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Qualcomm Technologies International, Ltd.

Qualcomm BlueCore™ Music Manager (E00F)

User Guide

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Document history

Revision	Date	Change reason
AA	October 2017	Original publication of this document. Alternate document number: CS-00406876-UG

1   Music Manager Introduction

The Music Manager is a Windows-based application that enables you to configure and monitor a Qualcomm Technologies International, Ltd (QTIL) Sink Digital Signal Processor (DSP) application:

■   Headset

■   Speaker

■   Soundbar

These applications are for manufacturers of mono/stereo Bluetooth-enabled products. This document refers to them as Sink DSP Applications.

The Music Manager is available with the appropriate version of an Application Development Kit (ADK) or as a Windows installer for a ROM-based chip. This document describes how to use the Music Manager application to perform basic tuning and monitoring activities.

QTIL provides audio effect algorithms for creating music-based products. Some decoders are licensed, such as AAC. The manufacturer must apply the correct licensing.

1.1   Mono/stereo sink ADK features

Decoder types are:

■   SBC Decoder

■   MP3 Decoder

■   AAC-LC Decoder

■   aptX-LL (low-latency) Decoder (security key required)

■   aptX Classic Bluetooth Decoder (security key required)

■   aptX HD Decoder (security key required)

> **NOTE**   Decoder types may require add-on installers.

Audio effects and input modes are:

■   Compander

■   User Parametric Equalization

■   ANC Parametric Equalization

■   Bass Enhancement (Bass boost and MeloD BASS+)

■   Speaker Parametric Equalization

■   Stereo Enhancement (MeloD and Virtual Spatial Enhancement)

- Cross Over

- Subwoofer Parametric Equalization

- Post Mastering (Noise-shaped Dither)

- Wired input modes. Wired modes can replace the Bluetooth radio (A2DP) inputs with:

  □ USB

  □ Analog (using the ADC inputs)

  □ Sony/Phillips Digital Interface (SPDIF)

  □ I^{2}S input

Audio effects are provided for 44.1 kHz, 48 kHz, (88.2 kHz, and 96 kHz for Analogue, S/PDIF and I^{2}S) input sample rates, and are supported for all the specified decoders described in this section. All audio effects are sample rate agnostic. For unsupported sample rates, no DSP frame-based effects are performed, the audio is strictly passed through.

1.2　Assumptions

This document assumes:

- The ADK software is built and downloaded to a suitable hardware platform. See the *ADK Release Note* for details on QTIL development board compatibility and supported ICs.

- You are using the correct versions of the DSP application software and Music Manager UFE.

- You are familiar with using common features of the Music Manager, such as:

  □ Connecting to the DSP

  □ Saving and loading `.psr` files

  □ Uploading and downloading parameters to PS Keys

  □ Monitoring values from the DSP

See Bass Enhancement for related voice and music applications supplied, which are also under the UFE Document tab, or contact your QTIL field support engineer.

2 Getting started

The basic steps for using the Music Manager application are:

1. Install the ADK or separate installer for ROM chips.

2. Access the Music Manager application.

3. Use parameters and metrics information for tuning and/or monitoring.

2.1 Installing the Audio Development Kit

QTIL designated a location for the download of ADKs or UFE Installer. Ensure that location is accessible.

To install an ADK or UFE Installer:

1. Locate and install the ADK. The ADK creates a subdirectory on the root drive of the PC.

2. Locate and install the UFE Installer. The installer creates a subdirectory on the **root/Program Files/CSR** drive of the PC.

2.2 Accessing the Music Manager application

To open the Music Manager application, the `UniversalFrontEnd.exe` file and the associated `*.dll` files must be installed in the same folder.

To access the Music Manager from the Universal Front End (UFE):

1. Open the UFE from the Windows Start menu:

    a. **Start\All Programs\<ADK>\Tools\Universal Front End**

    b. **Start\All Programs\ <UFE Installer Name>**

2. Select Music Manager from the drop-down menu.
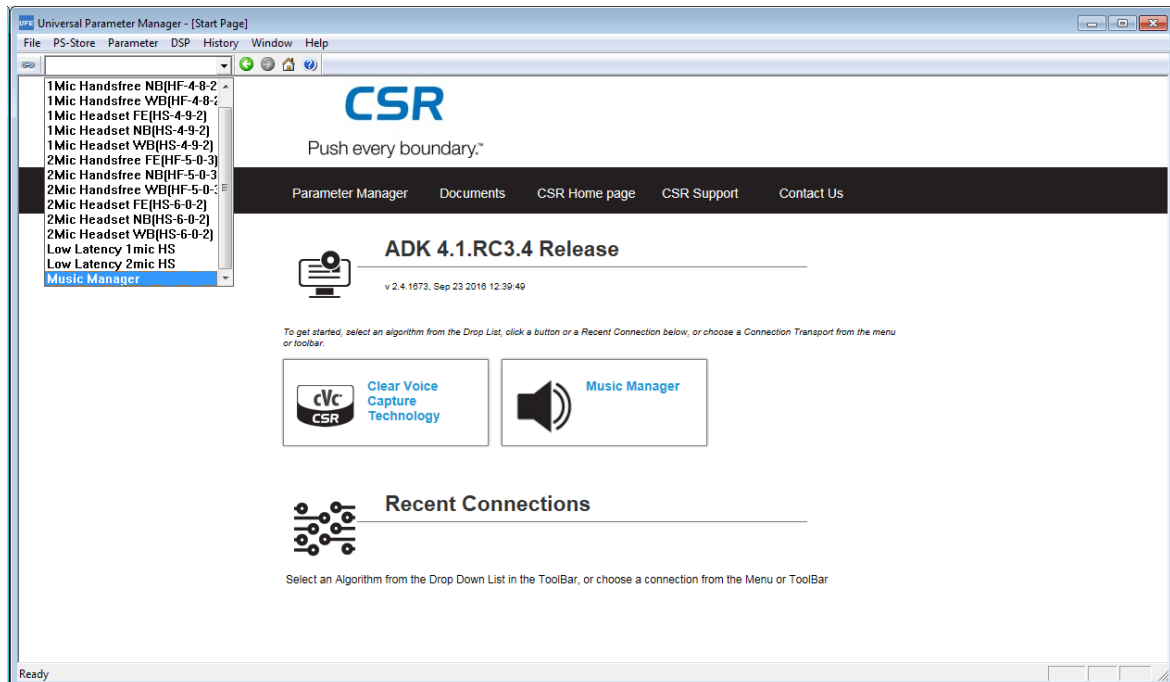
Figure 2-1   Selecting Music Manager application
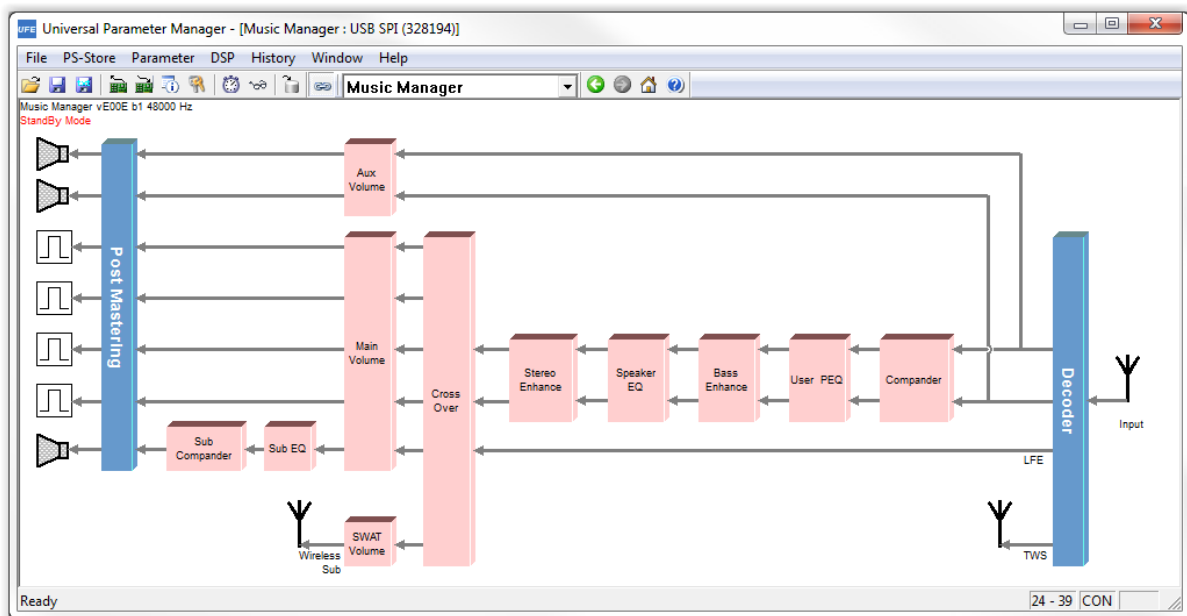
Figure 2-2   Music Manager application main screen

Processing blocks available for adjustment are:

■ Decoder

■ Compander

■ User EQ

■ ANC EQ

■   Bass Enhancement

■   Speaker EQ

■   Stereo Enhancement

■   Crossover

■   Main Volume

■   Aux Volume

■   SWAT Volume

■   Sub EQ

■   Post Mastering (Noise-Shaped Dither)

Table 2-1   Music Manager processing block descriptions

Option	Description
Gains, Adjustable Tuning Parameters	

Main Volume	■ One of three volume control schemes can be invoked. To define these schemes, use the Sink Configuration Tool.
	
	   ☐ Digital
	
	   ☐ Analog (DAC)
	
	   ☐ Hybrid
	
	   ☐
	
	■ For debug purposes, override and set the DAC gain during the tuning process (Monitoring mode).

Aux Volume	For debug purposes, you can override and set the Aux channel gain(s) during the tuning process (Monitoring mode)..
SWAT Volume	No parameters.
Crossover	■ Enabled when creating a Stereo 2.1 audio system, comprised of stereo front channels, and a wireless subwoofer. ■ A crossover filter enables the manufacturer to filter the frequency response to the individual speakers (such as, Tweeters, Midrange) and to the subwoofer. ■ Creates the signal path routing from the input channels to the Primary/ Secondary outputs.
Algorithms, Adjustable Tuning Parameters	
Speaker EQ	A single bank of up to a 10-stage parametric equalizer is applied to both the left and right channels.
Bass Enhancement	■ A single-stage parametric equalizer applied to both the left and right channels. ■ MeloD BASS+ provides dynamic bass enhancement for entry level speaker systems. ■ A user-driven feature, triggered by an on/off button press.
User EQ	Up to six banks of up to a five-stage parametric equalizer for both the left and right channels are possible.
ANC EQ	Up to a five-stage parametric equalizer for both the left and right channels are possible.

Table 2-1   Music Manager processing block descriptions (cont.)

Option	Description
Stereo Enhancement	■ Makes audio sound fuller by modeling the transmission of sound in a room environment. ■ Two separate effects are offered: MeloD Expansion and Virtual Spatial Enhancement (VSE).
Compander	Compresses or expands the dynamic range of streaming audio to deliver a more consistent listening level.
Post Mastering	Improves the DAC fidelity by providing noise-shaped dither (NSD).
Sub EQ	A single bank of up to a three-stage parametric equalizer.
Sub Compander	Compresses or expands the dynamic range of streaming audio to deliver a more consistent listening level.
Signal Interface Terminals	
DAC Output	Converts digital signals to analog signals.

