User manual for MERUS™ audio amplifier configurator

MA2304DNS/MA2304PNS

About this document

Scope and purpose

This document describes how to install and use the MERUS™ audio amplifier configurator graphical user interface (GUI), its DSP flows and algorithms.

Intended audience

Audio product design engineers, software engineers and tuning engineers.

Attention: ***Please read through this user manual before operating the board.***

Stuck or in need of help?

Support for Infineon's class D audio portfolio can be found quickly and easily by visiting the Class D Audio Amplifier IC Forum or by visiting community.infineon.com. The community forum features members of the audio applications team that are ready to provide timely support, helping you get your designs done quickly, reliably, and right the first time.

Safety precautions

Note: *Please note the following warning regarding the hazards associated with development systems.*

Table 1 **Safety precautions**

	Caution: *The evaluation or reference board contains parts and assemblies sensitive to electrostatic discharge (ESD). Electrostatic control precautions are required when installing, testing, servicing or repairing the assembly. Component damage may result if ESD control procedures are not followed. If you are not familiar with electrostatic control procedures, refer to the applicable ESD protection handbooks and guidelines.*

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1 Quick start

This section covers hardware configuration, software installation and running the MERUS™ audio amplifier configurator.

1.1 What's included

The evaluation kit box comes with:

- EVAL_AUDIO_MA2304xNS board
- Three interface boards (analog in, S/PDIF coax and S/PDIF optical)
- Micro USB cable
- 22 µH power inductor for performance measurements (refer to the hardware user manual for details)

The following items are required, in addition to the items in the evaluation kit, for basic UI and DSP evaluation:

- DC power supply: 10 to 20 V DC/6 A for BTL mode or 10 to 20 V DC/12 A for PBTL mode
- Loudspeaker(s): 2 to 8 Ω speaker impedance
- (Optional): 2 to 8 Ω power resistor, class D filter and audio analyzer

1.2 Hardware setup

For quick and easy evaluation, plug one of the interface boards into the J1 I/O header on top of the MA2304xNS board, as shown in Figure 1.

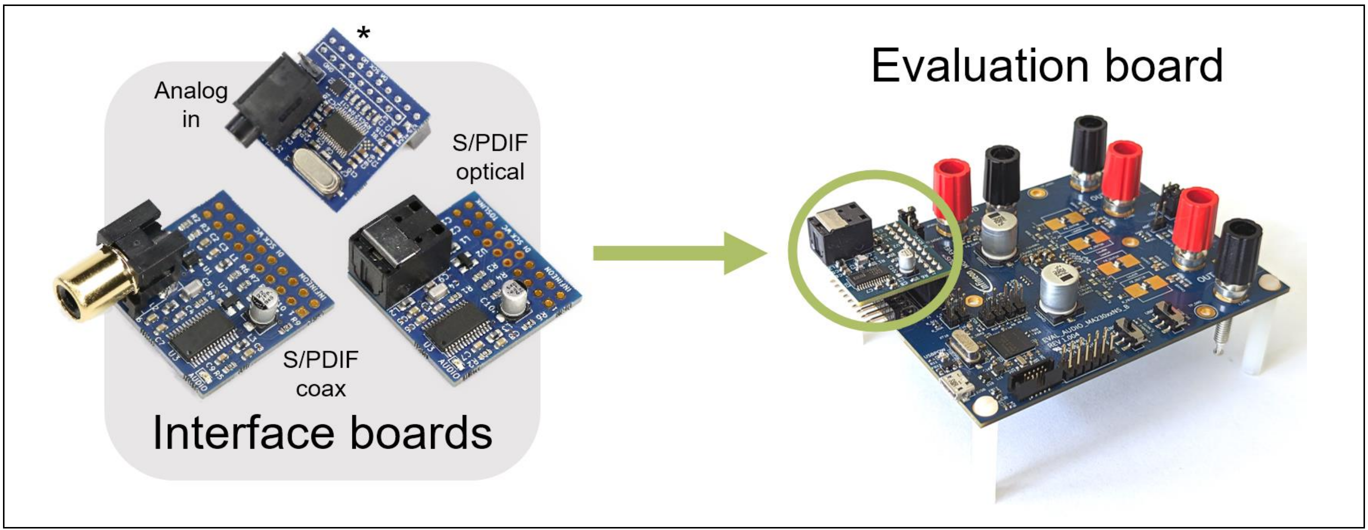

Figure 1 Interface boards and MA2304xNS evaluation board

An MA2304xNS evaluation setup with an interface board can play audio without using software, as the interface board's digital audio output format matches the MA2304xNS device default.

The MA2304xNS board's jumper and switch settings are preconfigured for basic BTL operation as default before shipping.

Refer to the hardware user manual for interface board specifications.

Connect the MA2304xNS evaluation board as shown in Figure 2:

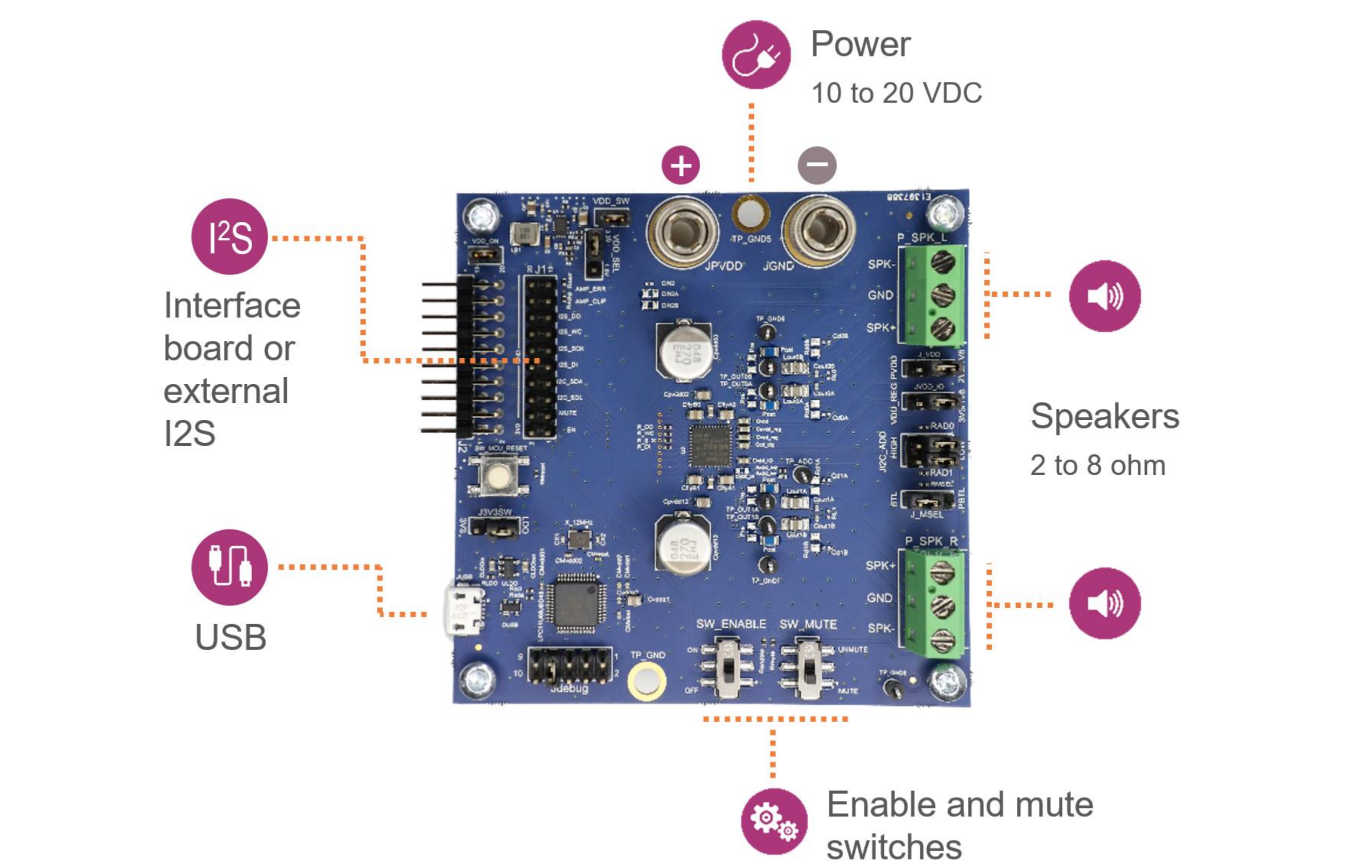

Figure 2 Basic MA2304xNS evaluation board configuration for use with the MERUS™ amplifier tool

Refer to the hardware user manual for default jumper and switch settings and other options.

1.3 Software installation

The MERUS™ audio amplifier configurator can be downloaded by visiting https://softwaretools.infineon.com/ and registering your board. Once registered, the software can be installed through the Infineon developer center launcher or downloaded as a separate installer file. This will also install all the necessary device drivers to communicate with the MA2304xNS evaluation board.

To install, search for “merus” under the Tools tab of the Infineon developer center website or launcher:

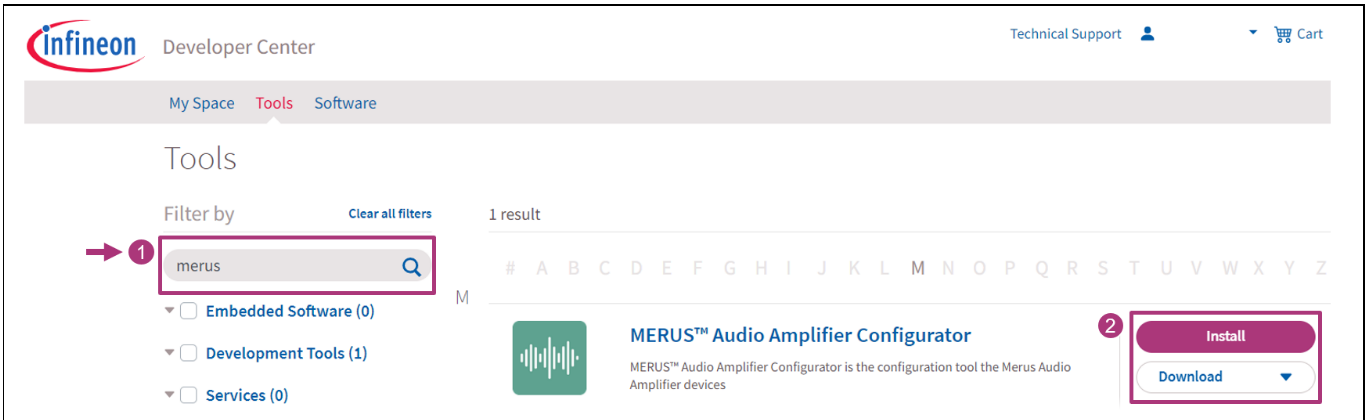

Figure 3 How to find the MERUS™ audio amplifier configurator

For proper operation, the software must be installed using administrative privileges, as shown in Figure 4.

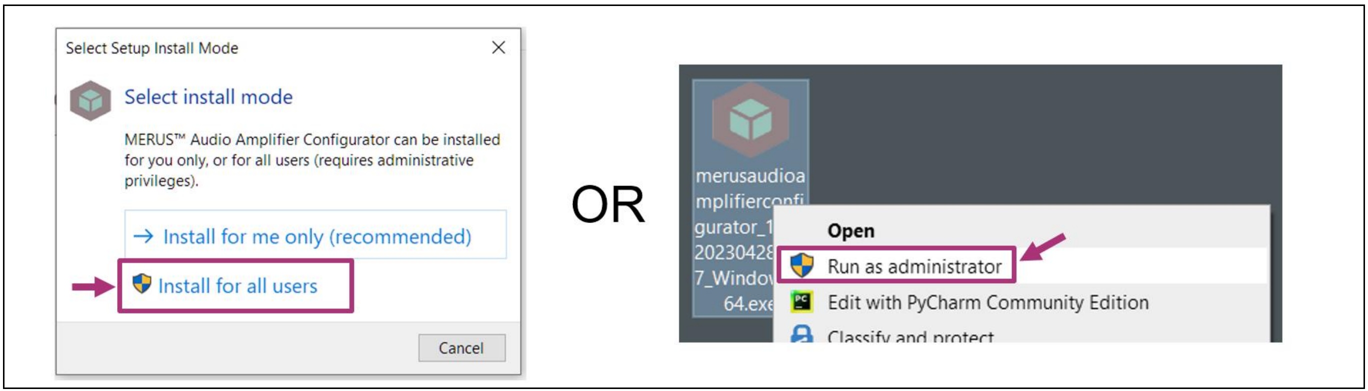

Figure 4 **How to install the software using administrative privileges**

1.4 Firmware flash for USB MCU

Please refer to the hardware manual for step-by-step instructions on how to update the board firmware.

1.5 Running the MERUS™ audio amplifier configurator

Prior to launching the software:

1. Power up the **P_{VDD}** DC power supply. 18 V is recommended for typical evaluation.
2. Connect the **Micro USB** cable from the computer to the MA2304xNS board.
3. Connect an audio source to the **interface board** input (analog, S/PDIF coax or optical) and keep it muted.
4. Set the **SW_ENABLE** switch to **ON** and **SW_MUTE** to **UNMUTE**, as shown in Figure 5.

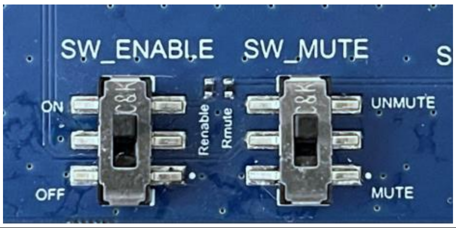

Figure 5 **MA2304xNS ENABLE and MUTE switches**

At this point, the MA2304xNS should be able to play audio. Set the audio source to a low volume and unmute it.

Once the hardware is configured, double-click the "Developer center launcher" or the "MERUS audio amplifier configurator" icon on your desktop to launch the application:

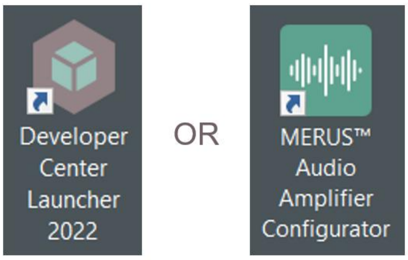

Figure 6 **MERUS™ audio amplifier configurator**

After a few seconds, the following screen should appear:

Figure 7 **MERUS™ audio amplifier configurator**

At this point, feel free to adjust the volume level or put the device on standby.

The Connected indicator means that the software can communicate with the MA2304xNS device.

To access advanced device features and DSP flows, click the Configuration button and select the **Configuration** option:

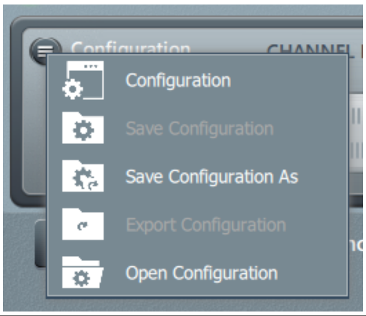

Figure 8 **Configuration dropdown menu**

The Configuration page should now appear. This page provides access to advanced device features, DSP flows, device registers and status indicators.

Figure 9 **Configuration page**

The following section describes the MERUS™ audio amplifier configurator in detail.

2 The MERUS™ audio amplifier configurator

The MERUS™ audio amplifier configurator is a GUI used to fully evaluate the MA2304xNS features. It provides access to the the MA2304xNS device registers through the use of simple controls as well as a register browser. The MA2304DNS can run various DSP flows specifically tailored for audio applications.

2.1 Demo mode

The MERUS™ audio amplifier configurator can run without having any board connected to the computer. In this case, the GUI will launch in demo mode, as shown in Figure 10. This mode can be used to evaluate the software and build user configurations. In this mode, multiple MA2304xNS devices can be added as well by clicking the "+" symbol on the bottom right of the window.

Figure 10 Initial screen in demo mode

2.2 Normal mode

If the MA2304xNS EVK is properly configured, powered up, connected and its ENABLE switch set to "ON", the following screen should appear when launching the software:

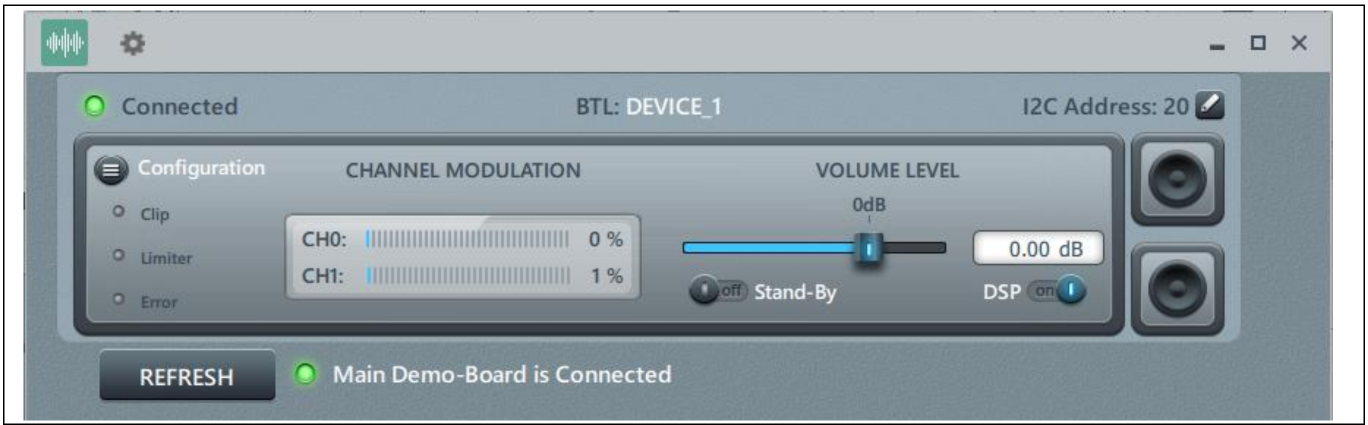

Figure 11 Initial screen in normal mode

2.3 MCU connected mode

If the MA2304xNS device is not detected, the following screen will appear instead:

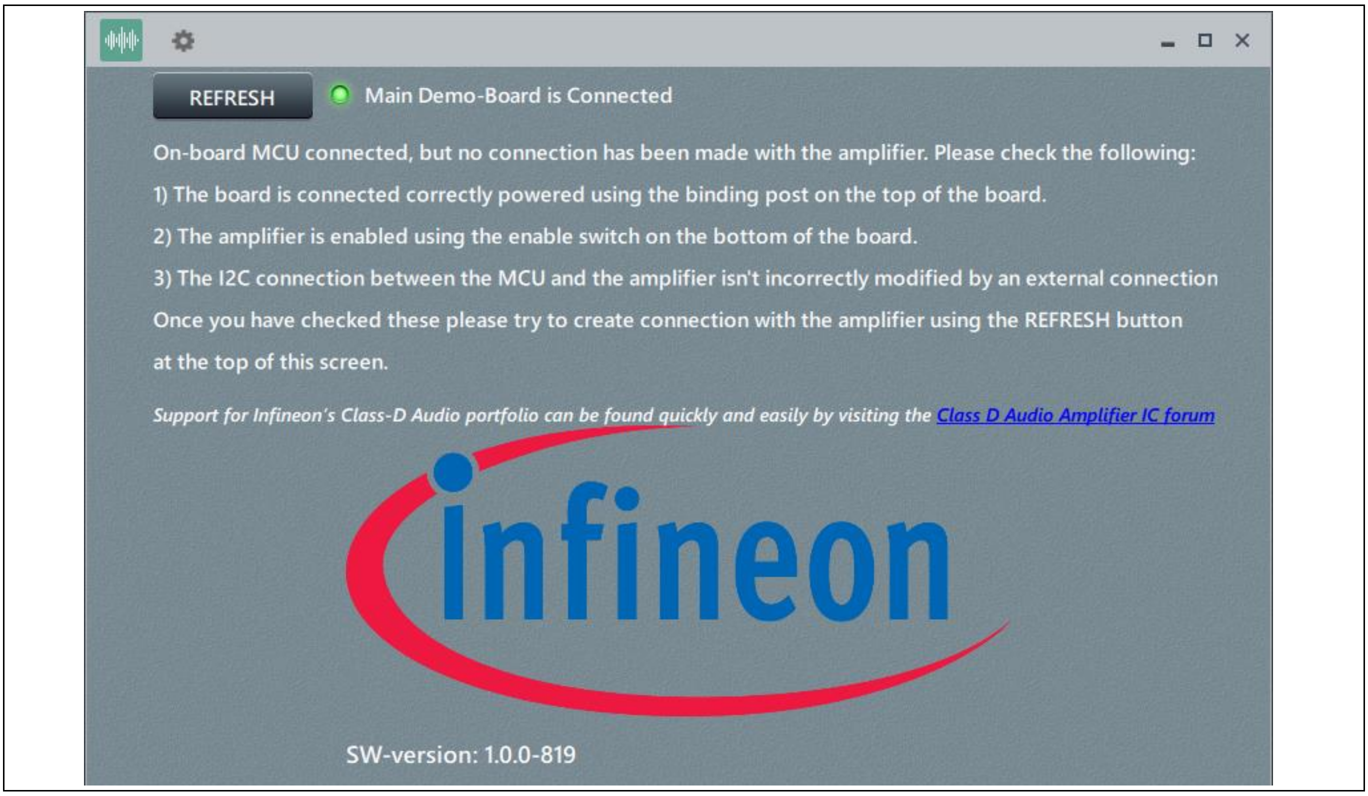

Figure 12 MCU connected mode

In this case, there is still connection to the onboard MCU, but not the amplifier.

