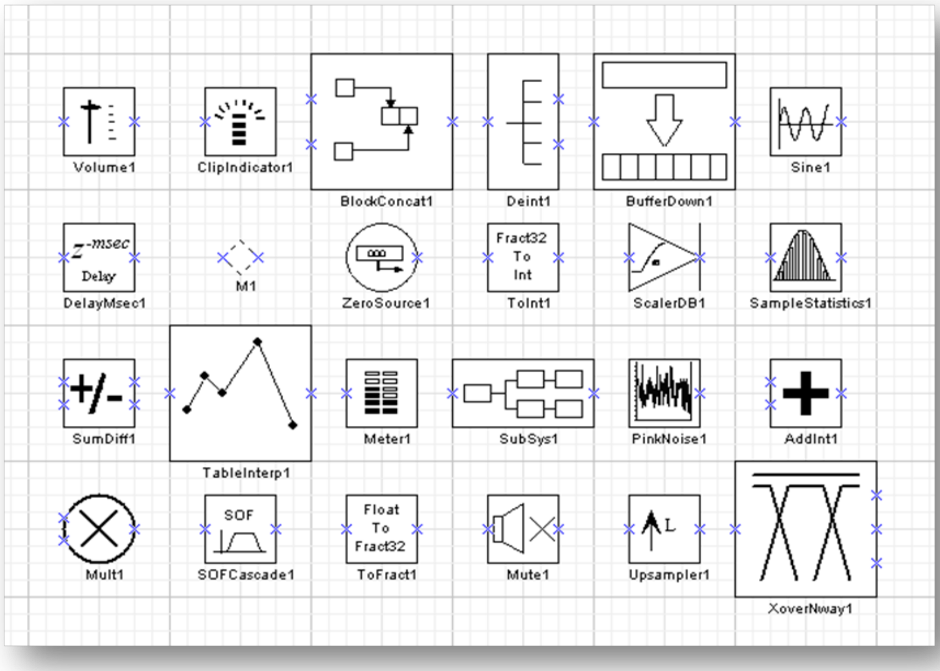
DSP CONCEPTS

AWE Core™ Integration Guide



Change Log

Version	Date	Description	Author
01	2016.Oct.11	Updated for AWE v5 functionality	MW
02	2016.Oct.28	Reorganization	MW
03	2016.Nov.1	Expanded Theory of Operation / Control Model	MW
04	2016.Nov.8	Added Figure cross-references and simplified control interface.	MW
05	2016.Dec.20	Fixed typos	AN
06	2017.Mar.24	Revised for AWE v6 functionality	CHP
07	2017.Mar.27	Added some doc-navigation, corrected text, updated naming conventions	MW
08	2017.Mar.29	Corrected prototype for **awe_fwPacketExecuteArray()**	MW
09	2017.Aug.10	Updated Control Interface section with proper source structure fields	MM
10	2017.Dec.17	Added detailed information on BSP prerequisities	PB
11	2018.Jan.3	Added missing AWE-6 changes	CHP
12	2018.Feb.11	Fixed typos and added missing details to audio processing code example	CHP
13	2018.Jun.21	Added rev to footer	MW

Table of Contents

Table of Contents..4

Table of Figures..5

1. Introduction...7

2. Theory of Operation............................7

  2.1. Data Interfaces.............................7

  2.2. Dynamic Instantiation..................8

  2.3. Tuning Model..............................9

  2.4. Control Model............................14

  2.5. Audio-Processing Model...........19

  2.6. Threading / Priority Model........22

3. Tuning Interface................................24

  3.1. Overview...................................24

  3.2. Integration Quick Guide............24

  3.3. Tuning API................................25

  3.4. Message Structure......................26

  3.5. RS-232 Protocol.........................26

  3.6. SPI Protocol...............................28

  3.7. USB Protocol.............................29

  3.8. Error Handling...........................29

4. Audio Interface..................................30

  4.1. Overview....................................30

  4.2. Audio Processing.......................31

  4.3. Audio API.................................32

4.4. Detailed Integration Steps..........33

5. Control Interface................................37

5.1. Overview...................................37

5.1. ObjectID allocation....................38

5.2. Control API................................38

5.3. Detailed Integration Steps..........39

6. Global Variables and Data Structures42

6.1. Memory Heaps...........................43

6.2. Module Table.............................43

6.3. Optimizing Memory Usage.......44

6.4. Target Info.................................45

6.5. Communication Buffer.............45

6.6. Memory Sections.......................46

7. Framework Prerequisites..................47

7.1. Basic Processor and Board Initialization 47

7.2. Simple "Audio Passthru" Application 47

7.3. Simple "Echo-Back" Application48

7.4. Basic Interrupts and Threading Configured 49

7.5. Optional: Flash memory............49

Table of Figures

Figure 1. The three data interfaces to the AWE Core....................7

Figure 2. Example commands that dynamically create a signal processing Layout at runtime......8

Figure 3. Standalone operation from compiled-in AWB....................11

Figure 4. InitCommands array generated by AWE Designer....................12

Figure 5. AWB stored in SPI FLASH....................12

Figure 6. External micro controller managing FLASH contents....................13

Figure 7. Example SOC implementation of Tuning interface....................14

Figure 8. Simple example of Control data in a layout....................15

Figure 9. Multi-input control convention....................16

Figure 10. Diagram of static DMA-blocksize versus 'tunable' processing-blocksize....................20

Figure 11. Channel-count is in the Layout is independent of the hardware....................21

Figure 12. An audio design with multiple block sizes....................23

Figure 13. Processor activity when audio processing occurs in multiple threads....................24

Figure 14. Tuning Command and Reply Format....................26

Figure 15. Tuning protocol CRC Calculation....................26

Figure 16. Block based audio processing using double buffered DMA....................31

Figure 17. Mapping ISR to interrupt on SHARC....................32

Figure 18. Simple system showing Source and Sink models as Layout Control I/O interfaces.. .36

Figure 19. Assign a static ObjectID to a control interface module....................37

Figure 20. Source and Sink Modules....................39

Figure 21. Example code that writes control data into the Layout....................40

Figure 22. Module Table....................41

Figure 23. List of Module Classes....................41

Figure 24. Module Classes used in a layout...42

1. Introduction

The AWE Core™ is an embedded, dynamically-configured audio-processing engine. This document describes the theory of operation and how to integrate this software-engine into a products' embedded software. The intended audience is embedded software engineers familiar with real-time embedded-audio systems.

2. Theory of Operation

2.1. Data Interfaces

The AWE Core™ is a library that's integrated into an embedded application. It is context-agnostic, so it can be used with an embedded RTOS or a simple priority-based (i.e. "bare metal" / "super loop") scheduling scheme. In either context, the processor's software interacts with the AWE Core by exchanging three types of data; Audio, Tuning, and Control.

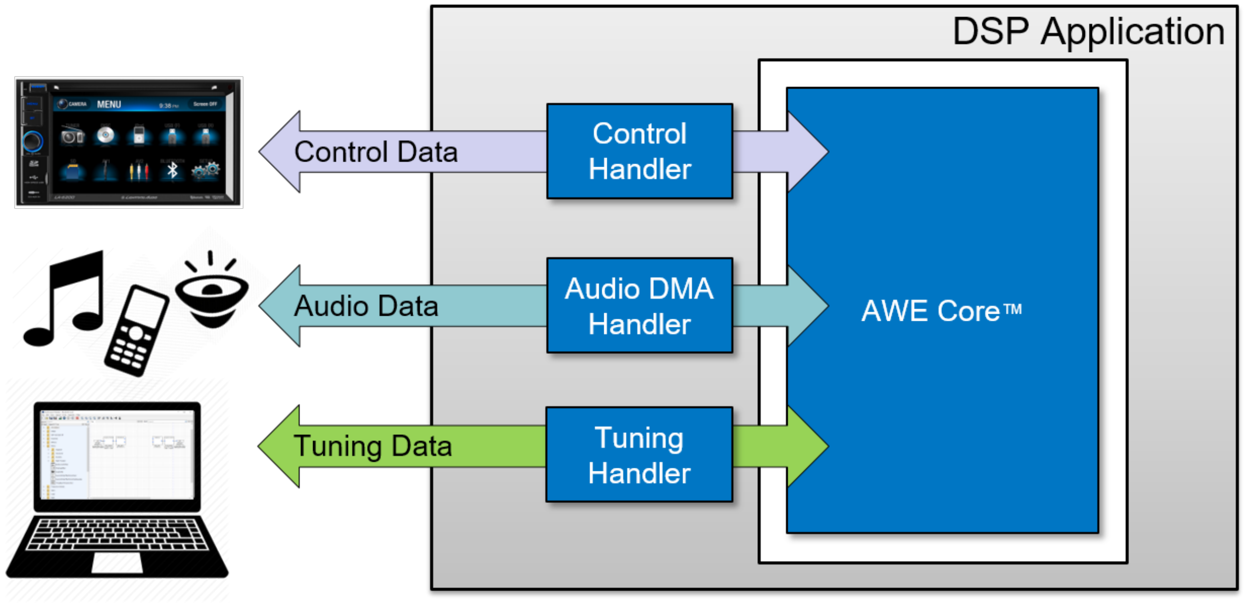


Figure 1. The three data interfaces to the AWE Core

Audio data is processed in blocks of 32 samples, 64 samples, or some other fixed size. The Audio Driver, part of the hardware-specific firmware, typically exchanges this data with the rest of the system using DMA from a peripheral or shared-memory buffers.

Tuning data is what dynamically configures the signal-processing, including the topology of the modules and the tuning parameters for each, such as filter coefficients or channel gain. This data is in the form of commands, typically issued by Audio Weaver Designer™ (or 3rd party tools / scripts). The Audio Weaver Server manages the tuning interface on the PC side and supports the following transport mechanisms:

- USB
- RS-232
- SPI
- Ethernet

The Tuning Driver, developed by the integrator, may control a peripheral, such as a UART or USB port, or it may receive tuning commands via sockets, shared memory, or some other IPC mechanism. The Tuning Driver exchanges commands and replies between the AWE Core using one of the tuning transports listed above.

Control data typically represents HMI (human-machine interface) settings, such as volume, balance, or system-state, such as battery-level. Being a bidirectional interface, data calculated in the AWE Core (e.g. RMS level, event-detection flags, or a direction-of-arrival estimate) may also be retrieved from the control interface.

2.2. Dynamic Instantiation

The AWE Core is a data-driven engine. Immediately after booting, it is inert with no pre-defined signal-processing path. Instead, the signal-processing, which we refer to as a '*Layout*', is constructed at runtime using a set of "*tuning*" commands. Commands add modules to the active layout, connect the wires between modules, write and read parameters, and start/stop audio processing; **a signal processing Layout is completely defined by a set of tuning commands**. We refer to this process as *dynamic instantiation.*

```
…
bind_wire,wire1,Input
bind_wire,wire2,Output
create_module,SYS_toFloat,ModuleFract32ToFloat,1,1,0,wire1,wire3
create_module,DCRemoval,ModuleButterworthFilter,1,1,0,wire3,wire4,1,1,10.0
write_float_array,DCRemoval.filt.coeffs[0],0.999345958,-0.999345958,0.0,-0.998691857,0.0
create_module,SourceEQ,ModuleSecondOrderFilterSmoothedCascade,1,1,0,wire4,wire4,2,10.0,1
write_float_array,SourceEQ.freq[0],250.0,250.0
…
```

Figure 2. Example commands that dynamically create a signal processing Layout at runtime

During product-development, tuning commands are automatically generated by Audio Weaver Designer™. (Matlab, 3rd party tools, or scripts [Python, etc] may also generate these tuning commands.

For a product to run stand-alone, the complete set of commands that defines a Layout (both in topology and tuning) is exported by AWE Designer as an "Audio Weaver Binary" (.AWB) file. This AWB data is then stored in a products' non-volatile memory and loaded (i.e. executed) during post-boot init.

Note: To facilitate dynamic instantiation, the AWE Core has its own memory manager, with three dedicated memory heaps: Fast, FastB, and Slow. As described in Section 6.1, the system developer is expected to place these three heaps in appropriate memory.

2.3. Tuning Model

A signal-processing Layout is instantiated in the AWE core through a set of Tuning commands. Fom the perspective of the integrator and system designer, there are two scenarios to understand - based on the origin of the tuning commands.

First, while a new product is being developed, DSP Engineers, Acoustic Engineers, and System Engineers will use PC-based tools like AWE Designer to create and refine the signal processing Layout. During this phase, commands are sent to the AWE Core from the PC in realtime.

Later, when developers are done tweaking and tuning a product's signal-processing Layout, it must be stored in a product's NVRAM so the product can operate standalone. The method of storing and loading a Layout from NVRAM will vary from product to product and is an engineering exercise for the integrator (though Reference Integrations of various forms are available for free from DSP Concepts).

Design-Time Operation

When using AWE Designer to create and tune a signal processing Layout, tuning commands (and the resulting replies) are exchanged between the PC and the AWE Core. The full path that these tuning commands travel is shown below in Figure .