Decoder
 	
(optional USB, I^{2}S, SPDIF, or Wired Analog)
	■ Supports decoder input types:
	  □ A2DP
	  □ USB
	  □ I^{2}S
	  □ S/PDIF
	  □ Wired audio
	■ The ADK version can support:
	  □ SBC
	  □ MP3
	  □ FastStream
	  □ AAC
	  □ aptX
	  □ aptX-HD
	■ The ROM versions support a subset and are chip-dependant.
	■ The decoder block name and bitmap changes if USB, I^{2}S, S/PDIF, or analog wired modes are active and the UFE is in Monitoring mode.
	■ Provides a selection matrix to define what audio effects are applied to each codec type.

2.3   Speaker and subwoofer Parametric Equalizers

The Music Manager application provides tuning support for two time domain Parametric Equalizer (EQ).

The first EQ is intended for the entire input stereo signal. When a subwoofer is connected, a second EQ is available for tuning. The only parameter tuning difference is the total number of stages available (Speaker EQ includes ten, Sub EQ offers three).

For the Speaker EQ, the Sink DSP application uses the bank of coefficients and is applied to the left and right channels. A different equalization curve cannot be selected for the left and right channels.

This Equalizer block helps offset any hardware considerations to meet overall product targets. The manufacturer configures the Speaker EQ to correct the speaker(s) frequency response. The user cannot alter the Speaker or Subwoofer EQ.

For the Sub EQ, the Sink DSP application uses the bank of coefficients and is only applied to the wired subwoofer signal (when configured).

2.3.1   Adjusting the Speaker EQ settings

To adjust the Speaker EQ settings:

1. From the **Music Manager** window, click the **Speaker EQ** block. The **Speaker Equalizer Settings** window opens with the **Default** preset selected as the default setting. To enable, check **Stage**.
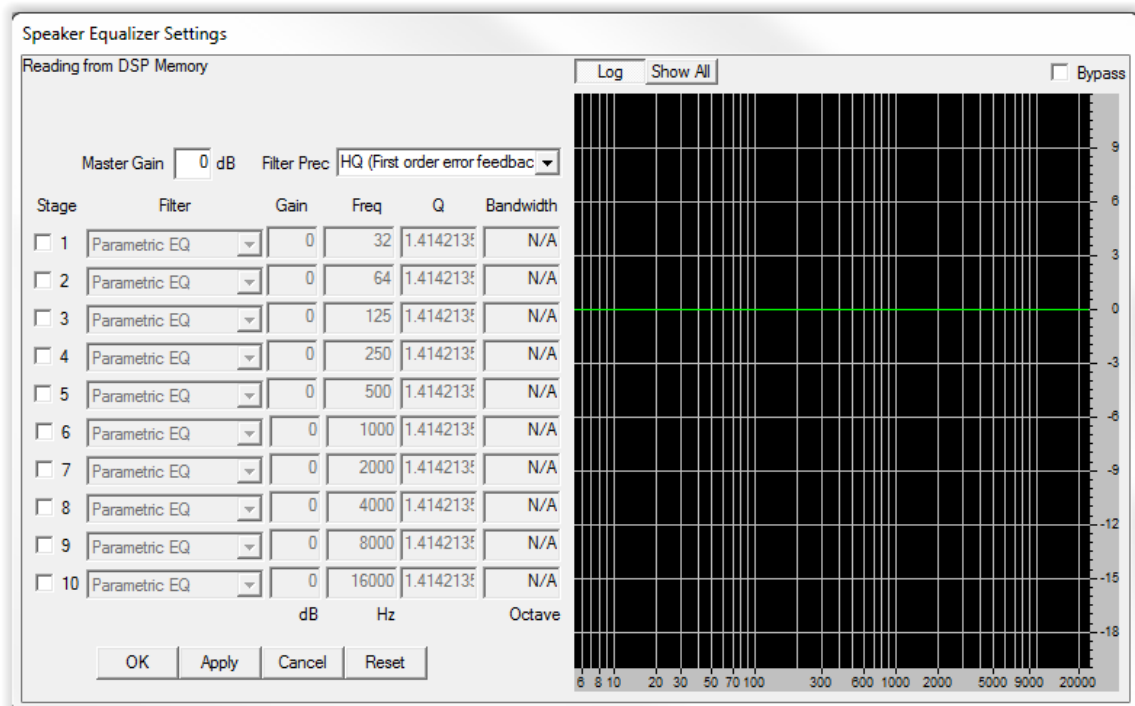
Figure 2-3   Speaker Parametric Equalizer (EQ) window

2. Figure 2-3 shows the state before the stages are selected. The line shown in the plot area represents the equalization curve applied to the audio.

3. To adjust this curve (bank) manually, either:

    □   Select the **Stage** check boxes and enter the appropriate filter parameters in the text box.

    □   Drag the filter stage handle in the Equalizer curve display area.

4. After adjusting the EQ to the required shape, click **Apply**.

    NOTE   To save the settings, click **OK** to exit the window or Download Parameters or Save PSR File, otherwise these settings are lost if the Music Manager is closed. The EQ interface supports multiple views of the Equalizer without needing to close the window.

5. Select the required settings based on the options listed in Table 2-3

2.4   Bass Enhancement

The Music Manager application provides tuning support for the time domain Bass Enhancement option.

The Sink DSP application uses the bank of coefficients and is applied to the left and right channels when Bass Boost is chosen from the Enhancement pulldown menu. A different equalization curve cannot be selected for the left and right channels. This Equalizer block helps offset any hardware considerations to meet overall product targets.

User events enable the user to toggle the Bass Enhance feature on and off with a button press. See the *ADK Sink Configuration User Guide* for configuration details of:

- `Bass Boost Enable Disable Toggle`
- `Bass Boost On`
- `Bass Boost Off`

2.4.1   Adjusting the Bass Enhancement settings

To adjust the Bass Boost settings:

1. From the **Music Manager** window, click the **Bass Enhancement** block. The **Bass Enhancement Settings** window opens with the **Bypass Bass Enhancemen**t selected as the default.
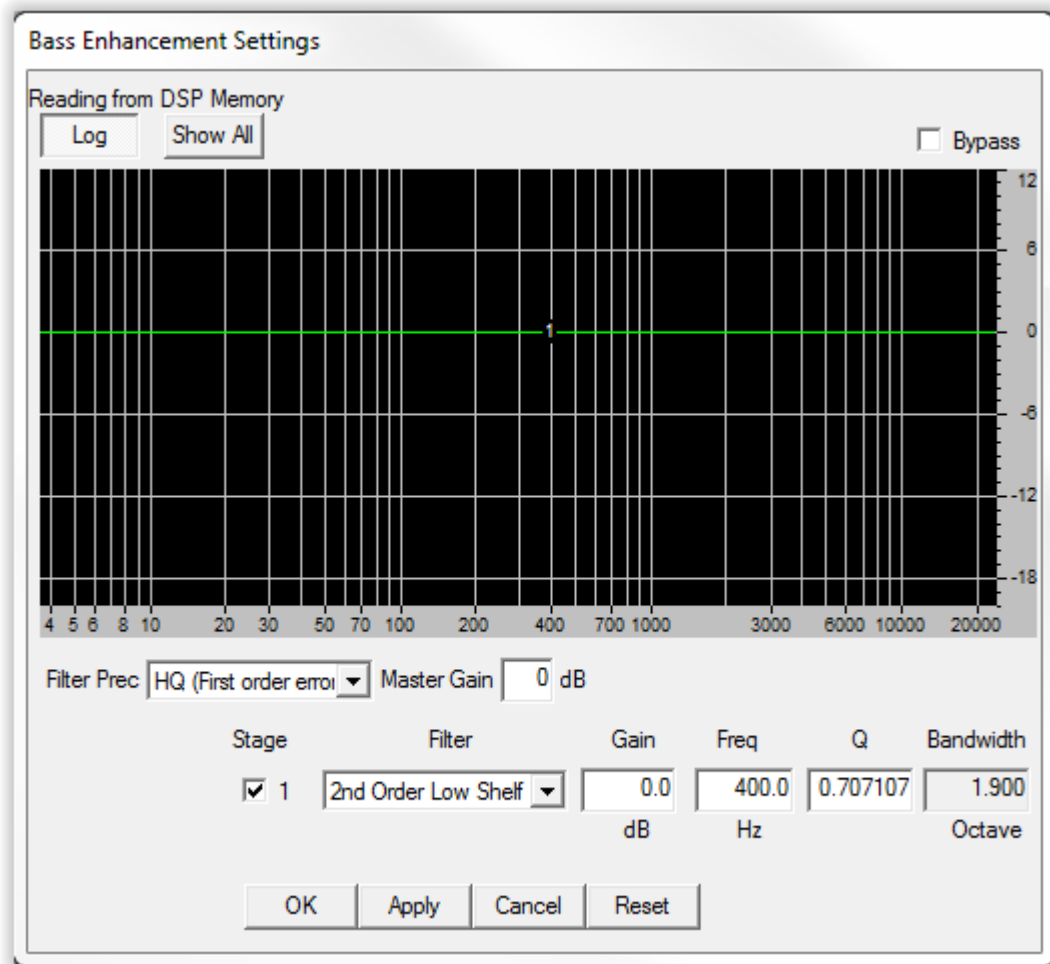
Figure 2-4   Bass Enhance Settings window, default state

The line shown in the plot area represents the equalization curve applied to the audio.

2. To adjust this curve manually, either:

   □   Enter the appropriate filter parameters in the text boxes.

   □   Drag the filter stage handle in the Equalizer curve display area.

3. After adjusting the EQ to the required shape, click **Apply**.

   NOTE   To save the settings, click **OK** to exit the window or **Download Parameters** or **Save PSR File,** otherwise these settings are lost if the Music Manager is closed.

The EQ interface supports multiple views without needing to close the window.
The Bass Enhancement also includes the option of adding Bass Plus for tuning. It may be used together with the Bass Boost or stand alone.
Select the required settings based on the options described in Table 2-3.

Table 2-2   MeloD Bass Plus options

Option	Description
Effect Strength	Maximum Bass Plus enhancement setting at 100% (range for this parameter is 0 to 100%).
Amplitude Limit	Maximum allowed amplitude at the output of the Bass Plus relative to 0 dBFS.
Low-Pass Corner Frequency	Upper corner frequency of the Bass Plus bandpass filter (Hz)
High-Pass Corner Frequency	Lower corner frequency of the Bass Plus bandpass filter (Hz).
Harmonic Content	Ratio of the harmonics added compared to the input signal.
Crossover Frequency	The crossover frequency that best matches the cut-off frequency of the speaker.
Mixing Balance	Controls the ratio of the Bass Plus output versus the high-passed input signal (0% results in only high-passed output).

2.5   User Parametric Equalizer

The Music Manager application provides tuning support for the time domain parametric Equalizer.

The Sink DSP application uses the active bank of filter parameters, and is applied to the left and right channels. A different equalization curve cannot be selected for the left and right channels.

Although the User EQ included with the Sink Music Manager DSP application is different from the Speaker EQ, the User EQ enables real-time selection of different banks of coefficients computed in the Qualcomm® Kalimba™ DSP at runtime. For more information on how the User EQ works, see the *User EQ* document.

The User EQ can support up to six banks of coefficients. The banks of coefficients can be user-selected for different types of music based on listening preferences. For example, a rock setting can apply gain to both the lower and upper frequencies, while a pop setting can apply gain to the mid-range frequencies.