This screen only contains a refresh button and a green connection light indicating that a connection is made with the onboard MCU.

If the connection light turns red, it means that the MCU is not detected.

To go to the initial screen (as shown in Figure 11) make sure that the amplifier is powered and its ENABLE switch is set to “ON”, then click the REFRESH button.

3 Using the software

3.1 Initial screen

The initial screen has various indicators and controls, as shown in Figure 13 and Figure 14. The name and description of each are also provided in these figures.

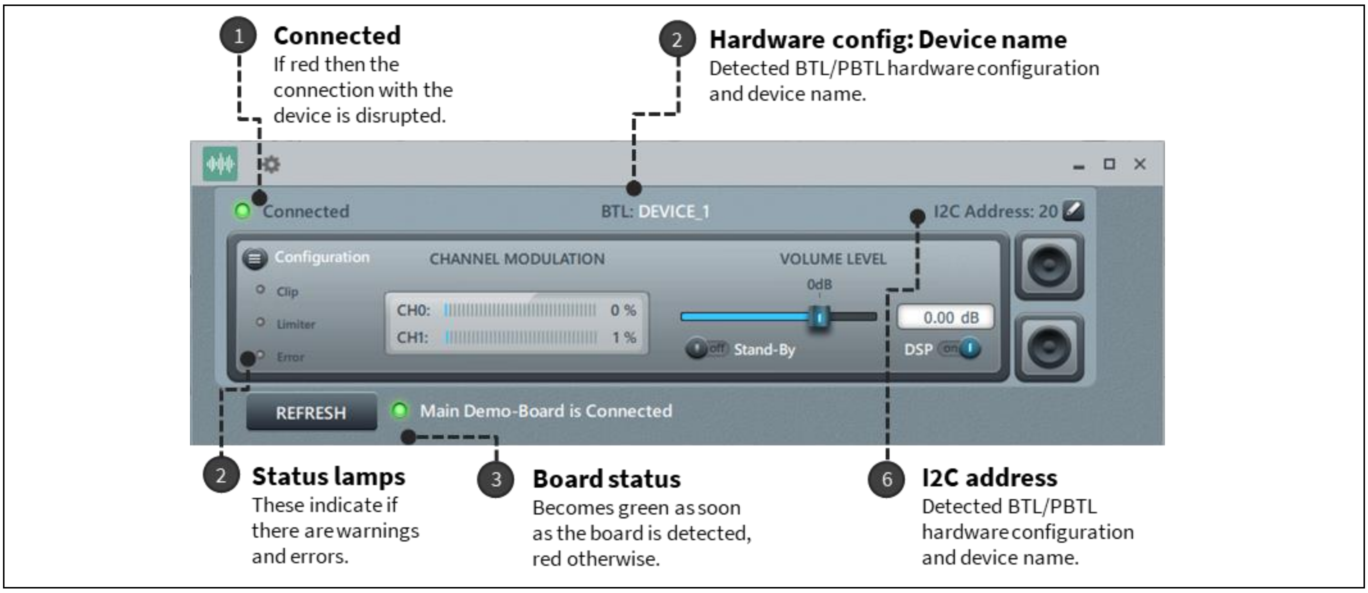

Figure 13 Initial screen indicators

To access advanced device features and DSP flows, click the  button and select the Configuration option:

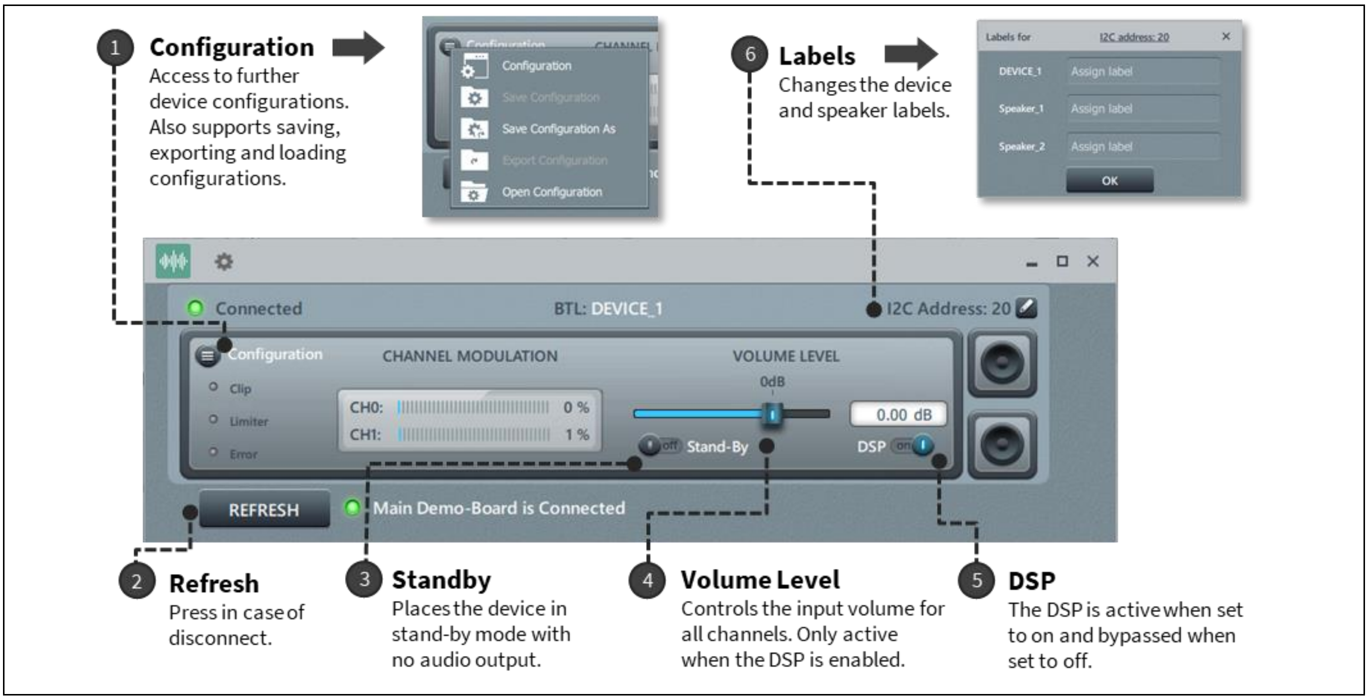

Figure 14 Initial screen controls

Additional information is available as tooltips by hovering the mouse cursor on top of a control or indicator, as shown below:

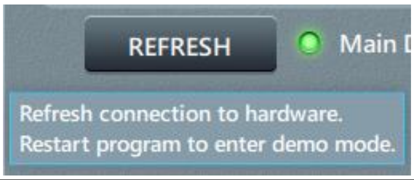

Figure 15 Tooltips

Tooltips can be enabled or disabled in the settings  menu:

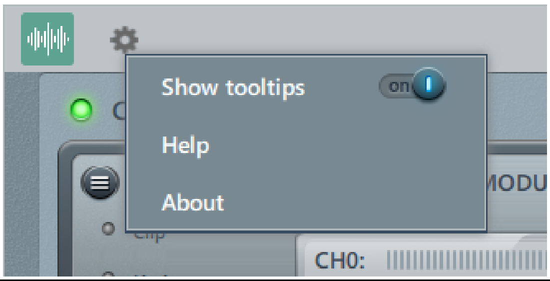

Figure 16 Enable or disable tooltips

The MERUS™ audio amplifier configurator detects if multiple MA2304xNS are connected on the same I²C bus and automatically stacks multiple instances of MA2304xNS devices, similar to a rack mount. This can be useful for systems with more than two speakers. Figure 17 is an example of a 2.1 speaker system (two speakers and a subwoofer). The Speakers rack is configured as stereo BTL, while the Subwoofer rack is mono PBTL.

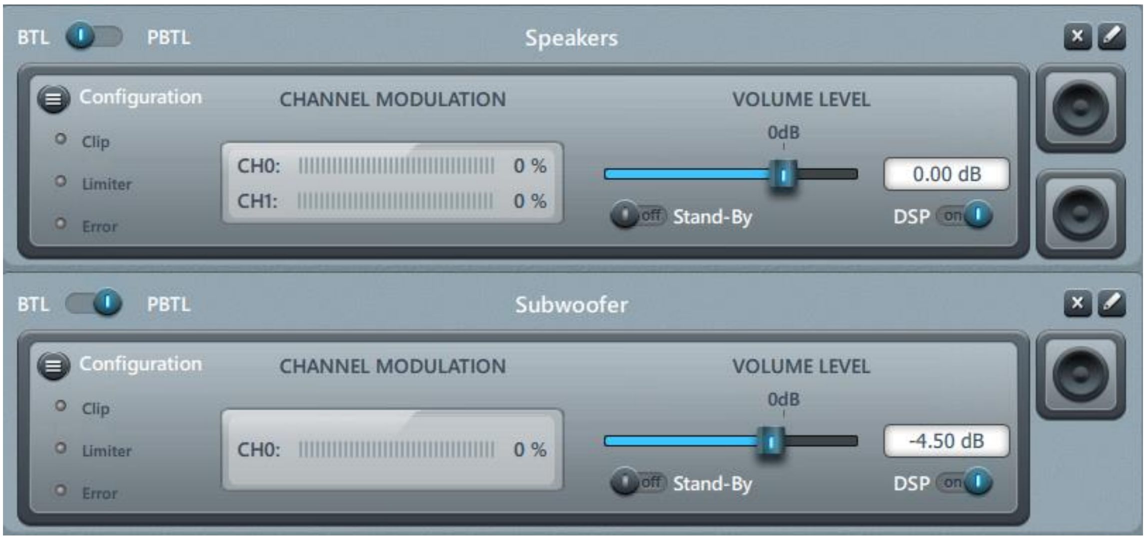

Figure 17 Example 2.1 system

Clicking the "Configuration" button opens the Configuration menu:

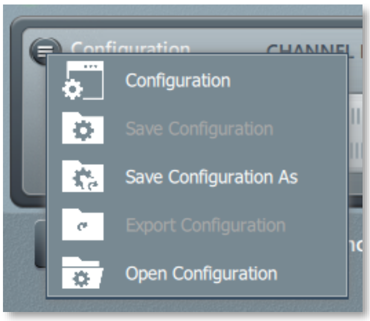

Figure 18 Configuration menu

Clicking "Configuration" opens the Configuration window, which is covered in the next section.

"Save configuration" saves the device state, including register settings and DSP flow into a folder. In addition, it generates .csv files, which can be used to import into a product.

"Export configuration" creates a header file that can also be used to import into a product.

The "Open configuration" option is used to browse to a previously saved configuration folder.

3.2 Configuration window

Use the Configuration menu in the initial screen to open the Configuration window.

The Configuration window provides access to device features, DSP flows (MA2304DNS only), a register browser and real-time status indicators. These items are covered in detail in the following sections.

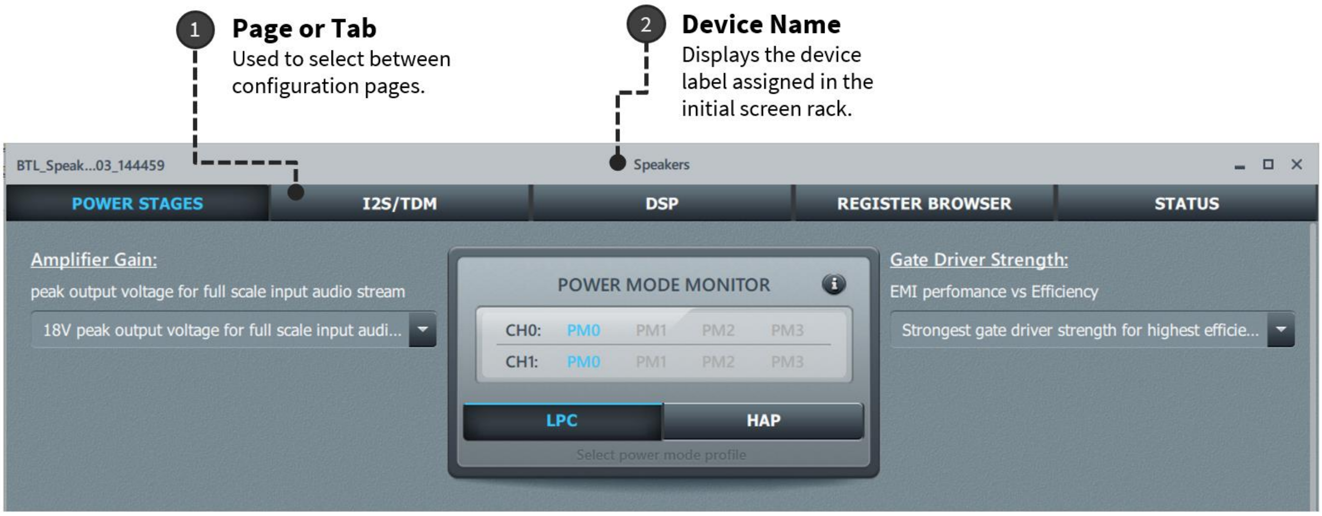

Figure 19 Configuration window

3.2.1 Power stages page

Through the power stage page, power mode profiles, amplifier gain and drive strength can be configured. Efficiency and THD+N plots for each power mode profile are provided as reference.

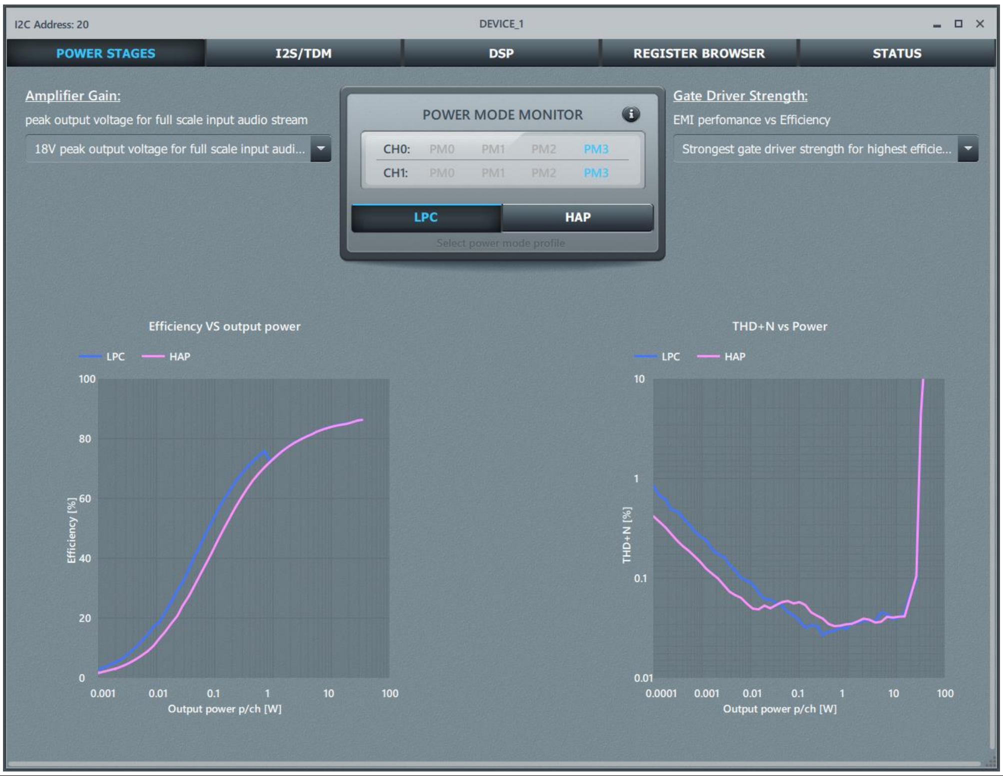

Figure 20 Power stages page

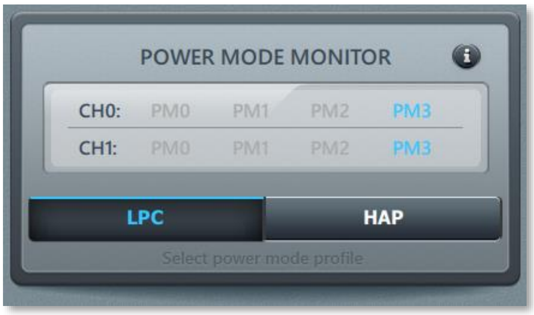

Figure 21 Power mode monitor

The power mode monitor shows the current active power mode per channel in real time.

Two power mode profiles can be selected:

- Low power consumption (LPC) – low power consumption, slightly higher noise
- High audio performance (HAP) – low noise, slightly higher power consumption

Power mode profiles trade off between power consumption and output noise.

Figure 22 Amplifier gain control

The amplifier gain is the relationship between a full-scale input (0 dB_{FS}) and output voltage (V_{peak}). Each one of these options has a recommended power supply voltage.

This control modifies the pvdd_scale register, as shown in Table 2.

Table 2 P_{VDD} voltage guideline

pvdd_scale	Recommended P_{VDD} voltage
00	10 V
01	12 V
10	15 V
11 (default)	18 V

Please refer to the device datasheet for pvdd_scale vs. P_{VDD} range plots.

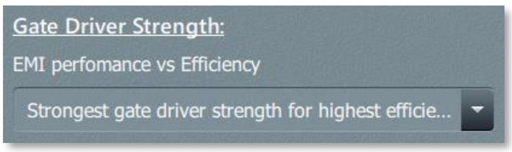

Figure 23 Gate driver strength control

The gate driver strength can be adjusted as a trade-off between EMI performance and efficiency.

3.2.2 I²S/TDM page

On the I²S/TDM page, communication-related configurations can be set up. The I²S configuration is default.

To route the I²S output out of the MA2304DNS, it is necessary to enable the "TX_ENABLE" control and load a DSP program that supports this.

