifiedValue,            /* Receives the notification value. */
185                                 portMAX_DELAY );             /* Block indefinitely. */
186
187         if ((I2S_EVENT_TX_EMPTY == g_i2s_event) || (I2S_EVENT_IDLE == g_i2s_event))
188         {
189            /* TX Empty and Idle Events: RX and TX FIFO unload/load was started in the SSI I2S Callback.
190               Increment buffer indices and calculate the audio left/right channel data frames available for the DSP process thread. */
191
192            /* Buffer enough RX Circular buffer data to start DSP processing */
193            g_rxSamplesToProcess = (rxdata_write_index >= g_rxdata_read_index) ?
194                                   (rxdata_write_index - g_rxdata_read_index) : ( g_rxdata_read_index - rxdata_write_index);
195            if ( g_rxSamplesToProcess < (BUF_ROW_SIZE >> 1) )
196            {
197                g_rxSamplesToProcess = 0;
198            }
199
200            /* Increment indices and keep Circular */
201            txdata_read_index++;
202            txdata_read_index %= BUF_ROW_SIZE;
203            rxdata_write_index++;
204            rxdata_write_index %= BUF_ROW_SIZE;
205         }
206         vTaskDelay (1);
207     }
208 }
```

Figure 41. Codec Thread Processing

The DSP Processing sub-system is covered next in Section 4.3 DSP Processing Thread.

4.3 DSP Processing Thread

DSP processing is handled in the DSP thread, which implements Audio EQ with the ARM CMSIS DSP Library. The entry function **dsp_thread_entry**(void *pvParameters) handles DSP filter initialization as well as the processing of data. The DSP processing runs continuously to filter audio data arriving from the I2S connected Audio Codec. See Figure 42. DSP Processing Thread.

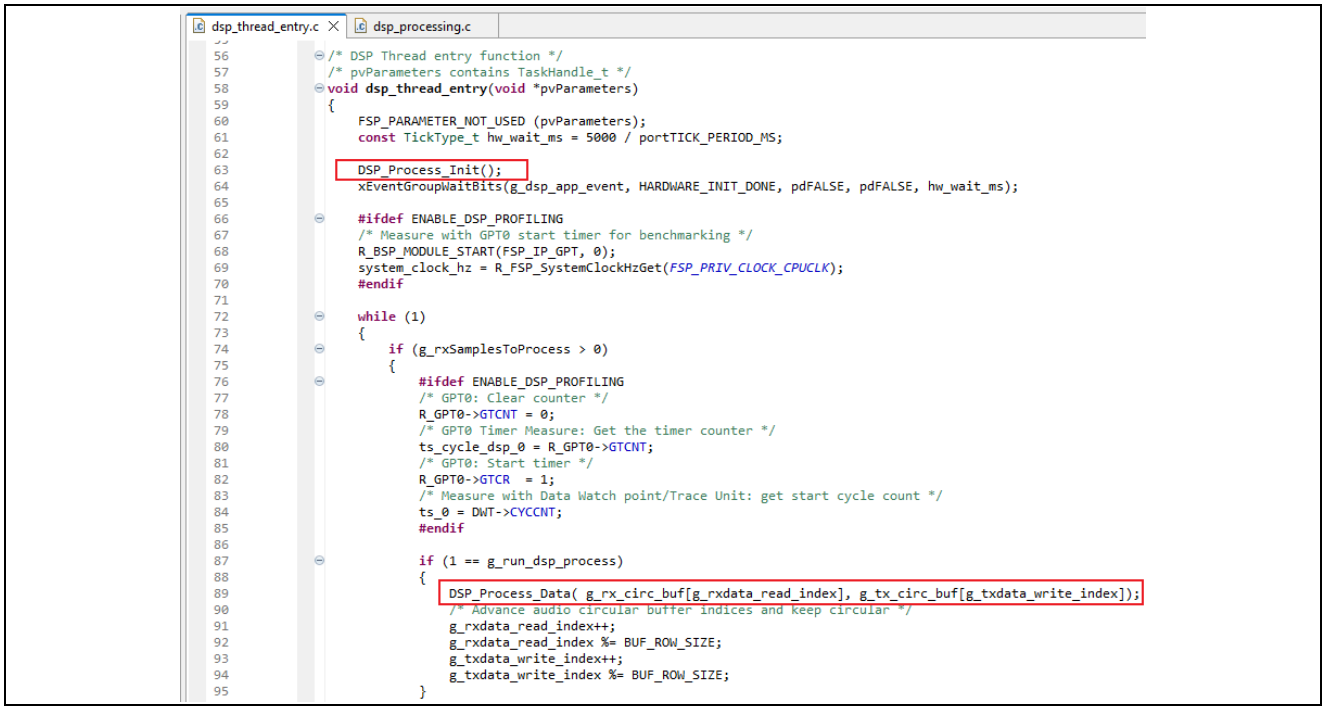


Figure 42. DSP Processing Thread

The function **DSP_Process_Init**() handles creating the Audio EQ filter coefficients, initializing the Biquad filter structures, and clearing the processing memory.

For more details, see Section 4.3.1 DSP Process Initialization.

Signal processing is performed by **DSP_Process_Data**() which is covered in Section 4.3.2 DSP Processing.

4.3.1 DSP Process Initialization

As observed in Figure 43. DSP Process Init, the DSP process initialization handles creating the Audio EQ filter coefficients, setting up the Biquad filter structures, and clearing the processing memory. The initialization is run once at MCU start-up.

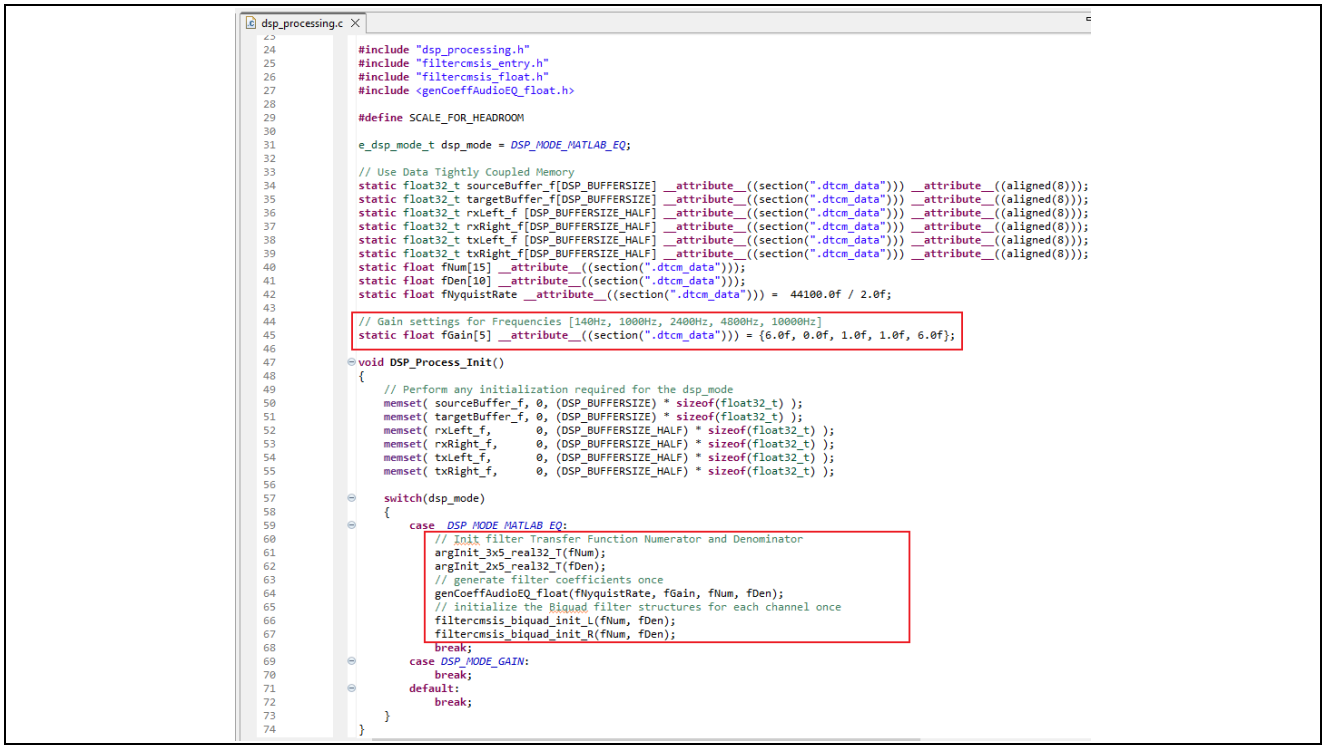


Figure 43. DSP Process Init

The coefficients are generated for the Audio EQ filter based on the audio band gains set with array **fGain[5]**. The function **genCoeffAudioEQ_float**(fNyquistRate, fGain, fNum, fDen) is generated by MATLAB as discussed previously in Section 3.4 Understanding the Generated Source Code. It creates the filter coefficients for the numerator and denominator of the filter transfer function.

The functions **filtercmsis_biquad_init_L**(fNum, fDen) and **filtercmsis_biquad_init_R**(fNum, fDen) are custom implementations that were factored out of the MATLAB generated code. The reason for the custom functions is to initialize the Biquad filter structures once in **DSP_Process_Init**() instead of calling the function each time a group of samples are filtered in the DSP Processing loop. This greatly improves the DSP processing latency.

See Section 3.5 Modify Source Code for Best Performance on RA8 MCUs for more details.

4.3.2 DSP Processing

In Figure 44. Audio EQ and Buffers, DSP processing of the audio data is implemented on a block of 32 samples after the RX Circular Buffer fixed-point data is deinterleaved into separate left- and right-channel data and converted to floating point data. After processing with the Audio EQ CMSIS DSP functions, the left- and right-channel data is converted back to fixed-point data and interleaved. Finally, the processed data is stored in the TX Circular Buffer to be transmitted to the Audio Codec.

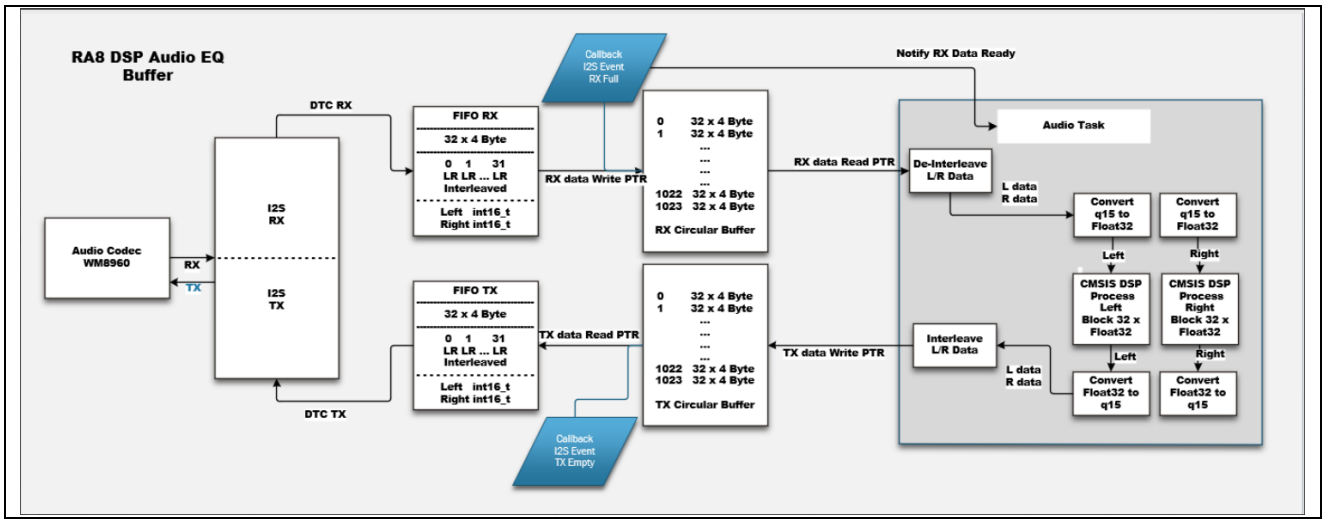


Figure 44. Audio EQ and Buffers

dsp_processing.c