■   Although there are six banks of coefficients available, only one bank can be applied to the audio stream at a time.

■   Under the VM control, the sink user can cycle through the EQ banks using the `Switch Audio Mode` user event. A typical user event configuration can be set up so that the user can cycle EQ banks by pressing the appropriate button during music play.

■   The EQ banks are cycled in the following order. This order assumes the Max is 6:

   a.   Flat (no EQ, if the Flat check box is selected)

   b.   EQ Bank 1

   c.   EQ Bank 2

   d.   EQ Bank 3

   e.   EQ Bank 4

   f.   EQ Bank 5

g.　EQ Bank 6

h.　Return to top of the list

2.5.1　Adjusting the EQ settings

To adjust the EQ settings:

1.　From the **Music Manager** window, click the **EQ** block. The **Equalizer Settings** window opens with the default setting having the **Bypass** checked. To enable, uncheck **Bypass**.
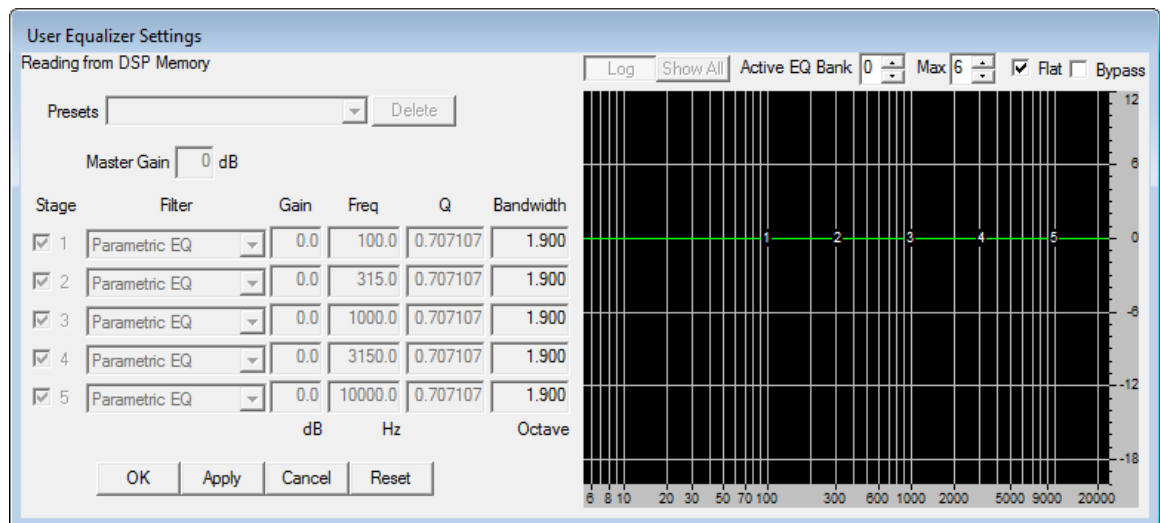
Figure 2-5　Parametric Equalizer (EQ) window

2.　The product manufacturer can configure up to six banks of filter parameters. The title bar of the window indicates:

☐　Which bank is active

☐　The maximum number of banks

☐　If a Flat EQ is required

3.　Figure 2-5 shows the default state before presets or custom banks are selected. The Presets field is blank by default. The line shown in the plot area represents the equalization curve applied to the audio.

4.　To adjust this curve bank manually:

a.　Uncheck **Bypass**

b.　Select the Max number of Banks to be supported

c.　Select **Flat** (optional)

5.　Select the **Active EQ Bank** index number to tune. To select a Preset from the drop-down menu, either

☐　Select the **Stage** check boxes and enter the appropriate filter parameters in the text boxes.

☐　Drag the filter stage handle in the **Equalizer** curve display area.

6.　After adjusting an EQ Bank to the required shape, click **Apply**.

7. Select the next Active EQ Bank index number, adjust the shape, click **Apply**, and repeat until all EQ banks are configured.

> **NOTE**   To save the setting, click **OK** to exit the window or **Download Paramet**ers or **Save PSR Fil**e, otherwise these settings are lost if the Music Manager is closed. The EQ interface supports multiple views of the Equalizer without needing to close the window.

8. Select the required settings based on the options described in Table 2-3.

Table 2-3   Equalization options

Option	Description
Master Gain	■ Only available when one of the Stage check boxes is selected ■ Applies a constant gain across all frequencies, shifting the curve up or down without changing the shape ■ Range for this field is -90 dB to 12 dB
Filter Prec	■ Allows selection of HQ (First order error feedback), or SH (Single precision feedback). ■ Using HQ results in more precise filter coefficients, while increasing MIPS.
Linear/Log	■ Controls the appearance of the plot area on the screen ■ Click to toggle the curve plot between the Linear (default) and Log (Logarithmic) views
Show All	Enables a view of all the stages for the current curve, in the plot area
Active EQ Bank	■ Enables you to view or adjust a specific active bank ■ Displays the current stages for the selected active bank
Max	■ QTIL enables the manufacturer to configure N banks where the maximum is N=6 ■ Configure each bank 1 through N with a valid curve, because the user event `Switch Audio Mode` cycles through the EQ banks in numeric sequence from 0 up to bank N, then repeats
Flat	■ Specifies that Active EQ Bank 0 provides a flat frequency response ■ This is an extra Bank beyond what is specified by **Max**
Bypass	■ Ignores all equalizer settings and bypasses the module ■ Does not consume any DSP resources (MIPS)
Presets	Provides a collection of sample curves. Available options include: ■ Default ■ Bass Boost ■ Treble Boost ■ Rock ■ Jazz Each Preset option can be altered using the stages listed in the Graphic EQ/Parametric EQ area on the screen.

Table 2-3   Equalization options (cont.)

Option	Description
Filter	The filter type drop list is a selection with the following types per stage:: ■ Bypass ■ 1st order low pass ■ 1st order high pass ■ 1st order all pass ■ 1st order low shelf ■ 1st order high shelf ■ 1st order tilt ■ 2nd order low pass ■ 2nd order high pass ■ 2nd order all pass ■ 2nd order low shelf ■ 2nd order high shelf ■ 2nd order tilt ■ Parametric EQ The Filter enables the Stage Filter type. A filter type can be set for each stage, which enables complex curves to be created In Parametric mode the fields in Gain, Freq, and Q can be edited. The Bandwidth is calculated based on the parameters entered, and is read-only. Enter or select data directly from the fields or from the plot area. Users can drag and drop the corresponding numbers on the curve to adjust the settings in the fields.
Filter Parameters: ■ Gain: Gain (dB) ■ Freq: Frequency (Hz) ■ Q: Sharpness of curve ■ Bandwidth: Read-only (octave)	Filter parameters vary based on filter type: ■ Bypass: Gain, Freq, and Q ■ x Order Low Pass: Freq, and Q ■ x Order High Pass: Freq, and Q ■ x Order All Pass: Freq, and Q ■ x Order Tilt: Gain: Freq, and Q ■ Parametric EQ: Gain, Freq, and Q
Reset	■ Reverts to the active EQ bank filter curve from the last saved state, which is the state last saved by clicking **OK**. ■ Click **Reset** to update the settings in the stages with the last saved settings.
Apply	■ In the Monitor mode, click **Apply** to write the EQ parameters to the DSP memory where the changes take immediate effect. Changes in the Monitoring mode only affect the DSP and PC memory, not the Persistent Store memory. ■ Apply can write unlimited parameters to the DSP memory without closing the User Equalizer Settings window. ■ Apply is not available in the Static mode.
OK	■ In the Monitor mode, click **OK** to write the EQ parameters to the DSP memory where the changes take immediate effect. The User Equalizer Settings window closes. ■ Changes made in the Monitoring mode only affect the DSP and PC memory, not the Persistent Store memory. ■ OK is not available in the static mode.
Cancel	Closes the User Equalizer Settings window without saving any of the latest changes.
Vertical and Horizontal Scale Bar User Controls	

Table 2-3   Equalization options (cont.)

Option	Description
Zoom In	To zoom in: 1. Use the mouse to select a starting zoom point on the gray scale bar in the plot area. 2. Left-click and drag to the required ending zoom point. 3. Release the mouse button. The plot area zooms to the selected area
Zoom Out	To return to the default scale, double-click the left mouse button on the gray scale bar in the plot area
Scale Drag	To change the curve in the plot area: 1. Right-click, hold, and drag the mouse button within the plot area to the required location 2. Release the mouse button. The horizontal scale only drags if the scale is zoomed.

2.6   Stereo Enhancement

The Music Manager application enables simple tuning of the MeloD Expansion and Virtual Spatial Enhancement algorithms in the Stereo Enhancement block included in the ADK.

The MeloD Expansion algorithm provides a fuller sound to the stereo audio, by modeling the transmission of sound in a room.

The Virtual Spatial Enhancement algorithm processes the stereo signals before sending them through closely spaced speakers such that it will give the impression that they emanate from more widely spaced virtual loudspeakers.

User events enable the user to toggle the 3D Enhancement feature on and off by pressing a button. See the ADK Sink Configuration User Guide for configuration details of:

■ `3D Enhancement Enable Disable Toggle`

■ `3D Enhancement On`

■ `3D Enhancement Off`

2.6.1   Adjusting the Stereo Enhancement settings

To adjust the MeloD Expansion settings:

1.   From the Music Manager window, click the Stereo Enhance block. The Stereo Enhancement Settings default window opens.

MAY CONTAIN U.S. AND INTERNATIONAL EXPORT CONTROLLED INFORMATION
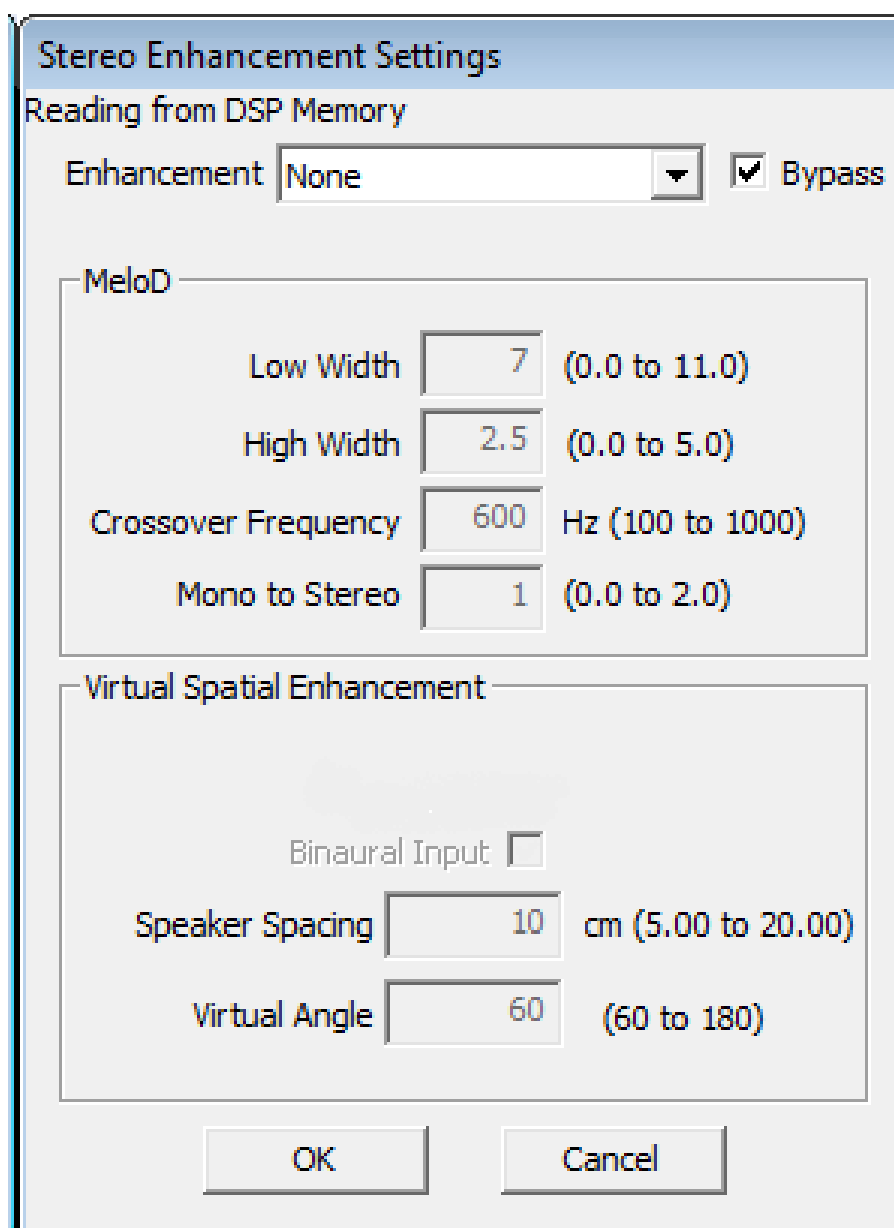
Figure 2-6   Stereo Enhancement Settings window