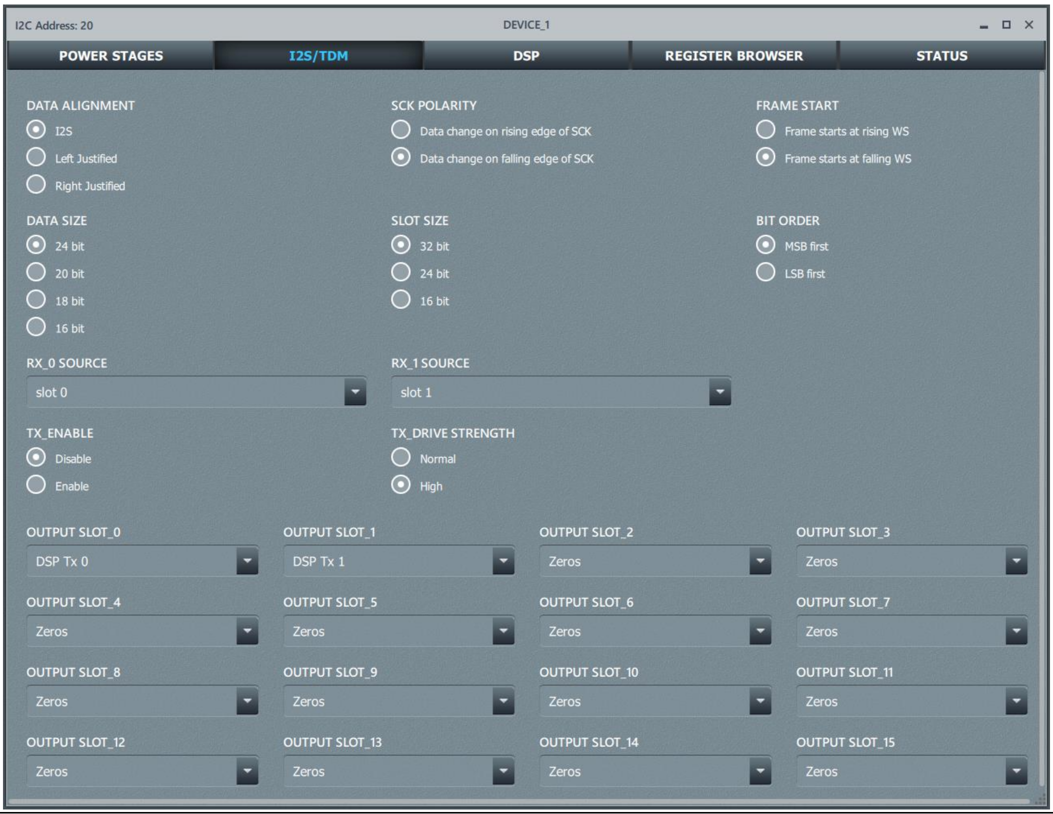

Figure 24 **I²S/TDM page**

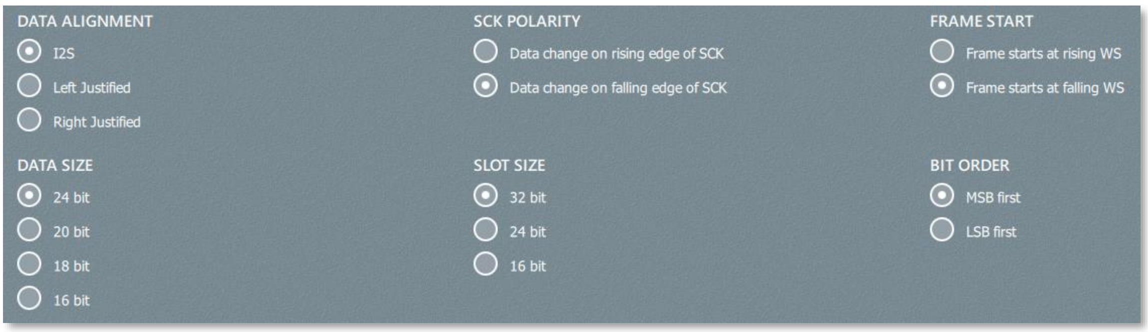

Figure 25 **Audio interface format controls**

These controls are used to configure the audio interface format. Refer to the device datasheet for details on audio interface format and timing.

The SLOT SIZE control affects the number of SCK cycles per channel.

The DATA SIZE control adjusts the bit depth of the data to and from the audio interface.

Figure 26 Rx source controls

Selects Rx_0 (left) and Rx_1 (right) DIN source. Each control has a range of 0 to 15 slots. These slots are usually sourced by an application processor or MCU.

Figure 27 TX_ENABLE and TX_DRIVE_STRENGTH controls

The TX_ENABLE control enables/disables I2S_DO output data from the DSP. The drive strength of the I2S_DO pin can also be configured through the TX_DRIVE_STRENGTH control.

Figure 28 Output slot controls

Each output slot control corresponds to a TDM time slot. In these, output refers to Tx (or I2S_DO) data.

Each control can be configured to output Zeros, High-Z, DSP Tx0 (left channel) or DSP Tx1 (right channel).

On multichannel applications, slots used by other MA2304xNS devices should be configured as High-Z to prevent contention. Configure unused channels as Zeros to save on bus power consumption.

3.2.3 DSP page

The DSP page has multiple audio DSP flows to choose from and program into the MA2304DNS internal DSP. It contains a graphical view of the current DSP flow and a plot of the expected amplitude and phase responses with options for viewing single channels or combined responses.

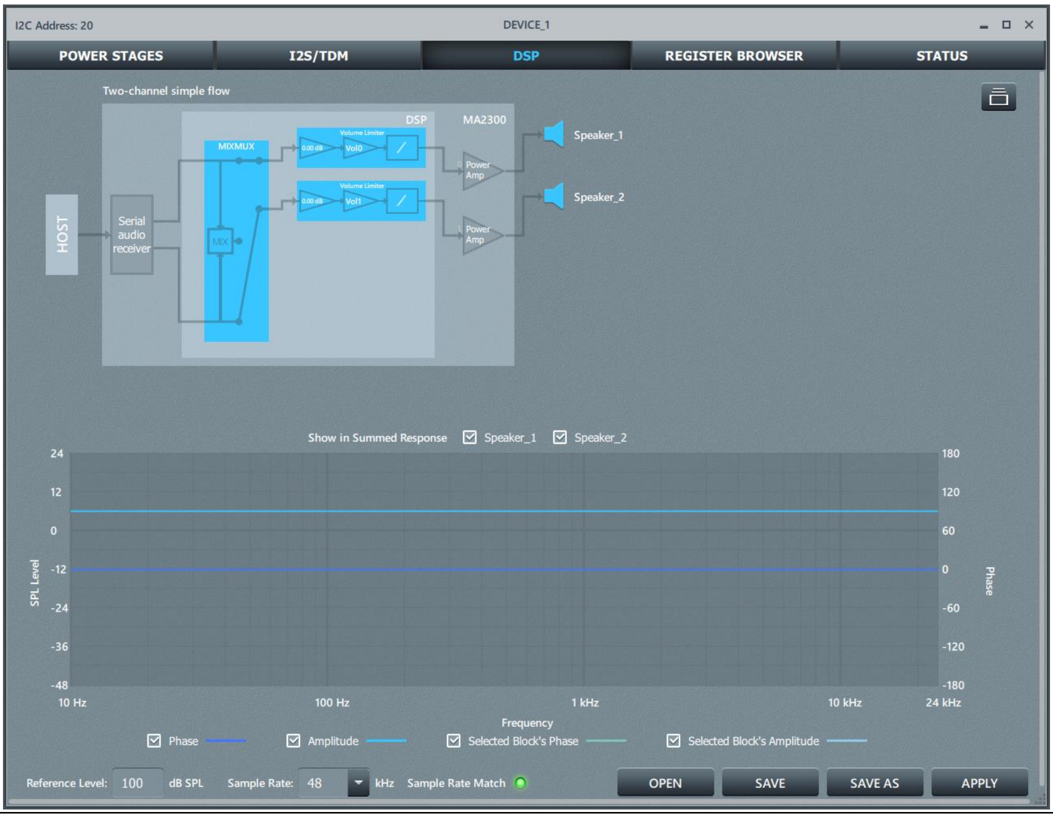

Figure 29 DSP page

Figure 30 Library button

To select a DSP flow, press the library button in the top right corner of the screen.

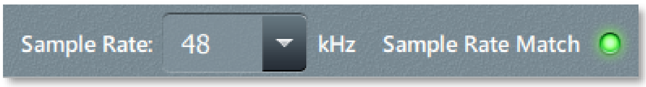

Figure 31 Sample rate control

After loading a DSP flow, please ensure that the sample rate selected in this control matches the sample rate of the incoming I²S signal to the device.

Figure 32 OPEN, SAVE, and APPLY buttons

A DSP flow can be saved and loaded through these controls. To program a DSP flow onto the device, press APPLY and wait until the warning symbol is gone. In some cases, it will be necessary to press twice. In earlier software versions it is necessary to mute the device before clicking APPLY. Otherwise, it may result in a loud popping sound and can potentially damage the speaker.

WARNING!

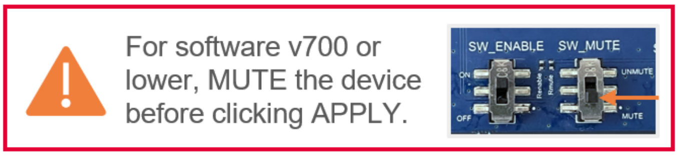

Figure 33 Warning

In other words, if the APPLY button is red, make sure to MUTE first before clicking.

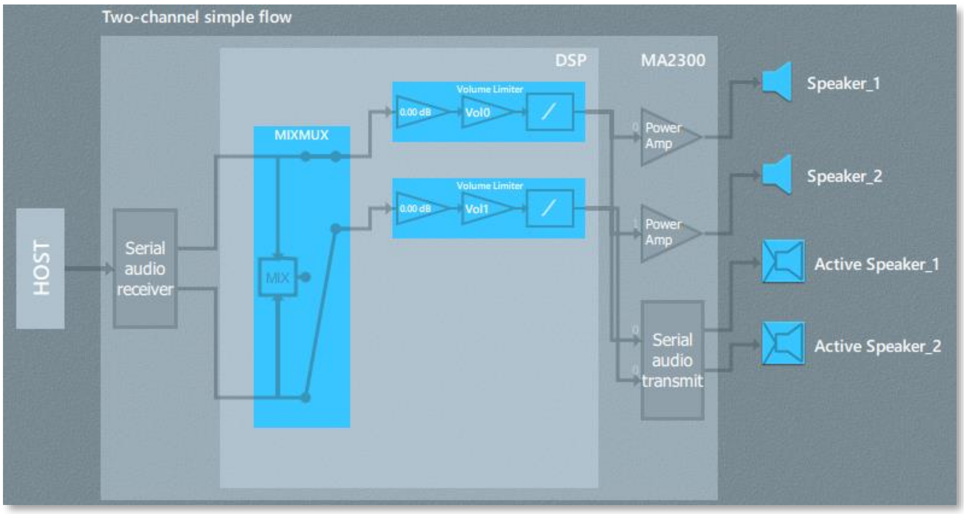

Figure 34 DSP flow diagram

The DSP flow is a block diagram of interconnected audio blocks that can be programmed into the MA2304DNS DSP. This screenshot shows the audio data flowing from the host (applications processor, MCU, etc.) into the digital audio interface of the MA2304xNS, through the DSP and its audio blocks, to the power amplifier and to the speaker(s).

If the TX_ENABLE control in the I^2S/TDM page is set to "enable" two additional outputs will be shown, labeled "Active Speaker_x". This will route the DSP output(s) to the I2S_DO pin for connection to external audio interfaces. This can be useful for multichannel applications or to send the output back to the host for echo canceling, as an example.

Detailed information on available DSP flows and its audio blocks is provided in depth in the DSP flows and DSP audio blocks chapters.

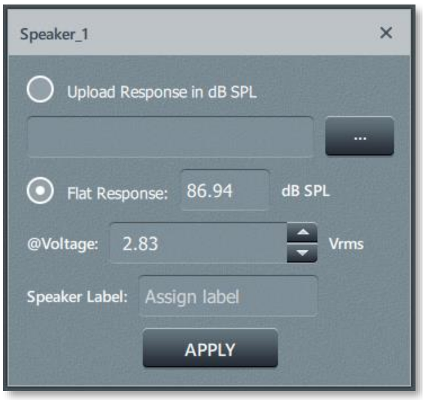

Figure 35 Speaker properties

Double-clicking any of the speakers  in the DSP flow will pop up a "speaker properties" window. In addition to assigning a custom label to the speakers, frequency response measurements (.csv format) can be imported by clicking the browse  button.

Each line of the .csv file must have a frequency and amplitude separated by a semicolon ";". The third column, phase, is optional:

Frequency(Hz);Amplitude(dB);Phase(deg)

Only numbers, period "." and semicolon ";" are allowed. Comments or spaces are not supported.

Below are two examples, with and without phase:

```
10;98
20.5;98.7
50;100.3
100;100.7
200;98.2
500;96.7
1000;94.9
2000;96.8
5000;98.1
10000;99.5
20000;92.3
```

Figure 36 Frequency and amplitude .csv file example

```
20.874025;82.843;-49.7109
50.170902;101.811;-177.9095
100.341805;110.373;117.8002
200.317399;110.449;26.0996
500.244181;105.225;-1.7379
1000.122152;103.390;-15.4097
2000.244303;100.241;-15.0307
5000.244548;88.649;-113.6205
10000.122884;104.190;-13.2627
20000.879557;76.711;-79.5115
```

Figure 37 Frequency, amplitude, and phase .csv file example

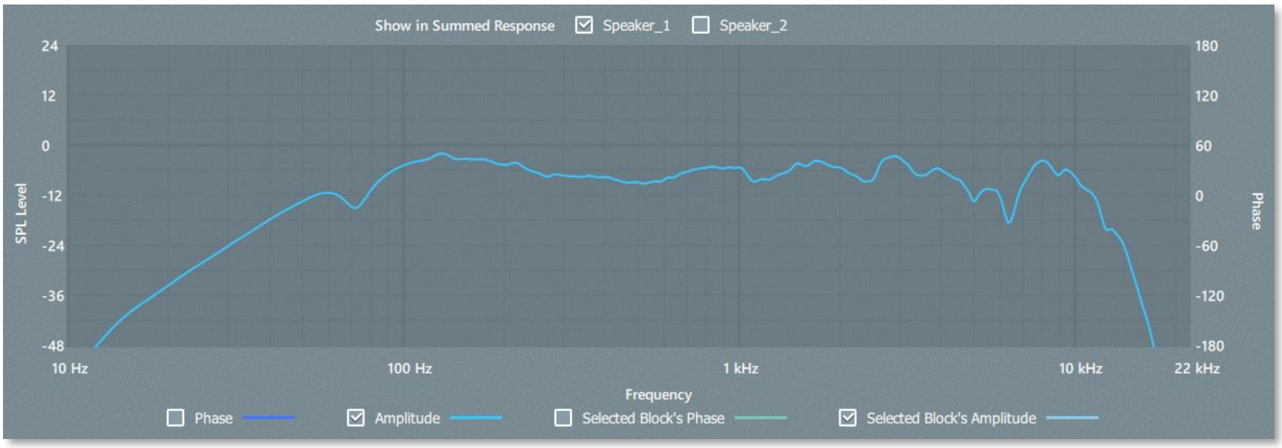

Figure 38 Frequency response plot

The frequency response plot can be configured to display the amplitude and phase of:

- The summed response of any combination of output channels
- Any individual block in the DSP flow diagram
- An imported custom speaker measurement

The summed response option can be useful to estimate the far-field response of two-way speakers, ported speakers and so on.

3.2.3.1 Example 1: EQ filter response plot

Let's open a two-channel flow and configure two left channel biquads as peaking filters (parametric EQ), as shown below:

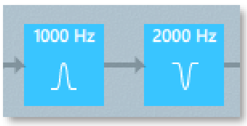

Figure 39 Two cascaded biquads

Now, let's enable the "Speaker_1" response and select the "Amplitude" option:

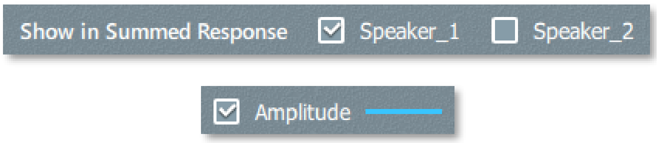

Figure 40 Enable "Speaker_1" response

The following peaks will then be shown in the frequency response plot corresponding to the peaking filters we just configured:

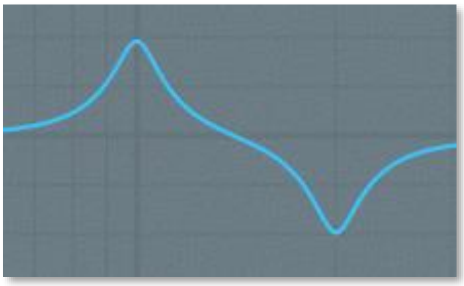

Figure 41 Frequency response plot

To observe the contribution of a single filter, check the "Selected Block's Amplitude" option:

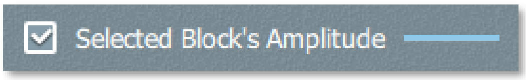

Figure 42 Selected Block's Amplitude

Then, left-click the desired biquad block:

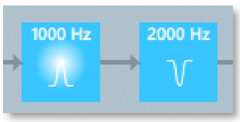

Figure 43 Biquad block

The plot will now display a combination of the first biquad response along with the overall DSP flow response:

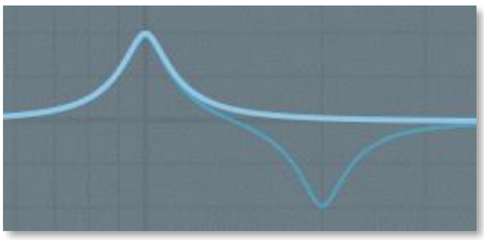

Figure 44 Plot showing first biquad response and overall DSP flow response

3.2.3.2 Example 2: Importing a custom speaker measurement and EQing

First, double-click the "Speaker_1" icon and import the .csv file of a speaker response measurement. Then, click "apply":

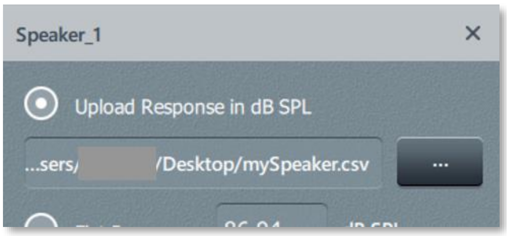

Figure 45 Import .csv file

Then, configure the checkboxes as shown below to show its amplitude response plot:

Figure 46 Amplitude response plot

We will now try to flatten the frequency response by configuring some biquads – an EQ cut, a bass shelf and a low-pass filter:

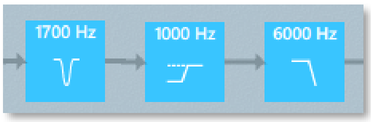

Figure 47 Configure biquads

A left click on the bass shelf filter reveals its contribution to the frequency response on the graph. A shelf filter response with a 15 dB cut below 1 kHz is shown in light blue:

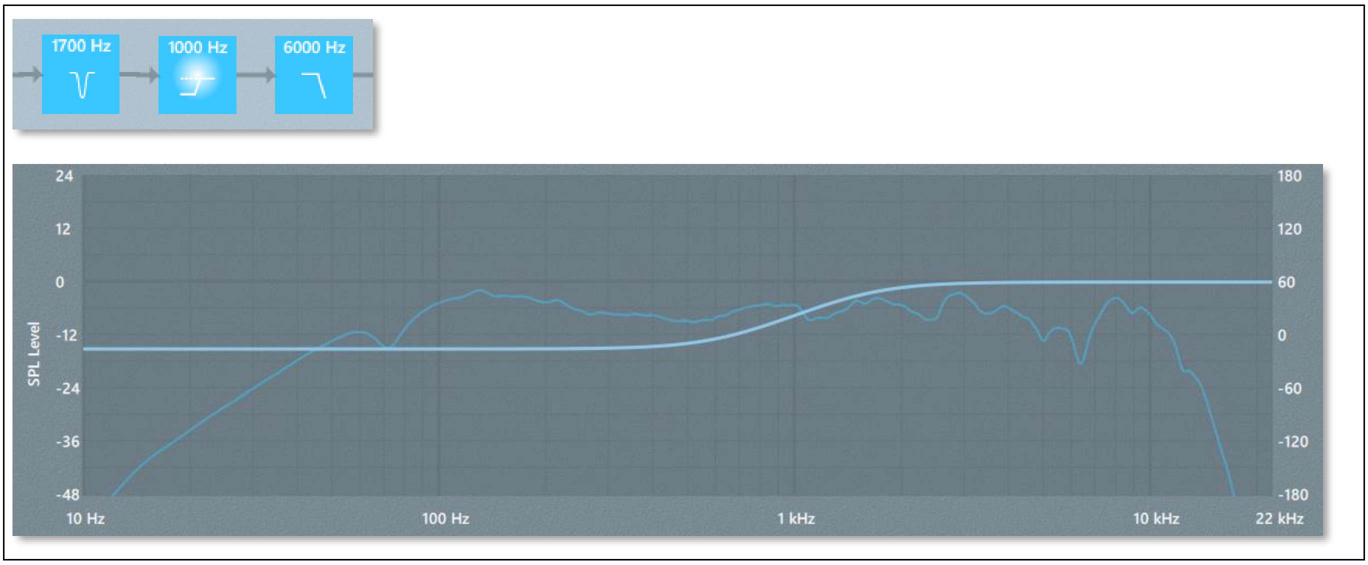

Figure 48 Frequency response graph

A left click on the "Speaker_1" icon reveals the unmodified speaker response (light blue) along with the flattened response (darker blue) for comparison:

