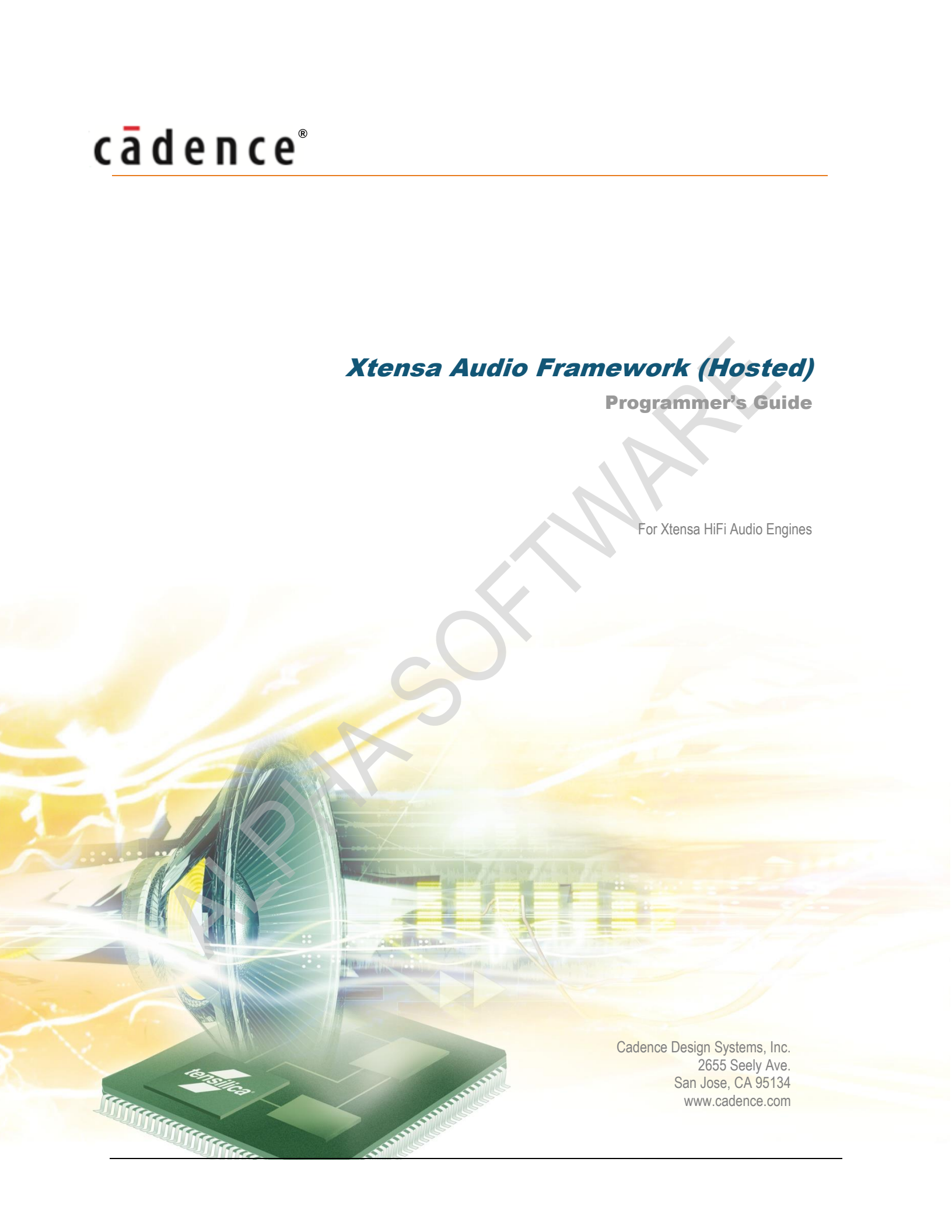
cadence®

Xtensa Audio Framework (Hosted)

Programmer's Guide

For Xtensa HiFi Audio Engines



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Document Change History

Version	Changes
0.6	Alpha release

1. Introduction to Xtensa Audio Framework

Xtensa Audio Framework (XAF) is a framework designed to accelerate the development of audio processing applications for the HiFi family of DSP cores. Application developers may choose components from the rich portfolio of audio and speech libraries already developed by Cadence and its ecosystem partners. In addition, customers may also package their proprietary algorithms and components and integrate them into the framework. Towards this goal, a simplified "Developer API" is defined, which enables application developers to rapidly create an end application and focus more on using the available components. XAF is designed to work on both the instruction set simulator as well as actual hardware.