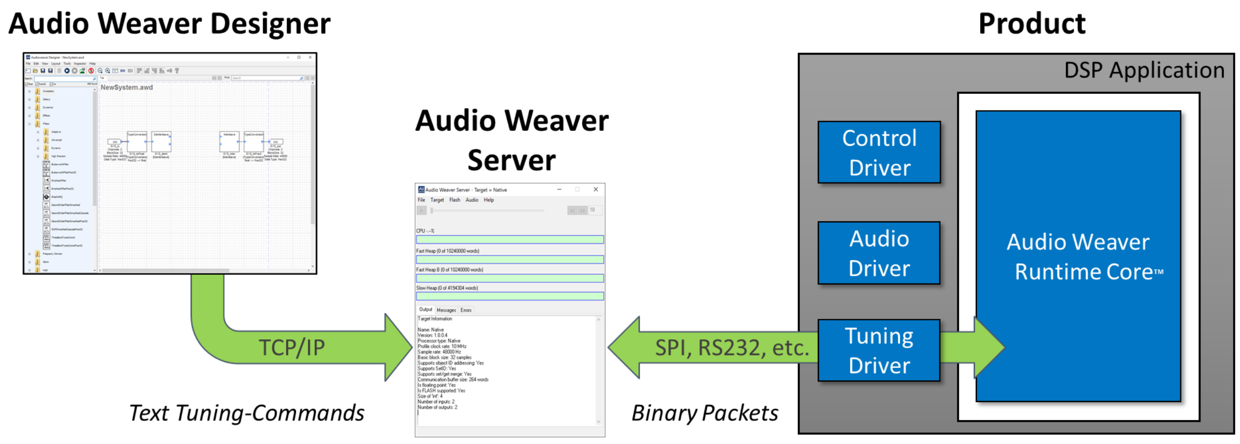


Figure 2. Overview of Design-time Tuning path

To communicate with the embedded AWE Core, AWE Designer (or a 3rd party tuning application / script) first connects to a PC application called the *Audio Weaver Server* [using TCP/IP]. The typical scenario is that both AWE Designer and AWE Server are running locally on the same computer using "localhost". Communication occurs using human-readable tuning commands. Scripts containing these commands are called Audio Weaver Scripts (AWS). AWE Server then converts each text command into a binary packet and sends them out over the specified tuning transport (e.g. Ethernet, USB, RS232, etc.) to the target platform. Scripts containing these binary commands are called Audio Weaver Binary (AWB) scripts.

Note: Figure 2 above shows some example AWS commands. The full list of commands and their arguments is detailed in the *Audio Weaver Server Command Syntax* document

In the Application software on the target, a Tuning Driver must be developed to handle the chosen tuning transport/peripheral. This tuning driver may respond to DMA interrupts from an on-chip peripheral, such as UART or SPI, or it may process packets received via USB, network, or shared-memory. The Tuning Driver passes the received tuning data to the AWE Core and sends replies back to the PC when they are ready. The Tuning Driver is simply a data-handler, exchanging messages between the tuning transport and the AWE Core. It may have to run at a fairly high priority in order not to miss communication events but the actual command handling must be done in a deferred procedure call at a priority lower than the signal processing. Note: this command protocol is driven by AWE Server. The server issues a command and will not issue another command until the original command is complete or times out. There are never unsolicited commands or replies.

The API that the Tuning Driver uses to interact with AWE Core is described below in section 3.

Standalone Operation

Once configuration and tuning is complete, a Layout can be saved on the product to allow stand alone operation. To do this, the Layout is exported as an Audio Weaver Binary (.AWB) file. (In AWE Designer, select *Tools / Generate Target Files*). AWB's may be stored in-product in two ways: either 'compiled-in', where the AWB is embedded as static data in the processor's executable, or as a separate data-object stored in NVRAM.

2.3.2.1. Compiled-in AWB

The simplest way to store and load an AWB is to initialize a C-array with the contents of the AWB. The array is then loaded (i.e. executed) immediately after booting, and audio processing starts immediately with no external activity. When integrating the AWE Core into an application, compiling-in the AWB is the recommended first step towards standalone operation. This is pictured below in Figure 3.

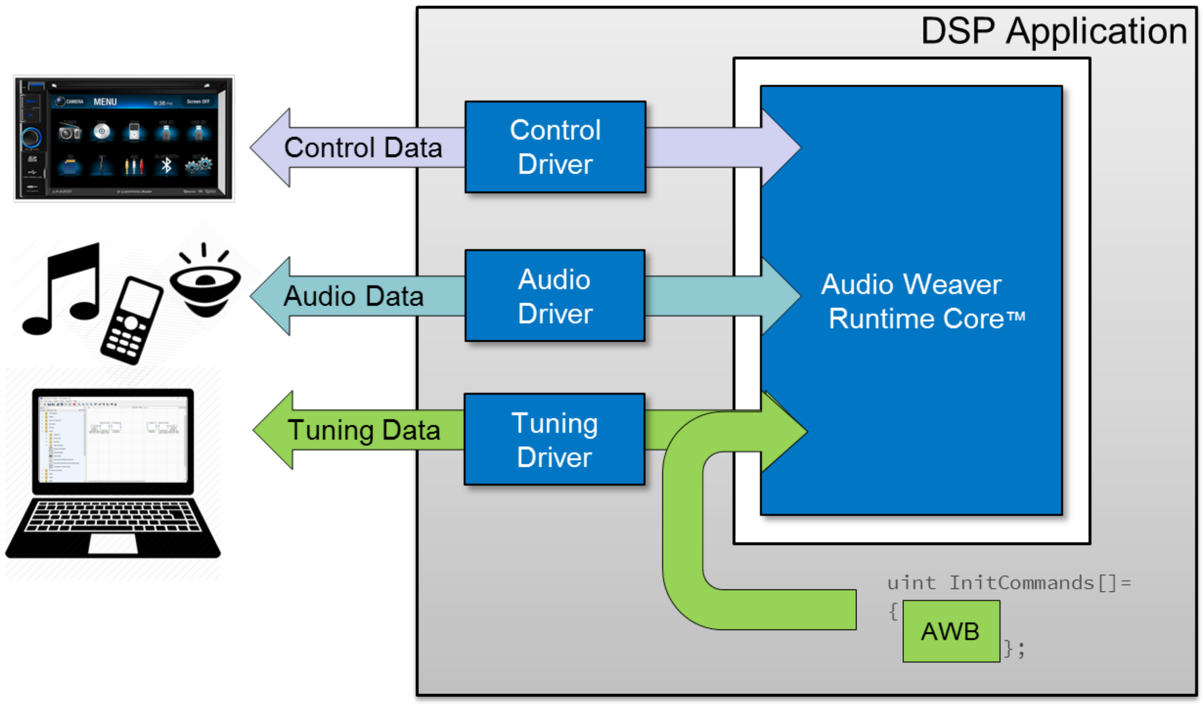


Figure 3. Standalone operation from compiled-in AWB.

Audio Weaver Designer will automatically generate this pre-initialized array in a .C file by selecting *Tools -> Generate Target Files*. An example of the resulting, generated source is shown below in Figure 4. It shows the AWB data, stored as hex values. Details of the command structure can be found below in section 3.4.

```
unsigned int InitCommands[] = {
0x00030037,      /* PFID_EnableAddressTranslation */
0x00000001,      /*           1 [enable] */
0x0002001c,      /* PFID_StopAudio */
0x0002000c,      /* PFID_Destroy */
0x00070003,      /* PFID_ClassPin_Constructor */
0x00000300,      /*         768 [blockSize] */
0x00000002,      /*           2 [numChannels] */
0x00000004,      /*           4 [sizeOfSamples] */
0x0000bb80,      /*       48000 [sampleRate] */
0x00000000,      /*           0 [isComplex] */
0x000a000f,      /* PFID_ClassModule_Constructor */
0xbeef0bf6,      /*        1014 [classID = ButterworthFilter] */
0x00000101,      /*  0x00000101 [nIO.  numIn=1. numOut=1. numScratch=0] */
0x00000003,      /*           3 [argCount] */
0x00048000,      /*  0x00048000 [wire 0] */
0x00050000,      /*  0x00050000 [wire 1] */
0x00000001,      /*  0x00000001 [arg 0] : filterType = 1 */
0x00000001,      /*  0x00000001 [arg 1] : order = 1 */
0x41200000,      /*  0x41200000 [arg 2] : cutoff = 10.0 */
0x000c004c,      /* PFID_P_SetValuesSetCall */
0x00118009,      /*  0x00118009 [Address] */
0x00000000,      /*           0 [offset] */
0x00118000,      /*    0x118000 [pModule] */
```

Figure 4. InitCommands array generated by AWE Designer

To load the Layout this AWB file represents, one simply calls the following function during post-boot init.

```
INT32 awe_fwPacketExecuteArray(AWEInstance *pAWE, UINT32 *array, UINT32 arraySize);
```

2.3.2.2. NVRAM-stored AWB

The other option for storing generated AWB files is to save them as unique objects elsewhere in your embedded system outside the Application executable. For example, they may be stored in a local SPI-FLASH, directly controlled by the audio-processor. In this case, code must be added to the application to manage data in the FLASH. When reading from FLASH, AWB data may be fetched one tuning command at a time and sent to the AWE Core using it's tuning API.

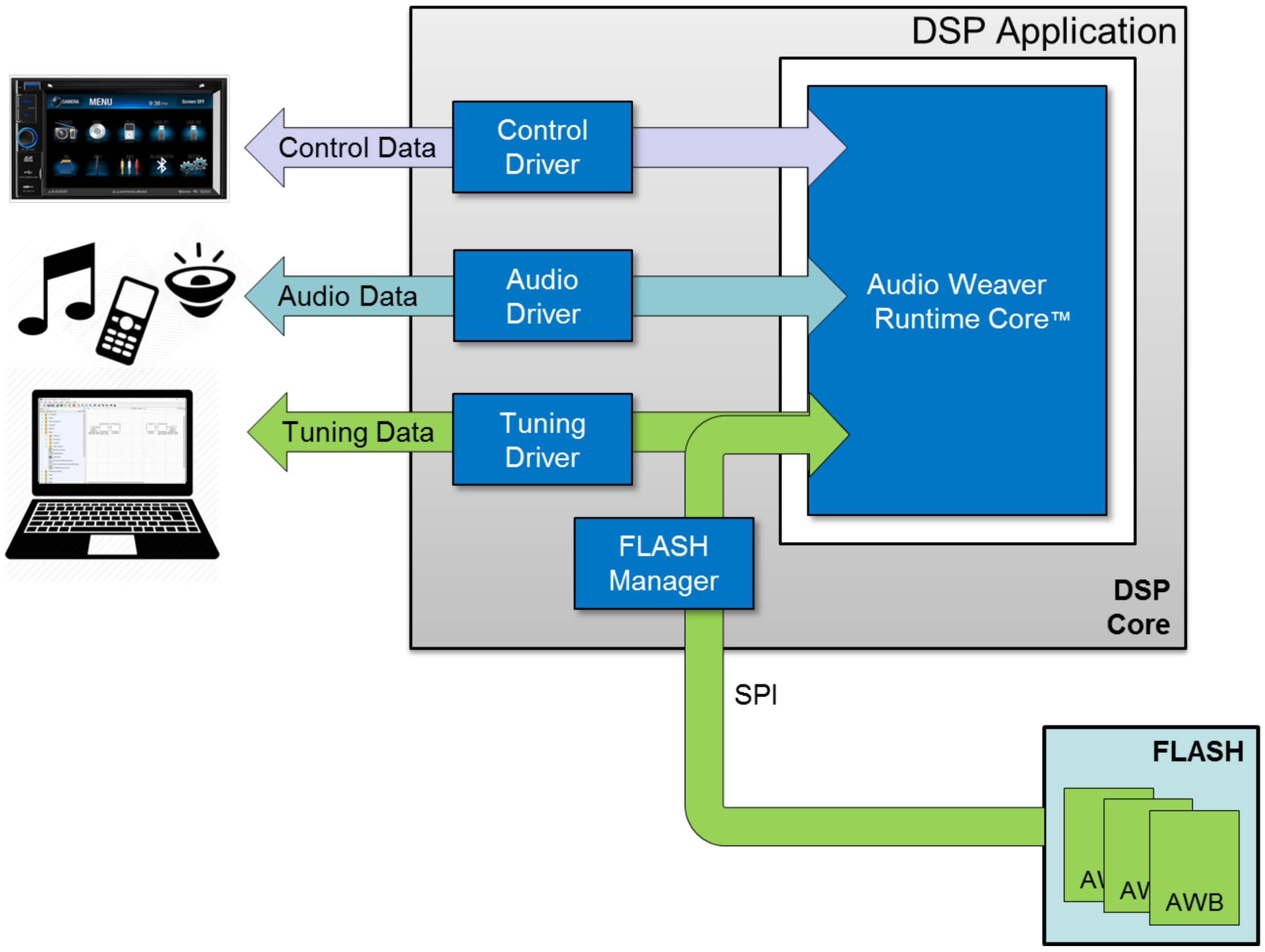


Figure 5. AWB stored in SPI FLASH

In other systems, FLASH will not be directly managed by the processor hosting the AWE Core. In this case, the software "plumbing" for the AWB packets must be developed to accommodate the unique architecture of the specific system. What follows are some typical architectures used in more complex systems.

In the example in Figure 6, a separate micro controller manages the FLASH device. In this case, the micro fetches AWB data from the FLASH and passes it along to the DSP via SPI. Here an IPC driver must handle the communication between the micro and the DSP. Tuning packets are passed over the IPC as a payload.