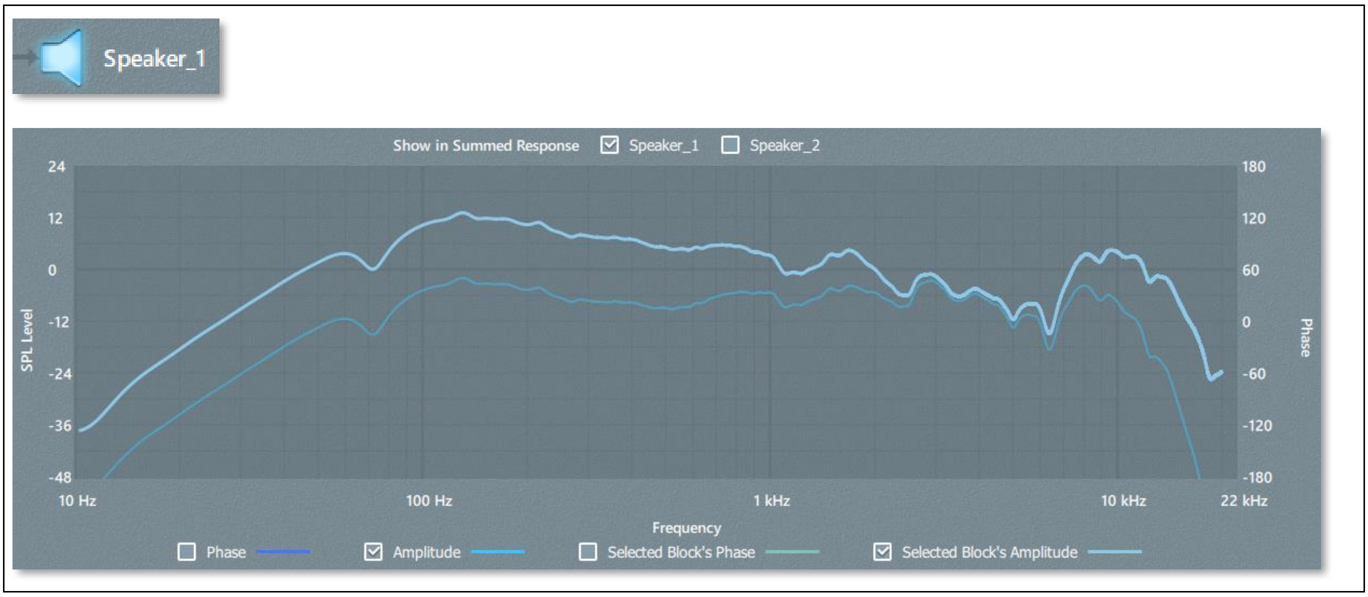

Figure 49 Speaker response comparison

3.2.4 Register browser page

The register browser contains detailed information about the many registers of the MA2304xNS. However, this should only be applied in rare cases where it is necessary to modify an uncommon setting.

If it is necessary to modify some registers in the register browser, it is best to use the search function in the upper left corner to find the registers.

Each row corresponds to a register address and/or group of bits.

Changes on any checkbox ☑ will be effective immediately (i.e., they will result in an I²C transaction) as long as there is connection with the MA2304xNS.

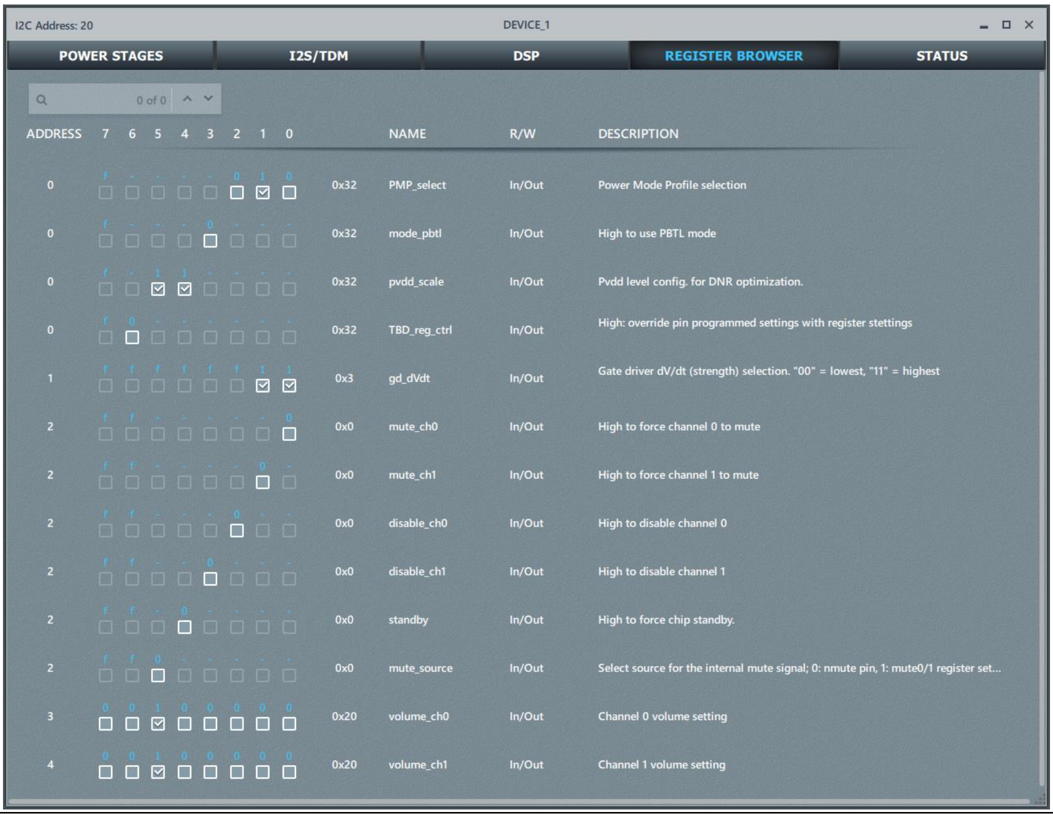

Figure 50 Register browser page

Figure 51 Search field

Type a register name or phrase in the search field and press enter to search for content in the register map. If there is a match, the corresponding register row will be highlighted in blue.

3.2.5 Status page

On the status page, device warnings and errors can be monitored.

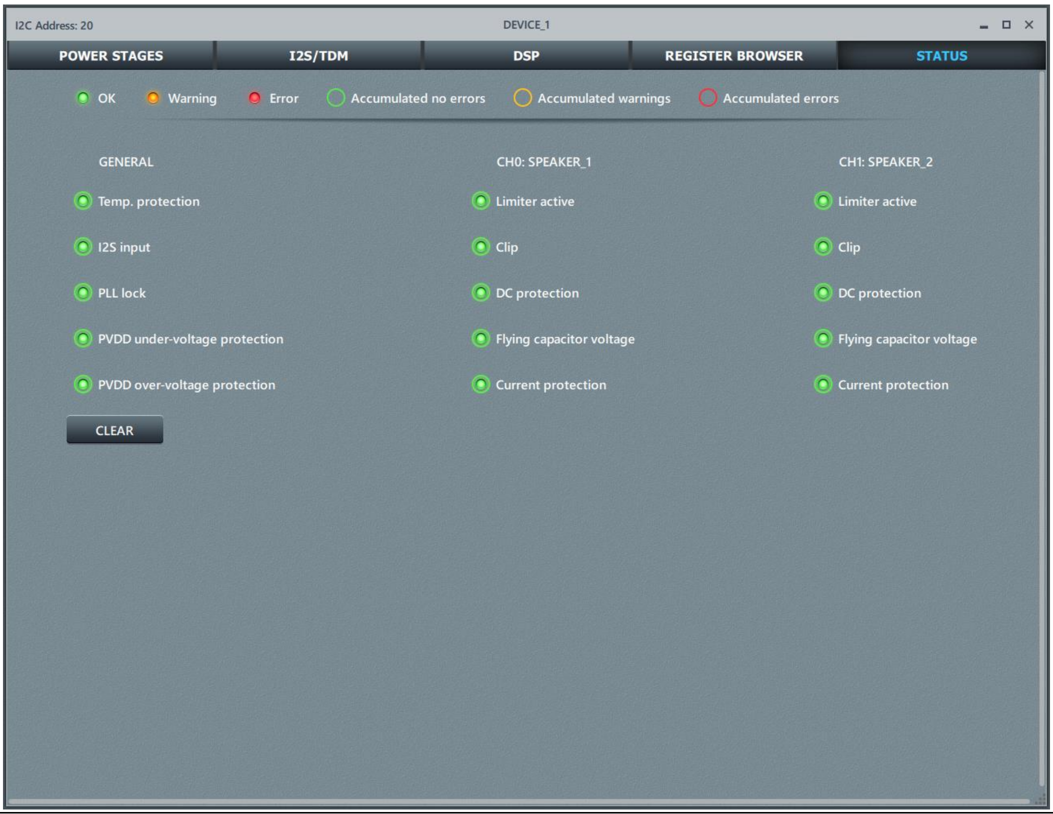

Figure 52 Status page

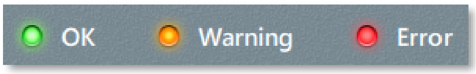

Figure 53 Real-time active indicators

These indicators show active warnings and errors as an orange or red light.

Figure 54 Accumulated (sticky) indicators

These warnings or errors will be shown as a colored circle around the indicator.

Figure 55 Clear button

Press the CLEAR button to clear previous states of the warning or error indicators.

4 DSP flows

This section covers additional details on the MA2304DNS DSP flows. As shown in the screenshot below, the DSP page block diagram is comprised of hardware level blocks, MA2300 blocks and a DSP flow. In addition, the DSP flow itself contains interconnected DSP audio blocks.

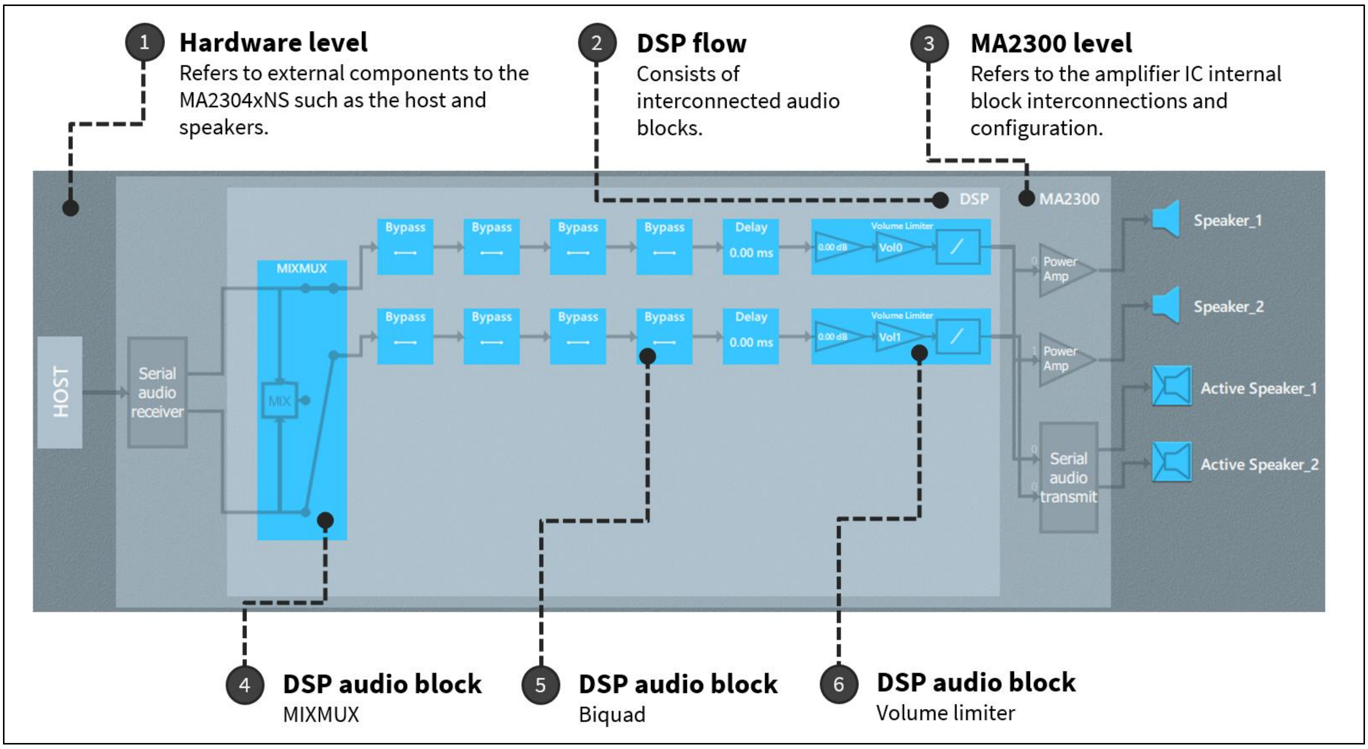

Figure 56 DSP page block diagram

To select a DSP flow, press the library  button in the top-right corner of the screen. A Library window will then appear, as shown in Figure 57 below:

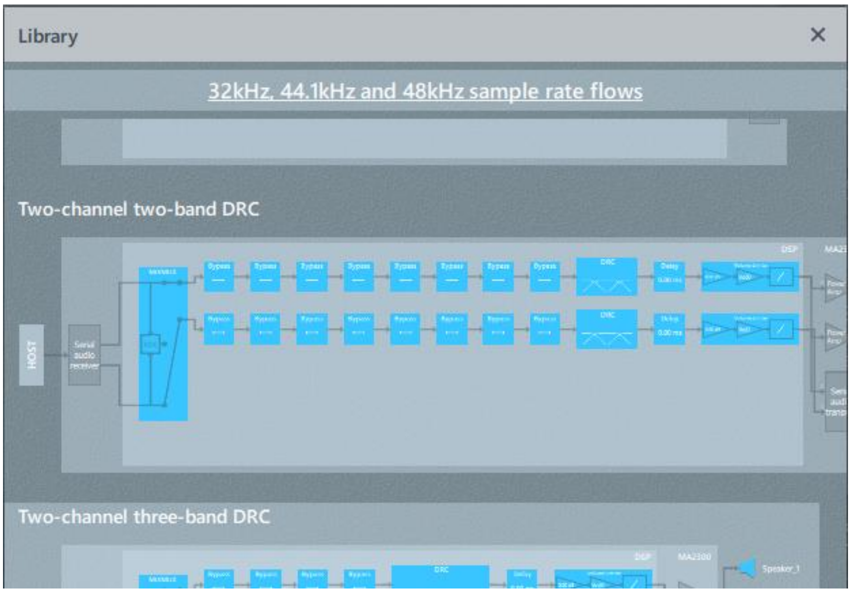

Figure 57 Library window

There are three categories of DSP flows to select from, based on sample rate:

- 32, 44.1, and 48 kHz
- 88.2 and 96 kHz
- 176.4 and 192 kHz

The higher the sample rate category, the higher the bandwidth, but fewer instructions per sample are available for DSP processing.

Detailed information about each audio block is provided in Chapter 5.

4.1 DSP flow application examples

This section goes through various application examples on how to configure, use and tune the blocks of a DSP flow. As mentioned previously, detailed information about each audio block is provided in Chapter 5.

A note on audio tuning

Detailed information on the limiters is available in Section 5.2. There is some tuning needed to keep the limiter effect as transparent as possible, while meeting the speaker power requirements. The peak limiter attack time can be set to 0 ms for a brickwall effect. However, the release time should be set long enough to prevent audible distortion caused by bass frequencies modulating. For smaller speakers, a biquad configured as a high-pass filter is recommended to allow more flexibility with release times.

EQing is highly dependent on the speaker enclosure and configuration. For small, single-driver speakers, EQing is mostly focused on flattening and/or matching a target response, improving bass and voice intelligibility, and solving some acoustic/driver resonance issues. The biquads can also be used to implement crossovers for two- and three-way speaker systems.

4.1.1 5 W mono wireless speaker

This hypothetical product has the following audio requirements:

1. The incoming stereo signal shall be mixed to mono.
2. Continuous output power shall be limited to 5 W average power.
3. Peak power shall not exceed the peak short-term power handling.
4. Power supply voltage level shall be at or above the peak power.

The loudspeaker specifications are as follows:

Table 3 Loudspeaker specifications

Parameter	Parameter	Value	Unit
P_1	Speaker continuous power handling	5	W
P_2	Speaker short-term power handling	20	W
Z_{nom}	Speaker nominal impedance	4	Ω

Requirement 1: stereo mixing

This requirement is straightforward. As explained in Section 5.5, left and right input channels can be mixed using the MIXMUX block. Output 0 can be configured as an input mix of the stereo input, as shown below:

Figure 58 **Output 0 configuration**

This setting scales the left and right input signals by two such that their sum results in unity gain.

Requirements 2, 3, and 4: output power limiting and P_{VDD} voltage

The goal for these requirements is to keep the loudspeaker RMS and peak power under control. Continuously driving 5 W into the speaker is small enough that a heatsink is not required to keep the IC temperature below the temperature warning.

The peak and RMS limiters of the MA2304DNS will then be configured to keep loudspeaker power (and temperature) under control.

The continuous and peak power handling specifications from a speaker manufacturer are typically derived from bandwidth-limited pink noise tests played for a specified duration. For larger speaker systems it is common to adjust the amplifier power supply voltage and/or amplifier gain setting to match the speaker power rating. In smaller speaker systems, intelligibility and loudness become important and, therefore, additional techniques such as limiting and compression may be used to achieve this purpose.

To select a power supply voltage and/or amplifier gain, we first calculate the equivalent RMS and peak voltages expected when the speaker is driven near its rated power levels.

First, let's calculate the equivalent RMS voltage, V_1, that corresponds to the continuous power handling, P_1, as previously provided in Table 3:

$$V_1 = \sqrt{P_1 \times Z_{nom}} \qquad [V_{rms}]$$

In this example, P_1 = 5 W and Z_{nom} is 4 Ω. V_1 is then 4.5 V_{rms}.

Similarly, V_2, the desired peak voltage, is calculated as follows:

$$V_2 = \sqrt{2P_2 \times Z_{nom}} \qquad [V_{peak}]$$

V_2 is then 12.6 V_{pk}. To be able to reproduce this voltage, a power supply greater than this voltage level is recommended.

Looking at the table below, a P_{VDD} of 15 V seems like a good choice.

Table 4 **Amplifier gain**

Parameter	Description	pvdd_scale	Recommended P_{VDD}	Value	Unit
G_{dB}	Amplifier gain in dB (V_{rms}/FS)	00	10 V	16.5	dBV/FS
		01	12 V	19	dBV/FS
		10	15 V	21.2	dBV/FS
		11	18 V	22.5	dBV/FS

To calculate the RMS and peak limiter thresholds, we use the corresponding G_{dB} from the table above:

$$Th_1 = 20\log(V_1) - G_{dB} \qquad [dBFS]$$

$$Th_2 = 20\log(\frac{V_2}{\sqrt{2}}) - G_{dB} \quad [dBFS]$$

This results in an RMS limiter threshold Th_1 = -8.2 dB and a peak limiter threshold Th_2 = -2.2 dB. These values can be entered into the volume limiter block window, as shown below:

Figure 59 Volume limiter block window

Attack and release times (as well as the threshold levels) depend on the audio tuning as well as the speaker characteristics and reliability tests.

A summary of the limiter calculations is shown in the worksheet table below:

Table 5 Limiter worksheet

Parameter	Parameter	Value	Unit
P_1	Continuous power handling	5	W
P_2	Short-term power handling	20	W
Z_{nom}	Nominal impedance	4	Ω
V_1	RMS voltage corresponding to P_1	4.5	V_{rms}
V_2	Peak voltage corresponding to P_2	12.6	V_{peak}
G_{dB}	Amplifier gain[1]	21.2	dBV/FS
Th_1	RMS limiter threshold corresponding to V_1	-8.2	dB_{FS}
Th_2	Peak limiter threshold corresponding to V_2	-2.2	dB_{FS}

[1]The pvdd_scale register must be configured for the amplifier gain setting to take effect

5 DSP audio blocks

This section explains each of the DSP flow audio blocks available in detail.