2. To enable, uncheck the Bypass check box. Select the required settings.

Table 2-4   MeloD expansion options

Option	Description
Low Width	☐ Increases the amount of stereo widening at low frequencies. ☐ The control ranges from 0 to 11 with 0 being no low frequency widening and 11 being the maximum. The default value is 7.0.
High Width	☐ Increases the amount of stereo widening at high frequencies. ☐ The control ranges from 0 to 5 with 0 being no low frequency widening and 5 being the maximum. The default value is 2.5.
Crossover Frequency	☐ Sets the frequency of the High Width / Low Width crossover. ☐ The default value is 600 Hz.
Mono to Stereo	☐ Gives a feeling of width and body to signals that are mixed to the center of a stereo. ☐ The control ranges from 0 to 2.0 with 0 being no increase in the width of mono signals and 2.0 being the maximum. The default value is 1.0
Enhancement	Select seither MeloD Expansion or Virtual Spatial Enhancement algorithm.
Binaural Input	Turns on processing allowing binaural audio to be properly heard by the user.
Speaker Spacing	The actual center-to-center spacing between Left and Right speakers (in cm.). The spacing may range from 5.00 to 20.00 cm. The default value is 10 cm.
Virtual Angle	The perceived angle of arrival desired by the user. ☐ The angle may range from 60 to 180 degrees. The default value is 60 degrees.
Bypass	Select this check box to disable the MeloD Expansion algorithm

2.7   ANC EQ

There is an additional 5 stage EQ available for use in conjunction with the ambient noise cancellation feature for headsets (ANC).

This EQ is activated when the ANC feature on the device is turned ON and bypassed when the ANC is turned OFF. This EQ is intended for providing compensation to any degradation that might be caused to the music content when ANC is turned ON.

In products where the ANC feature is not used, this EQ is not available for use.

2.8   Crossover

The Music Manager application enables tuning of the Crossover controls included in the ADK. The controls consist of:

- Bypass the entire module
- Control signal gains
- Selection of crossover filter types
- Change frequency of crossover filters

> **NOTE**   The speaker control system is a flexible speaker management system that enables multiway speaker systems. The crossover parameters are common to the 44.1 kHz and 48 kHz rates.

If a Subwoofer/sink connects, the VM automatically initializes the subwoofer connected parameters. Otherwise, subwoofer-not-connected parameters are used.

■ Systems using Qualcomm® cVc™ need to consider that the Primary output channels are used for voice. The speakers receiving voice audio through the Primary channels should be capable of handling such input.

■ The tone routing is performed after the music manager processing and as such does not go through the speaker crossover, so the tones are also directed to the primary audio channels in the same way as cVc. Speakers receiving indication tones through the Primary channels should be capable of handling such input.

2.8.1 Adjusting the Crossover settings

For the best result from the Crossover, the lowest frequency that the speakers are capable of producing must be known. These frequencies are then entered into the appropriate Crossover Settings window. The gain settings in the crossover are available to optimize the signal to noise ratio without clipping.

NOTE   Do not disable the filtering except for testing purposes. The system depends on this filtering for correct operation.

To adjust the Crossover settings:

1. From the Music Manager window, click the Crossover block. The Crossover Settings default window opens.

Figure 2-7   Crossover Settings window

2. To enable, uncheck the **Bypass Crossover** check box. Select the required settings.

3. To change system gains, update parameters in the Crossover window.

Table 2-5   Crossover options

Option	Description
Bypass	Completely disables the Crossover processing
Signal Gains	■ Provides modification of signal gains (in dBFS). Here, the input channels are listed on the column header Input Channels Left Right Bass. ■ Under each input channel, an attenuation factor (0 to -NF) may be applied for each output (Primary Left/Right, Secondary Left/Right, Wired Sub, and Wireless Sub).
Filter Types (Pull-down menus)	From the filter pull-down menus, the user has a choice of Butterworth (1 through 12th order), Linkwitz-Riley (2nd through 12th order, even orders only).
Primary Low Pass	Cutoff frequency used in the filter chosen for Primary Low Pass filter.
Primary High Pass	Cutoff frequency used in the filter chosen for Primary High Pass filter.
Secondary Low Pass	Cutoff frequency used in the filter chosen for Secondary Low Pass filter.
Secondary High Pass	Cutoff frequency used in the filter chosen for Secondary High Pass filter.
Filter Precision	■ Enables selection of HQ (First order error feedback), or SH (Single precision feedback). ■ Using HQ results in more precise filter coefficients, while increasing MIPS.
Invert	If channel inversion is selected, then all Crossover gains associated with that output should be inverted.
OK	■ In Monitoring mode, click **OK** to write the Crossover parameters to the DSP memory where the changes take immediate effect. The Crossover Settings window closes ■ Changes made in the Monitoring mode only affect the DSP and PC memory, not the Persistent Store memory ■ OK is not enabled in the Static mode
Cancel	Closes the Crossover Settings window without saving any changes

2.9   Compander and subwoofer compander

The Music Manager application provides tuning support for two time domain companders. The first Compander is intended for the entire input stereo signal, while the second Subwoofer Compander (Sub Compander) operates only on the wired subwoofer signal (when configured).

In the current configuration, the Sink DSP application only supports one Compander curve for the left and right channels.

The user cannot select a different curve for the left and right channels. The only tuning parameter difference between the two companders is the addition of the Volume Control Hard Limiter for the input stereo signal Compander. During the tuning phase, the Music Manager enables selection of different Compander curves or custom settings, in real-time.

The Compander processing varies the output level of a waveform, based on its input level. This process enables the user to automatically compress or expand the dynamic range of the audio so

perceived loudness is kept below a defined dB limit, or so that the waveform's overall dynamic range is kept at approximately the same level.

A Volume Control Hard Limiter is active by default with fixed settings (Threshold = -1 dBFS and Ratio = 20). The user may disable the Volume Control Hard Limiter.

The user can select a Compander curve from the set of factory Presets provided in the drop-down list. These are examples. The final curves can be adjusted and stored in PS for runtime use.

2.9.1   Adjusting the Compander settings

To adjust the Compander settings:

1. Click the **Compander** block in the Music Manager window. The **Compander** window opens. The default is **Bypass** and the Volume Control Hard Limiter is not bypassed. To disable, check **Bypass Volume Control Hard Limiter.**
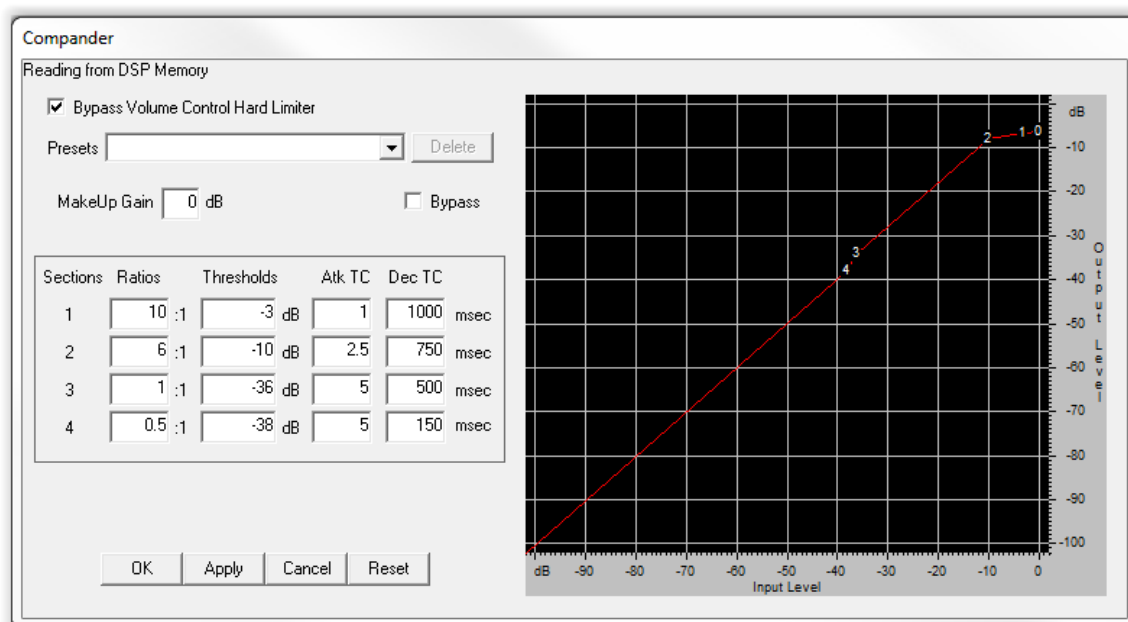
Figure 2-8   Compander window

2. One Compander curve can be configured. The **Presets** field indicates which curve is active or it can be customized.

3. Figure 2-8 shows the default state before **Presets** are selected. To disable tuning, check **Bypass**.

4. The red line shown in the plot area represents the Compander curve applied to the audio. Each section of the Compander process defines the curve between two adjacent points on the graph. To adjust the curve, either:

   □   Enter the appropriate parameters in the Sections area (for example, the **Ratios**, **Thresholds, Attack Time Constant,** and **Decay Time Constant** fields).

   □   Drag the section handles in the Compander I/O curve display area.