```
int DSP_Process_Data(int16_t *sourceBuffer, int16_t *targetBuffer)
{
    uint16_t index1, index2;
    int status;

    // convert I2S Codec sourceBuffer (64 samples int16_t = 32 Left + 32 Right audio) to
    arm_q15_to_float(
      (const q15_t *) sourceBuffer,
      (float32_t *)   sourceBuffer_f,
      (uint32_t)      DSP_BUFFERSIZE);

#ifdef SCALE_FOR_HEADROOM
    /* ------------------------------------------------------------------
    ** Scale down by 20%.  This provides additional processing head room so
    ** that the graphic EQ can apply gain at bands without clipping.
    ** ------------------------------------------------------------------- */
    arm_scale_f32( sourceBuffer_f,  0.80f, sourceBuffer_f,  (uint32_t) DSP_BUFFERSIZE);
#endif

    // De-Interleave source L/R data to separate left and right channel buffers
    for (index1 = 0, index2 = 0; index1 < DSP_BUFFERSIZE_HALF; index1++)
    {
        rxLeft_f [index1] = sourceBuffer_f[index2++];
        rxRight_f[index1] = sourceBuffer_f[index2++];
    }

    switch(dsp_mode)
    {
        case  DSP_MODE_MATLAB_EQ:
            filtercmsis_entry_L( rxLeft_f,  txLeft_f);   // process Left Channel
            filtercmsis_entry_R( rxRight_f, txRight_f);  // process Right Channel
            break;
        case DSP_MODE_GAIN:
        default:
            arm_scale_f32( rxLeft_f,  2.0f, txLeft_f,  (uint32_t) DSP_BUFFERSIZE_HALF);
            arm_scale_f32( rxRight_f, 2.0f, txRight_f, (uint32_t) DSP_BUFFERSIZE_HALF);
            break;
    }

    // Interleave Left/Right channel Data into targetBuffer_f
    for (index1 = 0, index2 = 0; index1 < DSP_BUFFERSIZE_HALF; index1++)
    {
        targetBuffer_f[index2++] = txLeft_f [index1];
        targetBuffer_f[index2++] = txRight_f[index1];
    }

    // convert targetBuffer (float32_t) to int16_t for I2S TX Codec
    arm_float_to_q15(
      (const float32_t *) targetBuffer_f,
      (q15_t *)           targetBuffer,
      (uint32_t)          DSP_BUFFERSIZE );
```

Figure 45. DSP Process Data

In Figure 45. DSP Process Data, the source code implementation of the Audio EQ DSP processing is shown.

The functions **filtercmsis_entry_L**(rxLeft_f, txLeft_f) and **filtercmsis_entry_R**(rxRight_f, txRight_f) handle the Audio EQ filtering. The function to process one channel of data is generated by MATLAB as discussed in Section 3.4 Understanding the Generated Source Code. In this case, two calls to the function are required to process each of the two audio channels.

4.3.3 Audio EQ Filter Implementation

The functions in **filtercmsis_entry.c** are generated by MATLAB and implement the Audio EQ filter.

To decrease the DSP processing latency, an optimization modification has been made to move the filter coefficient generation function call out of **filtercmsis_entry_L**(…) and **filtercmsis_entry_R**(…).

The filter coefficient generation call to **genCoeffAudioEQ_float**(fNyquistRate, fGain, fNum, fDen) is moved to **DSP_Process_Init**(…). For more information, see Section 4.3.1 DSP Process Initialization.

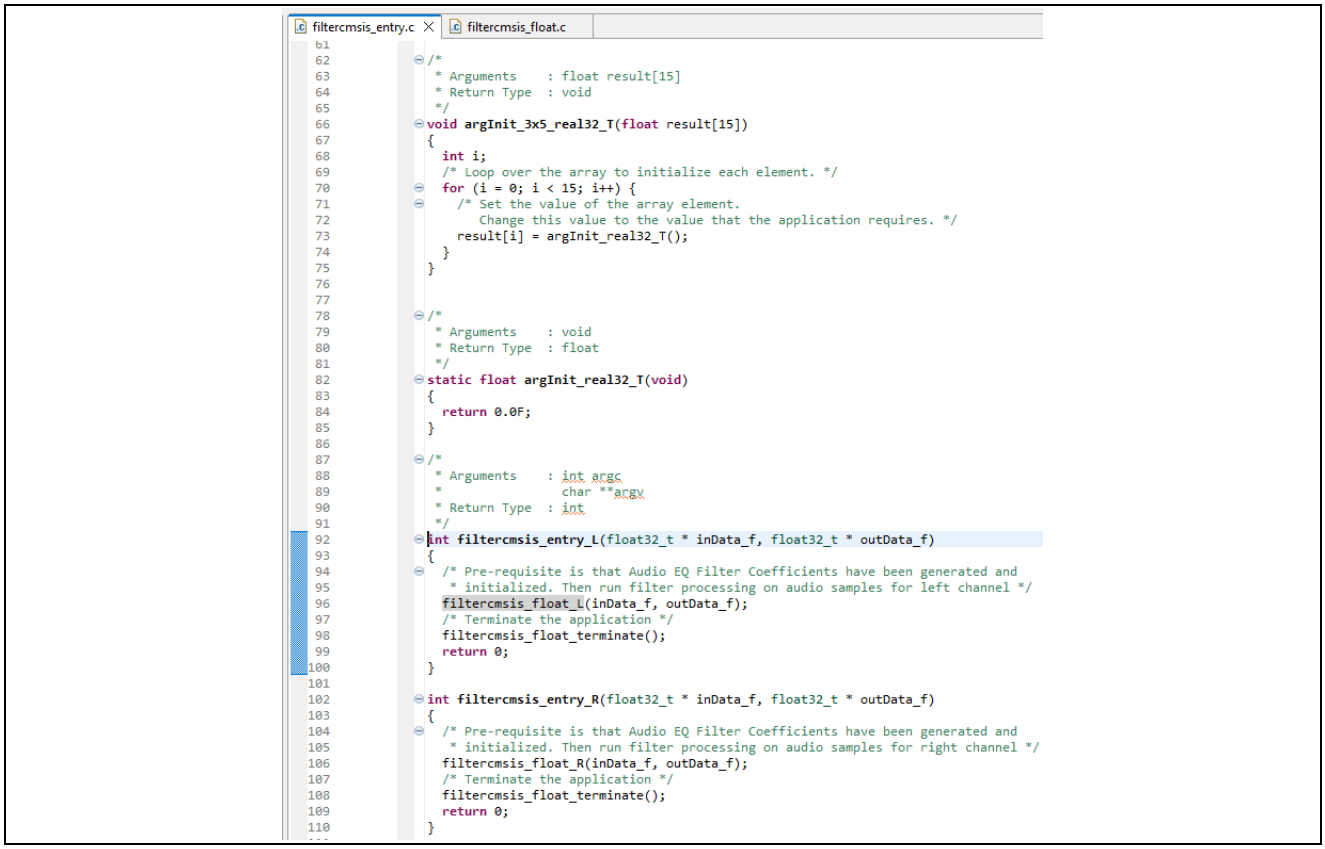


Figure 46. Audio EQ - Filter CMSIS Entry

In Figure 46. Audio EQ - Filter CMSIS Entry, the functions **filtercmsis_entry_L**(…) and **filtercmsis_entry_R**(…) call **filtercmsis_float_L**(input[32], output[32]) and **filtercmsis_float_R**(input[32], output[32]) that use the CMSIS DSP function **arm_biquad_cascade_df2T_f32**(…) to implement the Biquad cascade used in the five band Audio EQ.

In order to improve DSP processing latency, it is required to make optimization modifications to **filtercmsis_float_L**(…) and **filtercmsis_float_R**(…).

As observed in Figure 47. Audio EQ – Filter CMSIS Float Function, the first modification is to remove the call to **memcpy(&U0[0], &input[0], 32U * sizeof(float)),** which is unnecessary since all memory buffers are initialized in **DSP_Process_Init().**

The second modification is to the Biquad initialization, which is moved to **filtercmsis_biquad_init_L**(num[15], den[10]) and **filtercmsis_biquad_init_R**(num[15], den[10]).

The modified functions are called in **DSP_Process_Init**(). See Section 3.5 Modify Source Code for Best Performance on RA8 MCUs for more information.

As observed in Figure 47. Audio EQ – Filter CMSIS Float Function and Figure 48. CMSIS DSP Biquad Cascade Function, the last modifications are to move functions **filtercmsis_float_L**(…), **filtercmsis_float_R**(…), and **arm_biquad_cascade_df2T_f32**(…) to instruction tightly coupled memory (ITCM). For more information on how to use ITCM, see Section 5.3.2 Improve Performance Using DTCM and ITCM.

filtercmsis_float.c

```
/* Variable Definitions */
static bool filter_not_empty;
static dspcodegen_BiquadFilter filter_L;
static dspcodegen_BiquadFilter filter_R;

/* Function Definitions */
/*
 * implement the function with DSP Toolbox filter with float32 data type
 *  and use Cortex M CMSIS DSP functions
 *
 * Arguments    : const float input[1024]
 *                const float num[15]
 *                const float den[10]
 *                float output[1024]
 * Return Type  : void
 */
void __attribute__((section(".itcm_data"))) filtercmsis_float_L(const float input[32], float output[32]);

// I removed num,den since filtercmsis_biquad_init(num,den) called once to init biquad filter
void filtercmsis_float_L(const float input[32], float output[32])
{
    /*
    * I Modified the original Matlab codegen source code by moving biquad filter init to
    * separate function filtercmsis_biquad_init(const float num[15],const float den[10])
    */

    /*
     * Optimization: I modified to not use memcpy, instead pass input[0] to arm_biquad_cascade_df2T_f32(...)
     */
   // memcpy(&U0[0], &input[0], 32U * sizeof(float)); // not used, reduces latency
   arm_biquad_cascade_df2T_f32(&filter_L.cSFunObject.S, &input[0], &output[0], 32U);
}

void __attribute__((section(".itcm_data"))) filtercmsis_float_R(const float input[32], float output[32]);

// I removed num,den since filtercmsis_biquad_init(num,den) called once to init biquad filter
void filtercmsis_float_R(const float input[32], float output[32])
{
    /*
    * I Modified the original Matlab codegen source code by moving biquad filter init to
    * separate function filtercmsis_biquad_init(const float num[15],const float den[10])
    */

    /*
     * Optimization: I modified to not use memcpy, instead pass input[0] to arm_biquad_cascade_df2T_f32(...)
     */
   // memcpy(&U0[0], &input[0], 32U * sizeof(float)); // not used, reduces latency
   arm_biquad_cascade_df2T_f32(&filter_R.cSFunObject.S, &input[0], &output[0], 32U);
}
```

Figure 47. Audio EQ – Filter CMSIS Float Function

arm_biquad_cascade_df2T_f32.c

```
void __attribute__((section(".itcm_data"))) arm_biquad_cascade_df2T_f32(const arm_biquad_cascade_df2T_instance_f32 *
                                                                        const float32_t * pSrc,
                                                                              float32_t * pDst,
                                                                              uint32_t blockSize);