5.1 Biquad block

Figure 60 Biquad block

The biquad audio block is a generic second-order IIR filter for high-pass, low-pass, parametric EQ (peaking), notch, shelf and custom filters. Parameters such as quality factor, corner frequency and gain can be adjusted. A typical DSP flow has multiple of these biquad blocks interconnected in cascade.

Double-clicking a biquad block opens a biquad configuration window, as shown below:

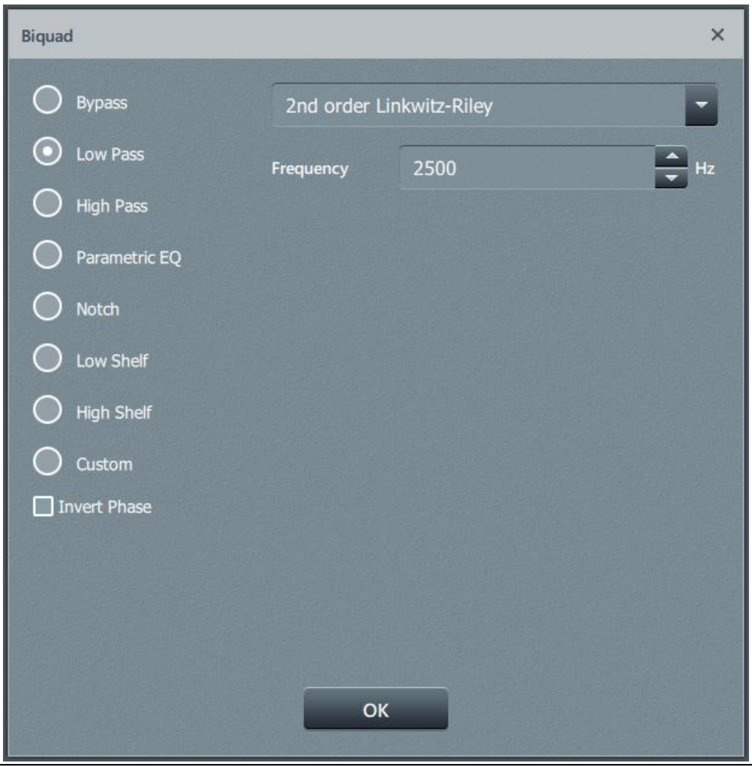

Figure 61 Biquad configuration window

Each biquad filter option has its own parameters to choose from. For example, a low-pass filter has the option to select the filter subtype and its cut-off frequency, while a notch filter has the option to select its frequency and Q-value.

For all biquad filter options, except bypass and custom, there is the option to invert the phase for that particular biquad using the “invert phase” control:

Figure 62 Invert phase control

This can be useful when designing second-order crossover filters, where there is a 180-degree phase shift between the low-pass and high-pass filter outputs.

Figure 63 Bypass

When configured as bypass, the input signal flows to the output unaffected.

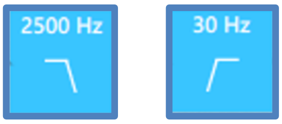

Figure 64 Low-pass and high-pass filters

The low-pass and high-pass filters have two controls: a filter subtype dropdown menu and a cut-off frequency control:

Figure 65 2nd order Butterworth controls

The image below shows the options that are available in the filter subtype dropdown menu:

Figure 66 Filter subtype dropdown menu

Each filter subtype represents one of many analog prototype filters. The single-pole filter option has a 20 dB/decade roll-off while the second-order filter options have a 40 dB/decade roll-off, and so on.

Using a single-pole high-pass filter (e.g., less than 50 Hz) is recommended to avoid any offset artifacts.

A fourth-order filter requires the user to configure two consecutive biquad blocks. For example, if a fourth-order Linkwitz-Riley filter is desired, the first biquad should be configured as "4th order Linkwitz-Riley (1st biquad)" and the second biquad should be configured as "4th order Linkwitz-Riley (2nd biquad)".

The second-order custom filter provides the option to manually enter the filter Q-value. For example, a Q = 0.5 results in a critically damped filter while a Q = 0.707 (e.g., Butterworth) results in a maximally flat (monotonic) filter.

Table 6 Parametric EQ filter controls

Control name	Description	Min.	Default	Max.	Unit
Filter subtype	Analog filter prototypes				
Frequency	Low-pass/high-pass cut-off (f_c) frequency	10	1000	$F_s/2$	Hz
Q-value[1]	Filter quality factor. Affects frequency response and damping in time domain.	0.5	1	10.0	

[1] Q-value only available in "2nd order custom" subtype.

Below is an example measurement of a second-order Linkwitz-Riley filter crossover:

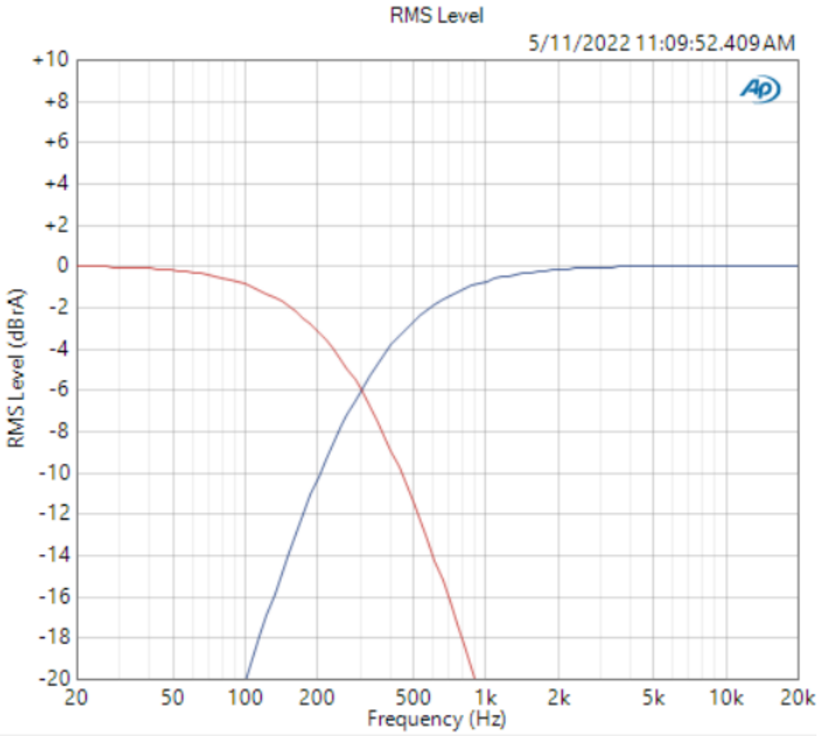

Figure 67 Second-order Linkwitz-Riley filters, f_c = 300 Hz, left channel blue, right channel red

Figure 68 Parametric EQ (peaking) filter

The parametric EQ filter has three controls: “frequency”, “gain”, and “Q-value”:

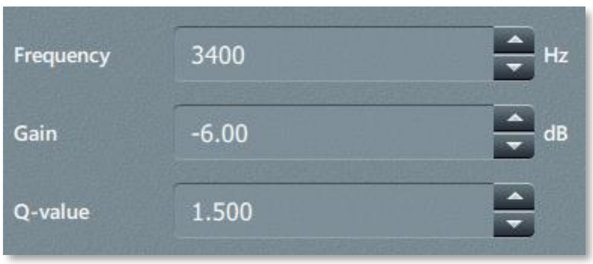

Figure 69 Parametric EQ controls

Table 7 Parametric EQ filter controls

Control name	Description	Min.	Default	Max.	Unit
Frequency	The center frequency (f_0) where the amplitude peak is observed	10	1000	$F_s/2$	Hz
Gain	The peak amplitude	-48	1	24	dB
Q-value	The filter quality factor	0.010	1	10.0	

An example family of EQ curves with varying gain settings, f_0 = 3 kHz and Q = 2, is shown below:

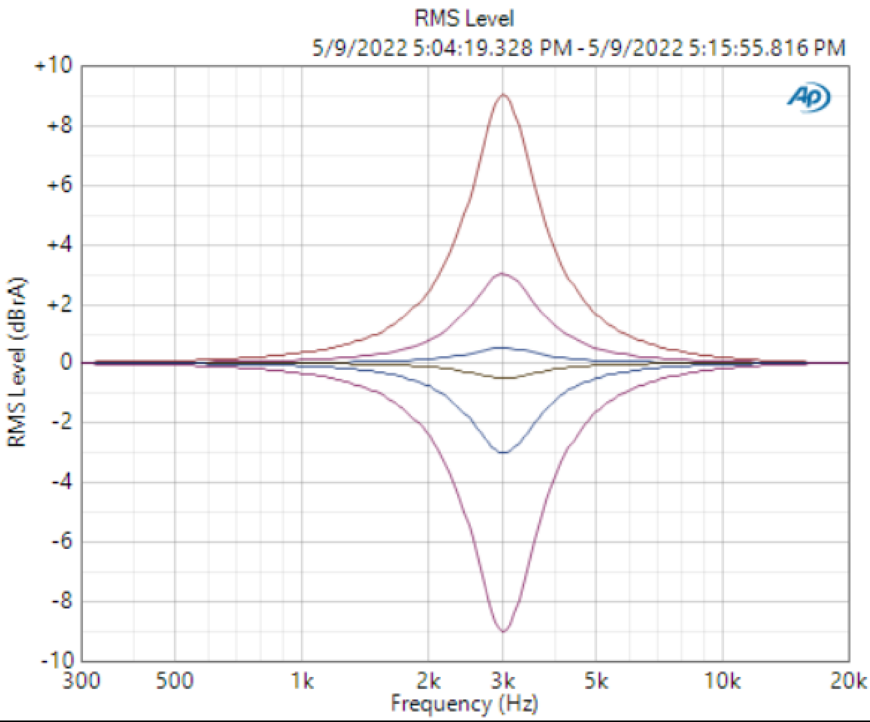

Figure 70 EQ filters with -9, -3, -0.5, 0.5, 3, and 9 dB gain, f_0 = 3 kHz and Q = 2

Figure 71 Low shelf and high shelf filters

A shelf filter boosts or cuts the magnitude below or above a frequency band. In the case of a low shelf, the lower frequency band is boosted/cut while, in the case of a high shelf, the higher frequency band is boosted/cut.

The “shelf filters” have two controls: “frequency” and “gain”:

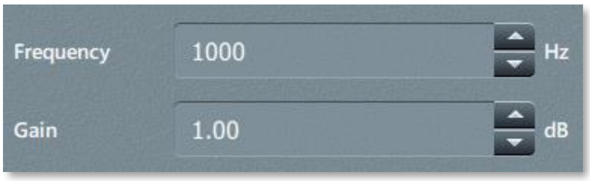

Figure 72 Low shelf and high shelf filter controls

Table 8 Shelf filter controls

Control name	Description	Min.	Default	Max.	Unit
Frequency	The frequency (f_0) in which the amplitude is half the gain difference between the boosted and unboosted region (in dB)	10	1000	$F_s/2$	Hz
Gain	The peak amplitude of the filter	-48	1	24	dB

An example family of shelf filter curves with varying gain settings, f_L = 300 Hz, f_H = 10 kHz, is shown below:

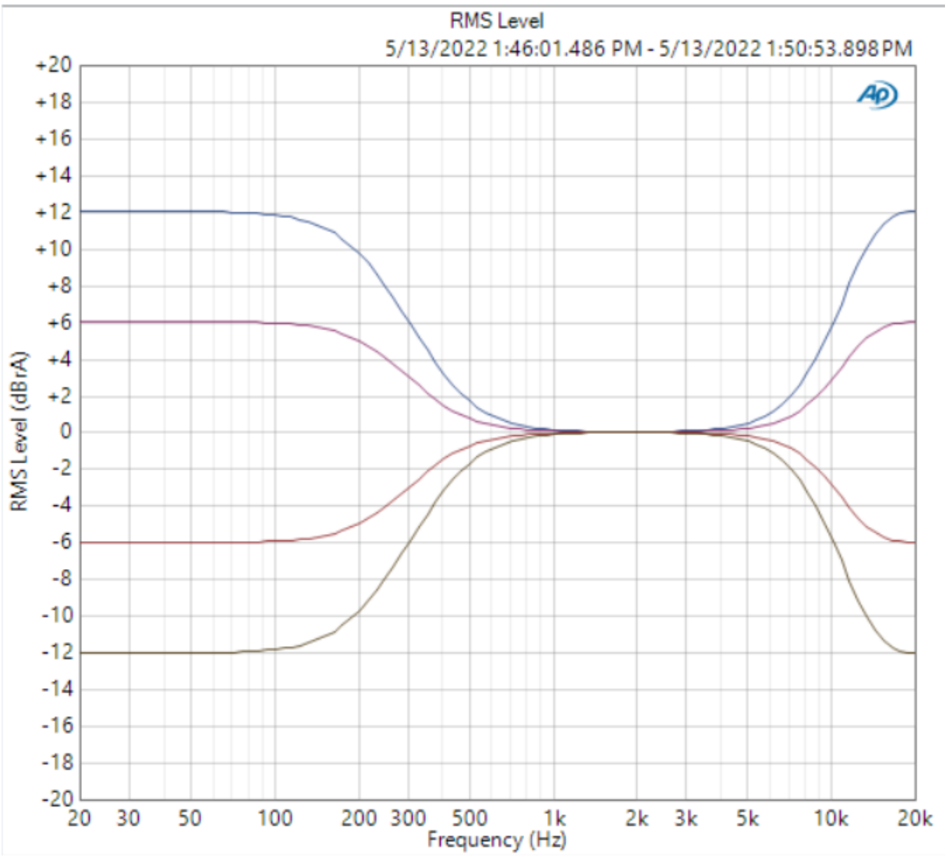

Figure 73 Shelf filters with -12, -6, +6, and +12 dB gain, f_L = 300 Hz, and f_H = 10 kHz

Figure 74 Notch filter

The “notch filter” is a band-rejection filter with a narrow stop-band (typically) and high attenuation. This filter is typically used to remove unwanted/problematic noise (e.g., humming).

The notch filter has two controls: “Frequency” and “Q-value”, as shown below:

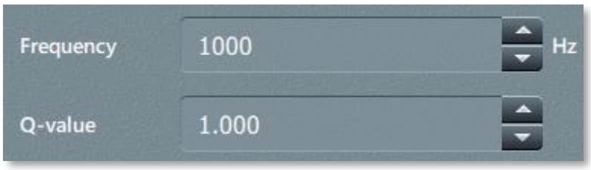

Figure 75 Notch filter controls

Table 9 Notch filter controls

Control name	Description	Min.	Default	Max.	Unit
Frequency	The notch center frequency (f_0)	10	1000	$F_s/2$	Hz
Q-value	The filter quality factor	0.1	1	10.0	

An example family of notch filter curves with f_0 = 1 kHz and Q = 1, 2 and 3 is shown below:

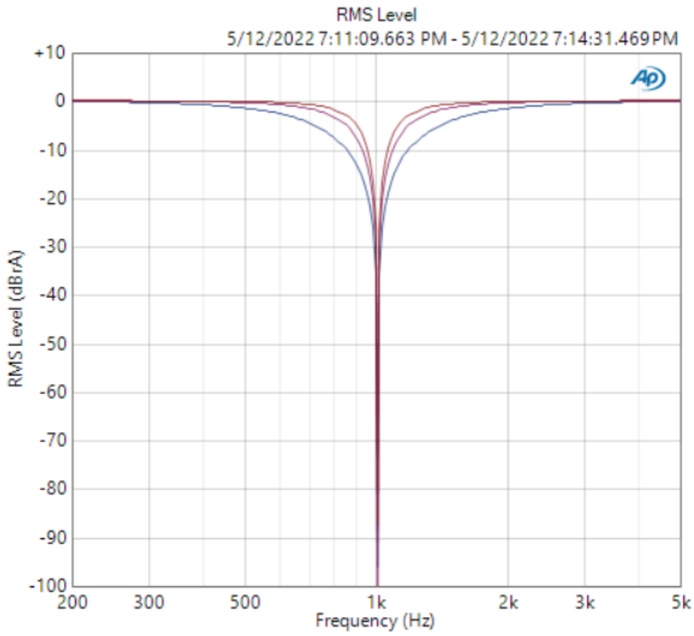

Figure 76 Notch filters with f_0 = 3 kHz and Q = 1, 2, and 3

As shown in this example, a higher Q (e.g., 3) results in a narrower stop-band while a lower Q (e.g., 1) results in a wider stop-band.

Figure 77 Custom filter

The “custom filter” allows the user to input biquad coefficients manually, giving flexibility to implement any biquad filter prototype in this block.

The custom filter has six input fields, with three corresponding to the biquad numerator and three corresponding to the biquad denominator:

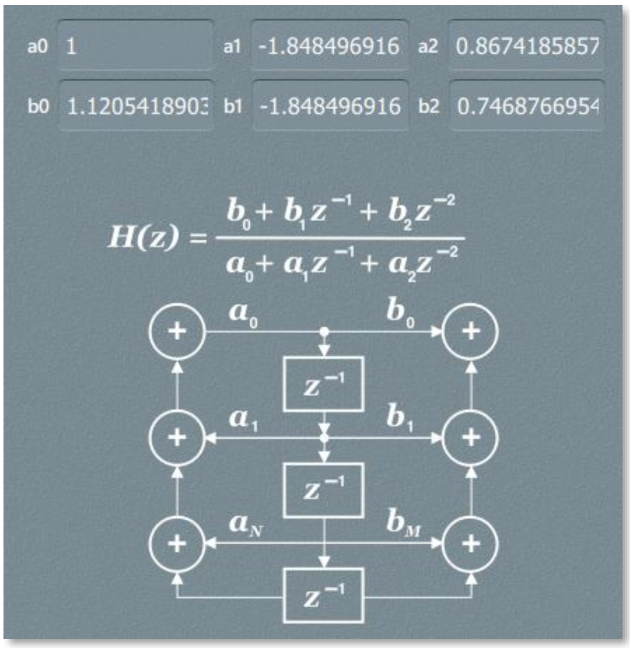

Figure 78 Custom filter controls

As shown in the previous image, the biquad transfer function is defined as:

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{a_0 + a_1 z^{-1} + a_2 z^{-2}}$$

Each coefficient and its range is defined in Table 10, below:

Table 10 Custom filter controls

Control name	Description	Min.	Default	Max.
a_0	Denominator coefficient. Typically set to 1 while other coefficients normalized by a_0.	0	0	10
a_1	Denominator coefficient	0	0	10
a_2	Denominator coefficient	0	0	10
b_0	Numerator coefficient	0	0	10
b_1	Numerator coefficient	0	0	10

Control name	Description	Min.	Default	Max.
b_2	Numerator coefficient	0	0	10

An example family of custom EQ curves with 9 dB gain, f_0 = 3 kHz and Q = 1, 2 and 3 is shown below:

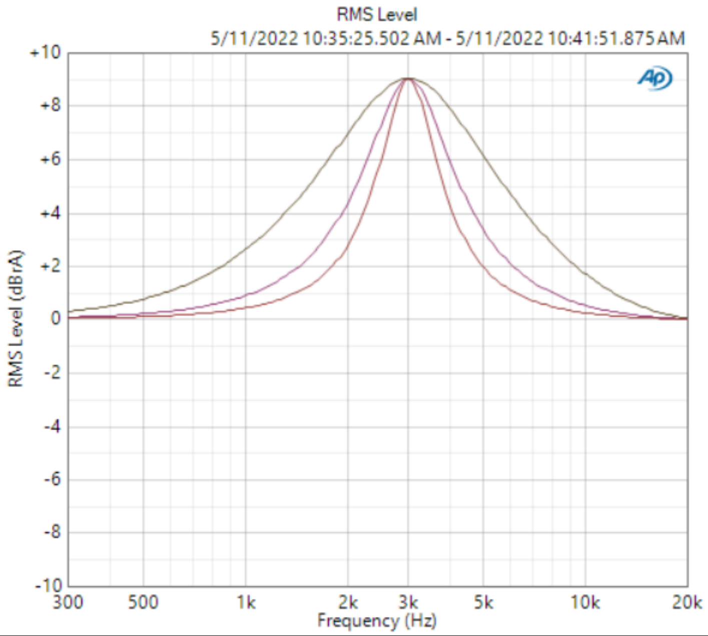

Figure 79 Custom EQ filters with 9 dB gain, f_0 = 3 kHz and Q = 1, 2, and 3

In this example, a higher Q results in a narrower peak and vice versa.