5. Select the required settings based on the options described in Table 2-6.

Table 2-6   Compander options

Option	Description
Presets	This drop-down list provides a collection of sample curves. Options include: ■ Default ■ Linear ■ Limit Hard -12 dB ■ Limit Hard -3 dB ■ 2:1 Compress -20 dB Hard Limit -6 dB ■ 2:1 Expand -30 dB 4:1 Compress -20 dB ■ Subwoofer Alter each preset option by entering appropriate parameters in the Sections area or by dragging the section handles in the Compander I/O curve display area. When a curve is adjusted, click Add to add it to the Presets list.
Delete/Add	This control is inactive until either: ■ A curve is edited: The Add button appears, enabling the user to create a curve then enter a name in the Presets drop-down list ■ A curve that has been added to the Presets list is selected: The Delete button appears, enabling the user to delete the curve from the Presets drop-down list
MakeUp Gain	■ This field enables a digital gain stage that is applied after the Compander curve ■ The value entered is applied to the whole curve and is automatically constrained to avoid saturation of the digital full scale (DFS)
Bypass Volume Control Hard Limiter	■ Disables the Volume Control Hard Limiter (threshold = -1 dBFS, ratio = 20)
Bypass	■ Disables the Compander algorithm ■ Reduces the total MIPs and results in lower overall power consumption
Ratio	■ Enables you to adjust how much the output signal amplitude increases for an increase in input signal amplitude ■ The value entered into the ratio field determines how much the input signal amplitude must increase for the output signal amplitude to increase by 1 dB. ■ Ratios greater than one compress the output signal in the given region, while ratios less than one expand the output signal in the given region. For example: □ 4:1: Compresses the output signal since the output signal only increases by 1 dB for a 4 dB increase in the input signal □ 0.5:1: Expands the output signal since the output signal increases by 1 dB for a 0.5 dB increase in the input signal □ 1:1: Is linear because the output signal amplitude increases by 1 dB for a 1 dB increase in the input signal amplitude
Thresholds	Enables at what input amplitude level, in dB, the compression/expansion for that region should begin, to be determined
Atk TC	■ The first order averaging filter time constant used when the instantaneous gain signal is greater than the averaged gain signal Determines how quickly the gain signal reacts when the calculated gain increases relative to the average ■ Range is 0 msec to 5000 msec

Table 2-6   Compander options (cont.)

Option	Description
Dec TC	■ The first order averaging filter time constant used when the instantaneous gain signal is less than the average gain signal ■ Determines how quickly the gain signal reacts when the calculated gain decreases relative to the average ■ Range of this parameter is 0 msec to 5000 msec
OK	■ In Monitoring mode, click **OK** to write the Compander parameters to the DSP memory where the changes take immediate effect. The Compander Settings window closes ■ Changes made in the Monitoring mode only affect the DSP and PC memory, not the Persistent Store memory ■ OK is not enabled in the Static mode
Apply	■ In the Monitoring mode, click **Apply** to write the Compander parameters to the DSP memory where the changes take immediate effect ■ Changes made in the Monitoring mode only affect the DSP and PC memory, not the Persistent Store memory ■ Apply can write unlimited parameters to the DSP memory without closing the Compander Settings window ■ Apply is not available in the Static mode
Cancel	Closes the Compander Settings window without saving any of the latest changes
Reset	■ Restores a Compander curve from the last saved state, which is the state last saved by clicking **OK** ■ Updates the settings in the Sections area with the last saved settings

2.10   Post Mastering (dither)

The Music Manager application enables simple tuning of the Post Mastering algorithms.

NOTE   Dither is the only Post Mastering algorithm implemented in this version of the Sink application.

The Dither algorithm reduces the distortion caused by truncation resulting from quantization by adding a barely audible amount of random background noise to the signal.

2.10.1   Adjusting the Post Mastering settings

To adjust the Post Mastering settings:

1. From the **Music Manager** window, click the **Post Mastering** block. The **Post Mastering Settings** window opens.
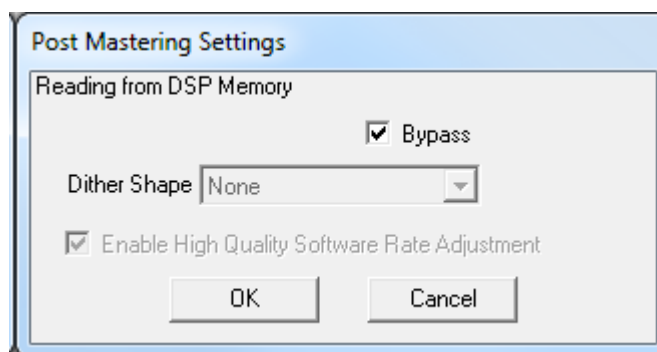
Figure 2-9　Post Mastering Settings window

2.　Select the required settings based on the options in Table 2-7.

Table 2-7　Dither options

Option	Description
Dither Shape	This drop-down list provides a collection of available dither configurations. Available options include: ■ None ■ Triangular ■ Shaped 　None effectively removes the dither. Each optional shape selects a different quantization technique. Select the technique that best suits your system.
Enable High Quality Software Rate Adjustment	Enables the use of higher definition coefficients for SBC, which improves frequency response (while increasing MIPS).

2.11　Main volume

The Music Manager application enables adjustment of the loudspeaker (DAC or gain) settings in monitoring mode. The left and right gains can be temporarily adjusted independently or linked together. To update and store the Volume Table, modify the default settings for Volume Table Size and Step Size. See Table 2-8 for Main Volume options.

2.11.1　Adjusting the volume control settings

To adjust the Main Volume settings:

1.　From the **Music Manager** window, click the **Main Volume** block. The **Main Volume Settings** window opens.
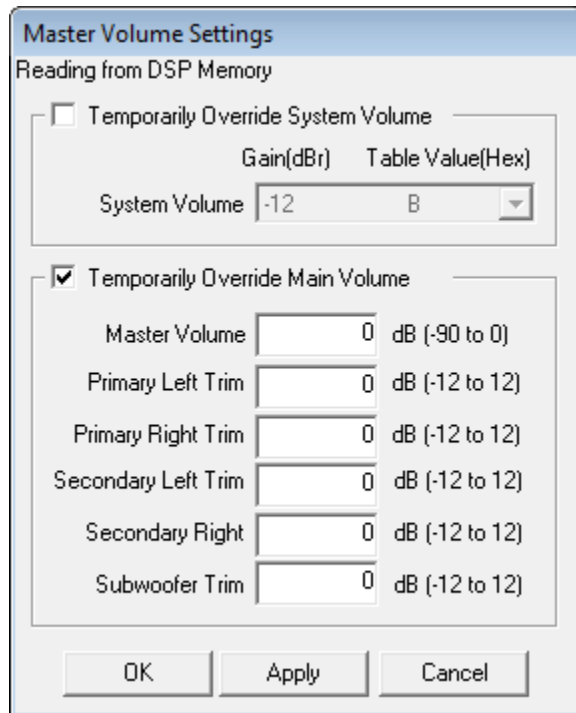
Figure 2-10   Main volume Settings window

2.   Select the required settings based on the options described in Table 2-8.

Table 2-8   Volume control options

Option	Description
Temporarily override System Volume	■ Enables the Gain Select field so a manual entry can be made for debugging purposes. Values are not retained. ■ A volume change can override the value in the Gain Select field ■ The check box must be selected for any volume or trim override to occur
System Volume	■ Enables you to temporarily set the DAC value if the DAC is used ■ Configures the left and right DAC settings with the value selected
Master Volume	Enables you to temporarily set the digital gain if digital gain is configured Applies the left and right volume settings with the value selected
Primary Left Trim	■ Selects the amount of gain/attenuation applied to the primary left channel when digital gain is used ■ The dB number indicates the Master Volume plus the dB trim value to be applied to the left channel
Primary Right Trim	■ Selects the amount of gain/attenuation applied to the primary right channel when digital gain is used ■ The dB number indicates the Master Volume plus the dB trim value to be applied to the secondary right channel
Secondary Left Trim	■ Selects the amount of gain/attenuation applied to the secondary left channel when digital gain is used ■ The dB number indicates the Master Volume plus the dB trim value to be applied to the secondary left channel

Table 2-8   Volume control options (cont.)

Option	Description
Secondary Right Trim	■ Selects the amount of gain/attenuation applied to the secondary right channel when digital gain is used ■ The dB number indicates the Master Volume plus the dB trim value to be applied to the secondary right channel
Subwoofer Trim	■ Selects the amount of gain/attenuation applied to subwoofer channel when digital gain is used ■ The dB number indicates the Master Volume plus the dB trim value to be applied to the subwoofer channel

2.12   Auxiliary Volume

The Music Manager application enables adjustment of the Aux Volume settings in monitoring mode. The left and right gains can be temporarily adjusted independently or linked together

When configured, the auxiliary output is a synchronous copy of the stereo input signal available for a variety of use cases. Other than volume control, no processing modules are available on the auxiliary signal chain

2.12.1   Adjusting the Aux Volume settings

To adjust the Main Volume settings:

1. From the **Music Manager** window, click the **Aux Volume** block. The **Aux Volume Settings** window opens.

Figure 2-11   Aux Volume settings window

2.   Select the required settings.

Table 2-9   Aux Volume settings

Option	Description
Temporarily override System Volume	■ Enables the Gain Select field so a manual entry can be made for debugging purposes. Values are not retained.A volume change can override the value in the Gain Select field. ■ The checkbox must be selected for any volume or trim override to occur
System Volume	■ Enables you to temporarily set the DAC value if the DAC is used ■ Configures the left and right DAC settings with the value selected
Master Volume	■ Enables you to temporarily set the digital gain if digital gain is configured. ■ Applies the left and right volume settings with the value selected

Table 2-9   Aux Volume settings (cont.)

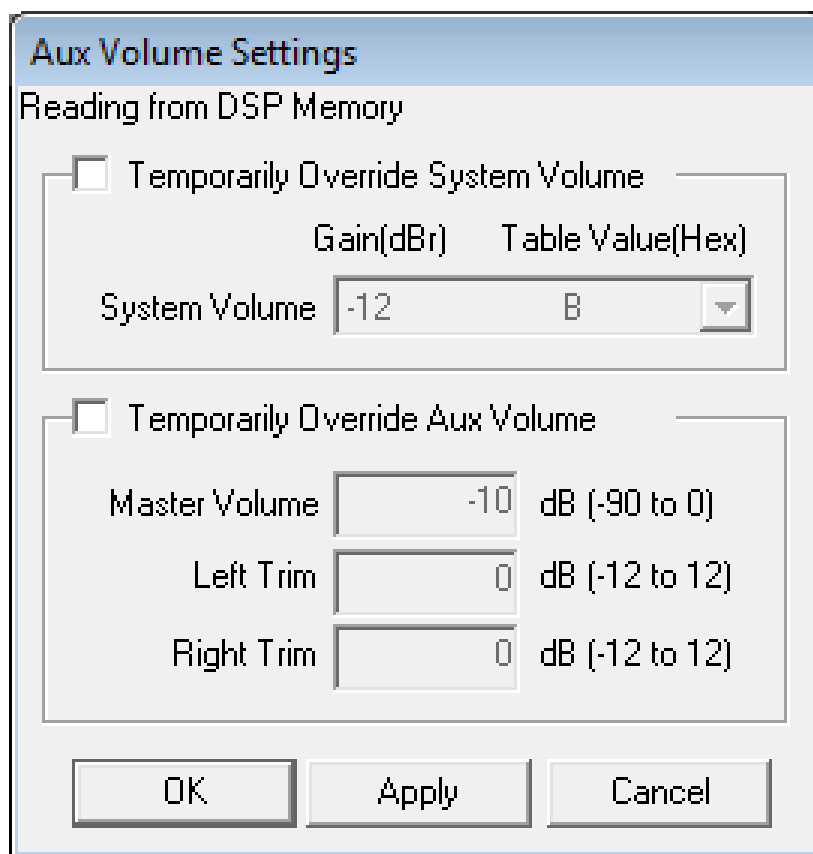| Option	Description
Left Trim	■ Selects the amount of gain/attenuation applied to the primary left channel when digital gain is used. ■ The dB number indicates the Master Volume plus the dB trim value to be applied to the left channel
Right Trim	■ Selects the amount of gain/attenuation applied to the primary right channel when digital gain is used. ■ The dB number indicates the Master Volume plus the dB trim value to be applied to the secondary right channel

2.13 Decoder selection matrix

The Decoder block provides a configuration matrix that defines which audio effects are applied to each codec type. When in Monitoring mode, the Decoder block automatically renames and changes the attached bitmap graphic to reflect the active Decoder, USB, or Wired mode.