ARM_DSP_ATTRIBUTE void arm_biquad_cascade_df2T_f32(
  const arm_biquad_cascade_df2T_instance_f32 * S,
  const float32_t * pSrc,
        float32_t * pDst,
        uint32_t blockSize)
{
  const float32_t *pIn = pSrc;                       /* Source pointer */
        float32_t *pOut = pDst;                      /* Destination pointer */
        float32_t *pState = S->pState;               /* State pointer */
  const float32_t *pCoeffs = S->pCoeffs;             /* Coefficient pointer */
        float32_t acc1;                              /* Accumulator */
        float32_t b0, b1, b2, a1, a2;                /* Filter coefficients */
        float32_t Xn1;                               /* Temporary input */
        float32_t d1, d2;                            /* State variables */
        uint32_t sample, stage = S->numStages;       /* Loop counters */

  do
  {
     /* Reading the coefficients */
     b0 = pCoeffs[0];
     b1 = pCoeffs[1];
     b2 = pCoeffs[2];
     a1 = pCoeffs[3];
     a2 = pCoeffs[4];

     /* Reading the state values */
     d1 = pState[0];
     d2 = pState[1];

     pCoeffs += 5U;

#if defined (ARM_MATH_LOOPUNROLL)

     /* Loop unrolling: Compute 16 outputs at a time */
     sample = blockSize >> 4U;

     while (sample > 0U) {
```

Figure 48. CMSIS DSP Biquad Cascade Function

4.3.4 CMSIS DSP Stack

The CMSIS DSP stack provides the CMSIS DSP functions required by the MATLAB-generated source code. There are no FSP configuration settings to make. As shown in Figure 49. FSP Configuration - CMSIS DSP Stack, just add the **Arm CMSIS DSP Library Source** stack to the project.

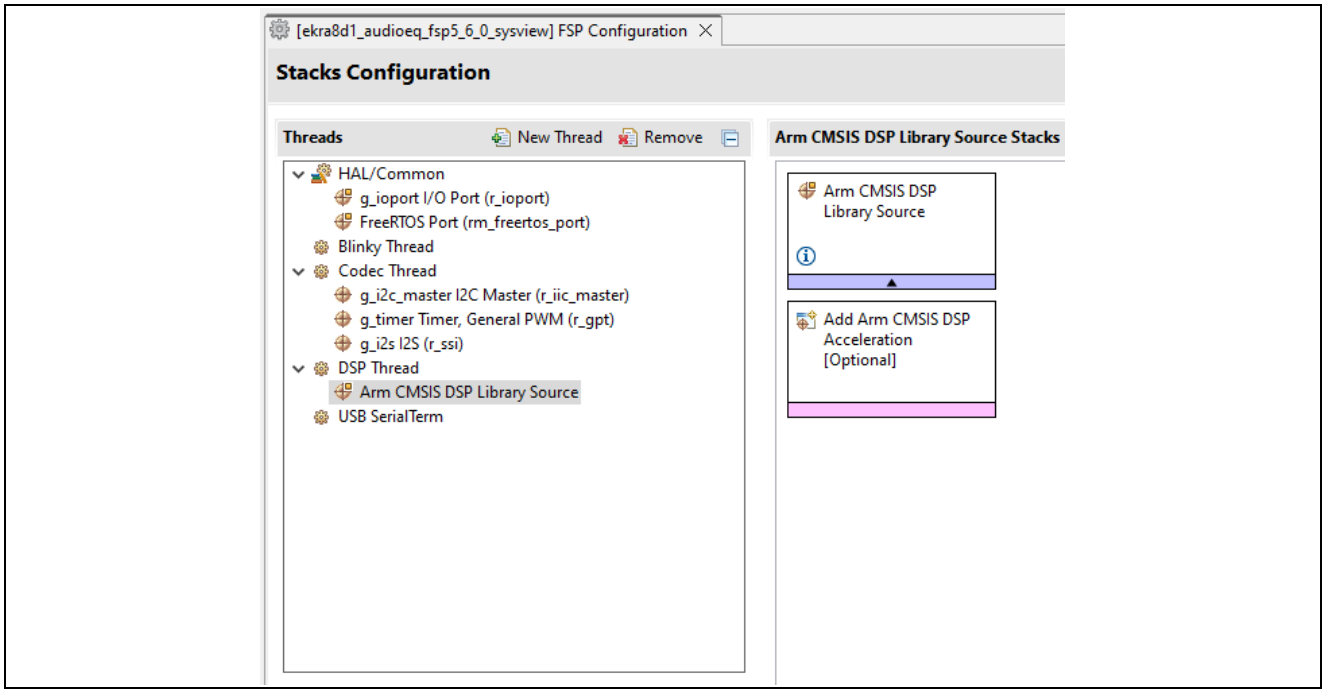


Figure 49. FSP Configuration - CMSIS DSP Stack

4.4 Command Line Interface Thread

The command line interface (CLI) is used to display the Audio EQ operation mode and processing latency. It is implemented with a Serial Terminal emulator over UART with Segger J-Link VCOM on-board through the Debug1 USB connection port.

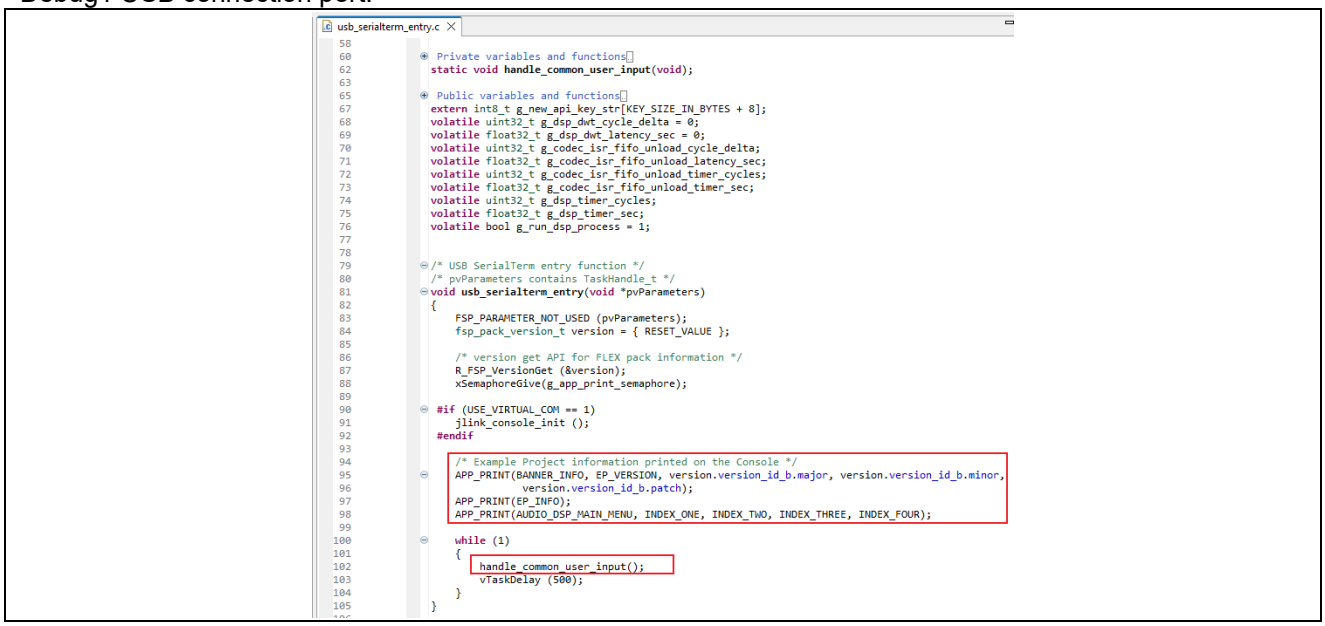
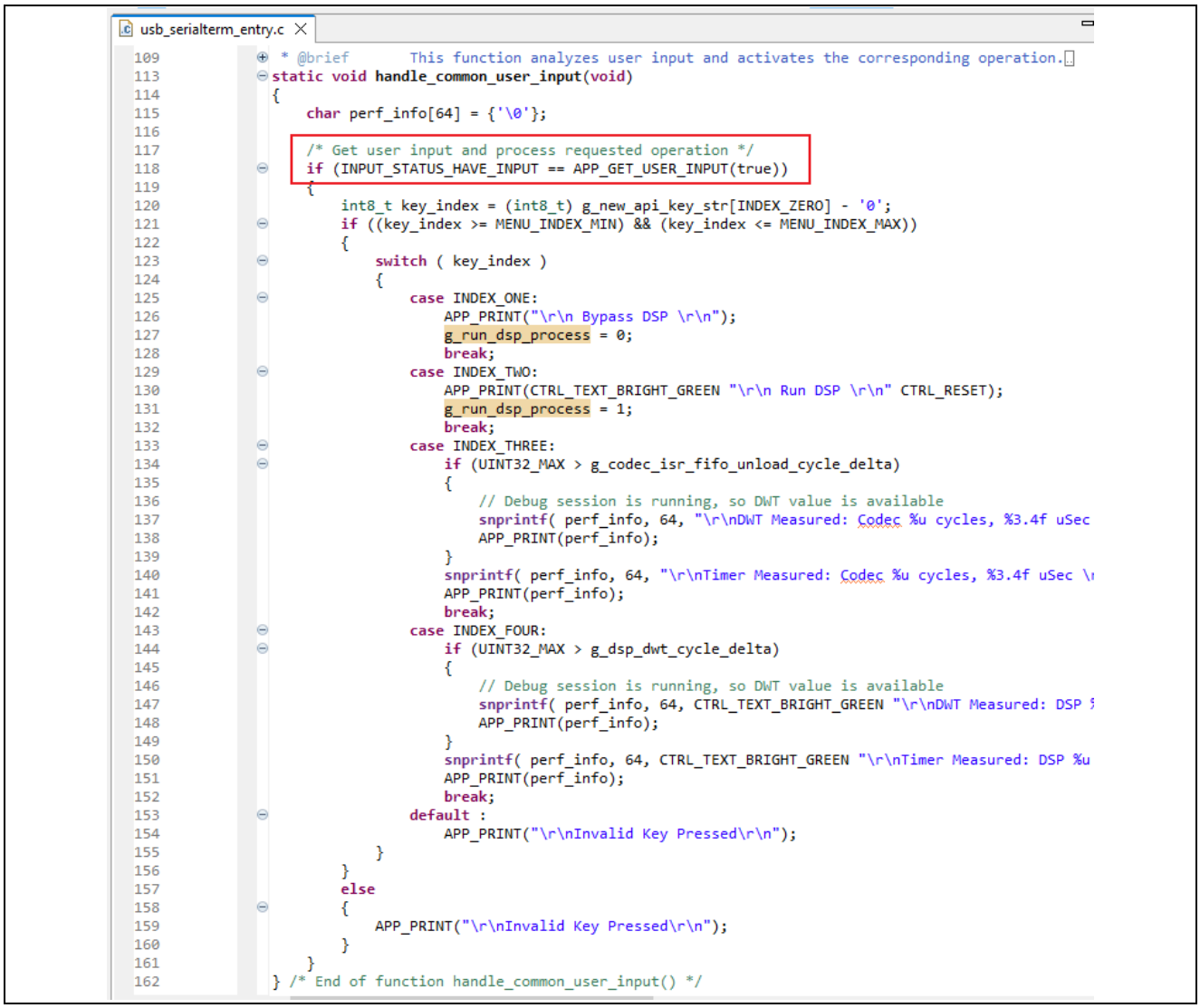


Figure 50. CLI - USB SerialTerm Thread

As shown in Figure 50. CLI - USB SerialTerm Thread, the CLI is implemented with the USB SerialTerm thread. The J-Link console starts with VCOM, and then the info banner and menu are displayed. The user interacts with the CLI by selecting a menu option.

As shown in Figure 51. CLI – Handle User Input Function, the user input handler is implemented with the function **handle_common_user_input().**

The menu option handlers are provided to bypass processing, activate processing, display the Audio Codec handler latency, and display the DSP processing latency.



usb_serialterm_entry.c