5.2 Volume control/limiter block

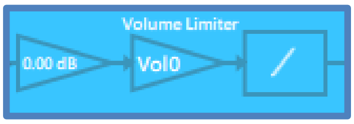

Figure 80 Volume control/limiter block

The "volume limiter" block is a peak and/or RMS-based limiter, typically at the end of a DSP flow, that helps keep signal levels, on average, below a certain level. A limiter can be extremely useful in audio applications; from driving loudspeakers within its rated power, keeping the amplifier IC temperature controlled to making music and voice sound louder and more intelligible.

5.2.1 Limiter configuration

Double-clicking the “volume limiter” block opens its configuration window, as shown in Figure 81. This window shows the peak and RMS limiter options as well a pre-gain and volume source controls. The input/output and time domain plots are shown for each limiter as well. These are described in this section in detail.

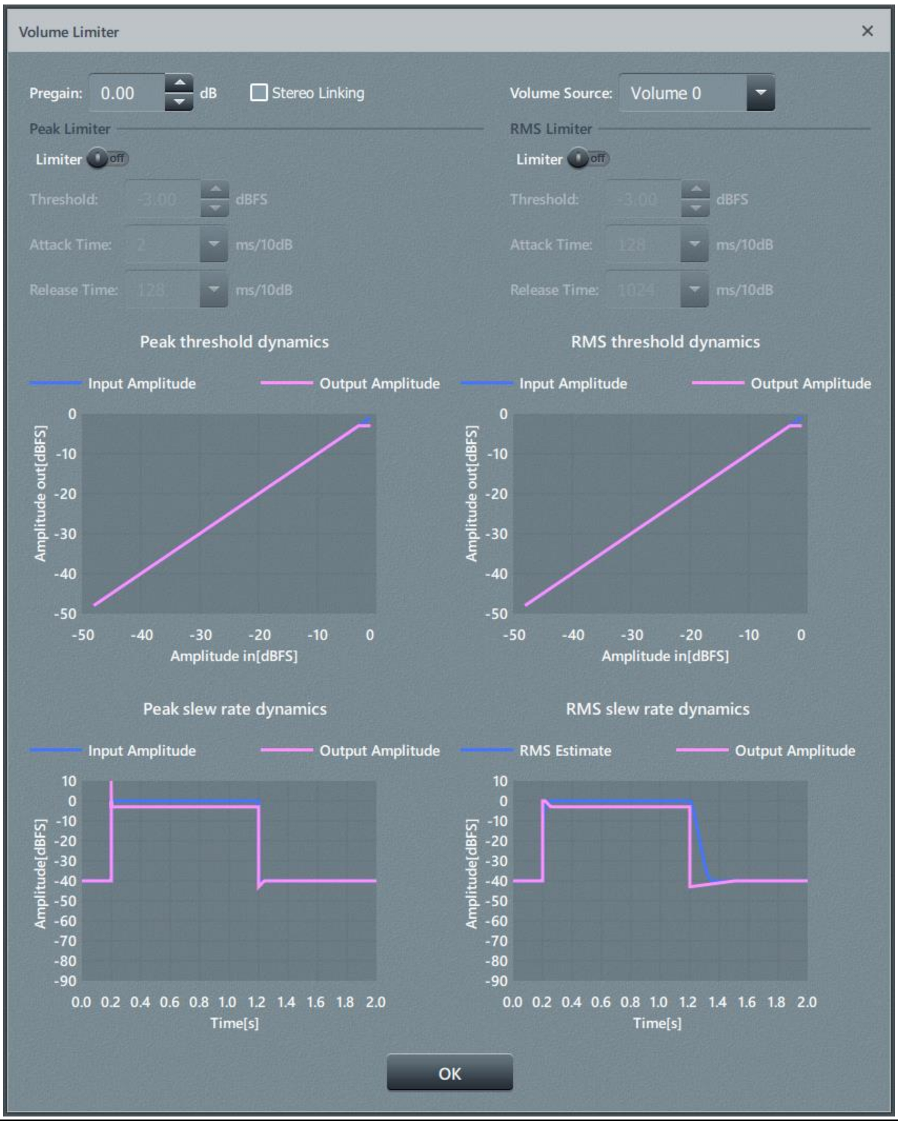

Figure 81 Volume limiter configuration window

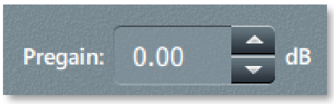

Figure 82 Pre-gain control

The pre-gain control is a global gain of the limiter block applied at the beginning of both the peak and RMS limiters. In terms of the limiter input/output curve, it shifts the curve horizontally, as shown in Figure 85.

Table 11 Pre-gain control

Control name	Description	Min.	Default	Max.	Unit
Pregain	Gain applied at the input of the limiter block	-48	0	24	dB

Figure 83 Stereo linking control

The stereo linking control is typically used to keep the stereo image intact while the left and right limiters are compressing. Setting this control to "on" compares the channel volume attenuation to the opposite channel attenuation and limits according to the worst-case.

Setting stereo linking to "on" on both left and right channels will configure both limiters to compress by the same amount.

Table 12 Stereo linking control

Control name	Description	Min.	Default	Max.
Stereo linking[1]	Compares the channel volume attenuation to the opposite channel attenuation and limits according to the worst case	Off	Off	On

[1]Needs to be applied to both left and right channels if true stereo linking is desired

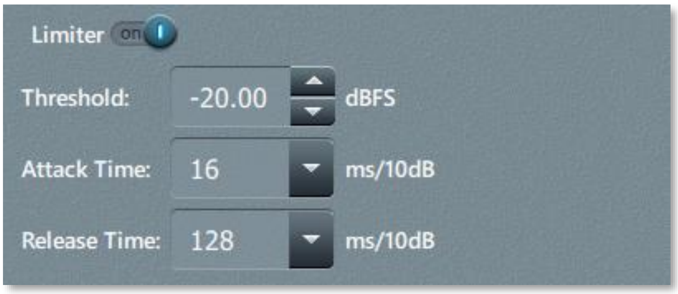

Figure 84 Peak and RMS limiter controls

Each limiter has four controls: limiter on/off, threshold, attack time and release time. When the signal input level is above the threshold, the limiter begins to decrease or increase gain, depending on the direction of the signal level that it detected. This input/output relationship is shown in Figure 85.

This change in gain does not happen instantaneously; the gain level keeps adjusting itself until the output estimate reaches its corresponding target level.

Limiter function details are explained in the next section.

Table 13 Peak and RMS limiter controls

Control name	Description	Min.	Default	Max.	Unit
Limiter	Enables or disables the limiter	Off	Off	On	
Threshold	The input level in which limiting begins	-48[1]	-3	24	dB_{FS}
Attack time	The time period it takes for the limiter to attenuate by 10 dB				
Peak		0	2	128	ms/10 dB
RMS		16	128	2048	ms/10 dB
Release time	The time period it takes for the limiter to increase gain by 10 dB				
Peak		16	128	2048	ms/10 dB
RMS		128	1024	16384	ms/10 dB

[1]The limiters have a maximum attenuation of 48dB each.

5.2.2 Limiter details

A limiter functions based on a static input-to-output relationship and time domain parameters.

The input-to-output relationship is typically represented by an input/output curve, in decibels. Any static gain setting applied to the input signal will shift the entire curve horizontally. Any gain applied to the output signal would shift the entire curve vertically (sometimes referred to as makeup gain). In case of a digital input speaker amplifier, like the MA2304xNS, the output limit is a fraction of the full-scale level (0 dB_{FS}) and the full-scale voltage. Remember that any adjustment to the full-scale voltage (e.g., pvdd_scale setting) will change the voltage limit at the output. This is especially important when limiting the output based on loudspeaker power handling specifications and/or IC temperature.

Figure 85 shows an input/output plot showing the effects of the pre-gain and threshold parameters.

Let’s assume that pre-gain = 10 dB and the threshold = -20 dB as shown in the plot. With these parameter values, an input level of -20 dB will now, effectively, result in a -10 dB input level from the limiter’s perspective, which makes the limiter kick in sooner (more aggressive).

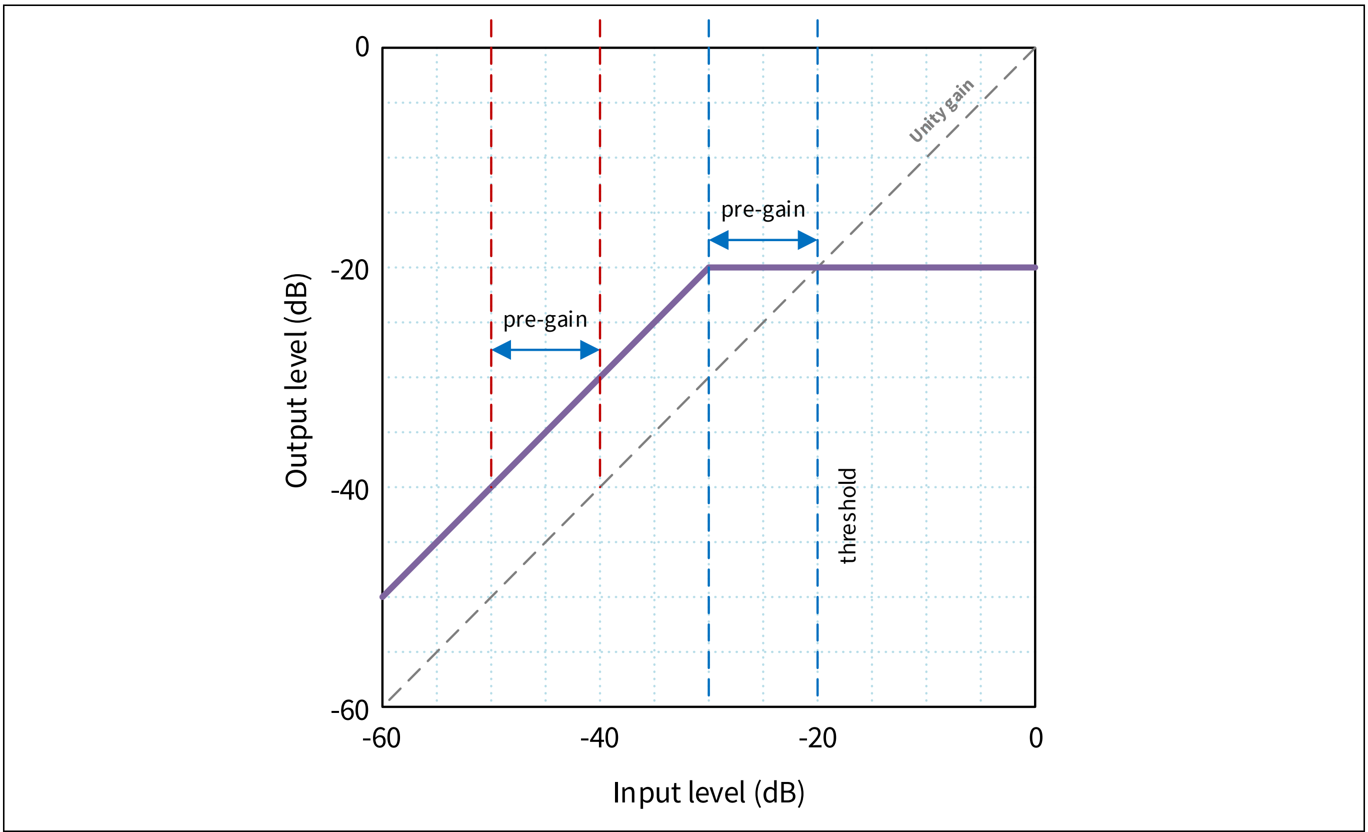

Figure 85 **Limiter input/output curve with threshold = -20 dB and a pre-gain = 10 dB**

The plot below shows an example of the effect the time domain parameters have on the limiter output waveform. In this example, a large input signal above the programmed limiter threshold is detected and is then attenuated to reach its corresponding limit over a period of time (attack phase). As the input signal changes to a smaller waveform below the threshold it is then amplified over a period of time (release phase) back to its original gain level (unity gain).

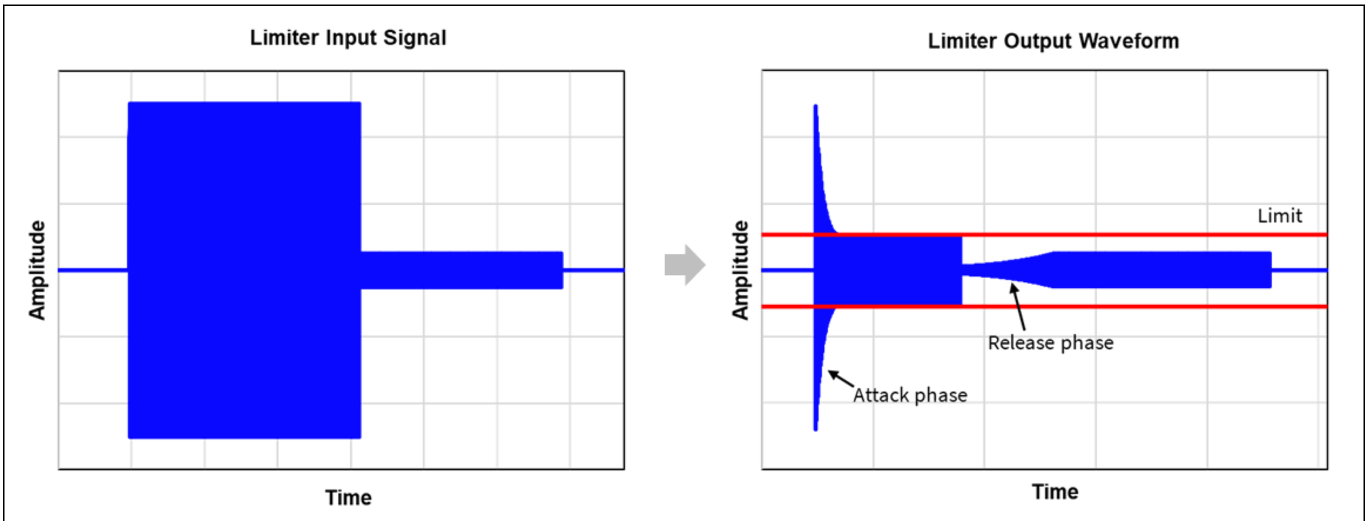

Figure 86 **Limiter waveform example (time domain)**

The limiter block can be configured as a "peak limiter" and as an "RMS limiter".

In the case of a peak limiter, its level detector follows the peak level of the input signal instead of the RMS or average level. This provides the flexibility to employ a "brickwall limiter" type of compression where the output is kept below a certain limit, as shown below:

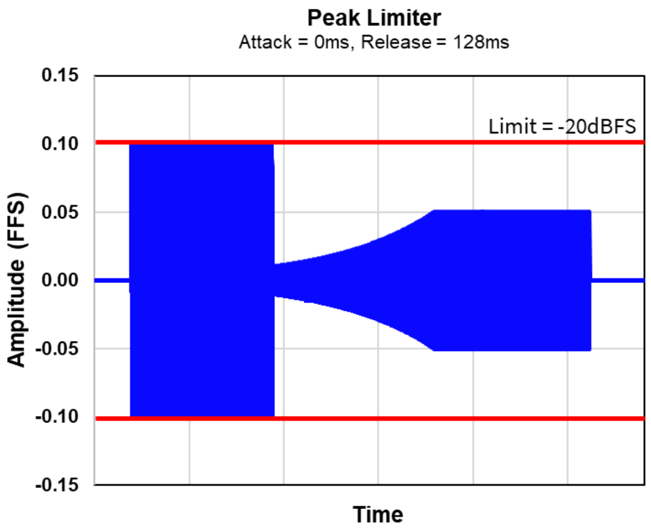

Figure 87 **Peak limiter configured as a brickwall limiter (attack = 0 ms)**

Increasing the peak limiter attack time higher than 0 ms allows the initial transient to go through the limit. This is shown in the example plot below:

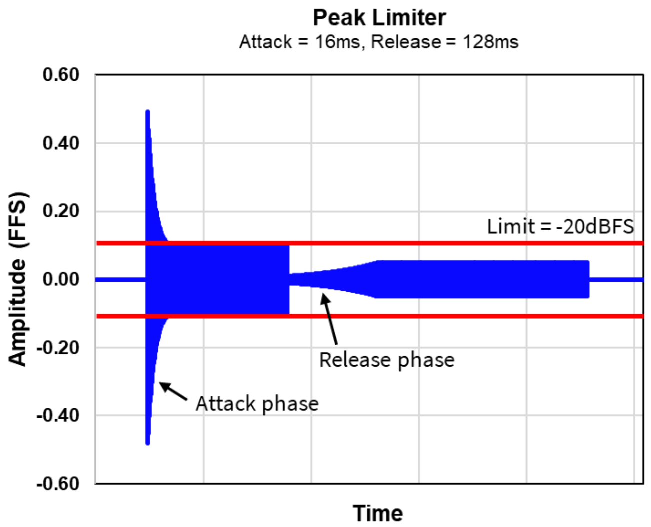

Figure 88 **Peak limiter with attack time greater than 0 ms**

Some applications benefit from allowing transients above the limiter threshold. Loudspeakers, for instance, heat up when power is applied to them, but not instantaneously. It takes some time for the speaker voice coil and magnet to heat up. This allows for short-duration peaks without heating the loudspeaker significantly. Typical music content has a large crest factor (peak-to-RMS ratio), which can be used as an advantage when using limiters by preserving the music dynamic range and keeping long-term signals (e.g., sine tones) controlled.

Lastly, in the case of an RMS limiter, its level detector follows the RMS level of the input signal. Measuring proper RMS levels require several periods of sampling which, by design, is not fast enough to stop peaks above the limit. However, an RMS limiter may sound more natural than a peak detector.

When observed in time domain, the RMS limiter will limit sinewaves 3 dB above the programmed threshold, as shown in the image below. This is because a sinewave's peak is 3 dB above its RMS level. However, a square-wave peak will be limited to the programmed threshold level since its peak is the same as its RMS.

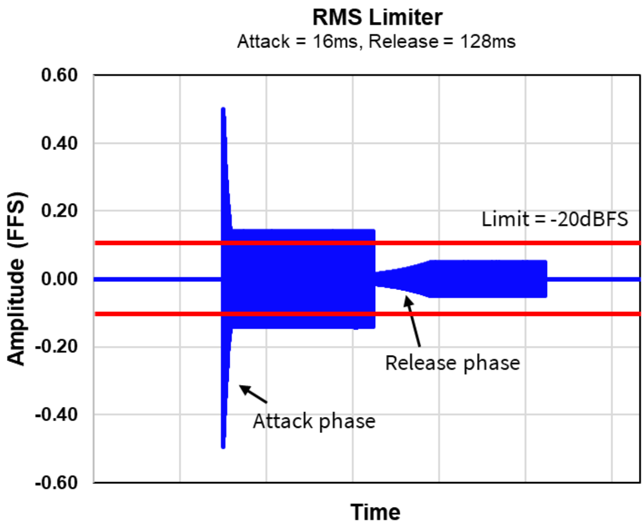

Figure 89 RMS limiter waveform

In addition to using the peak and RMS limiters individually, these can be combined by setting both enable controls to "on", as shown below:

Figure 90 Peak and RMS limiters enabled

For the following example, let's assume that the input waveform is a full-scale 0 dB_{FS} sinewave, pre-gain = 0 dB. With both limiters combined, the peak limiter can be configured as a brickwall limiter and, at the same time, have the RMS limiter settings more relaxed. This is shown in Figure 91. This can be especially useful to prevent harsh clipping when applying high levels of pre-gain to make the music louder while, at the same time, allowing peaks above the RMS limit.