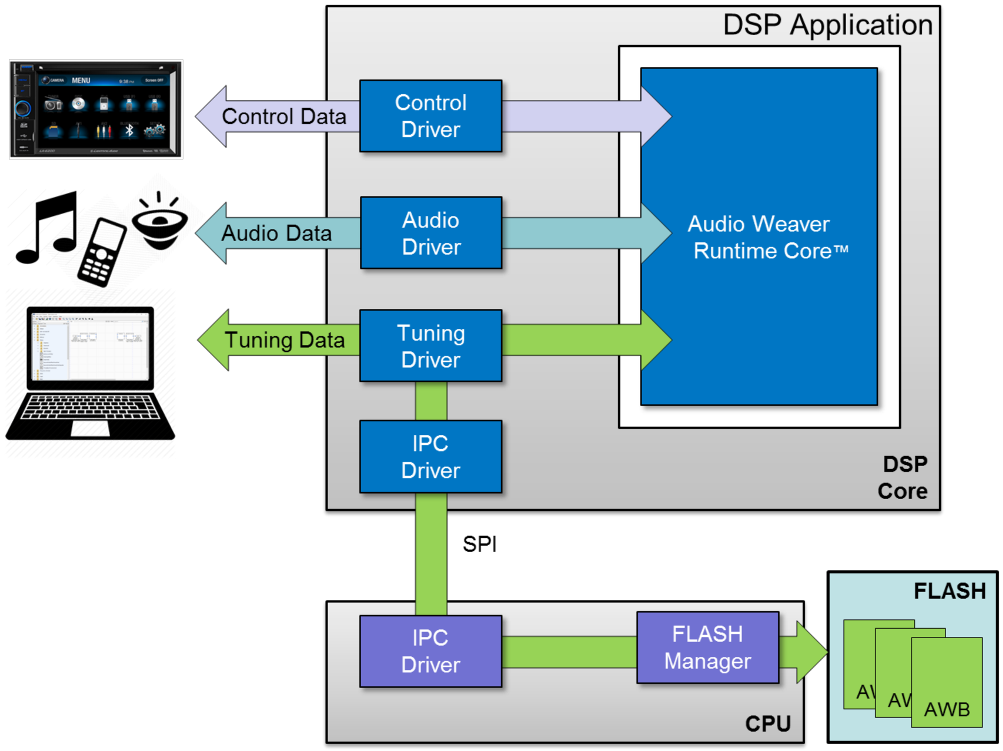


Figure 6. External micro controller managing FLASH contents

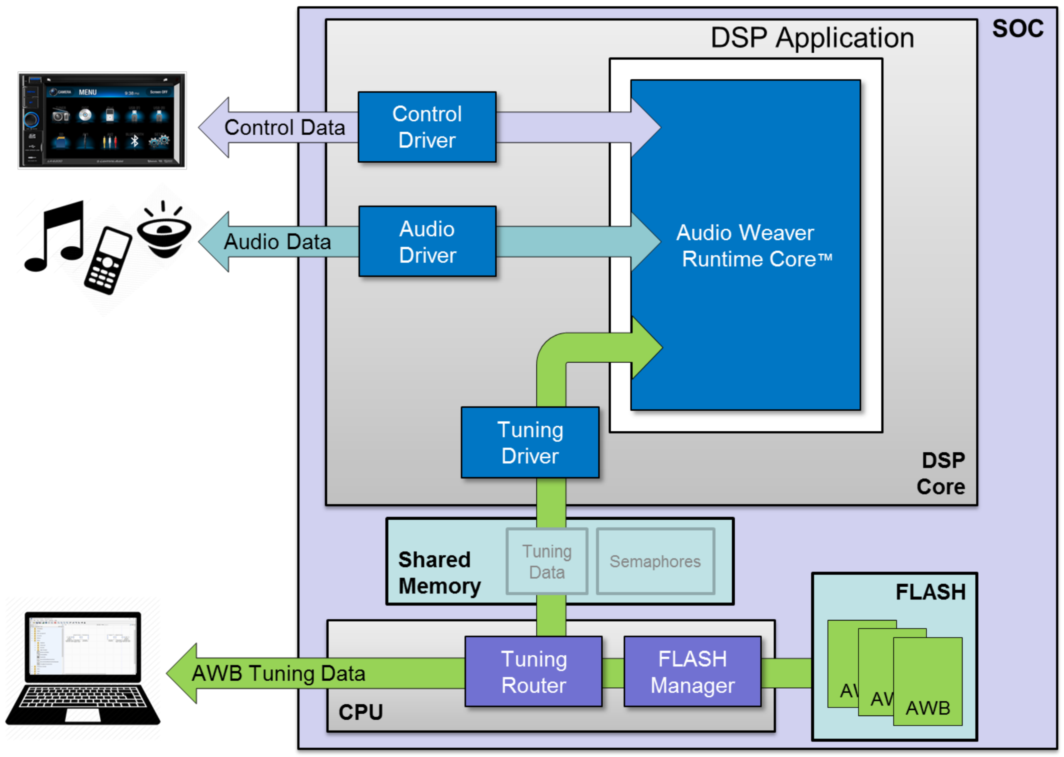


Figure 7. Example SOC implementation of Tuning interface

2.4. Control Model

Control data is defined as volume , mute, or other system-state that might affect audio processing like battery-level. Likewise, data calculated in the AWE Core (e.g. RMS level, an event-detection flag, or a direction-of-arrival estimate) may be retrieved using the control interface.

To move control data into and out of the Layout, the Application simply performs direct reads or writes of the state-data of module instances. As explained in further detail below, these writes/reads work by referencing module instances by their *ObjectID*. For the Application to access a module instance, its *ObjectID* must be known at application build-time. (At runtime *ObjectID*'s are used to fetch pointers using the function `awe_fwGetObjectByID()`.)

While the state/data of *any* module-instance may be directly accessed by the Application, it's highly recommended that the control data be exchanged using **Source** and **Sink** modules for the Layout / Application interface, as shown below in Figure 8.

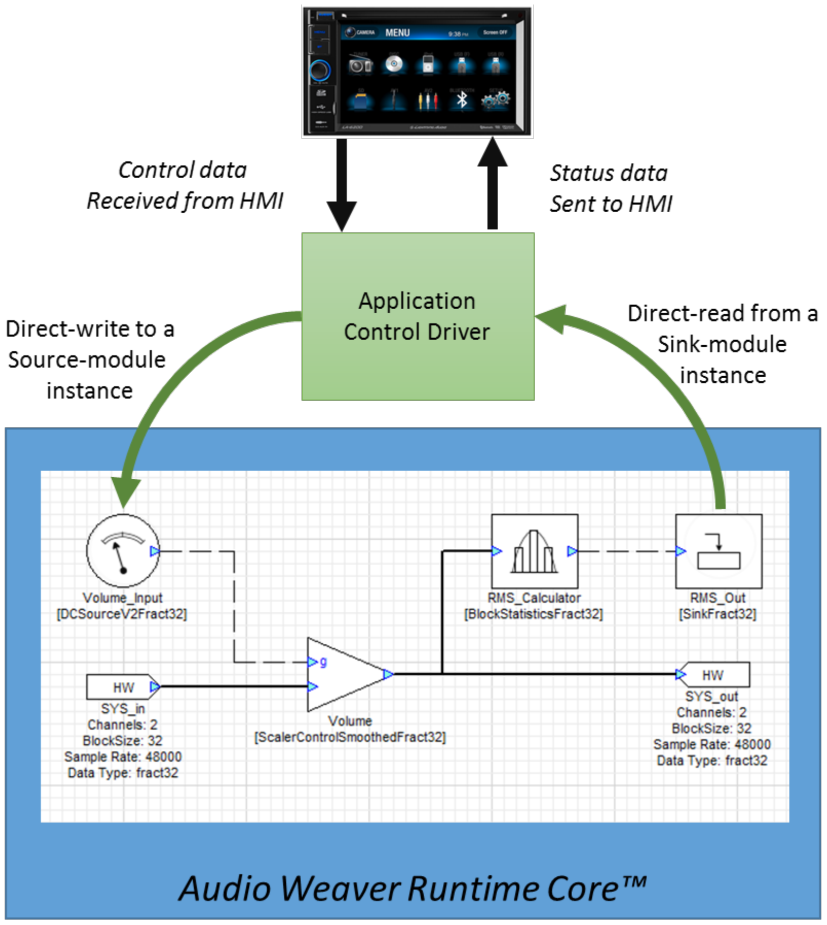


Figure 8. Simple example of Control data in a layout.

There are a number of advantages to using *Source* and *Sink* modules for your Layout control interface (as opposed to directly writing processing modules in the layout):

- *Greatly simplified firmware*: only a small number of ObjectIDs need to be known at build-time.
- *Increased Layout flexibility*: if the firmware isn't hard-coded to write to specific processing modules, then Audio Features may be built using any topology.
- *A reusable Control Driver*: If the Control Driver were to be hardcoded to interact with a specific implementation of an audio feature, it would need modification and testing for every Layout change. Instead, by using Source and Sink modules, the Control Driver can simply 'publish' control data in the Layout, and the DSP/Audio engineers can use those values as they see fit.
- *Increased Efficiency*: The Control Driver only needs to write the control-data to the

Layout once. It can then be distributed and used throughout the Layout via control wires.

Layouts typically use one of two conventions for Source and Sink modules as a Layout control interface. The first is to use unique Source/Sink modules for each control signal. The second is to use a ‘multi-channel’ Source to effectively create a “control bus”. Greater detail and tradeoffs are explained below:

Multi-input Control Convention

For less complex systems, where there isn’t much control I/O, it is convenient to use a separate ***DCSource*** module for each control input. Figure 8, above, shows this convention in use to provide an interface for a single input (Volume) and single output (RMS). In Figure 9, this convention is used to import Volume, Vehicle Speed, and Fan Speed.

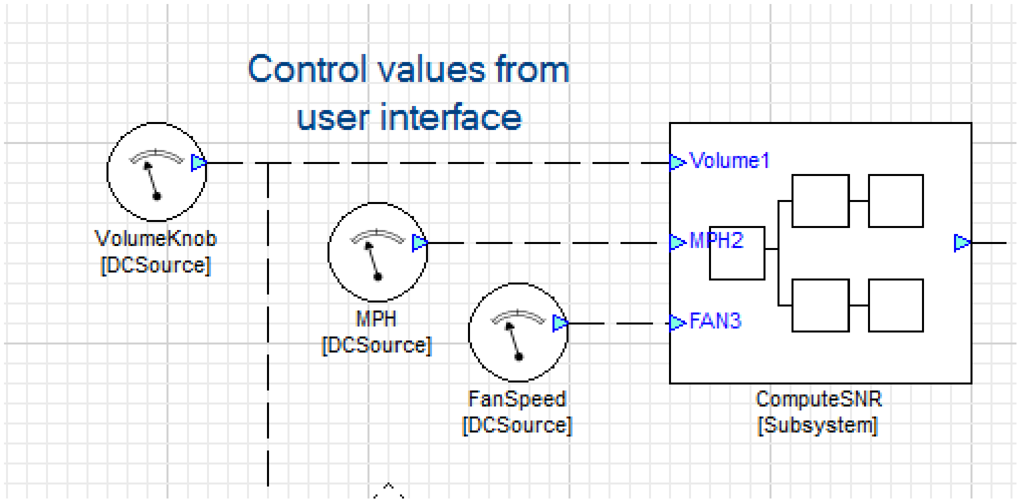


Figure 9. Multi-input control convention

An advantage to this convention is that control signals may be easier to work with in the layout. Using multiple ***DCSource*** modules may also make for a more visually clean Layout. A disadvantage is that the firmware becomes slightly more complicated, having to know the ObjectID’s for possibly many modules. Another potential disadvantage (again, for systems with many control inputs) is that Layout developers, to get access to all of a target’s control signals, will need to insert many individual ***DCSource*** modules and assign ObjectIDs to match the firmware’s expectations.

Control-Bus Convention

The other common way to create a control interface is to have a single, multi-channel ***Source*** module into which the Control Driver writes multiple control values. The output of this module is a multichannel signal which essentially forms a control bus that can be used throughout the layout. Router modules are then used to pull off the appropriate control channel.

Figure below shows an example of this convention. In this example, *View Wire Info* is enabled to show the channel count and blocksize; observe that control signals are dashed lines with a block-size of one, while audio signals are solid with a blocksize of 32. (Note: the thickness of the wire grows in proportion to the channel count; a 4-ch wire is thicker than a 1-ch wire)

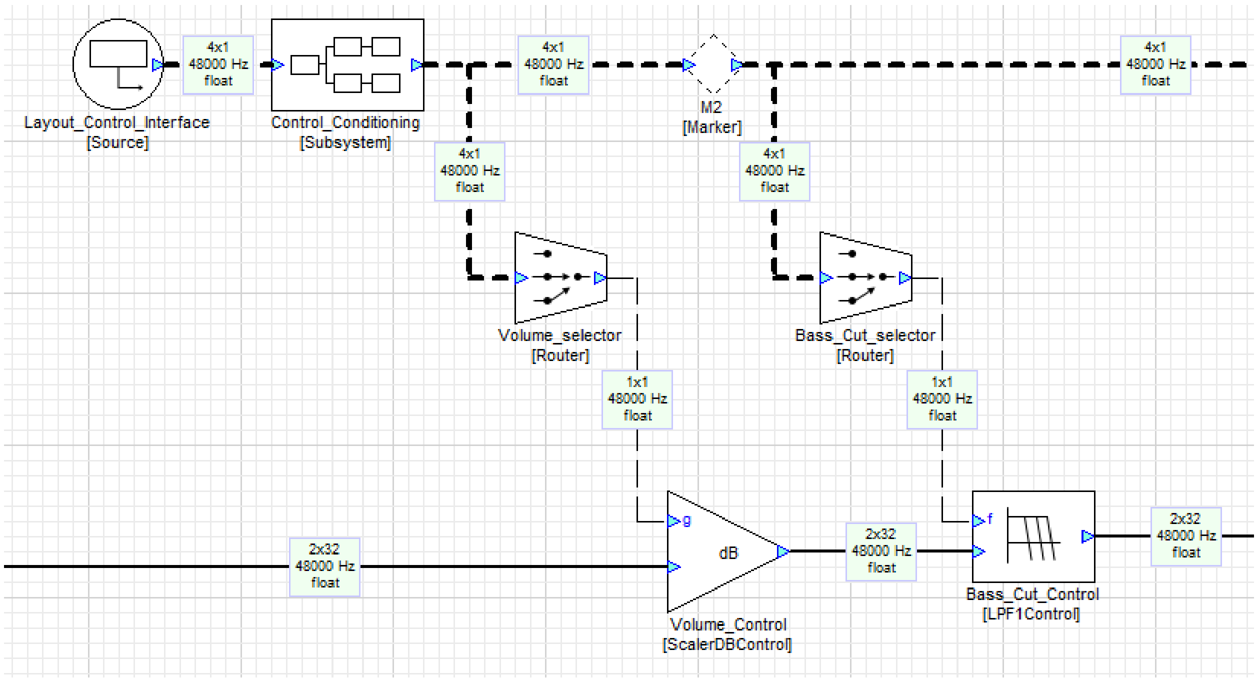


Figure 11. Control-Bus Example

The main advantages of using a control bus is that it makes for simplified and more reusable firmware. In this Control Convention, the Control Driver only needs to know one ObjectID, and writes all control data into that module (as an array). To get access to this control data, Layout developers simply insert one 'Layout_Control_Interface' source-module to get access to all control data published by that version of firmware. The main disadvantages of this approach is that one must take time to configure the routers appropriately and it's less clean, visually.