The version of XAF described in this guide is designed to work on actual hardware with Host CPU and HiFi DSP (that is, a "hosted" solution). Note, this version of XAF is ported and tested on HiKey960 Board where Host CPU runs Android operating system.

XAF is part of the "HiFi Integrator Studio" suite of tools.

1.1 Document Overview

This guide covers all the information required to create, configure, and run audio processing chains using XAF developer APIs. Section 2 briefly describes the XAF architecture, and Section 3 provides details about developer APIs available for the application developer. Section 4 provides details about building and running a sample application, which illustrates usage of the developer APIs. Section 5 provides a "How To" guide for adding support for a new component in XAF.

Instructions on how to run the XAF on the HiKey960 Board have been provided in a companion document [1].

1.2 Xtensa Audio Framework Specifications

This section provides XAF specifications, including the operating system.

1.2.1 Terminology

The following terms are used within this guide.

Audio Device: The software abstraction of a digital signal processor (DSP) core.

Component: A software module that conforms to a specified interface and runs on the audio device. It would implement some audio processing functionality.

Port: An interface through which a component can connect to other components and exchange data. Each port may be connected to only one port of another component. A component must have at least one port.

Input Port: A port through which a component can receive data from another component. A component may have 0 or more input ports.

Output Port: A port through which a component can send data to another component. A component may have 0 or more output ports.

Link: The connection between the output port of one component and the input port of another component.

Buffer: Memory block containing data that is transferred over a link between two ports.

Chain: A graph formed by connecting different components by links.

Framework: A software entity that enable the creation of an audio processing chain. It manages the transfer of buffers between ports as well as the scheduling of the different components in the chain.

Application: A software entity that uses the framework to create a chain. It is the responsibility of the application to provide input data to the chain and consume the output data generated by the chain.

Figure 1-1 shows the terms above in a diagrammatic form, with an example chain.