2.13.1 Adjusting the Decoder selection matrix

To adjust the Decoder Selection matrix:

1. From the **Music Manager** window, click the **Decoder** block. The **CODEC Configuration Settings** window opens.
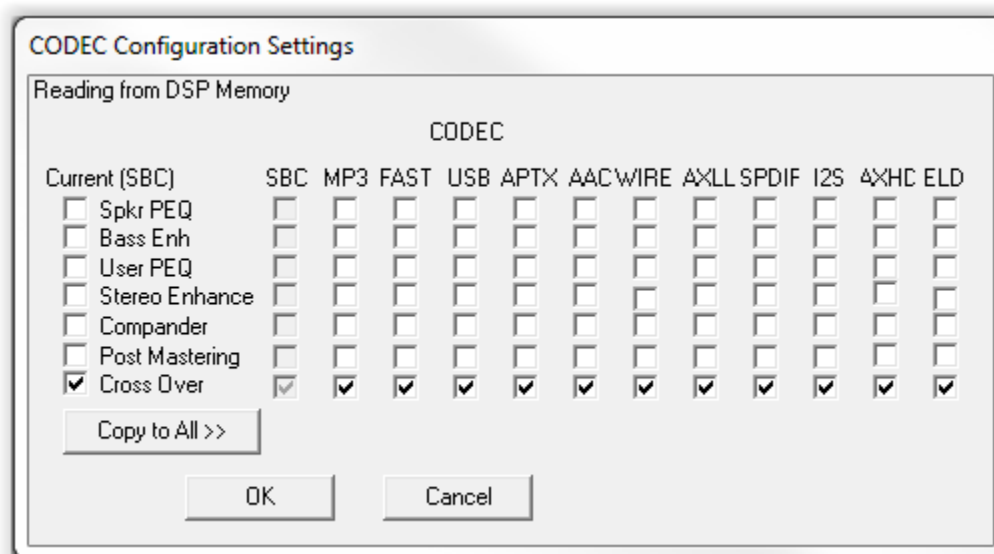
Figure 2-12   Codec Configuration Settings window

Table 2-10 lists the definitions for several abbreviations representing various codec options shown in the matrix.

Table 2-10　Codec options

Definition	Description
SBC	A2DP Streaming using the SBC decoder
MP3	A2DP Streaming using the MP3 decoder (not supported QCC300x)
FAST	A2DP Streaming using the low latency FastStream SBC decoder (not supported QCC300x)
USB	Streaming audio from a USB source
APTX	A2DP Streaming using the high quality APTX Classic Bluetooth decoder
AAC	A2DP Streaming using the AAC decoder
WIRE	Wired analog input source, digitized by the ADCs
AXLL	A2DP Streaming using the high-quality APTX-LL (low-latency) decoder
S/PDIF	Digital audio interconnection supporting PCM decoder format
I^{2}S	Digital audio interconnection supporting I^{2}S
APTX-HD	A2DP Streaming using the HD (24 Bit) APTX-HD Bluetooth decoder
AAC-ELD	A2DP streaming AAC Enhanced Low Delay

.

2. The CODEC options are all available in the Sink application using Flash ICs but not all are available for the CSR86xx family of ROM chips, which is chip-specific.

3. Select the required settings based on the options described in Table 2-11.

Table 2-11　Decoder options

Option	Description
Copy to All	Takes the current (far left set of check boxes) and applies those settings to all of the codec types
OK	Accepts all changes, applies the settings, and closes the window
Cancel	Cancels all changes and closes the window.

3 Sink DSP Application override modes of operation

The Sink DSP Application software supports these modes of temporary override operation:

- Stand-By
- Pass Through
- Full (processing)

The Music Manager application enables the mode to be changed in real time.

3.1 Manually selecting the mode of operation

1. From the Music Manager window, click the Mode box.
2. Select the appropriate mode from the drop-down list.
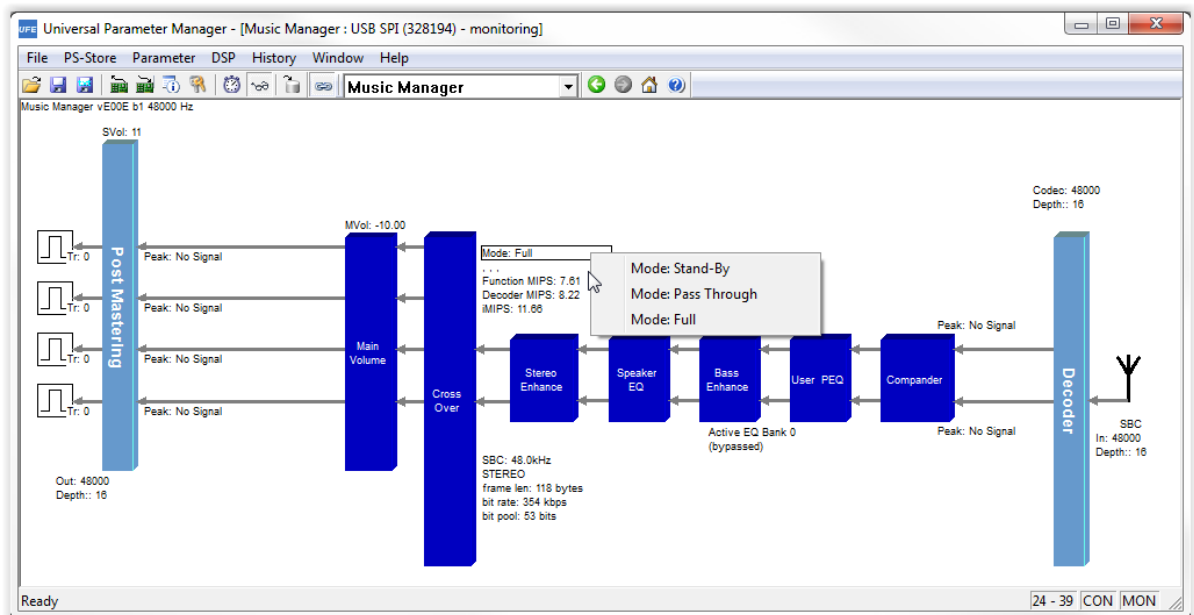

Figure 3-1   Manually selecting operation mode from Music Manager

3.2   Full processing mode

Full (processing) mode is the typical mode of operation. All the enhancement algorithms can be enabled. By default, all effects are set to bypass.
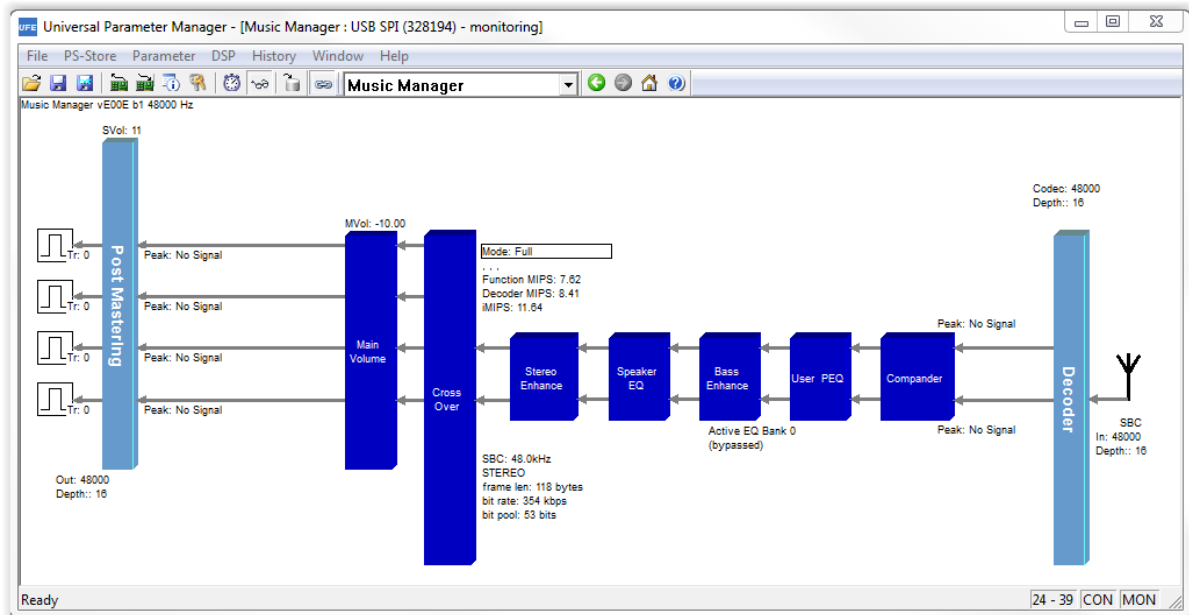

Figure 3-2   Music Manager in Full (processing) mode

3.3　Pass Through mode

In Pass Through mode, the decoded audio stream is routed directly to the outputs with no algorithms enabled. The software automatically switches to Pass Through mode if the sample frequency is not supported for audio enhancements.
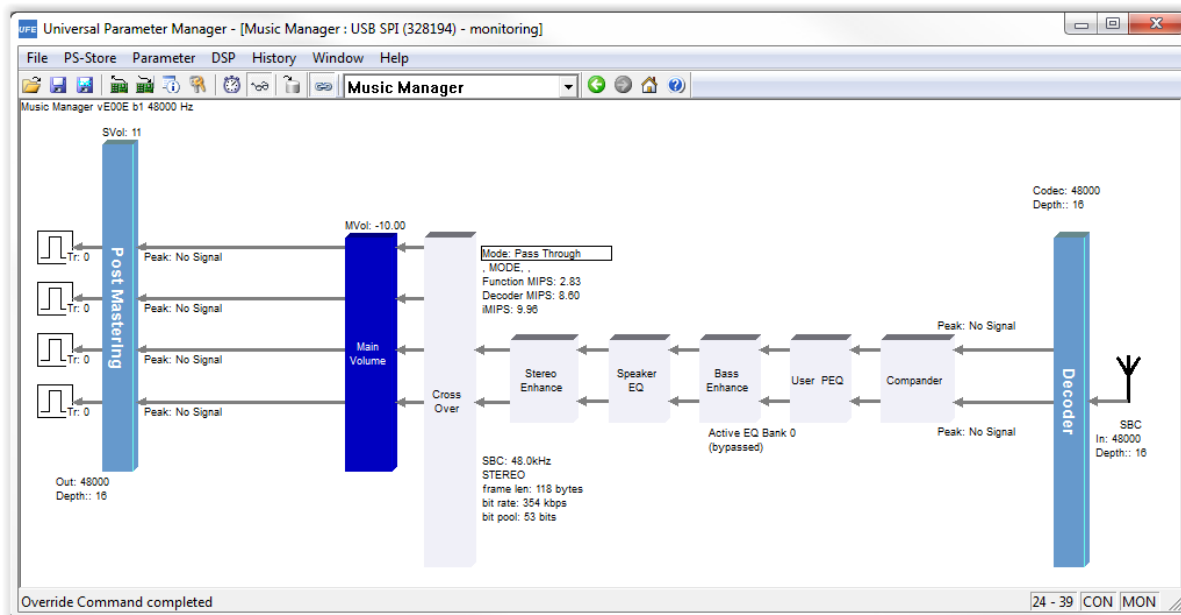
Figure 3-3　Music Manager in Pass Through mode

3.4   Standby mode

In Standby mode, the outputs are muted.