```
109         * @brief       This function analyzes user input and activates the corresponding operation.
113     static void handle_common_user_input(void)
114     {
115         char perf_info[64] = {'\0'};
116
117         /* Get user input and process requested operation */
118         if (INPUT_STATUS_HAVE_INPUT == APP_GET_USER_INPUT(true))
119         {
120             int8_t key_index = (int8_t) g_new_api_key_str[INDEX_ZERO] - '0';
121             if ((key_index >= MENU_INDEX_MIN) && (key_index <= MENU_INDEX_MAX))
122             {
123                 switch ( key_index )
124                 {
125                     case INDEX_ONE:
126                         APP_PRINT("\r\n Bypass DSP \r\n");
127                         g_run_dsp_process = 0;
128                         break;
129                     case INDEX_TWO:
130                         APP_PRINT(CTRL_TEXT_BRIGHT_GREEN "\r\n Run DSP \r\n" CTRL_RESET);
131                         g_run_dsp_process = 1;
132                         break;
133                     case INDEX_THREE:
134                         if (UINT32_MAX > g_codec_isr_fifo_unload_cycle_delta)
135                         {
136                             // Debug session is running, so DWT value is available
137                             snprintf( perf_info, 64, "\r\nDWT Measured: Codec %u cycles, %3.4f uSec
138                             APP_PRINT(perf_info);
139                         }
140                         snprintf( perf_info, 64, "\r\nTimer Measured: Codec %u cycles, %3.4f uSec \r
141                         APP_PRINT(perf_info);
142                         break;
143                     case INDEX_FOUR:
144                         if (UINT32_MAX > g_dsp_dwt_cycle_delta)
145                         {
146                             // Debug session is running, so DWT value is available
147                             snprintf( perf_info, 64, CTRL_TEXT_BRIGHT_GREEN "\r\nDWT Measured: DSP %
148                             APP_PRINT(perf_info);
149                         }
150                         snprintf( perf_info, 64, CTRL_TEXT_BRIGHT_GREEN "\r\nTimer Measured: DSP %u
151                         APP_PRINT(perf_info);
152                         break;
153                     default :
154                         APP_PRINT("\r\nInvalid Key Pressed\r\n");
155                 }
156             }
157             else
158             {
159                 APP_PRINT("\r\nInvalid Key Pressed\r\n");
160             }
161         }
162     } /* End of function handle_common_user_input() */
```

Figure 51. CLI – Handle User Input Function

By default, the Audio EQ processing is active at program startup, and processing can be bypassed by selecting Menu 1 option. See Section 2.3 Setup Command Line Interface and Section 2.4 Download and Operate Project for more information.

4.5 LED Blinky Thread

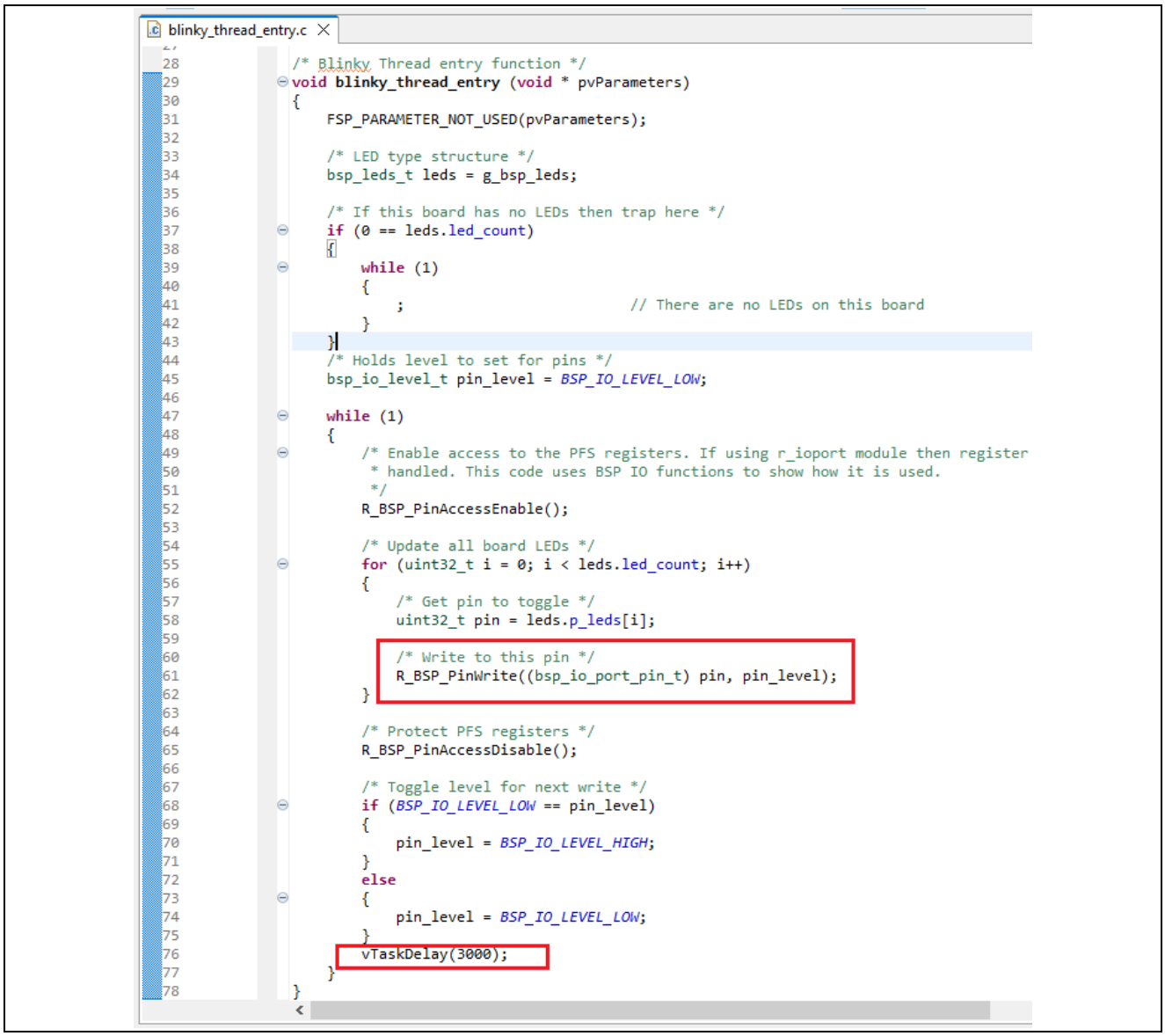


Figure 52. LED Blinky Thread

As observed in Figure 52. LED Blinky Thread, the LED Blinky thread operates Blue (LED1), Green (LED2), and Red (LED3) LEDs on the EK-RA8D1 board. The purpose of the thread is to indicate the board and software are operating properly, with the lighted LEDs cycling on and off about every 3 seconds. The blinking LED indicates that the MCU is operating properly. If the blinking LED stops, it is an indication that the MCU has halted due to an unhandled exception. If this occurs, press the red Reset button Switch S3.

4.6 Using Segger SystemView

A system level view of the Audio EQ project can be obtained with Segger SystemView. The tool is integrated into the project and enabled by default to show the process threads, thread priorities, thread timing, and MCU utilization. To disable Segger SystemView in the project, see Section 3.7 Changing Segger SystemView Support.

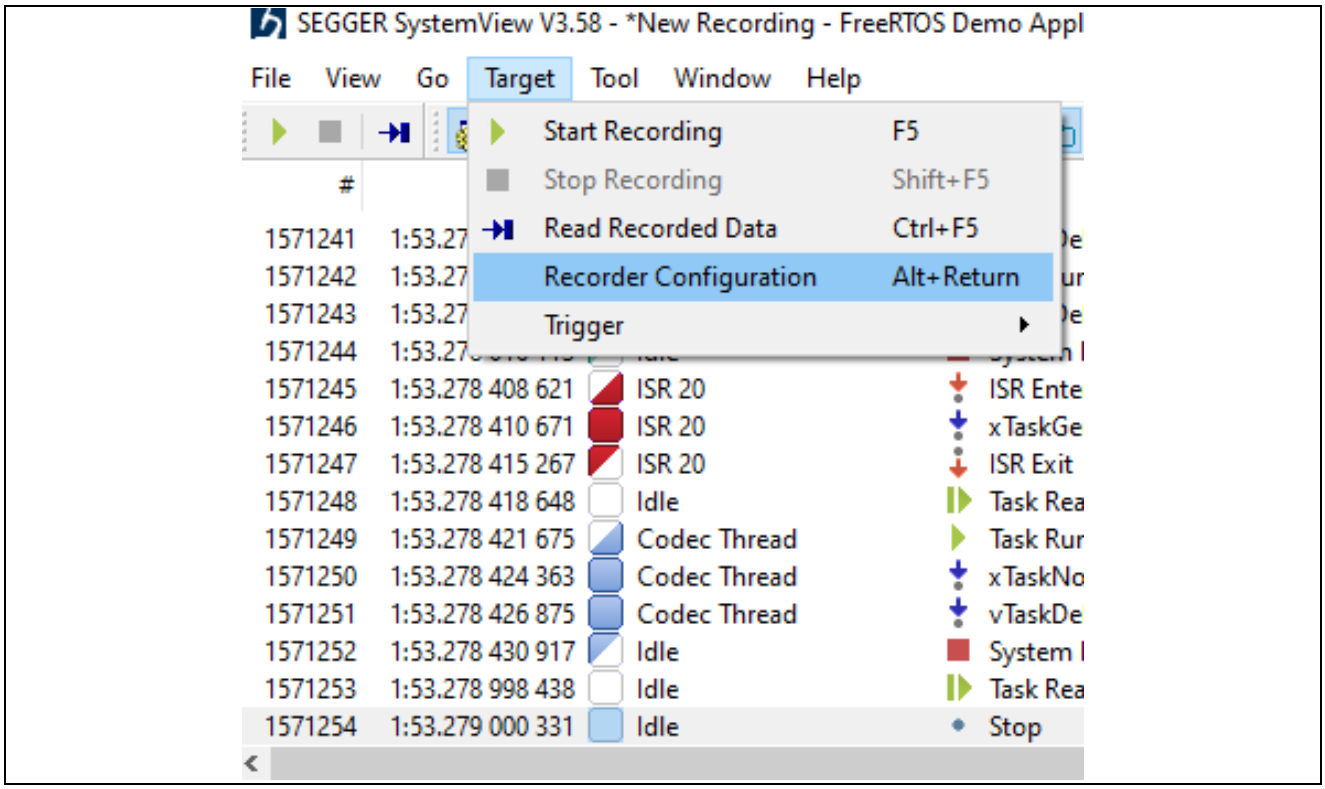


Figure 53. SystemView – Recorder Configuration

A recording of the system is obtained by configuring and starting a recording. In Figure 53. SystemView – Recorder Configuration and Figure 54. SystemView – Recorder J-Link Settings, use **Target -> Recorder Configuration** to configure the J-Link settings. If the project is modified, then find the updated RTT Control Block Address in /Debug/ekra8d1_audioeq_fsp5_6_0_system.map file. Look for the address assigned to **_SEGGER_RTT**.