Signal Conditioning

Often, the raw input from the target control interface will need to be adjusted before it can be used in a Layout. For example, a Volume Knob might provide an integer, in the range 0-100 that needs to be mapped to some dB value using a table-lookup (to create an appropriate volume taper). In products with multiple control inputs, it's common to add a 'Control Conditioning' subsystem in which such mathematical or logical operations are performed. This subsystem, present in Figure 11 above, is shown, opened, below in 12, where a volume taper is applied.

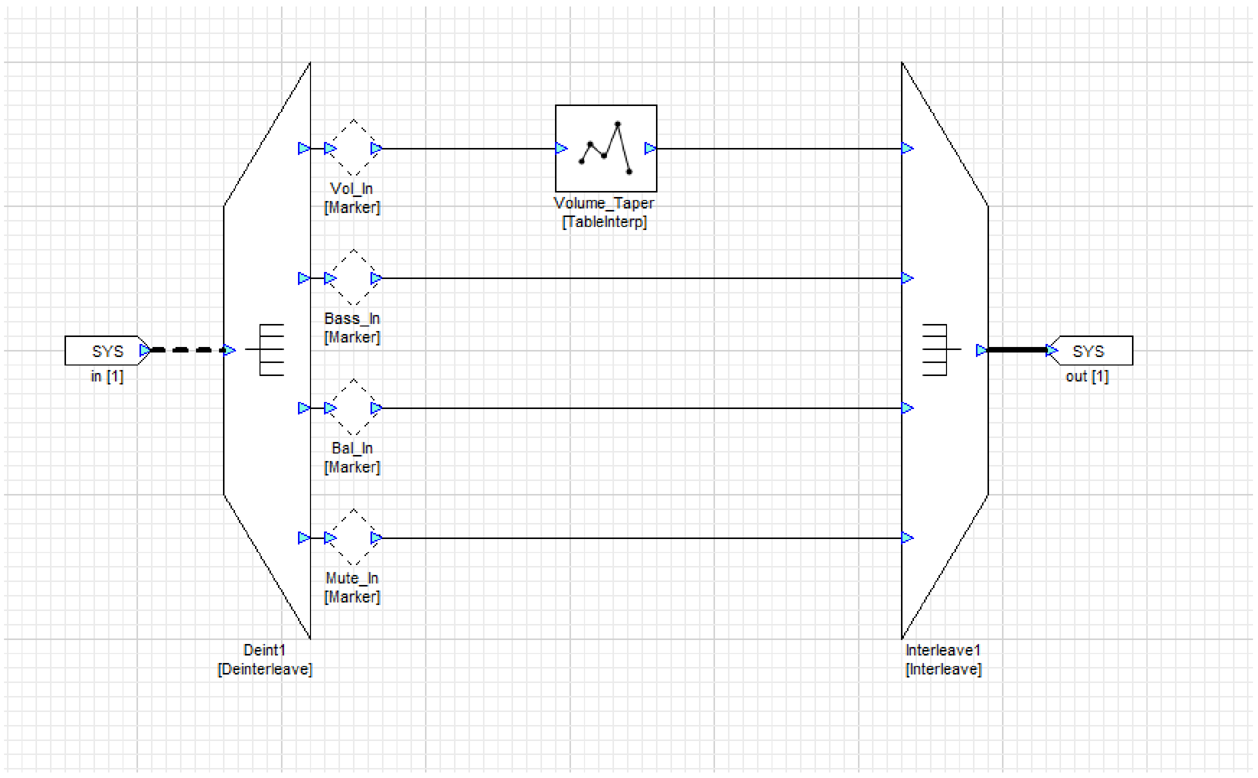
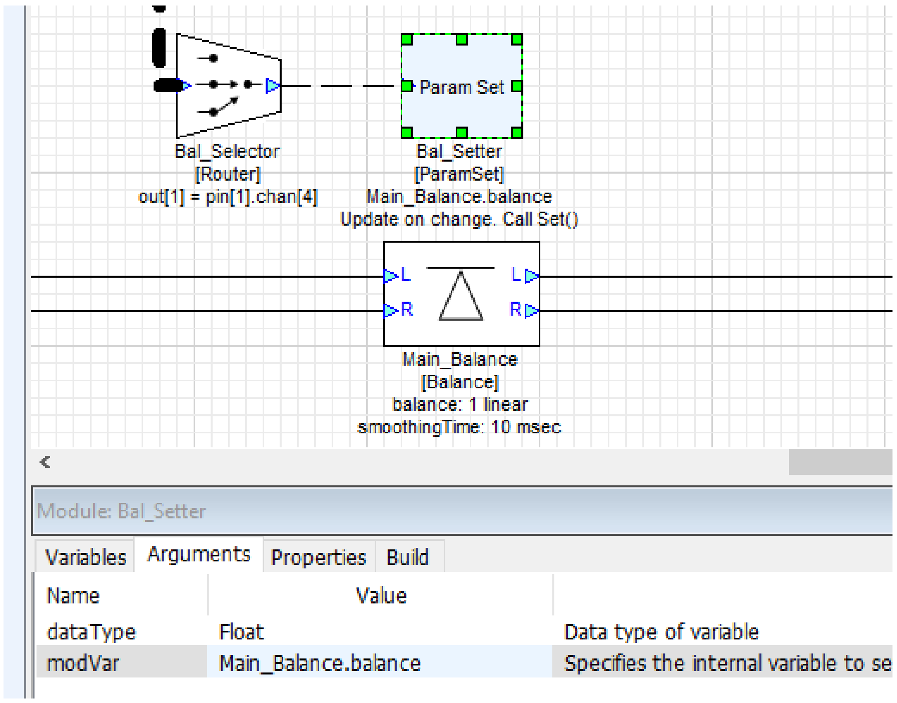


Figure 12. Control conditioning subsystem

Controlling Arbitrary Modules

While the AWE Core provides some modules with control pins, most modules do not have such inputs. To allow control wires to affect such modules, one should use the ***ParamSet*** module. This module has a control wire input, and, in the properties panel, can be configured to write the internal variable of any module. Likewise, the ***ParamGet*** module may be used to extract the current data from any module instance's inner variables. Figure below shows a ***ParamSet*** module being used to set the 'balance' variable in the 'Main_Balance' module.

Note: One can see the available Variables in module (that may be accessed by ParamSet/ParamGet) either by enabling 'View Module Variables' under the View menu or by clicking the 'Variables' tab in the Properties Panel.



Name	Value	
dataType	Float	Data type of variable
modVar	Main_Balance.balance	Specifies the internal variable to se

Figure 13. Controling balance using a ParamSet module

2.5. Audio-Processing Model

Block Processing

As audio samples are received, e.g. from an ADC via DMA, they are buffered into blocks. These blocks of audio samples are then copied into the Runtime Core, where they are processed, block by block.

A fundamental concept to the Audio-Processing Model is that the signal-processing block size is *independent* of the DMA block size. For example, an application may be configured for a DMA size of 32-samples per block, while a signal-processing Layout in the AWE Core may be tuned to operate on blocks of 64, 128, 256, etc. This decoupling allows the DSP engineer to easily make tradeoffs between MIPs, memory, and latency without having to modify or rebuild the Application executable. As shown in Figure 10, the AWE core has internal buffering to automatically handle this decoupling. In the example in Figure 104, the signal processing layout operates on blocks 4x the DMA block-size.

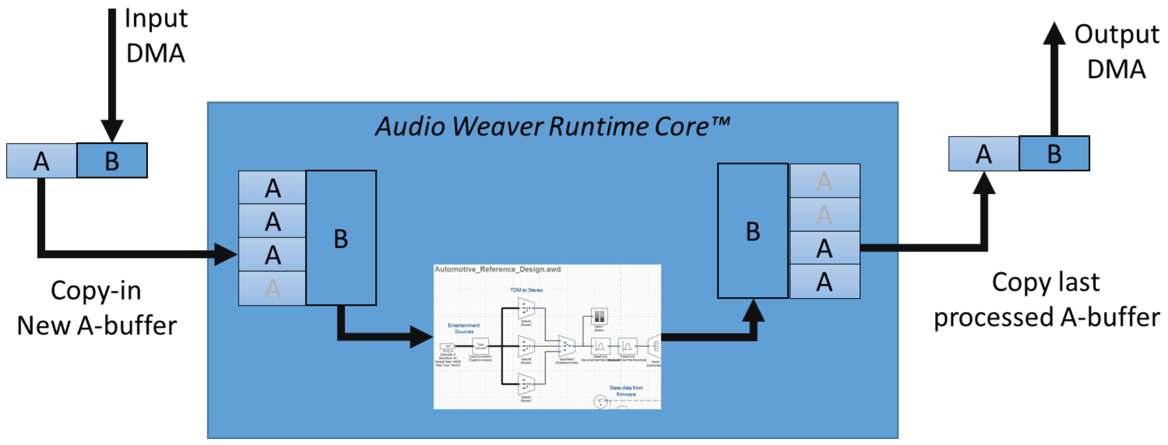


Figure 10. Diagram of static DMA-blocksize versus 'tunable' processing-blocksize.

To facilitate this design, the Audio Driver (in the ISR for a DMA-complete interrupt) interacts with the AWE Core's audio interface as follows:

1. As audio is received (i.e. as each DMA completes), it's copied from the DMA-receive buffer to the input-buffers of the AWE Core. (This copying operation is facilitated by some helper functions, described below in section 4.)

2. Likewise, since input and output are synchronous, newly processed audio is copied from the AWE Core's output buffers to the inactive DMA-transmit buffer.

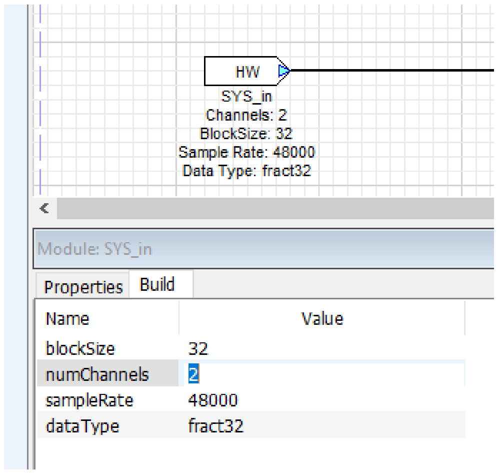
3. The Audio Driver tells the AWE Core how much audio was just exchanged by calling the function `awe_fwAudioDMAComplete(uint DMA_size)`. The return value of this function is a set of flags indicating whether the AWE Core has enough audio to process at the current processing block-size. (For example, if the DMA size is 32-samples per block, and the active layout is set to operate on blocks of 128 samples, then the AWE Core would indicate it's ready to run every 4th DMA-complete.)

4. When the AWE Core reports that enough audio has been accumulated (i.e. that the Layout is ready to be processed), the Audio Driver unblocks the Audio Processing thread. In bare-metal systems, this is done by raising the appropriate user-interrupt; in RTOS environments, this is typically done by posting the appropriate semaphore.

5. In the Audio Processing Thread, a call is made to the function `awe_fwPump()`, which processes the layout, refreshing the AWE Core's output buffers. (These buffers are then exported in step 2 above upon the next DMA-complete interrupt)

Channel Mapping

Just as Layouts and DMAs may use different *block sizes*, they may also have different *channel counts*. The number of channels supported by a target is fixed, based on the hardware and DMA configuration. Layouts, on the other hand, have a configurable number of input and output channels. (A Layout's input channel-count is configured in the *Properties* panel of the *SYS_in*, as shown in Figure 11. Likewise, a Layout's output channel count is independently configured in the properties of the *SYS_out* object.)



SYS_in
Channels: 2
BlockSize: 32
Sample Rate: 48000
Data Type: fract32

Name	Value
blockSize	32
numChannels	2
sampleRate	48000
dataType	fract32

Figure 11. Channel-count is in the Layout is independent of the hardware.

Along with the block-size decoupling, this channel independence allows any Layout to run on any hardware, e.g. a stereo-processing layout may run perfectly well on a board with 4-in, 8-out. To accommodate this, the Audio Driver should implement a simple convention to deal with channel mismatches. The most typical implementation is to use whatever audio is available and zero-fill the rest:

- If the hardware has *more input channels* (*N*) than the current Layout (*M*), then the Audio Driver only copies the first *M* channels into the AWE Core; the remaining hardware channels (*N-M*) are not processed.

- If the hardware has *fewer input channels* (*N*) than the current Layout (*M*), all *N* hardware inputs are copied into the AWE Core for processing, and the remaining Layout channels (*M-N*) are zero-filled.

- If the hardware has *more output channels* (*N*) than the current Layout (*M*), then all *M* Layout output channels are copied to the transmit DMA buffers; the remaining hardware output channels (*M-N*) are zero-filled.

- If the hardware has *fewer output channels* (*N*) than the current Layout (*M*), only the first *N* Layout output channels are copied to the transmit DMA buffers; the remaining Layout output channels (*M-N*) are not available as hardware outputs.

2.6. Threading / Priority Model

All threading and interrupt issues are managed outside of the AWE Core by the Application.

This approach allows Audio Weaver to run in virtually any embedded environment, from lightweight targets running bare-metal to more sophisticated systems with an RTOS.