Figure 3-4   Music Manager in Standby mode

4   DSP Monitoring mode

The Music Manager application enables real-time adjustment and monitoring of DSP variables.
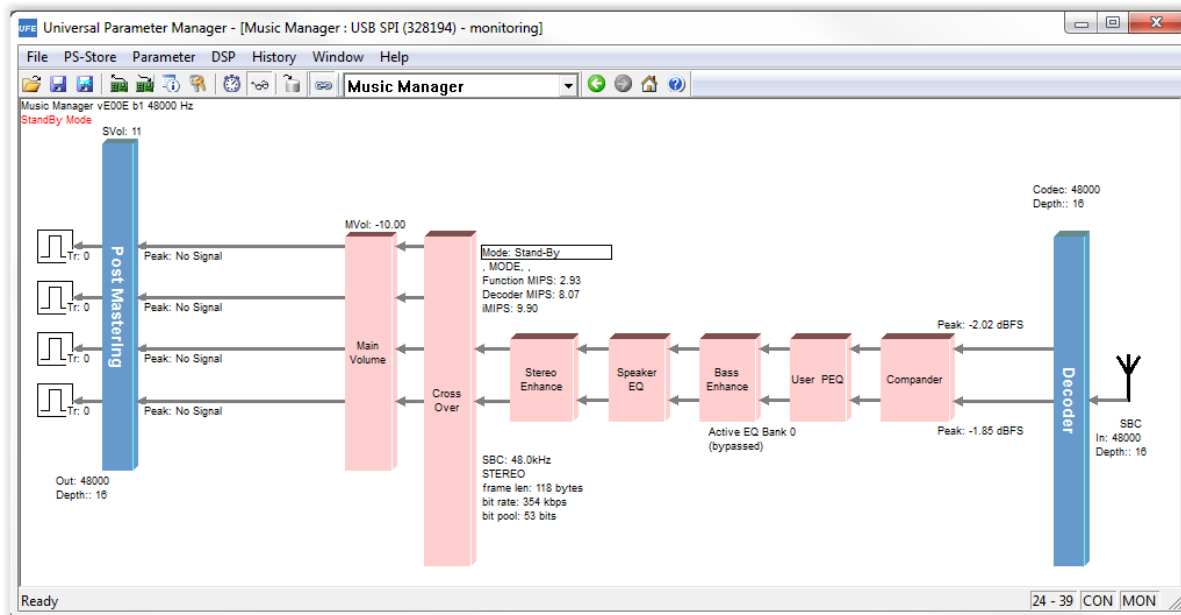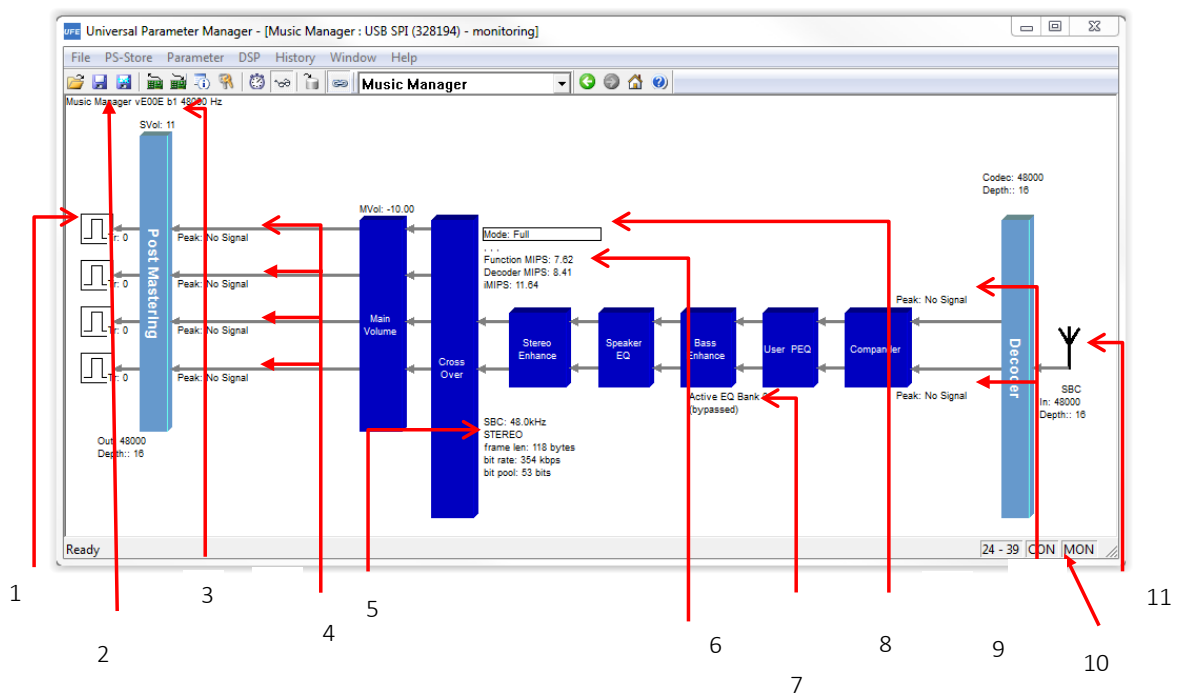

Figure 4-1   Music Manager main screen in Monitoring mode with MP3 dDecoder

The numbers correspond to:

1. Input interface identifier (Analog, I^{2}S, or S/PDIF) status.

2. Build Information indicates the Music Manager software version (E00F), build number (1) and sample rate (48 kHz).

3. Current volume displaying the Left Trim, System Master Volume, and Right Trim. If the user overrides this setting (for example, Volume Control), a green arrow displays on the Volume Control box.

4. Output peak level indicators in dBFS. Displays the left and right output levels after processing.

5. Subwoofer Peak Level indicates the peak level in dBFS of audio being sent to the subwoofer channel.

6. Decoder Parameters display the current parameters of the selected decoder. The values are specific to the selected decoder. The values shown display the parameters for the SBC decoder with the following parameters:

   □ Decoder and Sample Rate: SBC 48 kHz

   □ Channel Mode: STEREO

   □ Frame Length: 118 bytes

   □ Bit Rate: 354 kbps

   □ Bit Pool: 53 bits

7. LFE channel level indicator in dBFS.

8. MIPS display the MIPS required by the audio decoder algorithm (Decoder MIPs) and the MIPS required by the included functions (Function MIPS).

9. User EQ Status indicates the status of the User EQ. It displays the Active Bank of coefficients and whether or not the EQ is enabled or bypassed.

10. Mode Status indicates the current mode. If the user overrides the mode, MODE displays underneath the text box.

11. Input peak level indicators in dBFS. Displays the left and right input levels after the decoders.

12. Status Bar indicates the status of the UFE:

    □ 24-39: The parameters are stored in DSP_USER keys 24 to 39

    □ CON: Connected to the SPI and active □ MON: The UFE is in Monitoring Mode

13. Input Bitmap identifies the input source type when music is streamed. Options include:

    □ Bluetooth Antenna (shown)

    □ USB

    □ Wired connector

    □ I^{2}S

    □ S/PDIF

14. When configured, UFE displays the Stereo Aux Volume block and statistics displayed in dBFS.

5   Exiting the Music Manager application

To exit the Music Manager application, from the menu bar, select Exit from the File menu.

6   Connection source options

The information in this document is based on the music source being Bluetooth Advanced Audio Distribution Profile (A2DP) Decoders. With Music Manager vE00E, four extra source types are supported:

■ Bluetooth A2DP

■ USB

■ Wired Analog

■ I^{2}S

■ S/PDIF

6.1   USB

USB is supported as an optional source input for playing music. The Sink application must be correctly configured to support USB. When configured, a USB cable must be connected from the product to a suitable USB music source device such as a personal computer. The USB enumerates as a USB sound device and the music can be directed to the product for playback. If configured correctly, the USB cable provides DC power that can enable battery recharging while streaming music playback.

When a USB source is streaming music, the Music Manager changes to indicate this.
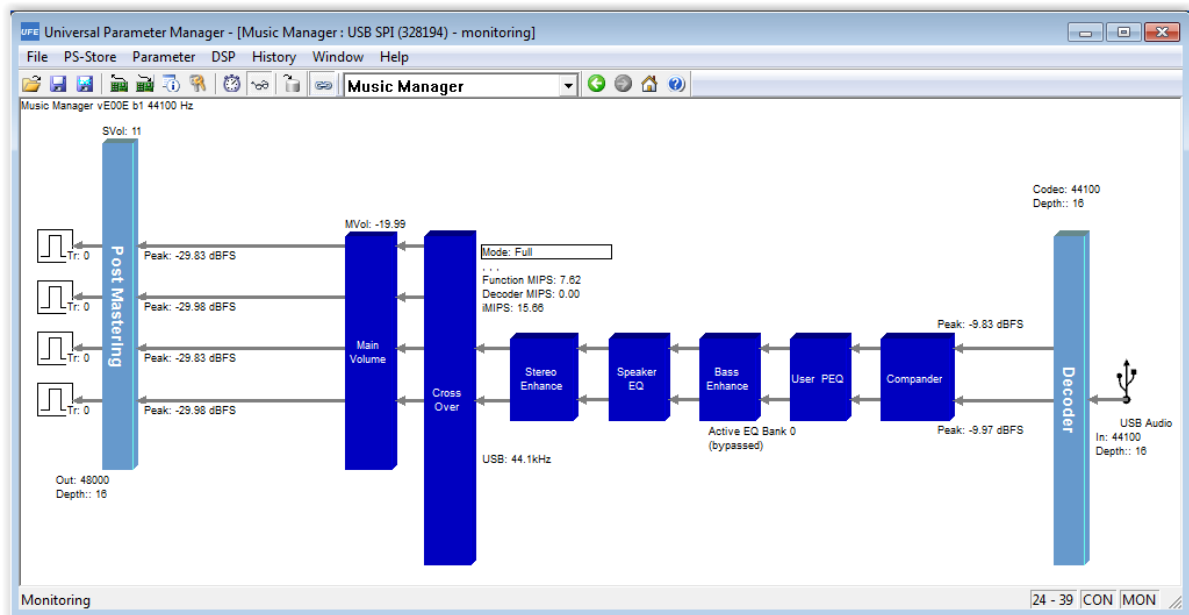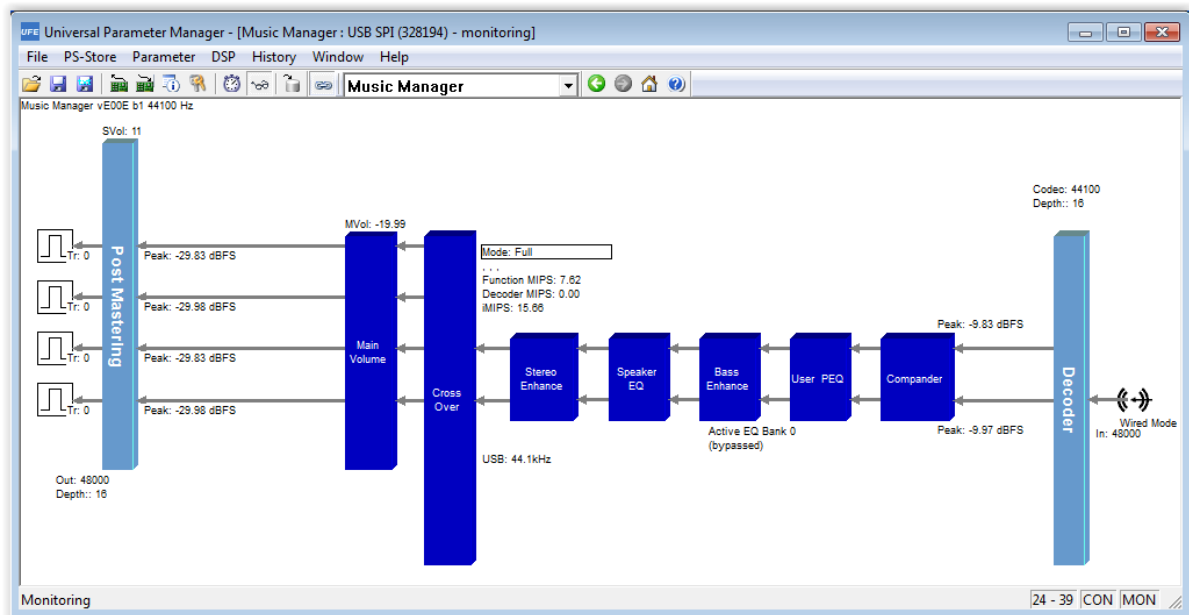