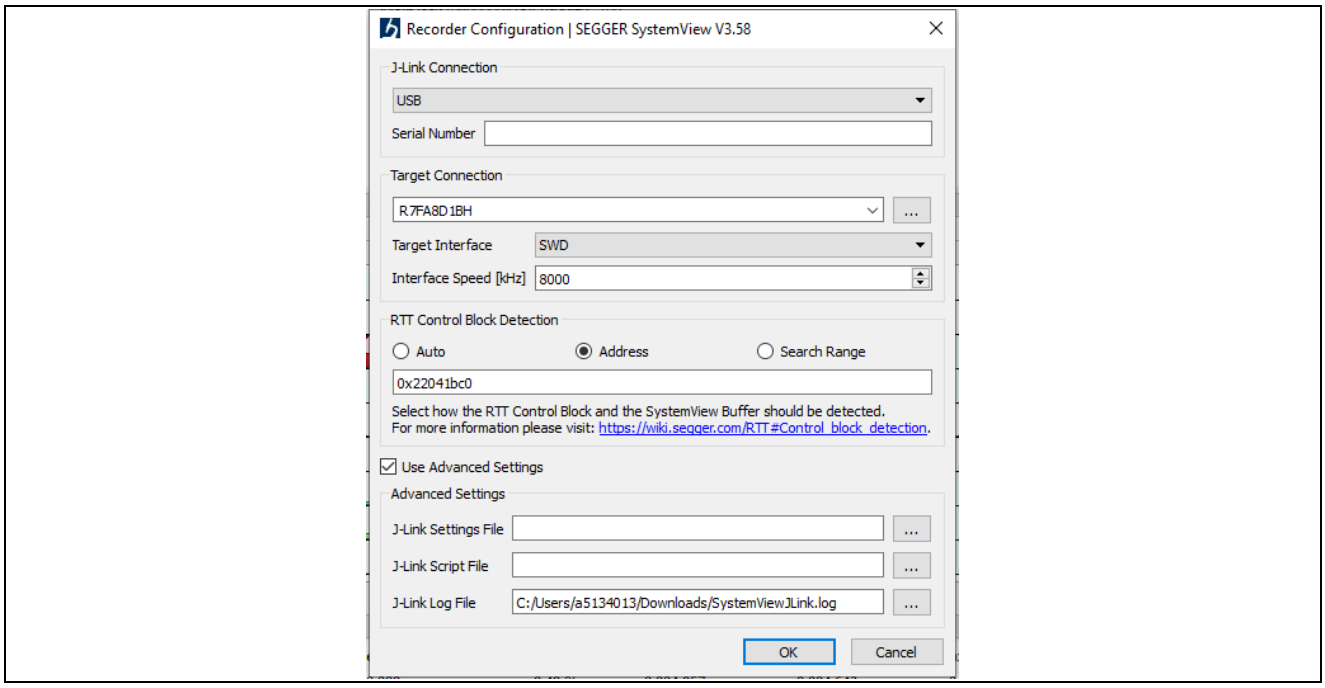


Figure 54. SystemView – Recorder J-Link Settings

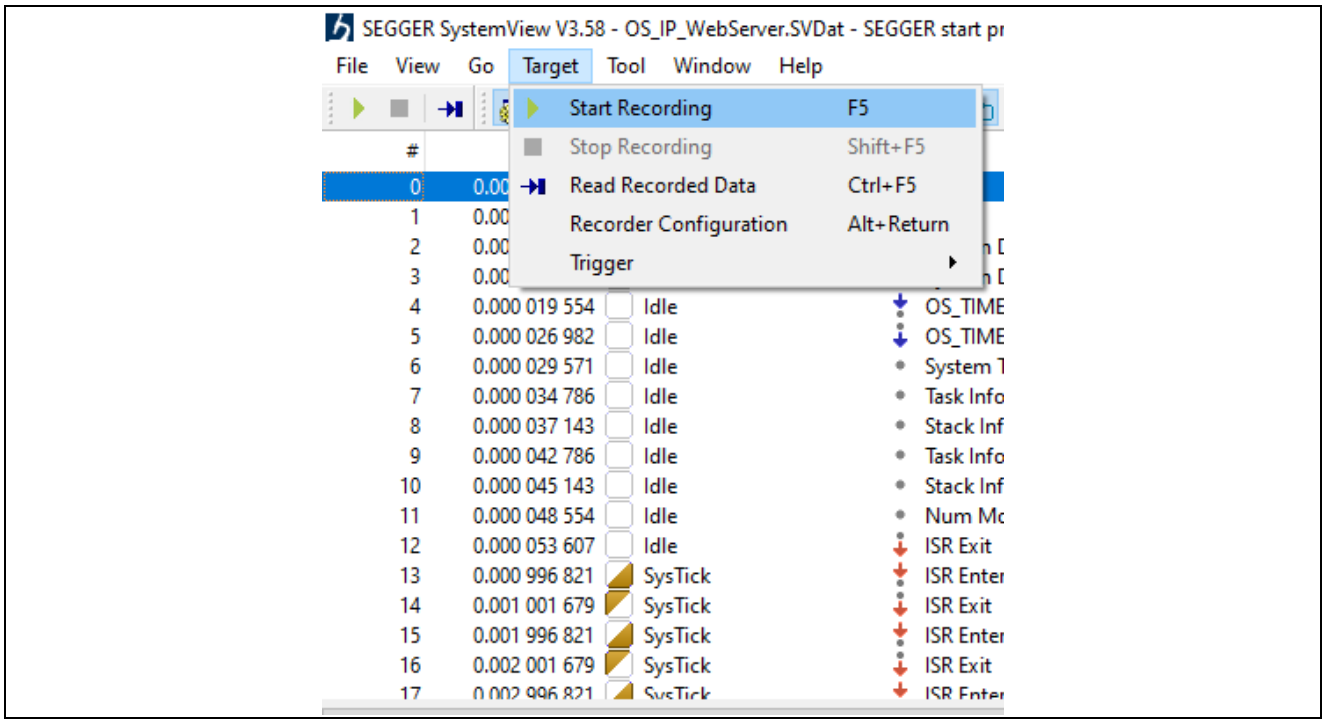


Figure 55. SystemView – Start Recording

Confirm the settings with SystemView and ensure that a debug session is running with e^2 studio before re-coding. As seen in Figure 55. SystemView – Start Recording, next, start a recording with **Target -> Start Recording.** Capture two or more minutes of system activity. The recording results are shown in Figure 56. SystemView – CPU Utilization.

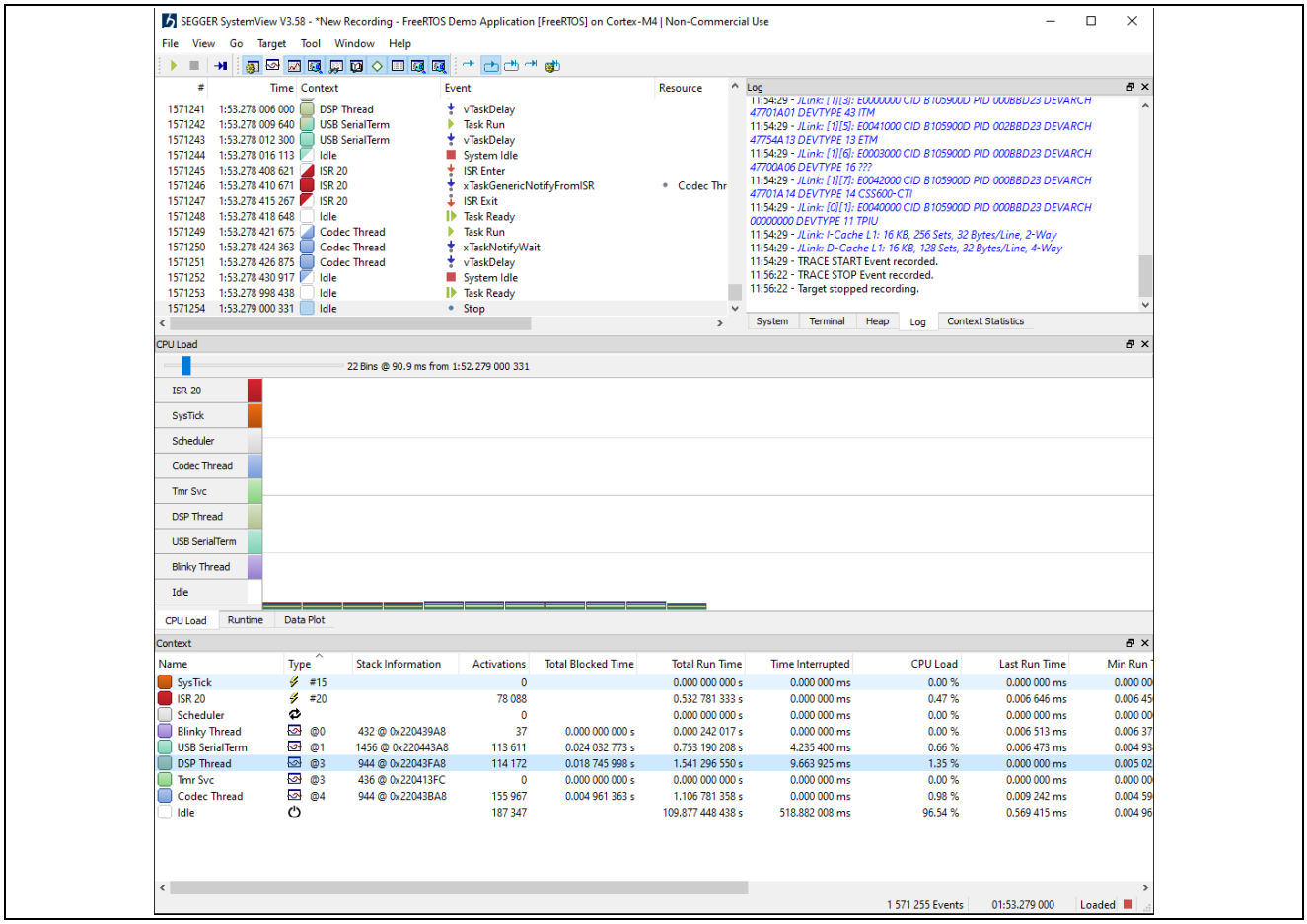


Figure 56. SystemView – CPU Utilization

The top window view shows the Audio EQ threads, ISRs, and Events with time stamps. The bottom window view shows the thread statistics that include the CPU load. The DSP, Codec, and Idle thread of the CPU load percent provides an indication of the MCU utilization.

In the Audio EQ project, the DSP thread is using 1.35% and Codec thread is using 0.98% of the CPU Load. From observing the idle task, the CPU load is 96.54%. More processing can be added to the DSP processing. The RA8D1 MCU is clearly not over-utilized in this project.

A good practice is measuring the system utilization before and after making changes to the project. It may be required to rebalance the priorities of the threads in the project if other threads and functionality are added to extend the Audio EQ project. An example of extended functionality would be adding a display for parameter control.

For other next steps to consider with this project, see Section 6 Next Steps.

5. Optimizing Audio EQ Performance with Arm® Helium™ on RA8 MCU

5.1 Introduction to Arm® Helium™

Arm® Helium™ technology is the M-profile Vector Extension (MVE) for the Arm Cortex-M processor series. It is part of the ARMv8.1-M architecture and enables developers to realize a performance uplift for DSP and ML applications. Helium™ technology provides optimized performance using Single Instruction Multiple Data (SIMD) to perform the same operation simultaneously on multiple data. Refer to the R01AN7127EU Application Note High Performance with RA8 MCU using Arm® Cortex®-M85 core with Helium™ for more details.

5.2 Arm® Helium™ Support in Renesas FSP and LLVM Toolchain

The LLVM Embedded Toolchain for ARM supports Helium™ instructions with the compiler settings by default. When generating a RA8D1 project using e² studio and Flexible Software Package (FSP), CPU settings and software settings are pre-optimized for Cortex®-M85 core and the CMSIS Helium™ support. See Figure 57. Create a new EK-RA8D1 project with LLVM Toolchain using e² studio.

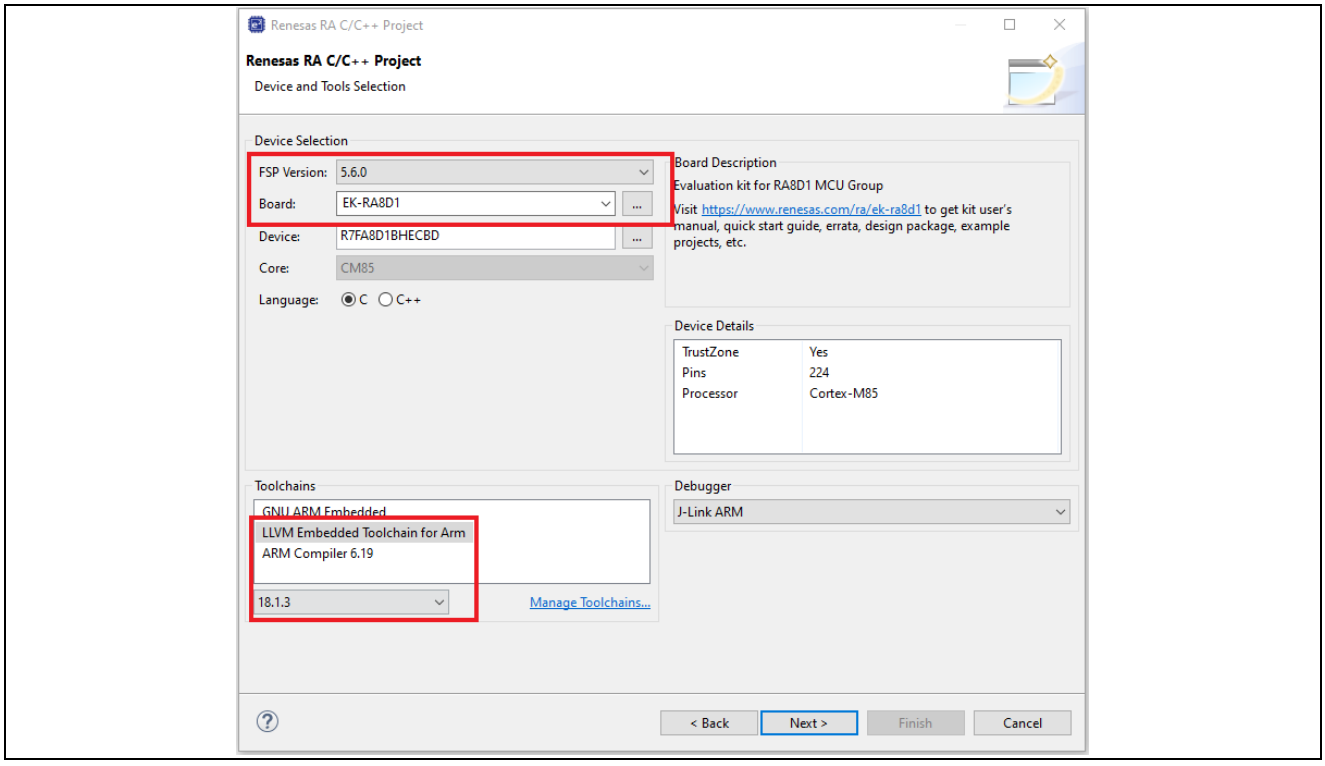


Figure 57. Create a new EK-RA8D1 project with LLVM Toolchain using e² studio

See Table 3. LLVM Toolchain Settings for the settings required with the LLVM Toolchain to get the best performance from the Cortex-M85 core with the CMSIS DSP extension.

Table 3. LLVM Toolchain Settings

Parameter	Default	Setting	Notes
-mfloat-abi	Hard	Hard	MCU has hardware support
-mfpu	Toolchain default	Toolchain default	Toolchain default
-march	Toolchain default	Toolchain default	MCU has DSP support
Optimization	-O3	-O3 or -Ofast	Optimize most or Optimize size
BSP/Main Stack Size	0x0	0x400	Size (bytes)
BSP/Heap Size	0x0	0x400	Size (bytes)

Depending on the memory resources required in your DSP project, the Main Stack and Heap size may need to be resized. In addition, consider testing your project with Optimization -O3 or -Ofast and measure processing latency before and after making the change. The optimization improvements are highly dependent on the CMSIS DSP functions that are used to implement the signal processing.

5.3 Improve DSP Performance

You can utilize Tightly Coupled Memory (TCM) and Cache together with Helium™ to achieve higher performance. Critical routines and data can be placed in TCM areas to ensure faster access. TCM does not use caches.

5.3.1 Improve Performance with Data Cache