Basic Threading Model

A basic Audio Weaver platform has 4 threads. From highest priority to lowest priority, the threads are arranged as:

- **Audio I/O** – Blocks of Audio Data are exchanged between DMA-buffers and the AWE Core's input and output buffers. Note: Audio is not *processed* in this thread; it's simply copied into and out of the AWE Core.
- **Tuning I/O** – Tuning Data, on a byte, word, or packet boundary is exchanged between the Tuning Driver and the AWE core. Note: Tuning data is not *processed* in this thread; it's simply moved into and out of the AWE Core.
- **Audio Processing** – In this thread, the active Layout is processed. This thread is raised / unblocked by the Audio I/O thread when sufficient audio data has been received.
- **Background Processing –** In the background / main() thread, non-real-time tasks are processed. The AWE Core has a single function that must be called from this context: `awe_fwTick();` This function handles all background tasks in AWE Core, including, among other things, the processing of tuning commands.

Advanced Threading Model

The AWE Core allows audio to be processed at multiple block sizes. For example, one path through the layout might do time-domain processing on a block size of 32 samples while another path through the Layout does frequency-domain processing on blocks of 1024 samples. A system like this is built using the BufferUp and BufferDown modules in the layout to connect the different block size domains as shown in Figure 12.

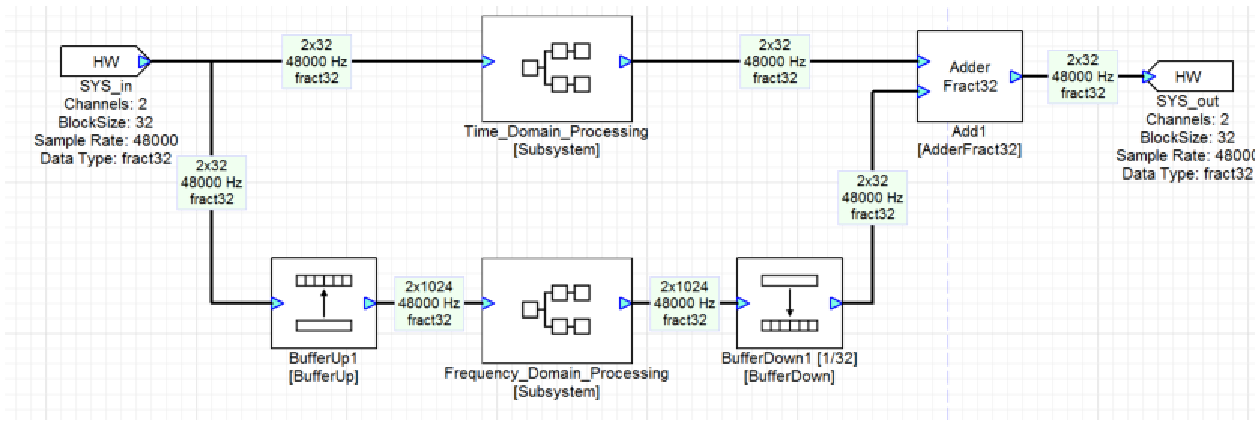


Figure 12. An audio design with multiple block sizes.

The processing of a large block-size path will take longer than the processing of small block-size paths. To prevent the longer-running (large block-size) path from consuming all processing resources, the AWE Core has a facility to allow different Layout paths to run in different priority threads. In the example in Figure 12, the Application would require two Audio-processing threads, each with its own priority. Generally, for each block-size in the Layout, the Application must provide an Audio-Processing thread with unique priority. (Smaller block-size paths require higher priority threads.)

With two block sizes, five total threads are typically used by the AWE Core with the following priorities:

- **Audio I/O** – Blocks of Audio Data are exchanged between DMA-buffers and the AWE Core's input and output buffers. Note: Audio is not *processed* in this thread; it's simply moved into and out of the AWE Core.

- **Tuning I/O** – Tuning Data, on a byte, word, or packet boundary is exchanged between the Tuning Driver and AWE core. Note: Tuning data is not *processed* in this thread; it's simply moved into and out of the AWE Core.

- **Audio Processing – Small Block** – In this thread, the Layout path with the small block-size of 32-samples is processed. This thread is raised / unblocked by the Audio I/O thread when sufficient audio data has been received.

- **Audio Processing – Large Block** – In this thread, the Layout path with the large block-size of 1024 samples is processed. This thread is raised / unblocked by the Audio I/O thread when sufficient audio data has been received.

- **Background Processing –** In the background / main() thread, non-real-time tasks are processed. The AWE Core has a single function that must be called from this context: `awe_fwTick();` This function handles all background tasks in AWE Core, including, among other things, the processing of tuning commands.

Figure 13 below shows the audio-related threads, with block sizes of 32 and 64. Because the processing for a large block-size may take a long time, the processing of path with smaller blocks is done at a higher priority than threads with larger blocks.

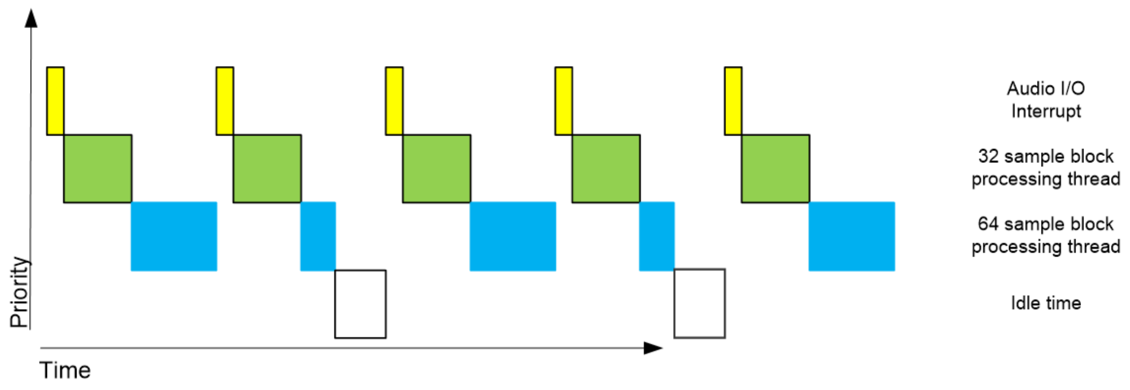

Audio I/O Interrupt
32 sample block processing thread
64 sample block processing thread
Idle time
Priority
Time

Figure 13. Processor activity when audio processing occurs in multiple threads.

Details of how AWE Core supports paths with different block-sizes is explained in further detail below, in section 4.4.5.

3. Tuning Interface

3.1. Overview

The tuning interface uses a single buffer for command and replies. The Application is responsible for instantiating this buffer, which is used for bi-directional interfacing between the AWE Core and the tuning transport.

3.2. Integration Quick Guide

To bring-up the AWE Core's Tuning Interface, the integrator should perform the following tasks:

1. In the Application, instantiate a 264-word buffer that the AWE Core can use for in-place packet processing, e.g. `UINT32 s_PacketBuffer[MAX_COMMAND_BUFFER_LEN] = {0};` Note that this buffer is used to host both the command and the resulting reply.
2. During post-boot initialization, initialize the AWE Core's tuning subsystem to use this buffer, e.g. `awe_fwTuningInit(s_PacketBuffer, MAX_COMMAND_BUFFER_LEN);`
3. Develop a driver for the tuning interface. This may mean configuring registers in a SPI or UART peripheral, or it could mean configuring a USB stack or opening TCP/IP socket.
4. Create a high-priority "Tuning I/O thread" to handle data from the tuning transport (e.g. UART, USB, etc.).
5. Verify driver, protocol, and hardware functionality by implementing simple 'echo-back' functionality. In this step, simply copy all received data back to the transmit buffer.

6. Modify the "Tuning I/O thread" to send received into the AWE core using whichever method is most convenient: `awe_fwTuningRxByte()`, `awe_fwTuningRxWord()`, or `awe_fwTuningRxPacket()`. Note: tuning commands are not processed in this thread.

7. In the background thread, add repeated calls `awe_fwTuningTick()` to continually process data as it arrives. This function will execute commands (once they're fully received), and will return `REPLY_READY` when a reply packet has been generated.

8. Develop a method to send the reply packet back over the tuning transport. The reply will always reside in the tuning buffer allocated in step one. As shown below in section 3.4, the length of the reply (in words) is in the upper 16-bits of the 32-bit header.

3.3. Tuning API

The API is defined in `TuningHandler.h`.

void **awe_fwTuningInit** (UINT32 *packet_buf, INT32 buf_length);

Initialize the AWE Core to use the specified buffer for communications. This buffer is used to hold the current command and is then overwritten with the reply data.

void **awe_fwTuningRxByte** (unsigned char ch);

Send received byte of tuning data into AWE Core. This is typically used for byte-based transports like UART.

void **awe_fwTuningRxWord** (unsigned char ch);

Send received 32-bit Word of tuning data into AWE Core. This is typically used for word-based transports like SPI.

void **awe_fwTuningRxPacket** ();

Set flag indicating that a new command has been received and is ready for processing by the AWE Core. This is typically used for network transports like Ethernet or USB.

UINT32 **awe_fwTuningtick** (AWEInstance * pAWEInstance);

Assemble tuning command, execute it when reception complete, and generate a reply. Returns REPLY_READY (0) when a command has been executed and reply is ready to be sent.

3.4. Message Structure

Audio Weaver Tuning Commands and Replies use 32-bit words, with a 1 word header and 1 word CRC wrapping the payload as shown in Figure 14:

16b Length	16b ID
Payload[0]	
Payload[1]	
…	

Payload[N-1]
CRC Word

Figure 14. Tuning Command and Reply Format

The length of the message includes the header and CRC word. Thus, the shortest possible message – one without a payload – is two words in length.

The CRC word is a 32-bit value and computed so that when all words of the message, including the CRC, are XOR'ed together, the result is 0. The following code computes the CRC of a packet prior to transmission:

```
nLen = g_PacketBuffer[0] >> 16;

DWORD crc=0;
for(i=0; i < (nLen-1); i++)
  {
    crc^=g_PacketBuffer[i];
  }

g_PacketBuffer[nLen-1] = crc;
```

Figure 15. Tuning protocol CRC Calculation

On the target side, the CRC calculations are handled in `awe_fwTuningTick()`. It checks both the CRC of the received message and computes the checksum of the reply.

3.5. RS-232 Protocol

Audio Weaver messages (commands and replies) are arrays of 32-bit integers. In the case of RS-232 communications, we add a lower level byte-by-byte protocol which adds another level of robustness to the communication link. This additional robustness is critical in RS-232 which is subject to buffer overruns and dropped bytes.

Each 32-bit integer s_msg_word within the message array is expanded into 5 bytes using the code:

```
ch = (unsigned char)(0x80 |  s_msg_word        & 0x7F);
ch = (unsigned char)(0x80 | (s_msg_word >>  7) & 0x7F);
ch = (unsigned char)(0x80 | (s_msg_word >> 14) & 0x7F);
ch = (unsigned char)(0x80 | (s_msg_word >> 21) & 0x7F);
ch = (unsigned char)(0x80 | (s_msg_word >> 28) & 0x7F);
```

7 data bits are taken at a time from each message word and the high bit is set. (For the last character, only the low 4 data bits are used.)

The data is then encapsulated within a series of protocol bytes:

Start Byte	0x02
Sequence Byte	0x30 to 0x39 (ASCII "0" to "9")
Message Bytes	0x80 to 0xFF. 5 bytes at a time.
Stop Byte	0x03

With this design, the protocol bytes are unique. That is, the protocol bytes (0x02, 0x03, 0x30-0x39) are never found within the data bytes since the data bytes always have the high bit set. This makes it easy to identify the start and end of data packets. The sequence byte starts at 0x30, increments 1 for each successful transmission, and then wraps from 0x39 to 0x30. The sequence byte is used to identify retransmissions.

For example, consider the command PFID_GetProfileValues with ID = 43. The message sent from the PC to the target processor is:

Message Length = 2	ID = PFID_GetProfileValue
CRC	

Which translates into the 32-bit words:

0x0002002b
0x0002002b (CRC is the same for 1 word payloads)

The 32-bit words are expanded into 5 bytes each:

0x0002002b → 0xAB 0x80 0x88 0x80 0x80
0x0002002b → 0xAB 0x80 0x88 0x80 0x80

Adding the remaining protocol bytes, the sequence sent is:

0x02 (start byte)
0x3X (sequence)
0xAB 0x80 0x88 0x80 0x80 (payload)
0xAB 0x80 0x88 0x80 0x80 (CRC)
0x03

3.6. SPI Protocol

The Audio Weaver SPI protocol mirrors the 32-bit packet structure. There are two differences:

1. The 32-bit synchronization word 0xDEADBEEF is sent before each message.
2. The 4 bytes within each 32-bit word as swapped. 0x12345678 it turned into 0x78563412.

For example, consider the command "PFID_GetProfileValues" with ID = 43 (0x2b). The message sent from the PC to the target processor is:

Message Length = 2	ID = PFID_GetProfileValue
CRC	

This translates into the 32-bit words:

```
0x0002002b
0x0002002b    (CRC is the same for 1 word payloads)
```

The overall message sent is:

```
0xDEADBEEF     →     0xEFBEADDE
0x0002002B     →     0x2B000200
0x0002002B     →     0x2B000200
```

After a message is received by the target processor and the message is being processed, the SPI output buffer will be set to the "not ready" word 0xA3A3A3A3. This will allow a host processor to poll the SPI interface waiting for the target processor to complete message processing. The target processor will continue to transmit the not ready word until the message has been processed. At this point, it will switch over to the sync word followed by the complete message.

If the host processor continues to read beyond the end of a reply from the target, then the target will return 0xFFFFFFFF.

3.7. USB Protocol

Audio weaver USB communication is based on the USB HID protocol. Audio Weaver commands and replies are encapsulated in one or more 56-byte HID report packets operating over a 64-byte USB pipe. Each HID report packet starts with the HID report ID. Audio Weaver uses HID Report ID 1. If USB HID communications are used for other firmware features, those features must use a HID Report ID other than 1.