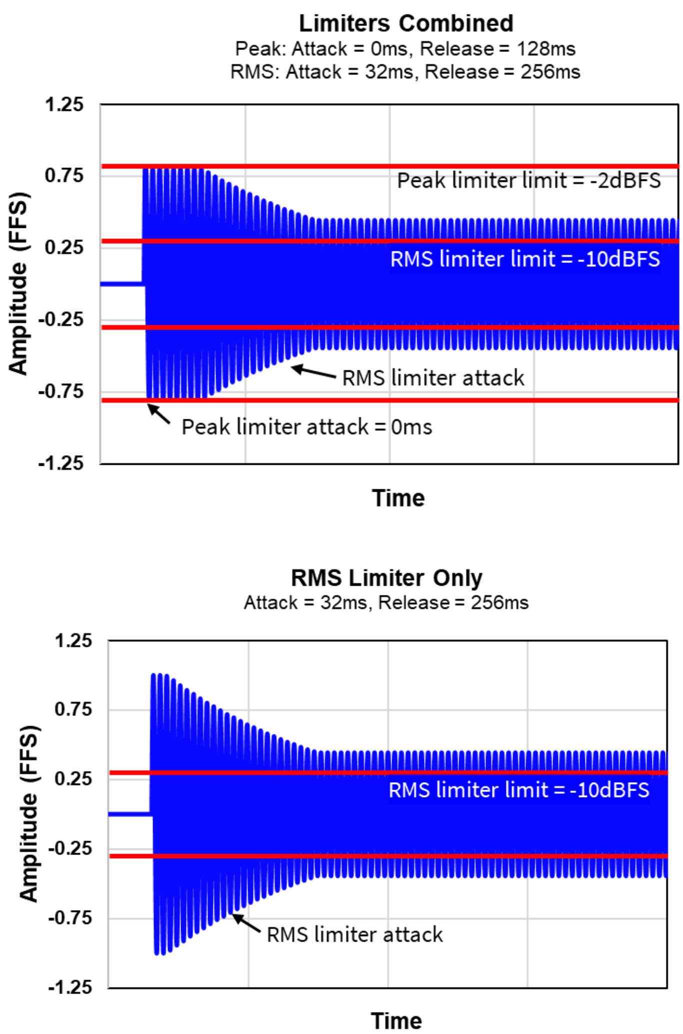

Figure 91 **Peak and RMS limiters combined**

5.3 Multi-band DRC blocks

Figure 92 **Multi-band DRC blocks**

The MERUS™ audio amplifier configurator features two- and three-band dynamic range compressors (DRCs). These can help music sound louder and more controlled, especially for smaller speaker systems and/or loud environments. The multi-band DRCs can be used in conjunction with the output limiters to shape the sound of the speaker system. The output limiters are most useful to keep IC temperature and speaker power in check. Once these safety mechanisms are in place, the DRCs can then be tuned in any way without the worry of exceeding these limits.

5.3.1 Two-band DRC configuration

Double-clicking the two-band DRC block opens its configuration window, as shown in Figure 93. This window has several controls to adjust the multi-band crossover frequency and DRC options for the low-pass and high-pass frequency bands. A frequency response and input/output plots are shown for each DRC band as well.

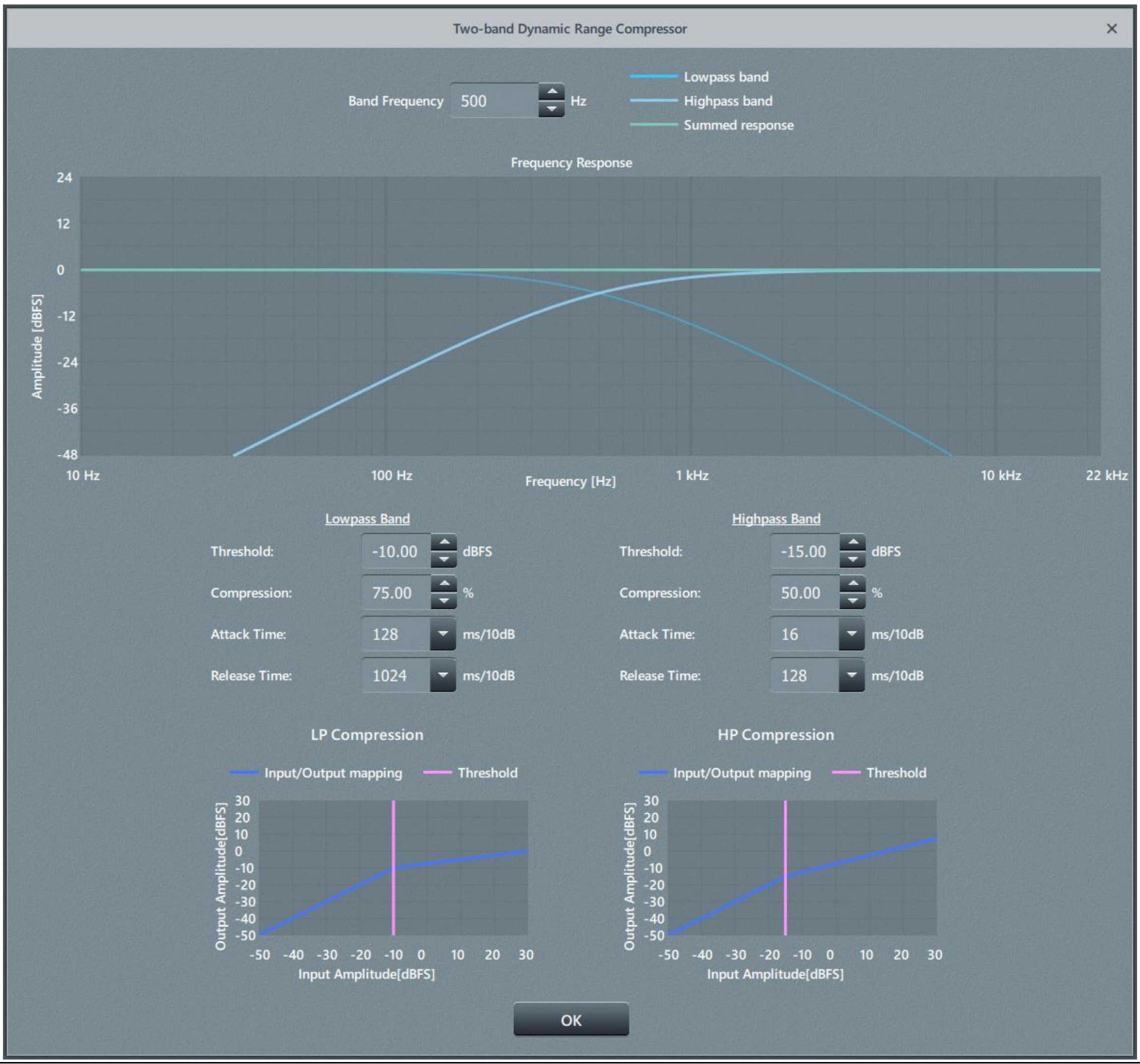

Figure 93 Two-band DRC configuration window

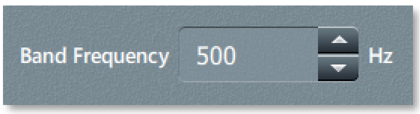

Figure 94 Band frequency control

The band frequency control adjusts the crossover frequency of the DRC bands. Second-order Linkwitz-Riley filters are used to maintain a flat frequency response when both bands are added back together.

Table 14 Band frequency control

Control name	Description	Min.	Default	Max.	Unit
Band frequency	Crossover frequency between the low and high bands	10	500	$F_s/2$	Hz

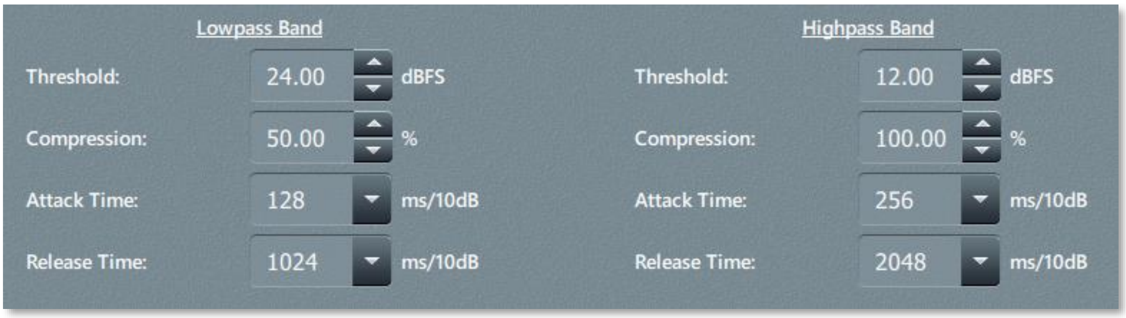

Figure 95 Two-band DRC controls

The two-band DRC has four controls per band. There is the option to adjust the threshold where compression begins, and the compression ratio (as a percentage) above the threshold, as well as attack and release times. The attack time is the time it takes the compressor to decrease gain by 10 dB. Similarly, the release time is the time it takes the compressor to increase gain by 10 dB.

Table 15 Two-band DRC controls

Control name	Description	Min.	Default	Max.	Unit
Threshold	The input level in which compression begins				
Low-pass band		-48	24	24	dB_{FS}
High-pass band		-48	12	24	dB_{FS}
Compression	The compression ratio above the threshold knee: 0 percent = 1:1, 75 percent = 4:1 …				
Low-pass band		0	50	100	%
High-pass band		0	100	100	%
Attack time	The time period it takes the compressor band to attenuate by 10 dB				
Low-pass band		0	128	2048	ms/10 dB
High-pass band		0	256	2048	ms/10 dB
Release time	The time period it takes the compressor band to increase gain by 10 dB				
Low-pass band		8	1024	16384	ms/10 dB
High-pass band		8	2048	16384	ms/10 dB

5.3.2 Three-band DRC configuration

Double-clicking the three-band DRC block opens its configuration window, as shown in Figure 96. This window has several controls to adjust the multi-band crossover frequencies and DRC options for the low-pass, mid-pass and high-pass frequency bands. A frequency response and input/output plots are shown for each DRC band as well.

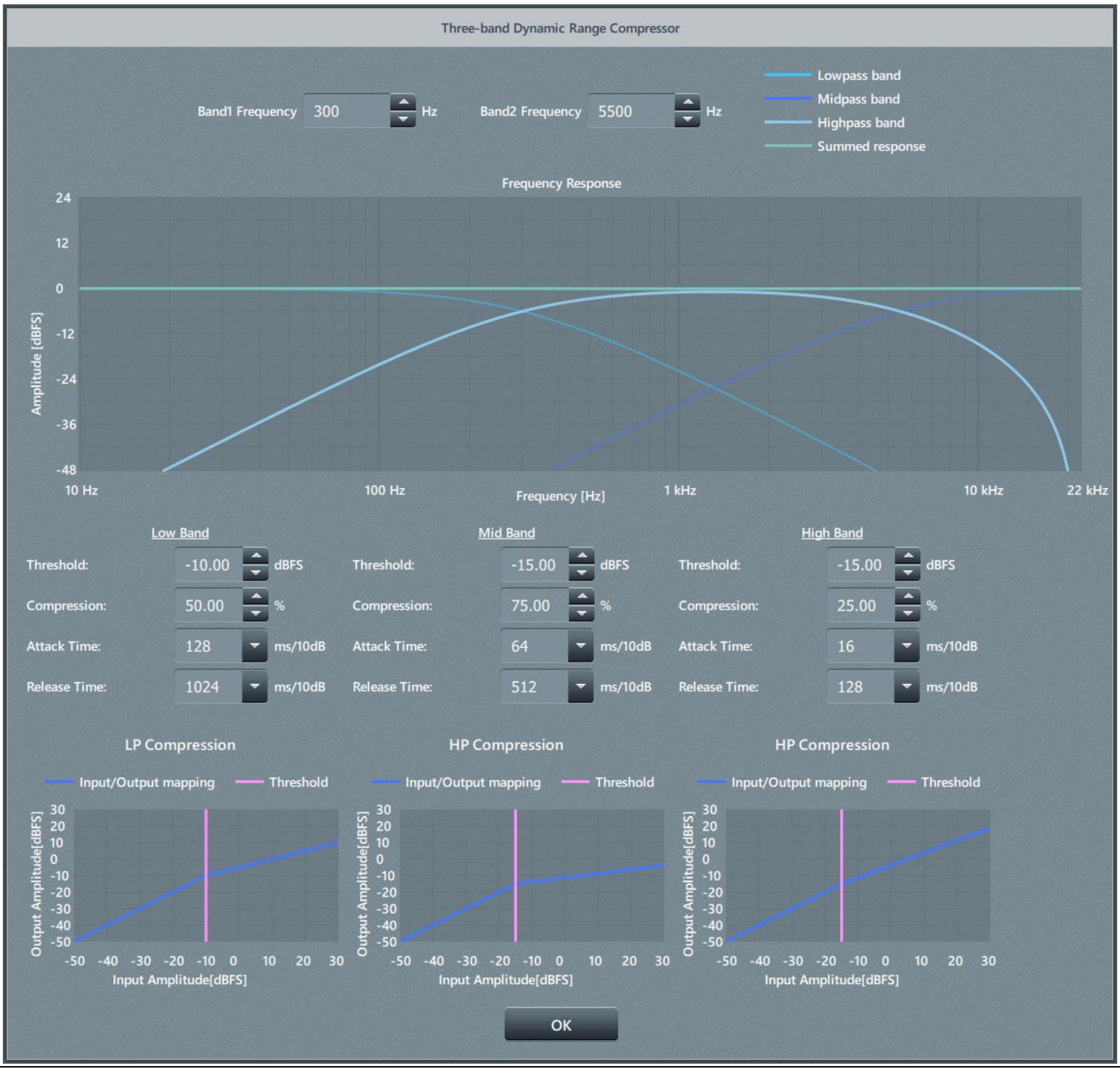

Figure 96 Three-band DRC configuration window

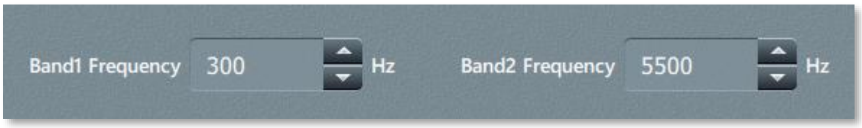

Figure 97 Band1 and band2 frequency controls

The Band1 and Band2 frequency controls adjust the crossover frequency between each of the three DRC bands. Second-order Linkwitz-Riley filters are used to maintain a flat frequency response when the three bands are added back together.

Table 16 Band1 and Band2 frequency controls

Control name	Description	Min.	Default	Max.	Unit
Band1 frequency	Crossover frequency between low band and mid band	10	500	$F_s/2$	Hz
Band2 frequency	Crossover frequency between mid band and high band	10	500	$F_s/2$	Hz

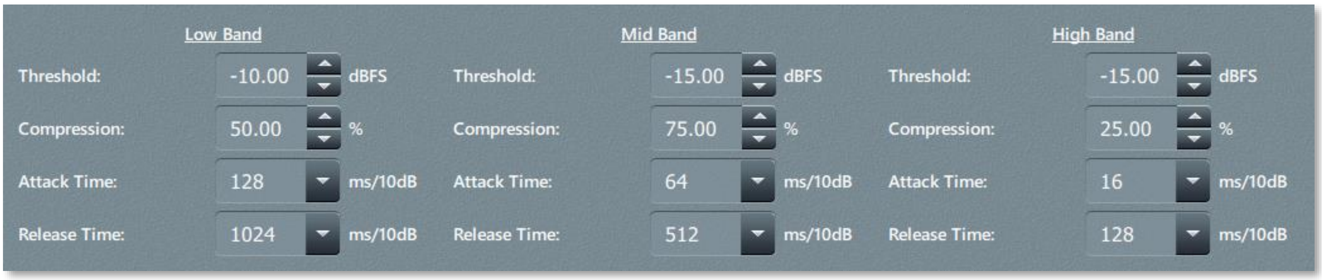

Figure 98 Three-band DRC controls

The three-band DRC has four controls per band. There is the option to adjust the threshold where compression begins, the compression ratio (as a percentage) above the threshold as well as attack and release times. The attack time is the time it takes the compressor to decrease gain by 10 dB. Similarly, the release time is the time it takes the compressor to increase gain by 10 dB.

Table 17 Three-band DRC controls

Control name	Description	Min.	Default	Max.	Unit
Threshold	The input level in which compression begins				
Low band		-48	0	24	dB$_{FS}$
Mid band		-48	0	24	dB$_{FS}$
High band		-48	0	24	dB$_{FS}$
Compression	The compression ratio above the threshold knee: 0 percent = 1:1, 75 percent = 4:1 …				
Low band		0	50	100	%
Mid band		0	50	100	%
High band		0	50	100	%
Attack time	The time period it takes the compressor band to attenuate by 10 dB				
Low band		0	0	2048	ms/10 dB
Mid band		0	0	2048	ms/10 dB
High band		0	0	2048	ms/10 dB
Release time	The time period it takes the compressor band to increase gain by 10 dB				
Low band		8	16384	16384	ms/10 dB
Mid band		8	16384	16384	ms/10 dB
High band		8	16384	16384	ms/10 dB

5.3.3 Multi-band DRC details

Similar to a limiter, a DRC functions based on a static input to output relationship and time domain parameters. In the case of a multi-band DRC, each DRC operates on its own frequency band, giving the flexibility to compress a frequency range independently of the others.

The input-to-output relationship of a DRC differs to a limiter in that a compression ratio between 1:1 and inf:1 is possible. This enables subtler compression as the signal level increases.

The graph in Figure 99 is an input/output plot showing several compression ratios. As an example, a 50 percent compression (2:1 ratio) shows a 10 dB output level increase per 20 dB of input signal, above the threshold.

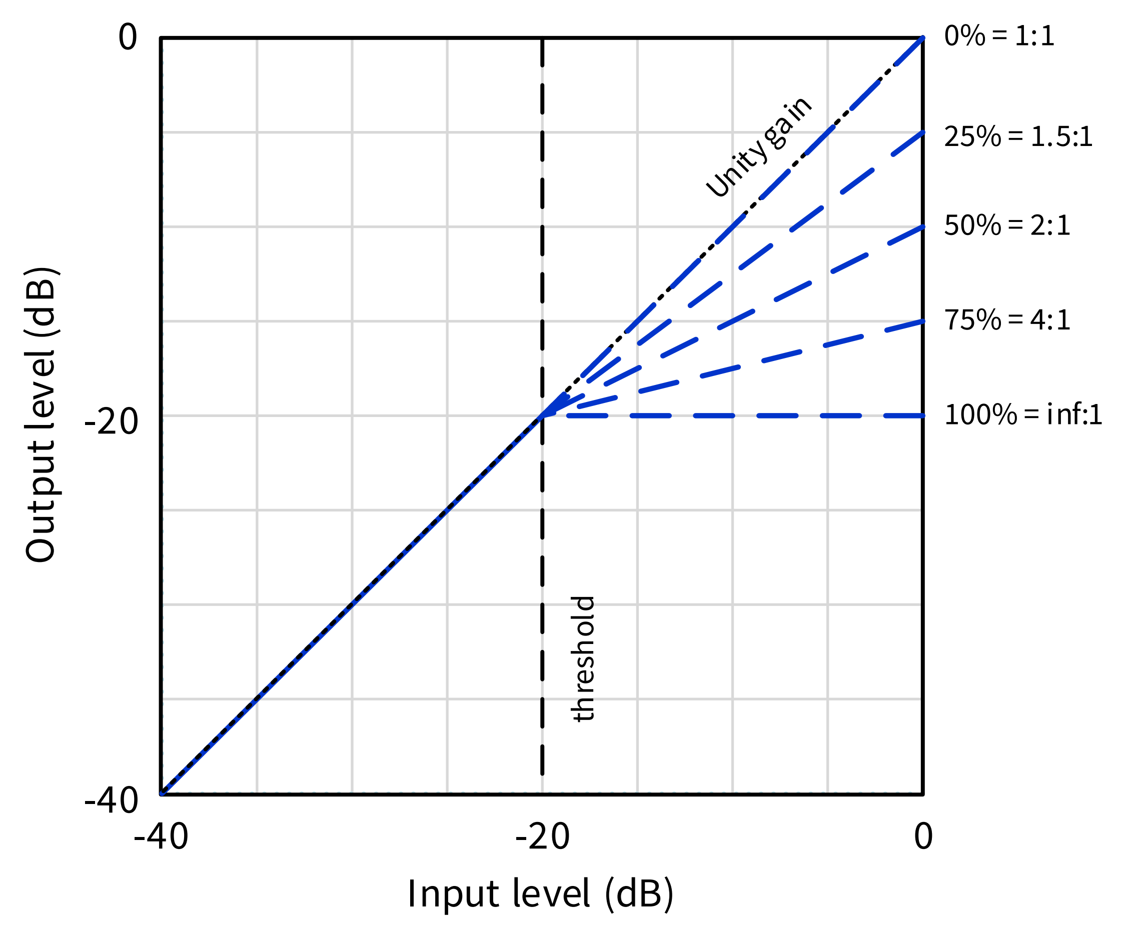

Figure 99 DRC input/output curve with threshold = -20 dB and several compression ratios

In time domain, this DRC implementation behaves similarly to that of the peak limiter, as explained in Section 5.2.2. However, the static input/output relationship will be different, as explained earlier.