Depending on the functionality of the DSP algorithm, the processing performance can be improved by enabling the cache in the FSP Configurator. Two example scenarios where Data Cache can improve performance include: a function that uses long loops to execute the same code repeatedly, or a function where data access is repeated and is sequential. To enable data cache with FSP Configurator, go to BSP -> Properties -> Settings -> RA8xx Family -> Cache settings -> Data cache. See Figure 58. FSP Configurator Data Cache Settings.

Even though enabling the data cache can sometimes improve performance, it can cause concurrency and coherency issues. In general, it is good practice to enable the cache for applications that have repeated access to the same set of data for processing. We recommend measuring the latency of the DSP process before and after enabling the data cache to understand the impact of the cache on the processing latency and to check for any potential caching issues mentioned above.

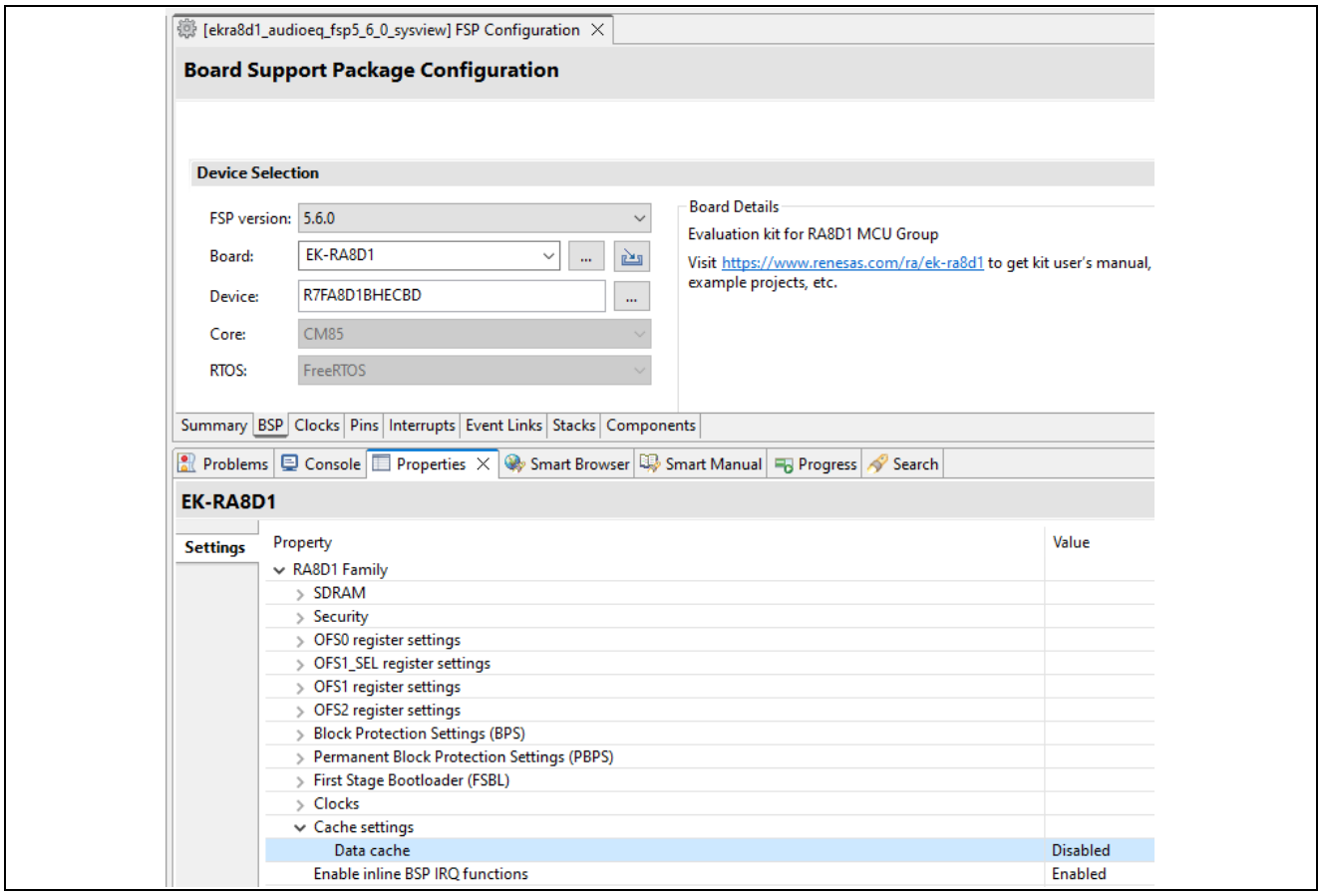


Figure 58. FSP Configurator Data Cache Settings

For the Audio EQ project, it was determined that no significant performance improvement could be made with data cache enabled. This is likely due to the processing of audio data streams that are continuously changing in the input and output circular buffers. However, using DTCM and ITCM did make a significant difference in performance. Continue to the next section to learn more about Tightly Coupled Memory (TCM).

5.3.2 Improve Performance Using DTCM and ITCM

The RA8 family of MCUs has 128 KB TCM memory that consists of 64 KB ITCM (Instruction TCM) and 64 KB DTCM (Data TCM). Note that accessing TCM is not available in CPU Deep Sleep mode, Software Standby mode, and Deep Software Standby mode. Refer to your specific MCU User's Manual for the TCM memory address areas.

Figure 59 shows ITCM and DTCM in the Local CPU Subsystem.

The FSP initializes both ITCM and DTCM areas by default. The linker script has defined sections for ITCM and DTCM areas, making it easy to utilize in user applications.

For the Audio EQ project with DTCM used for the DSP Processing subsystem local memory buffers, a change of 60.04% decrease in processing latency was measured.

For the Audio EQ, with the DSP functions placed in ITCM, no performance change was observed. Keep in mind any performance change is highly dependent on the application and the CMSIS-DSP functions that are used for processing. So, give it a try with your application, it may make an improvement in performance.

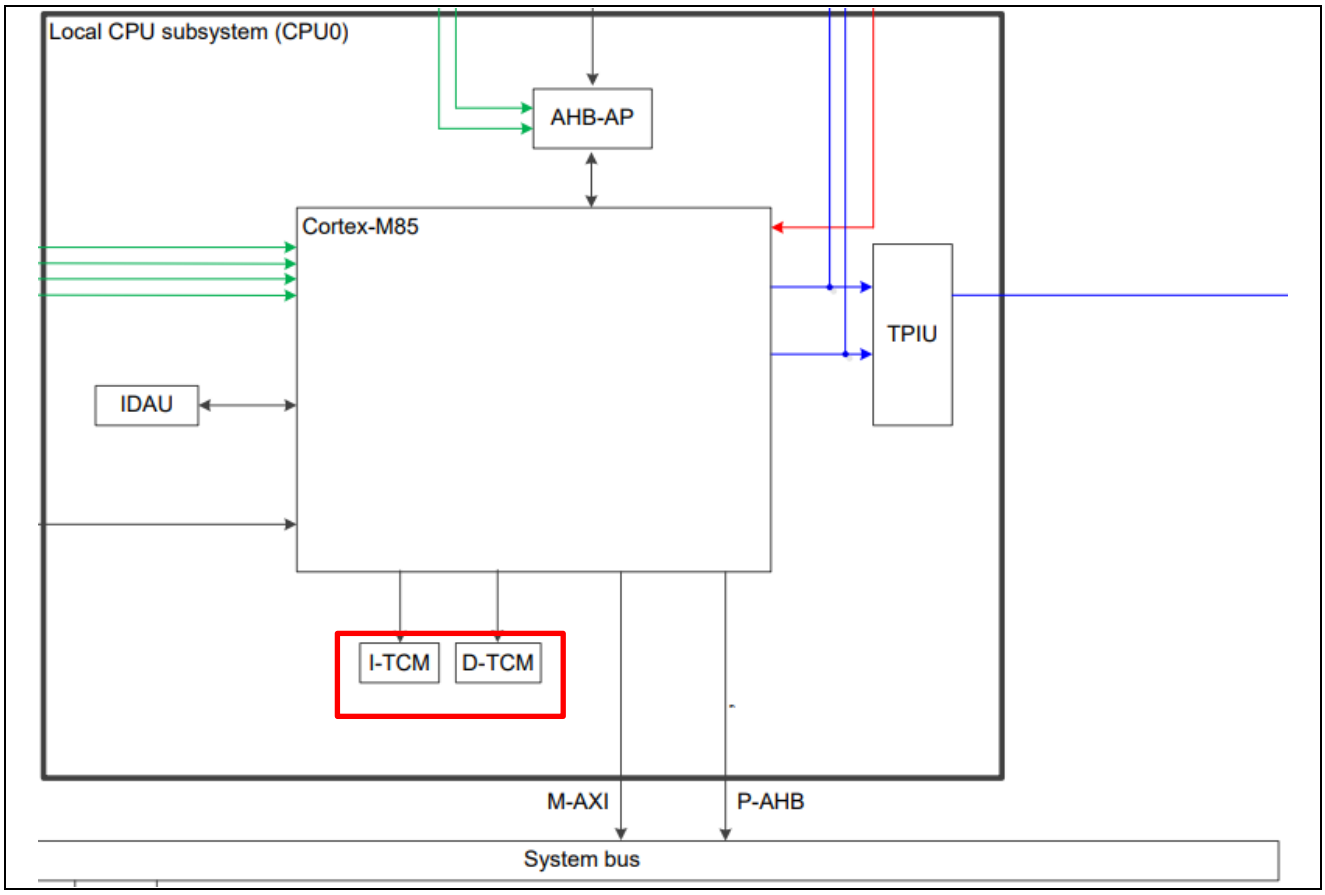


Figure 59. ITCM and DTCM in Local CPU Subsystem

You can place data in the DTCM section (.dtcm_data) in an FSP-based project using the __attribute__ directive, as shown in Figure 60. Placing Variables and Data in the DTCM Section.

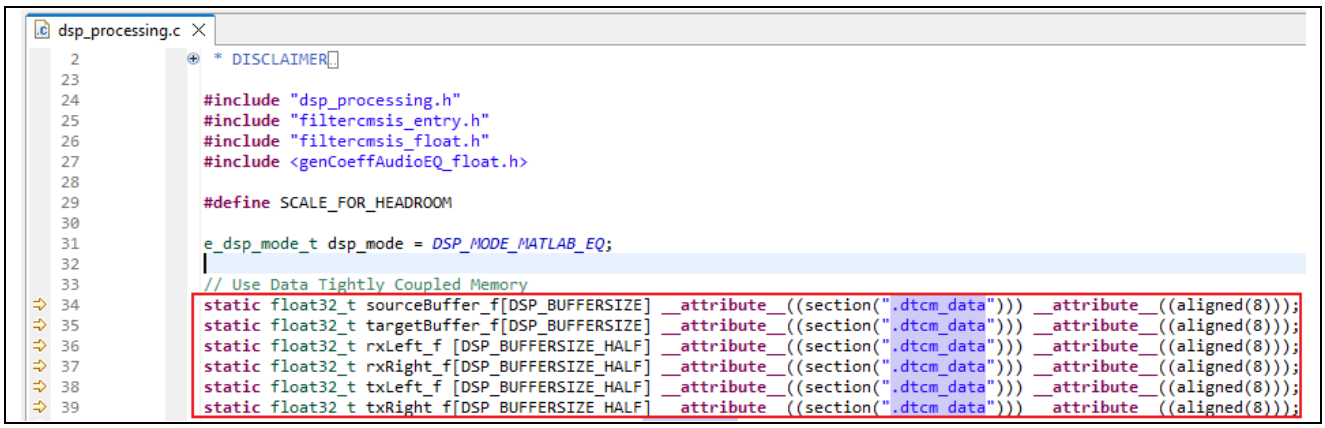


Figure 60. Placing Variables and Data in the DTCM Section

The above data placement in the DTCM section can be confirmed using the memory map (*.map) file generated by the compiler, as shown in Figure 61. Memory Map - Data Placed in DTCM Area

.

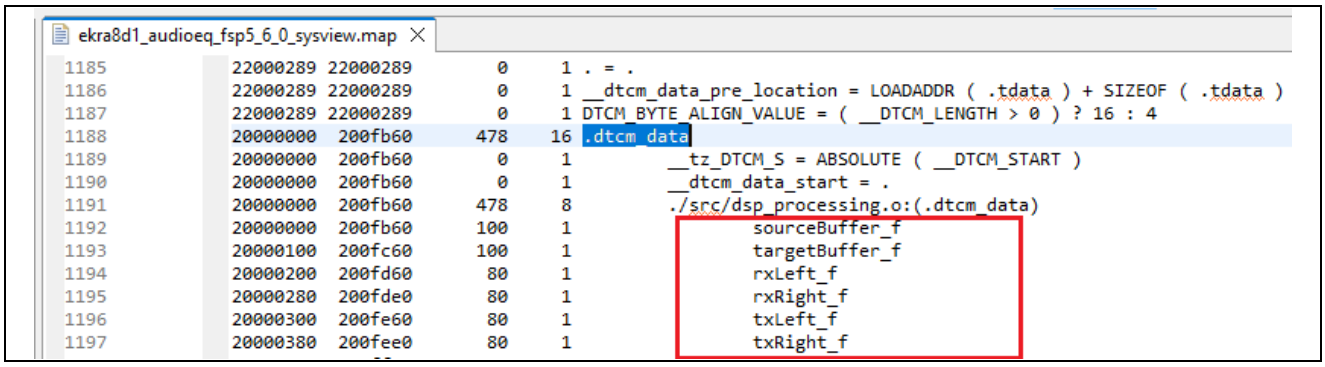


Figure 61. Memory Map - Data Placed in DTCM Area

You can also place program instructions in the ITCM section (.itcm_data) using the __attribute__ directive. Figure 62. Place a DSP Function in the ITCM Section shows an example of a modification to place the function

\ra\arm\CMSIS-DSP\Include\dsp\filtering_functions.h\arm_biquad_cascade_df2T_f32 in the ITCM area.

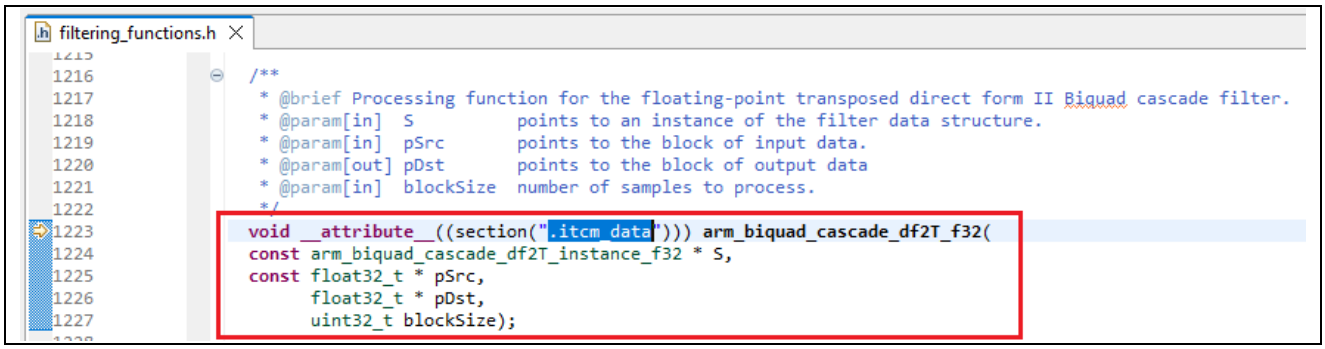


Figure 62. Place a DSP Function in the ITCM Section

You can confirm code placement using the memory map (*.map) file generated by the compiler, as shown in Figure 63. Memory Map - DSP Function in ITCM Area.

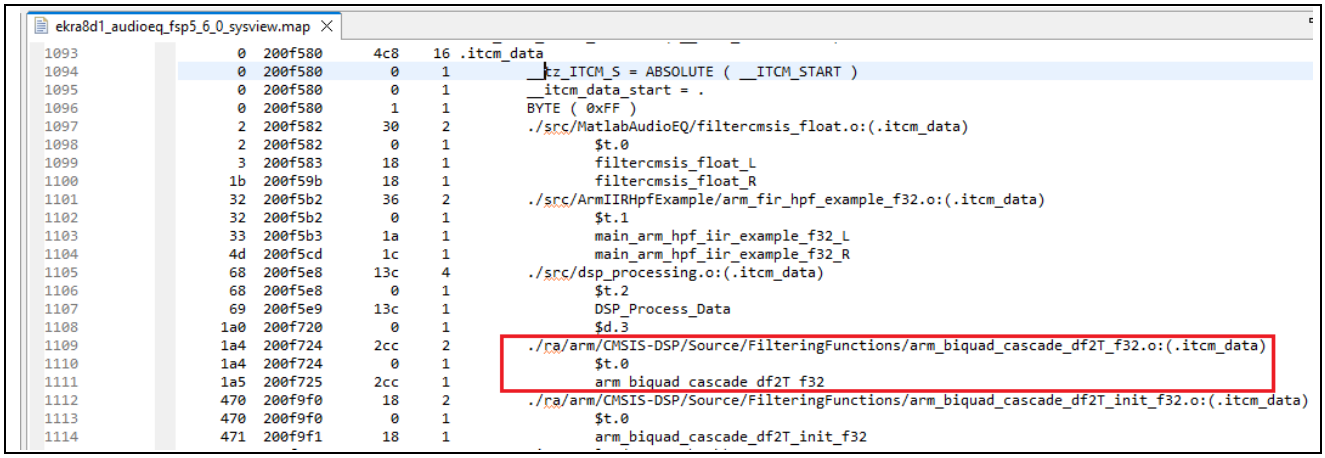


Figure 63. Memory Map - DSP Function in ITCM Area

If the CMSIS DSP Library source code is updated with a newer version of FSP, then you must add the "__attribute__ directive" to any affected CMSIS DSP Functions again, as shown in the example in Figure 62. Place a DSP Function in the ITCM Section.

5.3.3 Improve Performance Using ARM CMSIS DSP Optimizations