Following the HID report ID is a one byte seq number and a two byte length field.

HID Report ID 1	Packet Seq No	Command byte length		AWE packet bytes
1 byte	1 byte	hi byte	lo byte	upto 52 bytes

3.8. Error Handling

Several different types of errors can occur within the communication link.

Receive Errors (PC/Host → Target Processor)

If the received message has been formatted (such as wrong protocol bytes or the payload is not a multiple of 5 bytes in the case of RS-232) then the target processor silently ignores the message. The PC/Host will time out and then retransmit the message with the same sequence number.

If the received message has a CRC error, then the target processor will reply with message type PFID_Undefined. The message payload will be set to E_BADPACKET. When the PC/Host processor receives this reply, it will increment the sequence number and then resend the previous message.

There are some more error conditions defined. Take a look at PacketAPI.c. There are returned error codes now for: failing CRC, unknown message, and message length too long.

Reply Errors (Target Processor → PC/Host)

Assume that the message was properly received by the target processor and executed by the message handler in PacketAPI.c. The reply message is formed and then sent back to the host processor. The sequence number in the reply equals the sequence number of the original message from the PC/Host.

Now, assume that an error occurs in the transmission of the reply back to the PC/Host processor. When the PC/Host detects the error, it should resend the original message with the same sequence number. When the message is received by the target processor, the target processor sees that the sequence number is unchanged from the last message; this flags an error in the previous reply. *The target processor does not re-execute the last command.* Instead, it simply sends the reply again.

Once a message is received through the tuning interface, it is processed by in the function awe_fwTick(). The first step is to check the CRC. If there is a CRC error, the target processor returns a 3 word message to the host (PC). The message type is:

```
PFID_Undefined
```

and the message payload is

```
E_BADPACKET
```

When this error is received by the host (PC) the last message should be retransmitted.

The Audio Weaver Server application does extensive error handling on the received data. If an error occurs, the Server retransmits the packet 3 times before hard failing.

Time Outs and Resends

The Audio Weaver has several configurable timeouts. After a message is sent to the target processor the Server waits for a reply. When using the RS232 protocol, if the start of the reply is not received within 50 msec the Server declares a time out and resends the message (SingleCharTimeout). If the entire reply is not received within 150 msec then a timeout is declared (TotalTimeoutTimeout).

When a timeout occurs, the Server resends the message. If the same message has been sent 3 times (2 timeouts), then the Server will declare an error and inform the user.

The timeout periods can be modified to match your application. Edit the AWE_Server.ini file and add the lines:

```
[TimeOut]
SingleCharTimeout=200
TotalTimeoutTimeout=1000
```

All times are in milliseconds.

4. Audio Interface

4.1. Overview

Audio I/O takes place outside of the AWE Core, in the high-priority Audio I/O Thread. In this thread, audio-samples are buffered up into blocks and then passed into Audio Weaver. In typical implementations, audio I/O uses ping-pong buffers and DMA, as shown in Figure 16. The block-size of the DMA operation is called the *fundamentalBlockSize* and is typically 32 samples. The AWE Core has additional internal buffering which allows processing to run at any multiple of the *fundamentalBlockSize*.

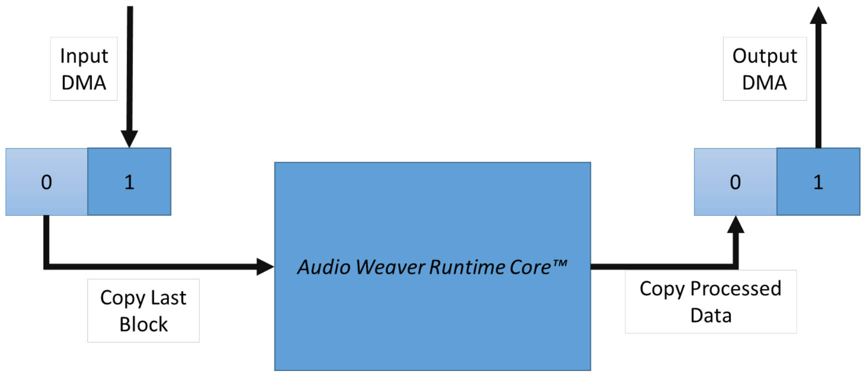


Figure 16. Block based audio processing using double buffered DMA.

The Application is responsible for all DMA setup and buffering, including configuring the audio codec, DMA, and managing interrupts. This section describes how audio data is exchanged with and processed by the AWE Core.

4.2. Audio Processing

The Audio Interface is typically the second thing in a system to be brought up during integration (after the Tuning Interface). To bring-up the AWE Core's Audio Interface, the integrator should perform the following tasks:

1. In the Application, instantiate DMA buffers for input and output.
2. During post-boot initialization, initialize the AWE Core's audio subsystem using `awe_fwInit();`
3. Develop a driver for the audio interface. This may mean configuring a TDM or I2S Serial-Port peripheral, A2B interface, or Shared-Memory IPC system.
4. Configure 'chained' Receive and Transmit DMAs.
5. Implement a basic high-priority "Audio I/O thread" (the ISR for a DMA-complete interrupt in a bare-metal system) with 'DMA pass-through' functionality. The goal of this step is to ensure that the rest of the Audio path (drivers, CODECs, network transceivers, amplifiers, etc) is functional by simply copying from the DMA-receive buffer to the DMA-transmit buffer.
6. Create a medium-priority 'Audio Processing thread' in which the audio-processing function, `awe_fwPump()` is executed. In a bare-metal system, this simply means

mapping a function to a user-interrupt. Figure 17 shows how this might be done on a SHARC DSP.

```
…
interruptcb(sig_usr0, RenderAudio);
…

void RenderAudio(INT32 sig_int)
{
    awe_fwPump(0); // Run the layout at user interrupt level
}
```

Figure 17. Mapping ISR to interrupt on SHARC

7. Plug the AWE Core into the audio path. To do this, modify DMA-passthrough code in your high-priority "Audio I/O thread" with code that exchanges audio data with the AWE Core. At a high-level, this is a 4-step process. This is described in greater detail below in section 4.4.

    a. Fetch details of the active Layout

    b. Copy audio from receive-DMA buffers to AWE Core's audio-input buffers

    c. Copy audio from AWE Core's audio-output buffers to transmit-DMA buffers

    d. Unblock the [lower priority] Audio Processing Thread if the Layout has enough data to execute.

4.3. Audio API

This API is defined in `Framework.h`

void	**awe_fwInit** (AWEInstance * pAWEInstance) Audio framework initialization function.
INT32	**awe_fwGetInputBlockSize** (AWEInstance * pAWEInstance) Returns the block size of the currently instantiated Layout.
INT32	**awe_fwGetInputSampleRate** (AWEInstance * pAWEInstance) Returns the sample rate of the currently instantiated Layout.
void	**awe_fwGetChannelCount** (AWEInstance * pAWEInstance, INT32 *inCount, INT32 *outCount) Returns the number of input & output channels in the currently instantiated Layout

INT32 *	**awe_fwGetInputChannelPtr** (AWEInstance * pAWEInstance, UINT32 pinNdx, UINT32 chan, INT32 *stride) Returns the address where the next block of input audio samples should be copied.
INT32 *	**awe_fwGetOutputChannelPtr** (AWEInstance * pAWEInstance, UINT32 pinNdx, INT32 chan, INT32 *stride) Returns the address from which processed, output audio samples may be read.
INT32	**awe_fwPump** (AWEInstance * pAWEInstance, UINT32 layout_no) Executes the currently instantiated Layout, 'pumping' audio from input to output.
INT32	**awe_fwTick** (AWEInstance * pAWEInstance) Must be called periodically to allow AWRC to operate its internal state machines.

4.4. Detailed Integration Steps

To plug the AWE Core into a functioning audio path (step 7 in section Error: Reference source not found above), the following steps must be taken. Note: To aid in understanding, it may be helpful to follow along in the `AudioDriver.c` from one of DSP Concepts' reference integrations.

Update DMA pointers

The Audio I/O thread runs when a new block of audio is available. This thread copies new audio samples from the DMA receive-buffer to the Layout; and copies freshly-processed audio from Layout to the DMA transmit-buffer. The first step in this process is to update pointers to the proper [inactive] DMA buffers.

Calculate the HW-to-Layout Channel Mapping

To do this, simply compare the number of Hardware channels vs the number of channels in the active Layout. The number of hardware channels is known to the integrator; the number of channels active in the Layout is fetched using the following method (which populates local variables with these counts):

```
awe_fwGetChannelCount(AWEInstance pAWEInstance, &layoutInChans, &layoutOutChans);
```

Note: If `layoutInChans==0` and `layoutOutChans==0`, this indicates that the audio system is not instantiated or that the system does not have any input or output pins.

```
if(HW_IN_CHAN < layoutInChans)
{
    chansToWrite = HW_IN_CHAN;
}
else //HW has enough input to populate all Layout input channels
{
    chansToWrite = layoutInChans;
}
```

Fill the Layout's input buffers one channel at a time

To copy audio into the Layout's input, fetch a pointer to each channel. As the Layout's input buffers are interleaved, we also get a pin stride to use:

```
    //fill available audio from HW
    for(chan=0; (chan < chansToWrite) ; chan++)
    {
      // If Source DMA buffer is interleaved
      Src_DMA_ptr = DMA_RX_Ptr + chan;

      // If Source DMA buffer is concatenated linear buffers
      //Src_DMA_ptr = DMA_RX_Ptr + AWE_FRAME_SIZE_IN_SAMPLES;

      AWE_In_Ptr = awe_fwGetInputChannelPtr(pAWEInstance, nPinNdx, ch
    an, &pinStride );
      for (Sample=0; (nSample < DMA_BLOCK_SIZE); Sample++)
      {
        // Convert from 16-bit PCM data to 32-bit fract32
        *AWE_In_Ptr  = *Src_DMA_ptr << 16;
         AWE_In_Ptr += pinStride;

         // If Source DMA buffer is interleaved
         Src_DMA_ptr += DMA_CHANNEL_COUNT;

         // If Source DMA buffer is concatenated linear buffers
         //Src_DMA_ptr++;
      }
    }
    //zero remaining extra layout channels
    for ( ; chan < (layoutInChans); chan++)
    {
      AWE_In_Ptr = awe_fwGetInputChannelPtr(pAWEInstance, nPinNdx, ch
    an, &pinStride );
      for(Sample=0; Sample<FW_BLOCK_SIZE; Sample++)
      {
        AWE_In_Ptr[Sample * pinStride] = 0;
      }
```

```
}
```

Fill the Transmit-DMA buffers one channel at a time

This process is the same as copying input-audio into the AWE Core. First, determine the output channel mapping:

```
if( HW_OUT_CHAN > layoutOutChans )
{
  chansToWrite = layoutOutChans;
}
else //Layout has more output channels than HW can handle
{
  chansToWrite = HW_OUT_CHAN;
}
```

Then, fill the DMA transmit buffer with whatever audio is available:

```
//fill transmit buffers from available audio
for(chan=0; chan < chansToWrite; chan++)
{
  // If Dest DMA buffer is interleaved
  Dst_DMA_ptr = DMA_TX_Ptr + chan;

  // If Dest DMA buffer is concatenated linear buffers
  //Dst_DMA_ptr = DMA_TX_Ptr + AWE_FRAME_SIZE_IN_SAMPLES;

  AWE_Out_Ptr = awe_fwGetOutputChannelPtr(pAWEInstance, nPinNdx,
chan, &pinStride);
  for (Sample=0; Sample < DMA_BLOCK_SIZE; Sample++)
  {
     // Convert from fract32 to PCM 16-bit
     *Dst_DMA_ptr = *AWE_Out_Ptr >> 16;

     // If Destination DMA buffer is interleaved
     Dst_DMA_ptr += DMA_CHANNEL_COUNT;

     // If Destination DMA buffer is concatenated linear buffers

     // Dst_DMA_ptr++;

     AWE_Out_Ptr += pinStride;
  }
}
//zero remaining extra layout channels
```

```
for ( ; chan < (layoutInChans); chan++)
{
  // If Dest DMA buffer is interleaved
  Dst_DMA_ptr = DMA_TX_Ptr + chan;

  // If Dest DMA buffer is concatenated linear buffers
  //Dst_DMA_ptr = DMA_TX_Ptr + AWE_FRAME_SIZE_IN_SAMPLES;

  AWE_Out_Ptr = awe_fwGetOutputChannelPtr(pAWEInstance, nPinNdx,
chan, &pinStride);
  for(samp=0;samp<FW_BLOCK_SIZE;samp++)
  {
    *Dst_DMA_ptr = 0;

     // If Destination DMA buffer is interleaved
     Dst_DMA_ptr += DMA_CHANNEL_COUNT;

     // If Destination DMA buffer is concatenated linear buffers

     // Dst_DMA_ptr++;
  }
}
```

Tell the AWE Core how many samples were added / consumed

After handling the audio exchange between the DMA buffers and the AWE Core, we inform the AWE core how many samples were exchanged using the following function:

```
unsigned int awe_fwAudioDMAComplete(AWEInstance pAWEInstance, int sampsPerTick);
```

The return value from this function indicates if any paths through the layout are now ready to run. (For an explanation of this, see section 2.5.1, Block Processing and Section 2.6.2 Advanced Threading Model)

Unblock the Audio Processing Thread(s) when appropriate

The return value from `awe_fwAudioDMAComplete` is a bit-field indicating when Layout paths are ready to run. Bit 0 being set `true` means the smallest block-size (that of the input and output pins) is ready to be processed. Bit 1 means the next larger block-size is ready to be processed, etc.