For the following example, a three-band DRC flow was loaded with the following configuration:

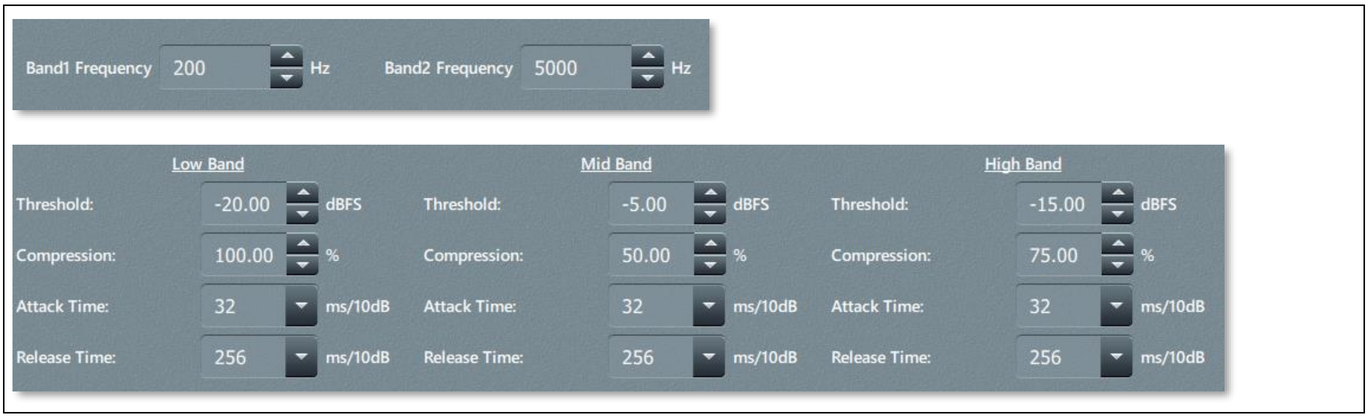

Figure 100 Three-band DRC flow configuration

The crossover frequencies are set to 200 Hz and 5 kHz. The low band is configured for heavy compression (100 percent) and a -20 dB_{FS} threshold. Little compression is applied to the mid band while the high band is configured for 75 percent compression.

Figure 101 shows a frequency sweep of this configuration. At a -20 dB_{FS} input level, the response is flat. As the input level is increased to -15 dB_{FS}, a reduction on the low band can be observed. Compression on the high band can be seen at the -10 dB_{FS} input level curve.

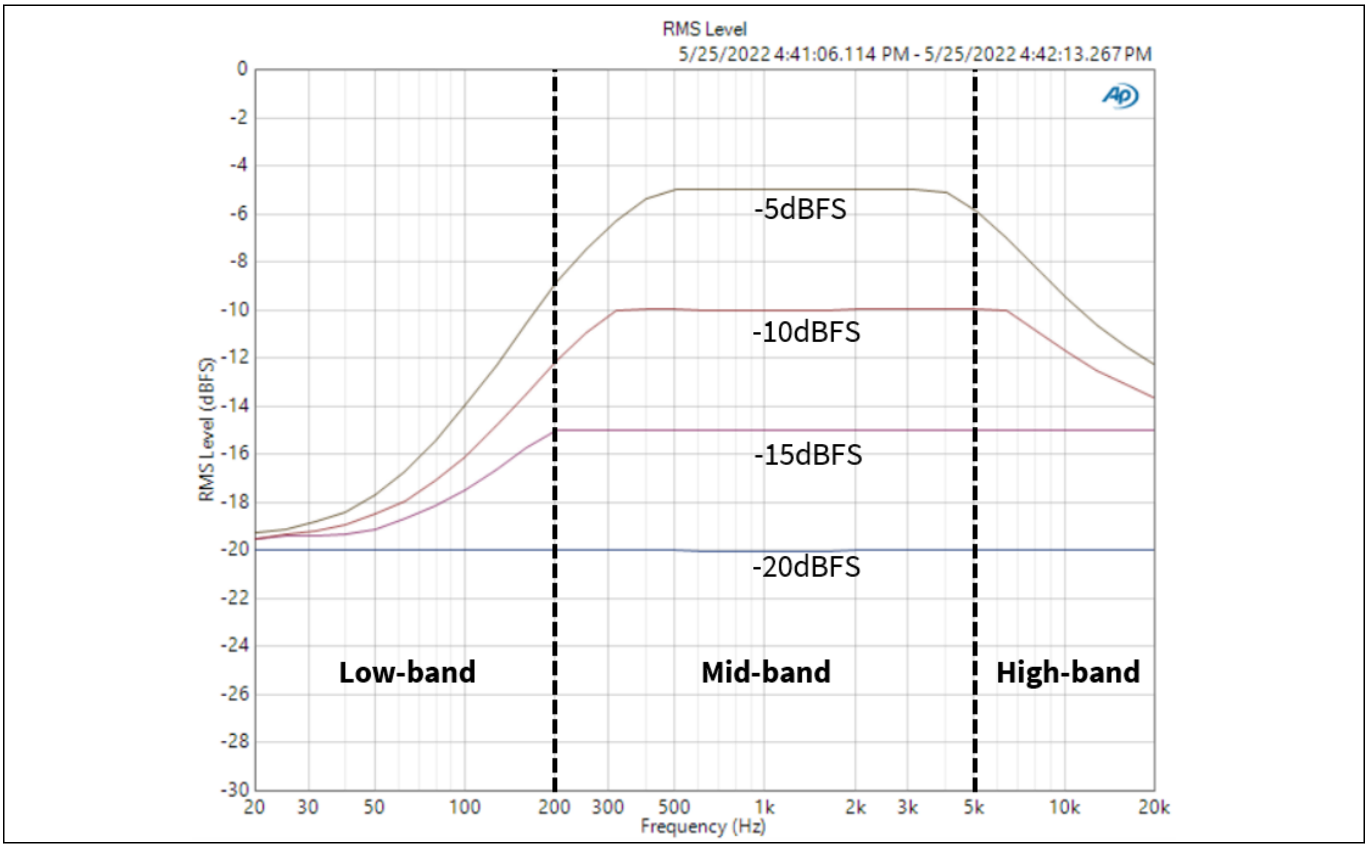

Figure 101 DRC frequency sweep at various input levels

5.4 Delay block

Figure 102 Delay block

The delay block simply adds delay to the signal path in per-sample increments. This block can be useful to sync audio and video in soundbar, home theater and other TV applications.

Double-clicking the delay block open its configuration window, as shown below:

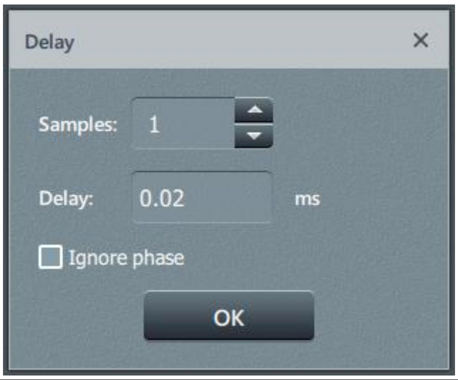

Figure 103 Delay configuration window

The delay block has a maximum delay of 860 samples across all sample rates. At 44.1 kHz sampling, this is equivalent to a maximum of 19.5 ms.

Table 18 Delay block control

Control name	Description	Min.	Default	Max.	Unit
Samples	Number of samples to delay	0	0	860	Samples
Delay	Delay time in ms	0.00	0.00	Samples/F_s x 1000	ms
Ignore phase	When enabled, the delay block phase reponse will not be taken into account in the GUI plot	Off	Off	On	

The plot below shows a one-sample delay applied to the left channel:

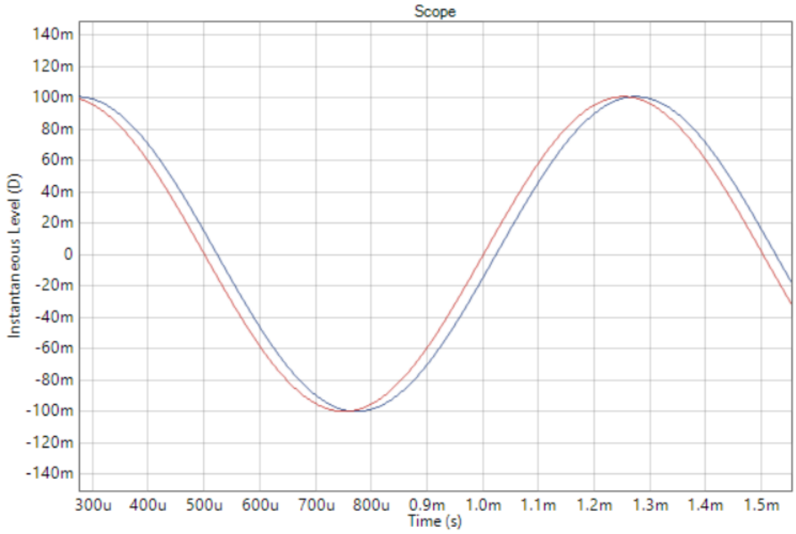

Figure 104 One-sample delay (0.02 ms) applied to the left channel

5.5 MIXMUX block

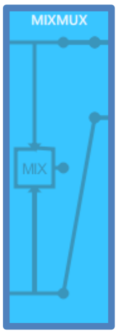

Figure 105 MIXMUX block

The MIXMUX block provides the option to select the input source (or a mix of sources) for each audio channel in the DSP flow. This can be useful for mono applications such as Bluetooth speakers, subwoofers and so on.

Double-clicking the MIXMUX block opens its configuration window, as shown below:

Figure 106 MIXMUX configuration window

The input source is derived from the RX_0 and RX_1 controls available in the I²S/TDM configuration page. Input 0 (left), input 1 (right) or a mix of inputs 0 and 1 can be selected as an option for each output channel.

The input mix option essentially divides the sum of input 0 and input 1 by 2.

Table 19 MIXMUX controls

Control name	Description	Options	Default
Output 0	Output channel 0 (left) source	Input 0 Input 1 Input mix	Input 0
Output 1	Output channel 1 (right) source	Input 0 Input 1 Input mix	Input 1
Output 2[1]	Output channel 2 source	Input 0 Input 1 Input mix	Input 0
Output 3[1]	Output channel 3 source	Input 0 Input 1 Input mix	Input 1

[1]Available in four channel audio flows only.

6 Saving and exporting configurations

Saving and exporting configurations is available on the configuration drop-down menu in the initial screen, as shown in Figure 107:

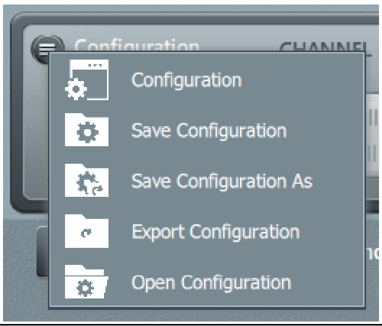

Figure 107 Configuration menu (initial screen)

6.1 Saving configurations

The "save configuration" options store register, DSP flow and GUI settings in a folder for later use. The files within this folder can be useful for debugging purposes as well.

Clicking "save configuration as" will prompt for a configuration name and location, as shown in Figure 108:

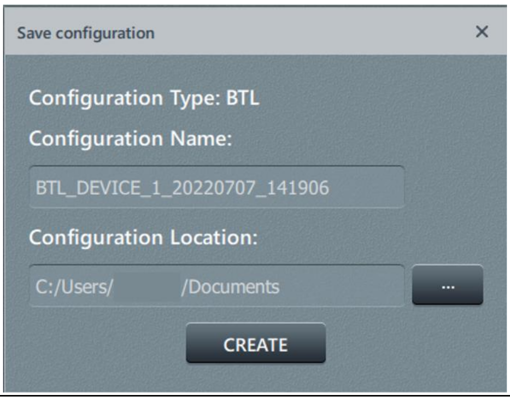

Figure 108 Save configuration window

To load a previously saved configuration, click the "open configuration" button in the configuration menu and select the same folder location.

Clicking the "create" button generates several files:

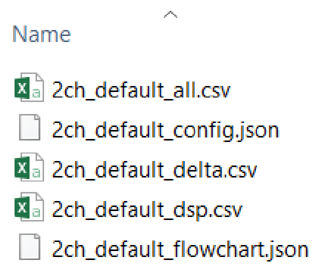

Figure 109 Files generated when saving a configuration

The following table provides a description for each of the generated files.

Table 20 “Save configuration” files

Filename	Description
xxx_all.csv	Contains a register dump of all relevant registers except DSP code
xxx_delta.csv	Contains any difference from the default register settings
xxx_dsp.csv	Contains the entire DSP program and data memory registers
xxx_config.json	Contains GUI settings
xxx_flowchart.json	Contains the DSP flow corresponding to the saved configuration

Each .csv file has two columns. The first column is the I^{2}C register address. Keep in mind that the MA2304xNS registers use a 2-byte address format. The second column is the corresponding 1-byte data.

Below is an example of a xxx_all.csv file:

```
0x0  0x34
0x1  0x3
0x2  0x0
0x3  0x20
0x4  0x20
0x5  0x2
0x6  0x0
...
```

Figure 110 xxx_all.csv snippet

Below is an example of a xxx_dsp.csv file:

```
0x1000 0x11
0x1001 0xf5
0x1002 0x3f
0x1003 0x0
0x1004 0x16
0x1005 0x50
0x1006 0x0
0x1007 0x0
...
```

Figure 111 xxx_dsp.csv snippet

The contents of the xxx_delta.csv and xxx_dsp.csv can be parsed and loaded via I^{2}C. The content of these files can be written into the device in any order. However, the MA2304xNS should be muted or put into standby while these data are being written to prevent audible clicks and pops.

6.2 Exporting configurations

To export a header file, select “export configuration” in the configuration drop-down menu. When prompted, select a folder to save the files into.

Clicking the “create” button generates several files:

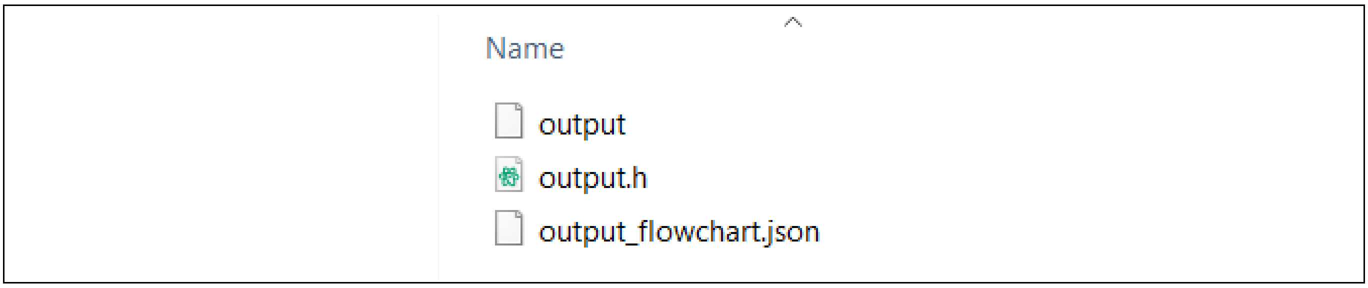

Figure 112 Files generated when exporting a configuration

The following table provides a description of each of the generated files:

Table 21 “Export configuration” files

Filename	Description
output	Contains the entire DSP program and data memory registers
output.h	Contains both the DSP code and device register settings
output_flowchart.json	Contains the DSP flow corresponding to the exported header file

The exported header file contains a PROGRAM_CODE[] array, a DATA_CODE[] array and REG_ADDRESS_X register settings:

```
#define PROGRAM_MEM_ADDR 0x1000
#define PROGRAM_MEM_SIZE 2048
#define PROGRAM_PACK_SIZE 1274

#define DATA_MEM_ADDR 0x2000
#define DATA_MEM_SIZE 2048
#define DATA_PACK_SIZE 160

uint16_t PROGRAM_CODE [PROGRAM_PACK_SIZE] = {
0x11f5, 0x3f00, 0x1650, 0x0000,
0x03f5, 0xff00, 0x5028, 0x0000,
0x10d8, 0xff00, 0x1f00, 0x0000,
...
0x13f0, 0x0400, 0x1552, 0x4000,
0x1050, 0x0000};

uint16_t DATA_CODE [DATA_PACK_SIZE] = {
0x0000, 0x0000, 0x0000, 0x0000,
0x0000, 0x0000, 0x0000, 0x0000,
0x0000, 0x0000, 0x0000, 0x0000,
...
0x0000, 0x0000, 0x0000, 0x0000,
0x0000, 0x0000, 0x0000, 0x0000};
```

Figure 113 output.h DSP code snippet

The REG_ADDRESS_X and REG_DATA_X macros are defined at the end of the file:

```
#define REG_ADDRESS_1    0x00bd
#define REG_DATA_1       0x0012

#define REG_ADDRESS_2    0x00be
#define REG_DATA_2       0x0000
```

Figure 114 output.h register settings snippet

These macros can be used to program device register settings in addition to the DSP code.

The PROGRAM_CODE and DATA_CODE arrays contain only data (i.e., no register addresses). Each array element is of 16-bit format.

6.2.1 Example MCU pseudo-code

To export a header file, select "export configuration" in the configuration drop-down menu. When prompted, select a folder to save the files into.

Below is an example MCU code snippet where the device is first muted, DSP data and device registers are written and then the device is unmuted for playback:

```
#include "output.h"
#define MA2304_ADDR 0x20
...

int main()
{
  amp_mute(true);
  amp_write_code(PROGRAM_MEM_ADDR, PROGRAM_CODE, PROGRAM_PACK_SIZE);
  amp_write_code(DATA_MEM_ADDR, DATA_CODE, DATA_PACK_SIZE);
  amp_write(REG_ADDRESS_1, REG_DATA_1);
  amp_write(REG_ADDRESS_2, REG_DATA_2);
  amp_mute(false);
  return 0;
}

int amp_write_code(uint16_t addr, uint16_t * code, uint16_t size) {
  int i = 0;
  // Split 16-bit register address into bytes
  uint8_t i2c_buffer;                    // Temporary storage
  i2c_buffer = addr >> 8 & 0xff;         // Register address MSB
  i2c_write (MA2304_ADDR, i2c_buffer);   // Send byte through I2C
  i2c_buffer = addr & 0xff;              // Register address LSB
  i2c_write (MA2304_ADDR, i2c_buffer);   // Send byte through I2C

  // Write data through I2C
  for (i = 0; i < size; i++) {
      i2c_buffer = code[i] >> 8 & 0xff;
      i2c_write (MA2304_ADDR, i2c_buffer);
      i2c_buffer = code[i] & 0xff;
      i2c_write (MA2304_ADDR, i2c_buffer);
    }
}
```

Figure 115 Example MCU pseudo-code

Revision history

Document revision	Date	Description of changes
V 1.0	2022-08-26	Initial release
V 1.1	2022-09-12	- Updated document title to: "MERUS™ audio amplifier configurator" - Renamed software name throughout the document accordingly - Added software download instructions
V 1.2	2022-10-28	- Modified section 1.3: Infineon developer center information added - Modified Figure 5
V 1.3	2023-09-25	- Modified Figure 1 - Added Figure 4

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