Figure 6-1   USB source

The extreme right side decoder block is renamed to USB Audio. The Receive In bitmap changes to a USB symbol. The statistic block reflects the USB sample rate.

6.2   Wired analog source input

Wired analog is supported as an optional source input for playing music. The Sink application must be correctly configured to support the wired source. When configured, a stereo cable must be connected from the product to a suitable analog music source. The wired inputs are routed to the Qualcomm chip ADC inputs, where the music is digitised. The Music Manager applies the audio effects to the digital stream. The audio is then routed to the output interface (DAC or I^2S) and onto the products speakers for playback.

When a wired source is streaming the music, the Music Manager changes to indicate this.
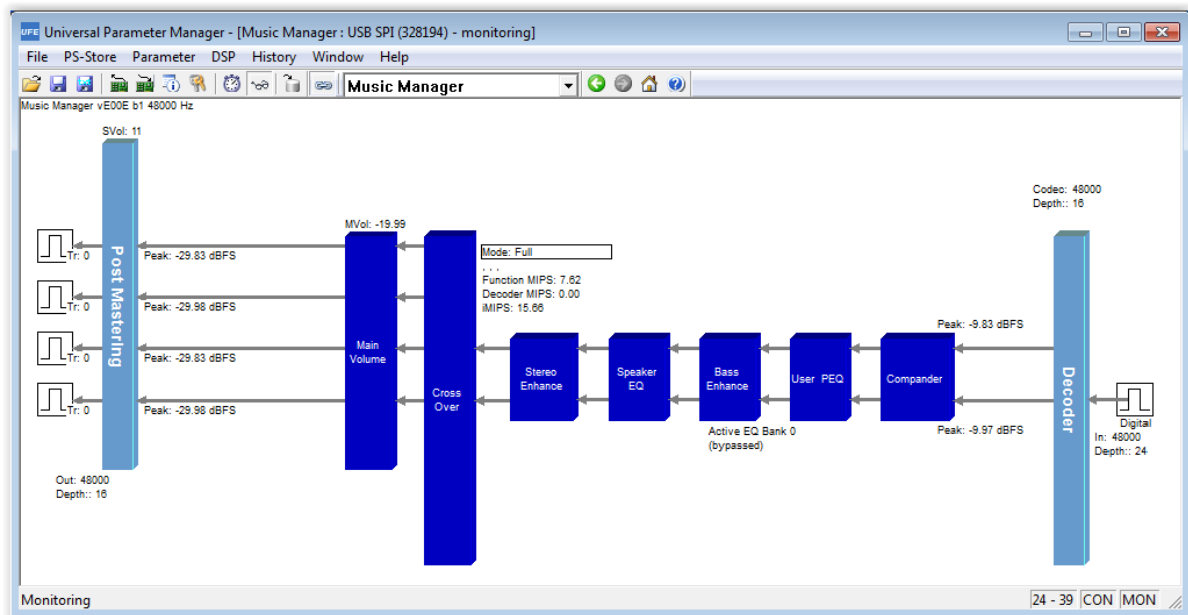

Figure 6-2   Wired analog source

The extreme right side decoder block is renamed to Wired mode and the Receive In bitmap changes to a 3.5 mm stereo jack symbol. The statistic block reflects the ADC sample rate.

6.3   I^{2}S source input

I^{2}S is supported as an optional source input for playing music. The Sink application must be correctly configured to support the I^{2}S source. The digital I^{2}S connections must be routed to the QTIL chip as configured. The Music Manager applies the audio effects to the digital stream. The audio is then routed to the output interface (DAC or I^{2}S) and onto the products speakers for playback.

When an I²S source is streaming the music, the Music Manager changes to indicate this.

Figure 6-3   I²S source

The extreme right side decoder block is renamed to Wired mode and the Receive In bitmap changes to indicate I²S connection. The statistic block reflects the sample rate and bit depth.

6.4   SPDIF source input

SPDIF is supported as an optional source input for playing music. The Sink application must be correctly configured to support the SPDIF source. The digital SPDIF connection must be routed to the QTIL chip as configured. The Music Manager applies the audio effects to the digital stream. The audio is then routed to the output interface (DAC or I²S) and onto the products speakers for playback.

When an SPDIF source is streaming the music, the Music Manager changes to indicate this.
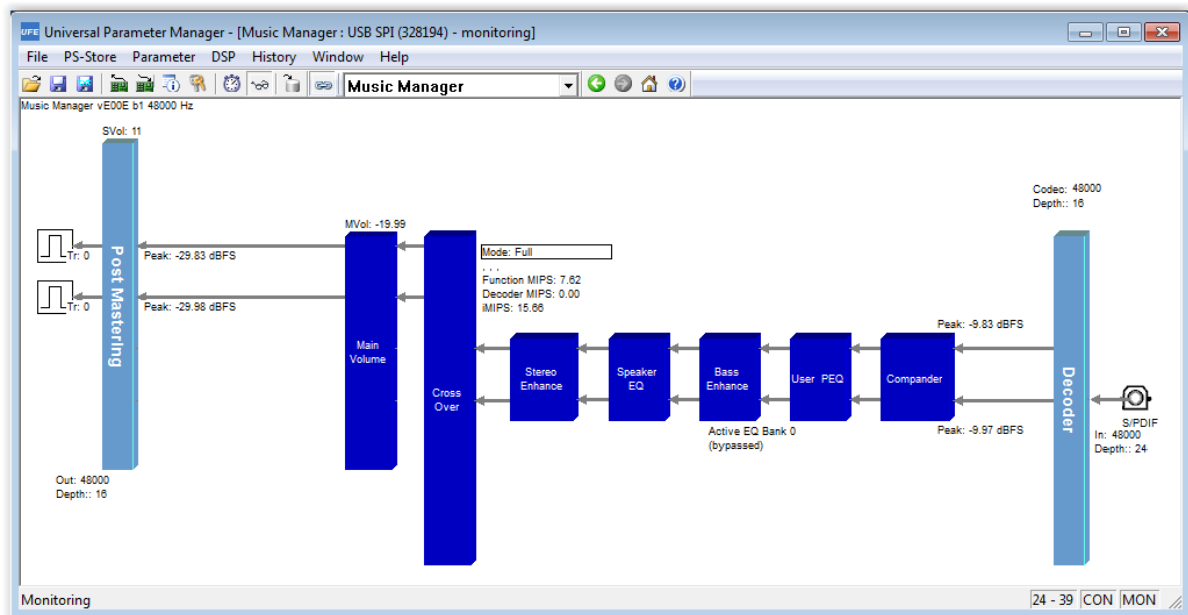
Figure 6-4   SPDIF source

The extreme right side decoder block is renamed to SPDIF and the Receive In bitmap changes symbol. The statistic block reflects the sample rate and bit depth.

Document references

Document	Reference
BCSW-CVC-HS-4-9-1 1M-HS Tuning Guide	80-CT419-1 /CS-00309814-UG
BCSW-CVC-HS-4-9-1 1M-HS Parameter Manager User Guide	80-CT420-1 /CS-00309816
BCSW-CVC-HS-6-0-1 2M-HS Tuning Guide	80-CT441-1/CS-309817-UG
BCSW-CVC-HS-6-0-1 2M-HS Parameter Manager User Guide	80-CT442-1/CS-00309818-UG
BCSW-CVC-HF-5-0-1 2M-HF Tuning Guide	0-CT412-1/CS-00309822-UG
BCSW-CVC-HF-5-0-1 2M-HF Parameter Manager User Guide	CS-00309823-UG
Wireless Subwoofer Audio Parameters E201	80-CT444-1/CS-309826-UG
BCSW-CVC-HF-4-8-1 1M-HF Tuning Guide	CS-329124-UG
BCSW-CVC-HF-4-8-1 1M-HF Parameter Manager User Guide	CS-329123-UG
ADK Audio Sink Configuration Tool User Guide	80-CT426-1 /CS-00309942-UG
User EQ	80-CT445-1 /CS-00309844-DC

Terms and definitions

Term	Definition
A2DP	Advanced Audio Distribution Profile
AAC	Advanced Audio Coding
ADC	Analog-to-digital Converter
AGC	Automatic Gain Control
ADK	Application Development Kit
Bluecore	Group term for QTIL's range of Bluetooth wireless technology chips.
Bluetooth	Set of technologies providing audio and data transfer over short-range radio connections.
Bluetooth SIG	Bluetooth Special Interest Group
Codec	Coder Decoder
cVc	Clear Voice Capture
DAC	Digital-to-analog Converter
DFS	Digital Full Scale
DSP	Digital Signal Processor
EQ	Equalizer
HPF	High Pass Filter
I^{2}S	Inter-Integrated Circuit Sound
I/O	Input/Output
IC	Integrated Circuit
LFE	Low-Frequency Effects
LPF	Low Pass Filter
MIPS	Million Instructions Per Second
NSD	Noise Shaped Dither
PC	Personal Computer
PCM	Pulse Code Modulation
PEQ	PEQ Parametric Equalization
PS	Persistent Store
SBC	Sub-band Coding
SCO	Synchronous Connection-Oriented
SNR	Signal to Noise Ratio

Term	Definition
SPI	Serial Peripheral Interface
SPDIF	Sony/Philips Digital Interface Format
UFE	Universal Front End
USB	Universal Serial Bus VM
VM	Virtual Machine