In a bare-metal system, where thread priorities are managed using user-interrupts, this step is done by raising the appropriate interrupt as follows:

```
LayoutMask = awe_fwAudioDMAComplete(pAWEInstance, FW_BLOCK_SIZE);
```

```
    if (layoutMask & 0x01) //hi prio (small block) path ready
    {
      raise(SIG_USR0);
    }
    if (layoutMask & 0x02) //lo prio (large block) path ready
    {
      raise(SIG_USR1);
    }
```

5. Control Interface

5.1. Overview

This section describes how to control the audio processing in the final product without being connected to a PC. Typical use cases are adjusting volume, bass, or treble based on some hardware event (like a volume knob) or perhaps dynamically changing filter coefficients based on changing environmental listening conditions.

A simple example of this is shown below in Figure 18.

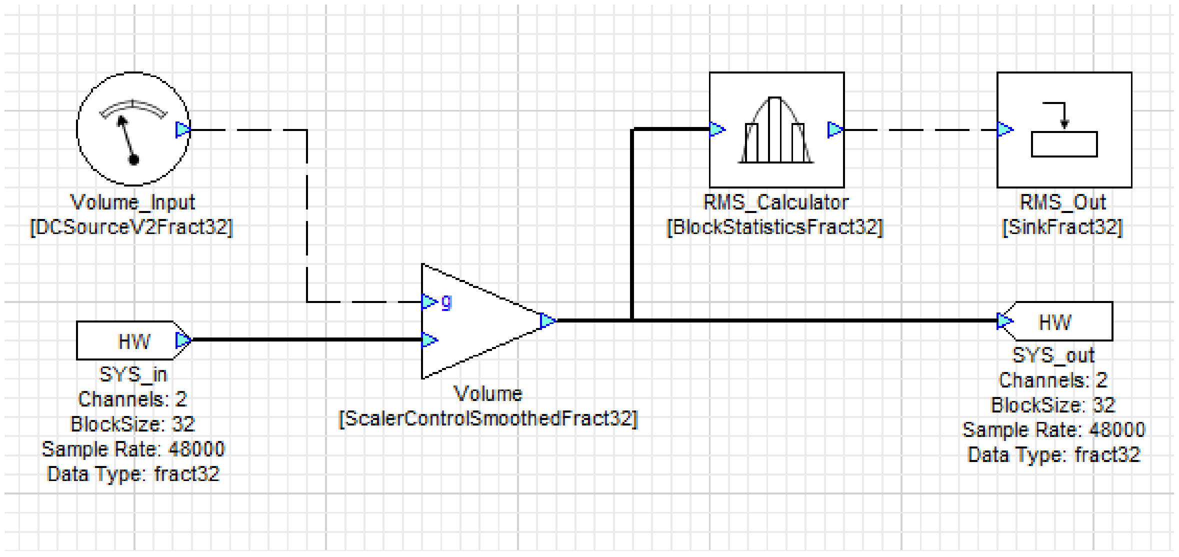
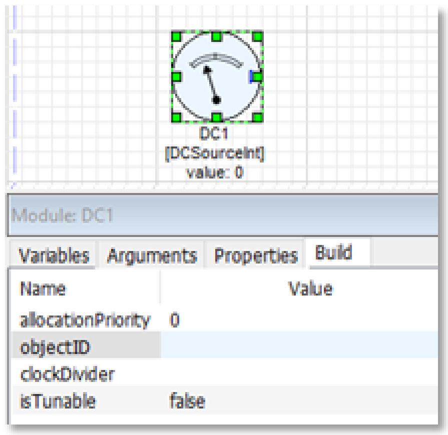


Figure 18. Simple system showing Source and Sink models as Layout Control I/O interfaces.

The API for the control interface is simple; firmware may directly read or write the state-data of any module instance. Because layouts are dynamically instantiated, the Control Driver must get the address of a module's state-data using the Control API's only function, **awe_fwGetObjectByID()**. Once it has this pointer, the application can directly read and write data.

1.1. ObjectID allocation

Every module in a Layout has an *ObjectID*. By default, *ObjectIDs* are dynamically assigned by the AWE Core during run-time instantiation. However, for modules that the integrator wishes to access, a static *ObjectID* may be assigned to a module instance. To assign a static *ObjectID* in the Layout, select a module, open the *Properties* panel, select the *Build* tab, and set the *ObjectID*, as shown in Figure 19. (When the *ObjectID* field is left blank, it will be dynamically assigned by the AWE Core.)



Name	Value
allocationPriority	0
objectID	
clockDivider	
isTunable	false

Figure 19. Assign a static ObjectID to a control interface module.

The AWE Core reserves *ObjectID's* 0-29999. Statically assigned *ObjectIDs* must start at 30000.

5.2. Control API

```
INT32 awe_fwGetObjectByID(AWEInstance *          pAWEInstance,
                          UINT32                 ID
                          InstanceDescriptor **  pObject,
                          UINT32 *               pClassID);
```

Locate the object based on its objectID.

Parameters

[in]	**pAWEInstance**	AWE Instance pointer
[in]	**ID**	ObjectID to locate
[out]	**pObject**	Pointer to object reference to be updated with address of module
[out]	**pClassID**	Pointer to object reference to be updated with class ID of object

Returns

E_SUCCESS or E_NO_MORE_OBJECTS

5.3. Detailed Integration Steps

Choose a control convention

As described above, in section 2.4, most integrators choose either the *multi-input* or *control-bus* convention.

Add Interface modules to the Layout

The source and sink modules in Figure 20 below can be used to create a Layout Control Interface.

On the input side, there are essentially two classes of modules to choose from, *Source* and *DCSource* modules. A *Source* module can hold multiple unique inputs, making that data available to the layout as a multichannel control-bus. To the firmware, *Source* modules look like an array, with each entry corresponding to an output channel in the layout. A *DC Source* module can only hold one value. To the firmware, *DC Source* modules look like a single `int` or `float`.

On the output side, one may use either a *Sink* module or a *Triggered Sink* module. Both of these modules can be multichannel and inherit the channel-count based from the properties of the input wire. The *Sink* module's value is updated each time the Layout is processed, while the *Triggered Sink* module is updated whenever the input control wire has a non-zero value.

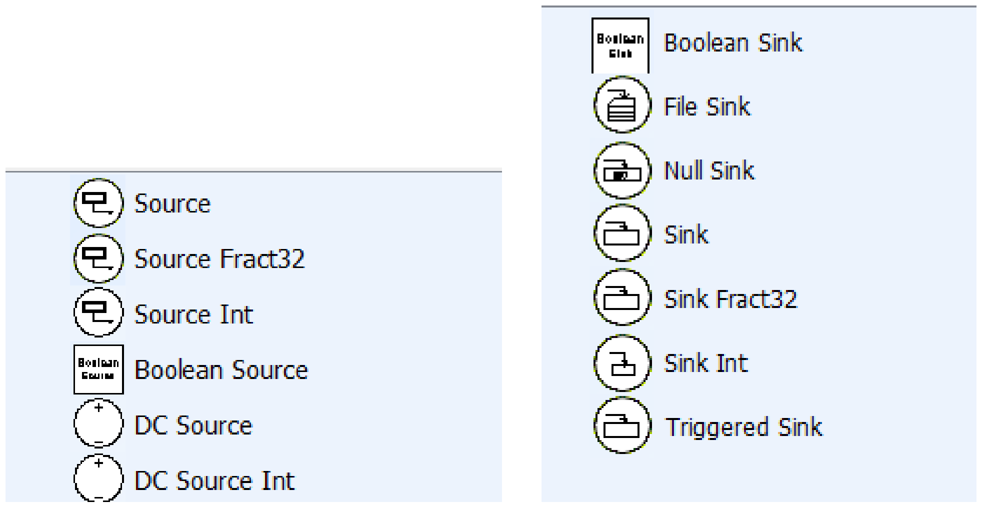


Figure 20. Source and Sink Modules

Assign Static ObjectID's to interface modules

As shown above in Figure 19, each module that will be accessed by the Control Driver is manually assigned a static ***ObjectID*** (in the Build tab of the Properties panel) by the AWE model author**.** The target BSP has pre-assigned ***ObjectIDs*** to specific controls.This assignment must be documented so that the AWE model author will know what ***ObjectID*** to assign for the desired control.

On the application side the ***ObjectID*** for each control is hard-coded in a table in the Control Driver. These ***ObjectIDs*** must be greater than 30000 and less than 40000. The ***ObjectID*** to be assigned to the control object in the Properties panel must match the ***ObjectID*** statically assigned to the control in the Control Driver.

Update your Application to interact with the interface modules

Somewhere in your application, code must be added to fetch pointers to the Layout Control Interface modules and write data in. Control data may then either be updated on a periodic basis in the background thread, or alternatively be updated only when changes occur, e.g. in response to an IPC message from a host processor.

The exchange of control data with the Source and/or Sink modules occurs either in the command processing loop or in a thread running at a lower priority than the real time processing thread. The rational for this is that communication with control peripherals is typically much too slow to run in the real time loop. This becomes obvious when multibyte communication is required as is typical with messaging over I2C or CAN. When data is exchanged between the peripheral and the module the real time processing must be momentatily locked out to maintain the data consistency. This is typically done using semaphores or disable/reenable interrupts.

The Control driver consists of the following three methods:

```
void ControlInit(void);
void InitializeControlPeripheral(UINT32 nControlID);
void ProcessControlIO(void);
```

ControlInit discovers what controls the running model is interacting with and creates a table of pointers to those modules along with any peripheral information associated with that control.

ControlInit calls *InitializeControlPeripheral* to perform any setup need to access that peripheral.

ProcessessControlIO is the method called from the command processing loop to do the actual interaction with the control when the model is running.

```
    // Find any source control modules instantiated in the model
    for (n = 0; n < CTRL_NUM_INPUTS; n++)
    {
        // Does the current AWE model have a source module with this control object ID?
        if (awe_fwGetObjectByID(&g_AWEInstance,
                                InputCtrlID[n], &pInstance, &classID) != OBJECT_FOUND)
        {
            // No such object.
            continue;
        }

        // Check that module assigned this object ID is of module class DC Source
        if (classID != CLASSID_DCSOURCEV2)
        {
            continue;
        }

        // Save the module instance in the control info structure
        g_InputControl[g_nInputControlCount].pModule = (ModClassModule *)pInstance;

        // Save the I/O pin number in the control info structure
        g_InputControl[g_nInputControlCount].nPinNo = InputCtrlPinNo[n];

        g_nInputControlCount++;

        InitializeControlPeripheral(InputCtrlID[n]);
    }

    g_nOutputControlCount = 0;

    // Find any sink control modules instantiated in the model
    for (n = 0; n < CTRL_NUM_OUTPUTS; n++)
    {
       // Does the current AWE model have a sink module with this control object ID?
        if (awe_fwGetObjectByID(&g_AWEInstance,
                                OutputCtrlID[n], &pInstance, &classID) != OBJECT_FOUND)
        {
            // No such object.
            continue;
        }

        // Check that module assigned this object ID is of module class Sink
        if (classID != CLASSID_SINK)
        {
            continue;
        }

        // Save the module instance in the control info structure
        g_OutputControl[g_nOutputControlCount].pModule = (ModClassModule *)pInstance;

        // Save the I/O pin number in the control info structure
        g_OutputControl[g_nOutputControlCount].nPinNo = OutputCtrlPinNo[n];

        g_nOutputControlCount++;
        InitializeControlPeripheral(OutputCtrlID[n]);
    }
/**
 * @brief  Perform control I/O
 */
void ProcessControlIO(void)
{
    UINT32 nPinValue;
    UINT16 n;

    // Process any source control modules
    for (n = 0; n < g_nInputControlCount; n++)
    {
        // Get the AWE module instance for this source control
        awe_modDCSourceV2Instance * pModule =
                                   (awe_modDCSourceV2Instance *)g_InputControl[n].pModule;

        // Which GPIO pin is this control attached to
        UINT32 nPinNo = g_InputControl[n].nPinNo;

        // Get the current state of the control
        awe_pltGPIOGetPin(nPinNo, &nPinValue);

        // Update the AWE module state
        pModule->value = (FLOAT32)nPinValue;
    }

    // Process any sink control modules
    for (n = 0; n < g_nOutputControlCount; n++)
    {
        // Get the AWE module instance for this sink control
        awe_modSinkInstance * pModule = (awe_modSinkInstance *)g_OutputControl[n].pModule;

        // Which GPIO pin is this control attached to
        UINT32 nPinNo = g_OutputControl[n].nPinNo;

        // Set to 1 if greater than 0 otherwise set to 0
        nPinValue = (UINT32)(*(pModule->value) ) > 0 ? 1 : 0;

        // Change the ouput state of this control
        awe_pltGPIOSetPin(nPinNo, nPinValue);
    }

} // End ProcessControlIO
```

Figure 21. Example code that writes control data into the Layout

6. Global Variables and Data Structures

An application that integrates the AWE Core must define a number of global variables and data structures.

6.1. Memory Heaps

The AWE Core has its own memory manager that dynamically allocates memory for audio processing modules and wires. There are 3 different memory heaps that modules will request in their constructors, two internal and one external. This allows zero-overhead, dual-data fetches on some architectures, with coefficients being stored in one internal heap and samples being stored in the other.

Modules typically specify a priority of memory heaps. For example, allocate from the internal DM heap. If it is full, allocate from the internal PM heap. If it is full, allocate from external memory. This overflow behavior lends itself to processors that only have two different memory types (e.g., internal and external) or to cache based architectures where no distinction is made between memory types.

```
AWE_FW_SLOW_ANY_CONST UINT g_master_heap_size = MASTER_HEAP_SIZE;
AWE_FW_SLOW_ANY_CONST UINT g_slow_heap_size = SLOW_HEAP_SIZE;
AWE_FW_SLOW_ANY_CONST UINT g_fastb_heap_size = FASTB_HEAP_SIZE;

section("awe_heap_fast")
UINT g_master_heap[MASTER_HEAP_SIZE];

#pragma section("awe_heap_slow",NO_INIT)
UINT g_slow_heap[SLOW_HEAP_SIZE];

section("awe_heap_fastb")
UINT g_fastb_heap[FASTB_HEAP_SIZE];
```

6.2. Module Table

The module table specifies which modules in the AWE Core library the linker will include during building, thus making them available in the executable for dynamic allocation. The module table is an array of pointers to module class objects.

```
AWE_FW_SLOW_ANY_DATA
const ModClassModule *g_module_descriptor_table[] =
{
    LISTOFCLASSOBJECTS
};

AWE_FW_SLOW_ANY_DATA
UINT g_module_descriptor_table_size =
sizeof(g_module_descriptor_table) /
sizeof(g_module_descriptor_table[0]);
```

Figure 22. Module Table

This macro, LISTOFCLASSOBJECTS, is defined in **ModuleList.h** as shown in Figure 23.