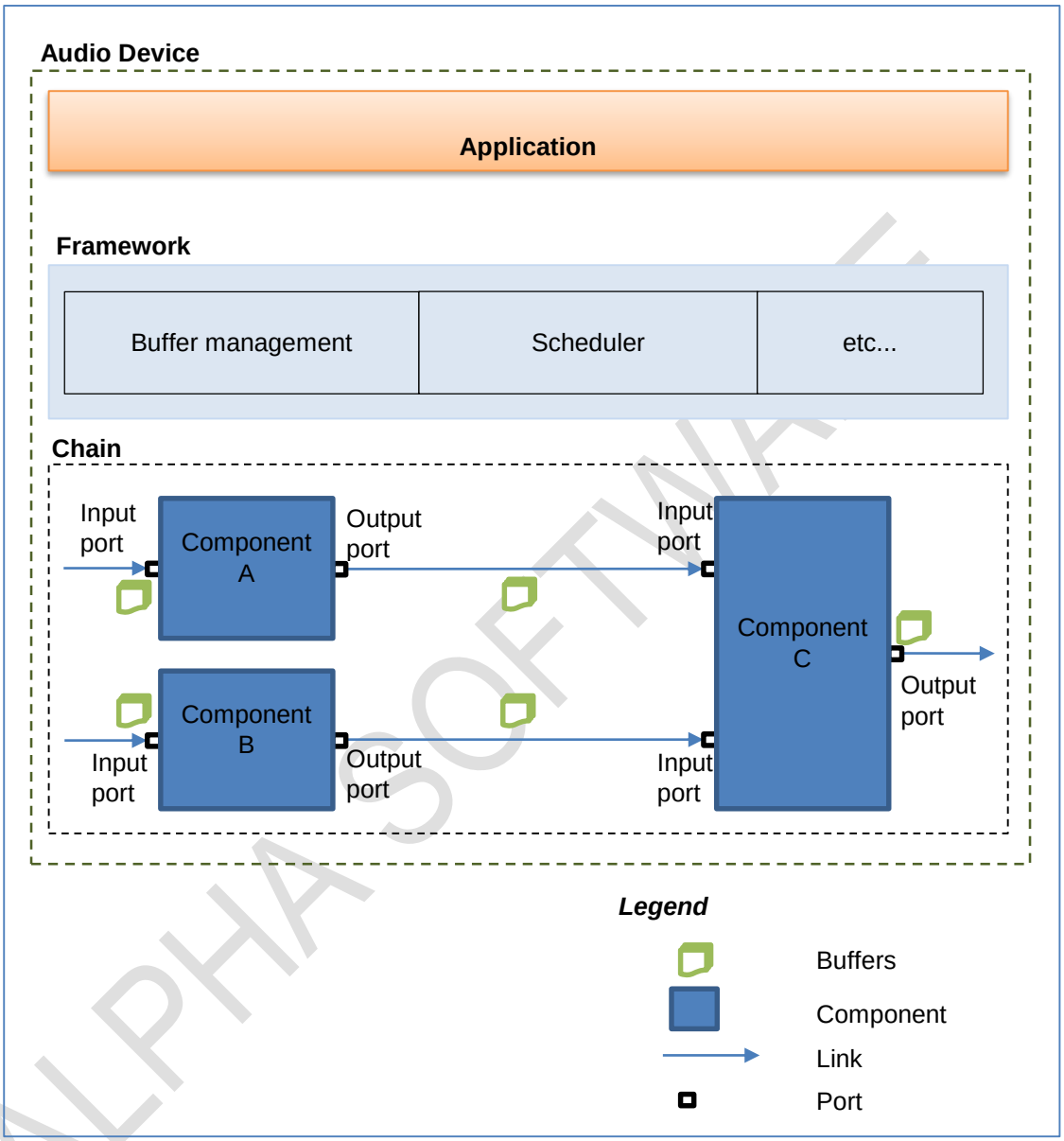


Figure 1-1 Terminology

1.2.2 Feature Set

API features:

Ability to create components and connect them in a chain.

Ability to read and write component configuration parameters.

Ability to read component status and trigger component processing.

XAF features:

Manages the scheduling of components in the chain. No explicit restriction on the complexity of the component chain, i.e., the number of components/links is restricted by the memory/MHz resources available and not by XAF.

Manages the allocation of memory for data buffers. Number of data buffers is 0, 1, or 2 for input ports and 0 or 1 for output ports. The number of buffers on a link between two ports can be increased at component connection stage.

Manages the allocation of memory for itself and created components. Dynamic memory allocation within XAF is done through an allocation function registered by the application. This allows the application to control the memory type/region for the allocation.

Manages the data transfer between components. The buffering of data to match the different block sizes between two connected components is also managed by XAF. As XAF merely transfers the data between components, there is no restriction on the actual format of the data.

Component types supported (See Table 1-1), depending on the number of ports and the type of data transferred across the ports (PCM or non-PCM).

Table 1-1 Component types

Component Type	Input		Output		Component Description
	Ports	PCM	Ports	PCM	
Decoder	1	N	1	Y	Decodes input compressed data to generate output PCM data.
Encoder	1	Y	1	N	Encodes input PCM data to generate output compressed data.
Mixer	4	Y	1	Y	Combines input PCM data from multiple ports to generate one output PCM data.
Pre-processing	1	Y	1	Y	Pre-processes input PCM data to generate output PCM data.
Post processing	1	Y	1	Y	Post-processes input PCM data to generate output PCM data.

Package features:

One example test application provided to demonstrate various use-cases.

Example code to demonstrate the integration of two Cadence audio libraries (Ogg Vorbis decoder, PCM gain library) into XAF is included in this package. Note that some of these libraries *may* need to be licensed separately.