You can improve performance further with the CMSIS DSP Library by using compiler preprocessor macros (Defines -D) to select additional function implementation optimizations when the library is built, as shown in Figure 64. Compiler Optimization - Macro Defines for CMSIS DSP and Figure 65. Assembler Optimization - Macro Defines for CMSIS DSP.

Examples of preprocessor macros are:

- ARM_MATH_LOOPUNROLL
- ARM_MATH_HELIUM
- ARM_MATH_MVEF

In the Audio EQ project, the ARM_MATH_LOOPUNROLL optimization made the most significant improvement in DSP processing with a 24.32% decrease in latency.

The ARM_MATH_HELIUM optimization made an improvement with a 6% decrease in latency.

The ARM_MATH_MVEF optimization made a very small improvement with a 0.09% decrease in latency.

Under certain use cases, the optimizations provide further latency improvements but are MCU (ARM core) and CMSIS DSP function-dependent. It is recommended to use the latency measurement in this project to take a measurement before and after enabling an optimization option. Not all MCUs support the possible CMSIS function optimizations. Consult the CMSIS DSP documentation for specific details on which support is available for your MCU and the CMSIS DSP functions in use.

Arm Software GitHub CMSIS_5 DSP

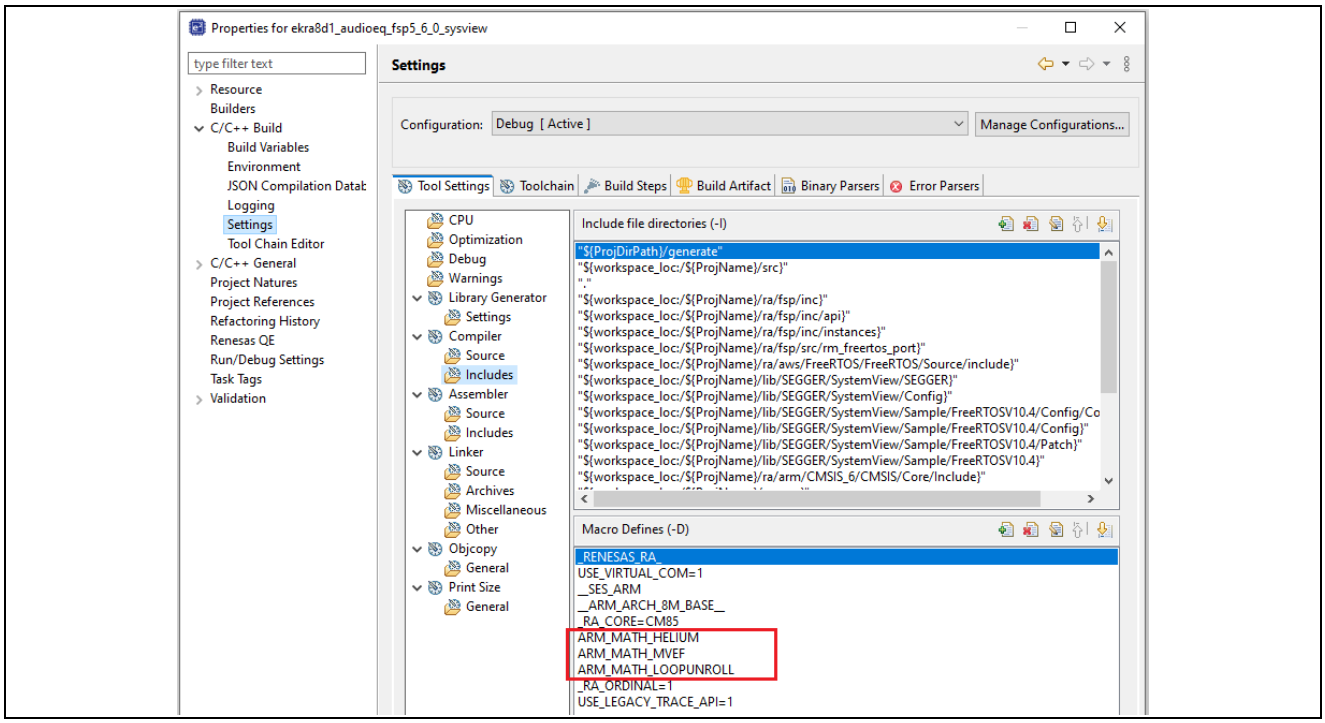


Figure 64. Compiler Optimization - Macro Defines for CMSIS DSP

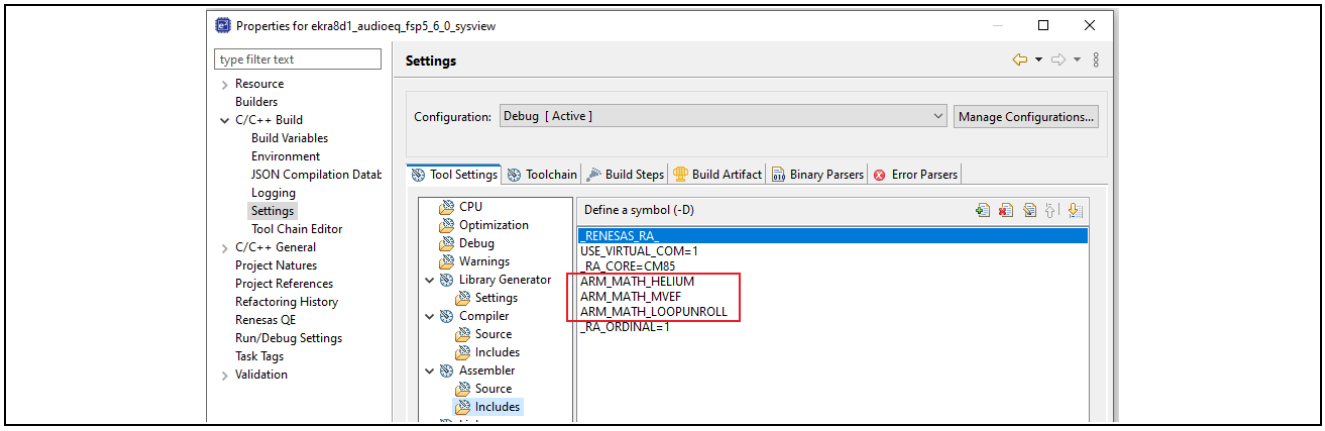


Figure 65. Assembler Optimization - Macro Defines for CMSIS DSP

6. Next Steps

After following the design workflow presented in this application note, the next steps that should be considered for your project are:

1. Replace the Audio EQ with your DSP filters or algorithms and observe changes in system bandwidth and processing latency.
2. Analyze the impact of the DMAC/DTC on system bandwidth and latency by removing its usage from the reference project.
3. Consider processing multiple audio streams using multiple threads and then synchronize the results at the end. Consider off-loading the data movement performed with the CPU by using DMAC/DTC transfers for audio stream data to and from memory.
4. Consider using DSP processing for multi-channel audio systems (for example, Dolby 5.1 Surround or Dolby Atmos).
5. Drive a display in parallel to control the DSP filter parameters.
6. Use a different audio codec or switch to an ADC/DAC for cost savings.
7. Move the audio circular buffers from internal SRAM to external SDRAM or OSPI RAM and evaluate the performance.
8. Consider Firmware OTA techniques for updating audio algorithms with a cloud server.

7. References

RA8D1 MCU Hardware User's Manual

FSP Software User's Manual

ARM Cortex M85 CPU Technical Reference Manual

MATLAB Documentation

Arm Helium Technology M-Profile Vector Extension (MVE) for Arm Cortex-M Processors Reference Book - Jon Marsh - ISBN: 978-1-911531-23-4.

8. Website and Support

Visit the following URLs to learn about key elements of the RA family, download components and related documentation, and get support:

RA Product Information	renesas.com/ra
RA Product Support Forum	renesas.com/ra/forum
RA Flexible Software Package	renesas.com/fsp
Renesas Support	renesas.com/support

Revision History

Rev.	Date	Description	
		Page	Summary
1.00	Feb.24.25	—	Initial release

General Precautions in the Handling of Microprocessing Unit and Microcontroller Unit Products

The following usage notes are applicable to all Microprocessing unit and Microcontroller unit products from Renesas. For detailed usage notes on the products covered by this document, refer to the relevant sections of the document as well as any technical updates that have been issued for the products.

1. Precaution against Electrostatic Discharge (ESD)