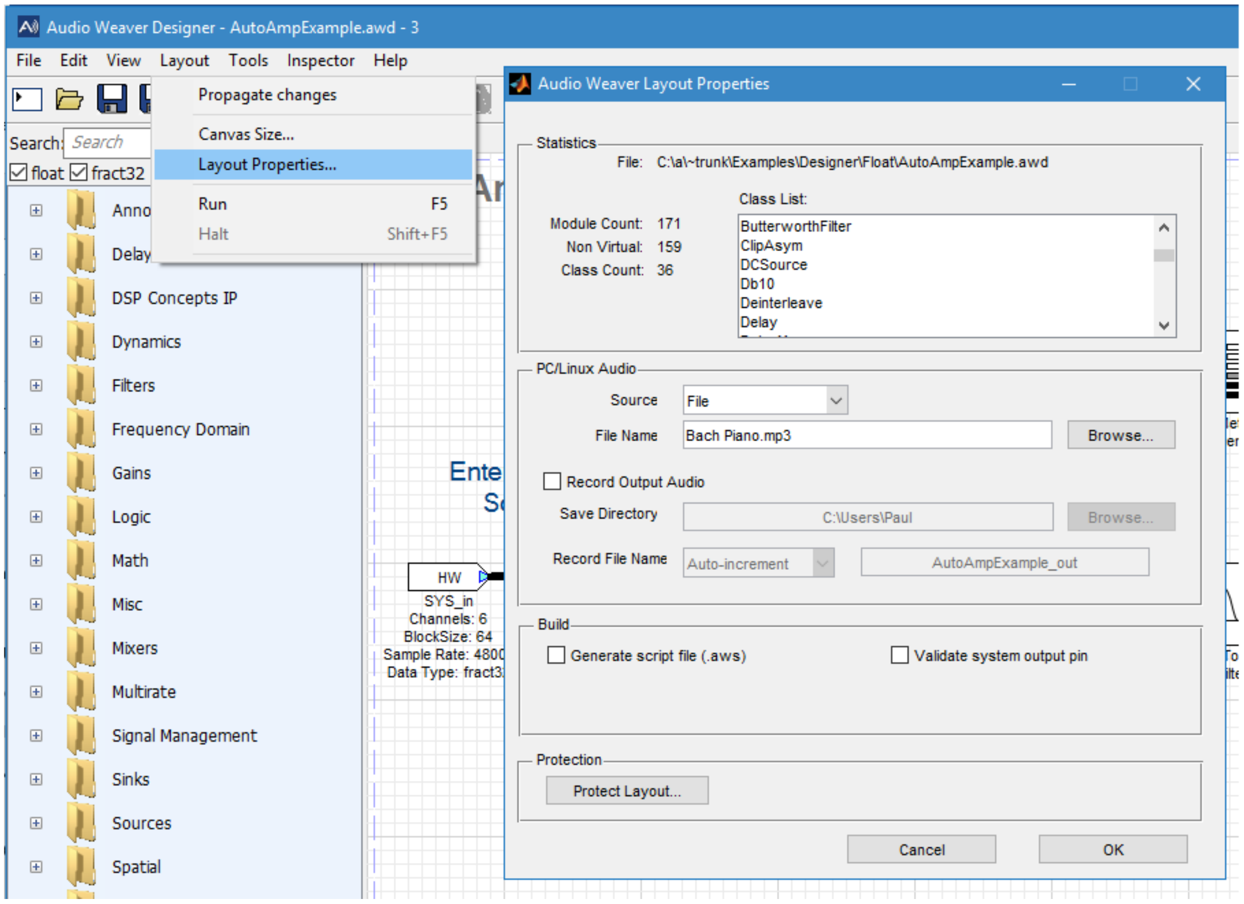
```
#define LISTOFCLASSOBJECTS \
&awe_modAGCAutoAttackReleaseClass, \
&awe_modAGCCoreClass, \
&awe_modAGCCoreARClass, \
&awe_modAGCGainComputerClass, \
&awe_modAGCLimiterCoreClass, \
...
```

Figure 23. List of Module Classes

6.3. Optimizing Memory Usage

By default the AWE Core enables all modules available on a given processor. This means that memory will be used to allow instantiation of modules that will never be used. Should you need to optimize memory usage, the simple way to do so is to modify the LISTOFCLASSOBJECTS macro in TargetInfo.h.

To make modules unavailable in your executable, simply remove them from this Macro. At build time, with Linker-elimination enabled, they will not be built into the executable. Note: The list of modules used in a given layout can be seen in AWE Designer by selecting the *Layout / Layout Properties*, as shown in Figure 24.



Statistics
File: C:\a\~trunk\Examples\Designer\Float\AutoAmpExample.awd
Module Count: 171
Non Virtual: 159
Class Count: 36
Class List:
ButterworthFilter
ClipAsym
DCSource
Db10
Deinterleave
Delay

PC/Linux Audio
Source: File
File Name: Bach Piano.mp3
Record Output Audio
Save Directory: C:\Users\Paul
Record File Name: Auto-increment AutoAmpExample_out

Build
Generate script file (.aws)
Validate system output pin

Protection
Protect Layout...

Cancel OK

Figure 24. Module Classes used in a layout

6.4. Target Info

The application maintains a master include file named “TargetInfo.h” that defines the capabilities of the target system. This file contains firmware version information, device capabilities, heap sizes and which AWE modules are supported. When the Server connects, it reads this data structure and displays the information in the output window.

```
/* ------------------------------------------------------------------
** Target Definition
** --------------------------------------------------------------- */

// Version Information
#define VER_DAY 02
#define VER_MONTH 4
#define VER_YEAR 17

#define CORE_ID 0
#define CORE_SPEED 168e6f
#define SAMPLE_SPEED 168e6f
#define HAS_FLOAT_SUPPORT 1
```

```
#define HAS_FLASH_FILESYSTEM 1
#define NO_HW_INPUT_PINS 1
#define NO_HW_OUTPUT_PINS 1
#define IS_SMP 0
#define NO_THREADS_SUPPORTED 2
#define FIXED_SAMPLE_RATE 48000.0f
#define IS_COMPLEX 0
#define SAMPLE_SIZE_IN_BYTES 4

/* ------------------------------------------------------------------
** Specifies the sizes of each of the heaps on the target
** ---------------------------------------------------------------- */
#define MASTER_HEAP_SIZE          (1024*18)
#define SLOW_HEAP_SIZE            (1024*16)
#define FASTB_HEAP_SIZE           (1024*32)

#define MAX_COMMAND_BUFFER_LEN 272

// Audio Weaver modules
extern const ModClassModule awe_modAGCCoreClass;
extern const ModClassModule awe_modAGCLimiterCoreClass;
```

6.5. Communication Buffer

The communication buffer is an array of 32-bit unsigned integers. The Platform essentially owns the communication buffer. The Platform allocates the buffer, receives messages over the tuning interface, calls the Framework communication handler to process the message, and then returns the resulting message back to the PC or host.

```
AWE_FW_SLOW_ANY_DATA DWORD s_PacketBuffer[MAX_COMMAND_BUFFER_LEN];
```

6.6. Memory Sections

The Audio Weaver module libraries and Framework code utilize a large number of memory sections. The memory sections are defined using macros in Framework.h and appear within the Linker Definition File (LDF). The sections related to code placement:

Macro	Section Name	Description
AWE_MOD_SLOW_CODE	awe_mod_slowcode	Non time-critical code used by the audio modules.
AWE_MOD_FAST_CODE	awe_mod_fastcode	Time-critical code used by the audio modules.
AWE_FW_SLOW_CODE	awe_fw_slowcode	Non time-critical Framework code.
AWE_FW_FAST_CODE	awe_fw_fastcode	Time-critical Framework code.

There are 24 memory sections related to data placement. The naming convention used by the macros is:

```
AWE_{MOD/FW}_{SLOW/FAST}_{PM/DM/ANY}_{DATA/CONST}
```

There are 2 x 2 x 3 x 3 = 24 different permutations and each subfield has a specific meaning:

MOD/FW – specifies whether the section applies to modules (MOD) or to the Framework (FW).

SLOW/FAST – specifies the time criticality of the memory section. SLOW memory is referenced infrequently or from non-real-time code. It is suitable to be placed in external memory with little loss of performance. FAST memory is time-critical and should be placed internally. If internal placement of a FAST section is not possible, then there will be a performance penalty.

PM/DM/ANY – specifies internal memory placement on the SHARC. PM and DM refer to two distinct internal memory blocks and enable parallel memory accesses. Traditionally, these are called "PM" and "DM" spaces on the SHARC. More recent processors with more than 2 internal memory blocks require placement in distinct blocks. ANY indicates that placement in PM or DM is unimportant; the section can go anywhere.

DATA/CONST – DATA sections must be placed in RAM and hold variables that are being updated. CONST sections store initialized constant data and are suitable for inclusion into ROM.

The corresponding section names (which must appear within the LDF) are lower case. For example, AWE_MOD_SLOW_DM_DATA is placed in the section awe_mod_slowdmdata.

7. Framework Prerequisites

We are often asked about integrating the Audio Weaver Core into an existing real-time framework. This section lists the features and capabilities required of a framework to support Audio Weaver. You can think of this information as a summary of the requirements and APIs previously listed.

7.1. Basic Processor and Board Initialization

1. Processor can boot from NVRAM
2. PLL and all clocking subsystems initialized
3. Memory and Cache's configuredAudio Subsystem working
4. Audio drivers implemented (TMD/I2s serial ports configured with proper clock

polarities, etc)

5. DMA channels configured, with ping-pong buffers for DMA RX and TX

6. If there are multiple audio peripherals, all serial ports and DMA should be started synchronously

7. Audio devices (CODECs, ADCs, DACs, network interfaces) configured and verified to be passing clean audio

8. Proper sample rates, clock polarities, etc

9. CODEC gains, filters, and mixers have to be properly configured

10. Communications subsystem working

11. Communication drivers (SPI, USB, RS232, TCP/IP, etc) configured and verified to be exchanging clean communications data with Windows PC.

12. Proper baud rates, clock polarities, etc

7.2. Simple “Audio Passthru” Application

To ensure we're integrating the AWE Core into a known-good environment, it's required that a simple "Audio Passthru" application be functional. This "hello world" application should simply pass audio from input to output, demonstrating that the PCB, audio peripherals, components, and baseline configuration are all functional and stable.

1. CODEC (etc) configured properly

2. I2S/TDM serial port I/O is serviced by double buffered ("ping-pong"), block based DMA. (The DMA needs to run in the background without needing processor intervention.)

3. The DMA must generate an interrupt at the end of each block transfer. (Polling for the completion of a block transfer in the background/main loop won't work. The main loop is reserved for non-real-time critical operations.)

4. In the DMA Interrupt handler, ping-pong buffers are toggled, and audio is simply copied from RX to TX buffer

At this point, the audio I/O APIs described in Section 4 can be called from the DMA interrupt handler.

7.3. Simple “Echo-Back” Application

Tuning commands (and replies) are exchanged in realtime between Audio Weaver Designer on the PC and the AWE Core library on the embedded target. This communication can happen over a number of different protocols, listed below in order of throughput (higher bandwidth interfaces provide a better user experience).

1. Ethernet - standard TCPI/P interface.
2. USB - HID device
3. SPI - custom 32-bit protocol
4. UART - custom byte-based protocol

Similar to the requirements for the Audio subsystem, the example application should also include 'Echo-back' functionality on the chosen communications device. Here, each byte/word/packet received should be immediately transmitted back to the PC, demonstrating that the PCB, comms peripherals and components and baseline-configuration are all functional and stable.

- Comm peripheral is enabled and interrupts are triggered on a byte/word/packet boundary (depending on protocol)
- In the Comm Interrupt handler, RX data should be copied to the TX buffer to be sent back to PC.
- Real-time Audio and Communications I/O must be able to happen at the same time.
- The tuning interface should be low latency. Ideally, we like a round trip message time of < 5 msec. (This latency is a soft requirement and mainly affects user-experience; Audio Weaver Designer uses this channel to facilitate module "inspectors", and some, such as meters, tuning knobs and sliders, etc., may be less than smooth if there's a communications bottle-neck.)

7.4. Basic Interrupts and Threading Configured

Systems with the AWE Core integrated typically have code that executes at four different priorities. (For more details see Section 2.6 of this document.) These tasks may be run as actual threads in an RTOS, or simply interrupt-handlers in a bare metal implementation. From highest to lowest priority, these threads are typically as follows:

Highest	Communication I/O - Tuning data is simply moved between the RX/TX buffers and the AWE Core's message-queue (where it's processed at a lower priority).
	Audio I/O - Audio samples, in blocks of 32 e.g., are moved between the DMA RX/TX buffers and the AWE Core's input/output pins.

	Audio Processing - In this thread the Audio Weaver signal processing Layout is executed.
Lowest	Background - In this thread, non-realtime tasks are handled. Audio Weaver Tuning commands are executed and the general machinery of the AWE Core is maintained.

7.5. Optional: Flash memory

To integrate the Embedded File Manager, we need to have the following functions implemented:

- Flash Read
- Flash Write
- Flash Erase