   A strong electrical field, when exposed to a CMOS device, can cause destruction of the gate oxide and ultimately degrade the device operation. Steps must be taken to stop the generation of static electricity as much as possible, and quickly dissipate it when it occurs. Environmental control must be adequate. When it is dry, a humidifier should be used. This is recommended to avoid using insulators that can easily build up static electricity. Semiconductor devices must be stored and transported in an anti-static container, static shielding bag or conductive material. All test and measurement tools including work benches and floors must be grounded. The operator must also be grounded using a wrist strap. Semiconductor devices must not be touched with bare hands. Similar precautions must be taken for printed circuit boards with mounted semiconductor devices.
2. Processing at power-on

   The state of the product is undefined at the time when power is supplied. The states of internal circuits in the LSI are indeterminate and the states of register settings and pins are undefined at the time when power is supplied. In a finished product where the reset signal is applied to the external reset pin, the states of pins are not guaranteed from the time when power is supplied until the reset process is completed. In a similar way, the states of pins in a product that is reset by an on-chip power-on reset function are not guaranteed from the time when power is supplied until the power reaches the level at which resetting is specified.
3. Input of signal during power-off state

   Do not input signals or an I/O pull-up power supply while the device is powered off. The current injection that results from input of such a signal or I/O pull-up power supply may cause malfunction and the abnormal current that passes in the device at this time may cause degradation of internal elements. Follow the guideline for input signal during power-off state as described in your product documentation.
4. Handling of unused pins

   Handle unused pins in accordance with the directions given under handling of unused pins in the manual. The input pins of CMOS products are generally in the high-impedance state. In operation with an unused pin in the open-circuit state, extra electromagnetic noise is induced in the vicinity of the LSI, an associated shoot-through current flows internally, and malfunctions occur due to the false recognition of the pin state as an input signal become possible.
5. Clock signals

   After applying a reset, only release the reset line after the operating clock signal becomes stable. When switching the clock signal during program execution, wait until the target clock signal is stabilized. When the clock signal is generated with an external resonator or from an external oscillator during a reset, ensure that the reset line is only released after full stabilization of the clock signal. Additionally, when switching to a clock signal produced with an external resonator or by an external oscillator while program execution is in progress, wait until the target clock signal is stable.
6. Voltage application waveform at input pin

   Waveform distortion due to input noise or a reflected wave may cause malfunction. If the input of the CMOS device stays in the area between V_{IL} (Max.) and V_{IH} (Min.) due to noise, for example, the device may malfunction. Take care to prevent chattering noise from entering the device when the input level is fixed, and also in the transition period when the input level passes through the area between V_{IL} (Max.) and V_{IH} (Min.).
7. Prohibition of access to reserved addresses

   Access to reserved addresses is prohibited. The reserved addresses are provided for possible future expansion of functions. Do not access these addresses as the correct operation of the LSI is not guaranteed.
8. Differences between products

   Before changing from one product to another, for example to a product with a different part number, confirm that the change will not lead to problems. The characteristics of a microprocessing unit or microcontroller unit products in the same group but having a different part number might differ in terms of internal memory capacity, layout pattern, and other factors, which can affect the ranges of electrical characteristics, such as characteristic values, operating margins, immunity to noise, and amount of radiated noise. When changing to a product with a different part number, implement a system-evaluation test for the given product.

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