Compile time flags provided to support detailed analysis and debugging

- Tracing: Enables printing of detailed component scheduling information (Set `TRACE = 1` while building, disabled by default).
- Debugging: Disables optimization and enables debugger information (Set DEBUG = 1 while building, disabled by default).
- Profiling: Measurement of component and XAF MHz. (Set XAF_PROFILE = 1 while building, enabled by default).

Support for HiKey960 Board[1] along with detailed instructions on board setup.

RTOS:

XAF requires a real-time operating system (RTOS) on Host CPU for multithreaded execution, Inter Processor Communication (IPC) interface, mutual exclusion for accessing shared resources, etc. This version of XAF supports Android [1] (only) as the RTOS. Support for other RTOS variants will be added in future versions.

Limitations:

Code for the components needs to be linked into the application. Dynamic loading of code is not supported.

API enhancements to disconnect components from a graph and to pause/resume the graph are planned in future releases.

Only one instance of XAF can run at a time.

Known Issues:

1.3 Xtensa Audio Framework Performance

The performance was characterized on the 5-stage Audio Engine processor cores. The memory usage and performance figures are provided for design reference.

1.3.1 Memory

Table 1-2 Library Memory

XAF Module	Text (Kbytes)	RODataa (Kbytes)
HiFi3 DSP Firmware Library	39.5	9.0
ARM64 XAF Host Library	20.4	0.7

Note Above memory measurements are for XAF on HiKey960 Board set up.

Note The HiFi3 DSP Firmware Library measurements were done with Version 7.0.5 of the Xtensa tool chain and it excludes memory required by audio components (i.e. Ogg Vorbis Decoder, PCM Gain and Mixer components).

The total runtime memory allocated can be divided into two categories:

1. Local memory used by XAF on AP: This memory is allocated by AP for XAF data structures.
2. Local memory used by XAF on DSP: This memory is allocated by DSP for usage by audio components.
3. Shared memory between AP and DSP for IPC: This memory is shared by AP and DSP for Inter Processor Communication (IPC) between AP and DSP (i.e. for sharing messages and audio data).

Table 1-3 shows the runtime memory allocated by XAF for a simple processing chain.

Table 1-3 Runtime Memory

No	Memory breakup	RAM (Kbytes)
		HiFi 3
1	Local memory used by XAF on AP	4
2	Local memory used by XAF on DSP	1030
3	Shared memory between AP and DSP for IPC	256
	Total memory	1290

Note Above memory measurements are for XAF on HiKey960 Board set up.

Note 'Local memory used by XAF on AP' mentioned above is for use cases demonstrated by xaf-dec-test and xaf-dec-mix-test.

Note 'Local memory used by XAF on DSP' and 'Shared memory between AP and DSP' are allocated to higher values for XAF on HiKey960 Board set up. To compute exact sizes required by any audio processing chain, please refer to Appendix 7.

Note The HiFi3 DSP measurements were done with Version 7.0.5 of the Xtensa tool chain.

1.3.2 Timings

Table 1-4 contains details for the MCPS usage by the audio processing chain on DSP. The "Average CPU Load" MCPS are the total MHz consumed by the DSP for executing the use case through XAF.

Table 1-4 MCPS

Testbench	Use Case	MCPS for	Average CPU Load (MHz)
			HiFi 3
xaf-dec-test	Ogg-Vorbis Decoder (Stereo, 48 KHz, 16 bits/sample)	XAF DSP	46
xaf-dec-test	PCM gain (Stereo, 48 KHz, 16 bits/sample)	XAF DSP	14
xaf-dec-mix-test	Ogg-Vorbis Decoder (2 instances) + Mixer (Stereo, 48 KHz, 16 bits/sample)	XAF DSP	83
xaf-dec-mix-test	PCM Gain (2 instances) + Mixer (Stereo, 48 KHz, 16 bits/sample)	XAF DSP	42

Note Above memory measurements are for XAF on HiKey960 Board set up.

Note The measurements were done with Version 7.0.5 of the Xtensa tool chain.

2. Xtensa Audio Framework Architecture Overview

2.1 Xtensa Audio Framework Building Blocks

The following figure shows various building blocks of applications based on XAF. Note that in this figure the gray blocks are not part of XAF.

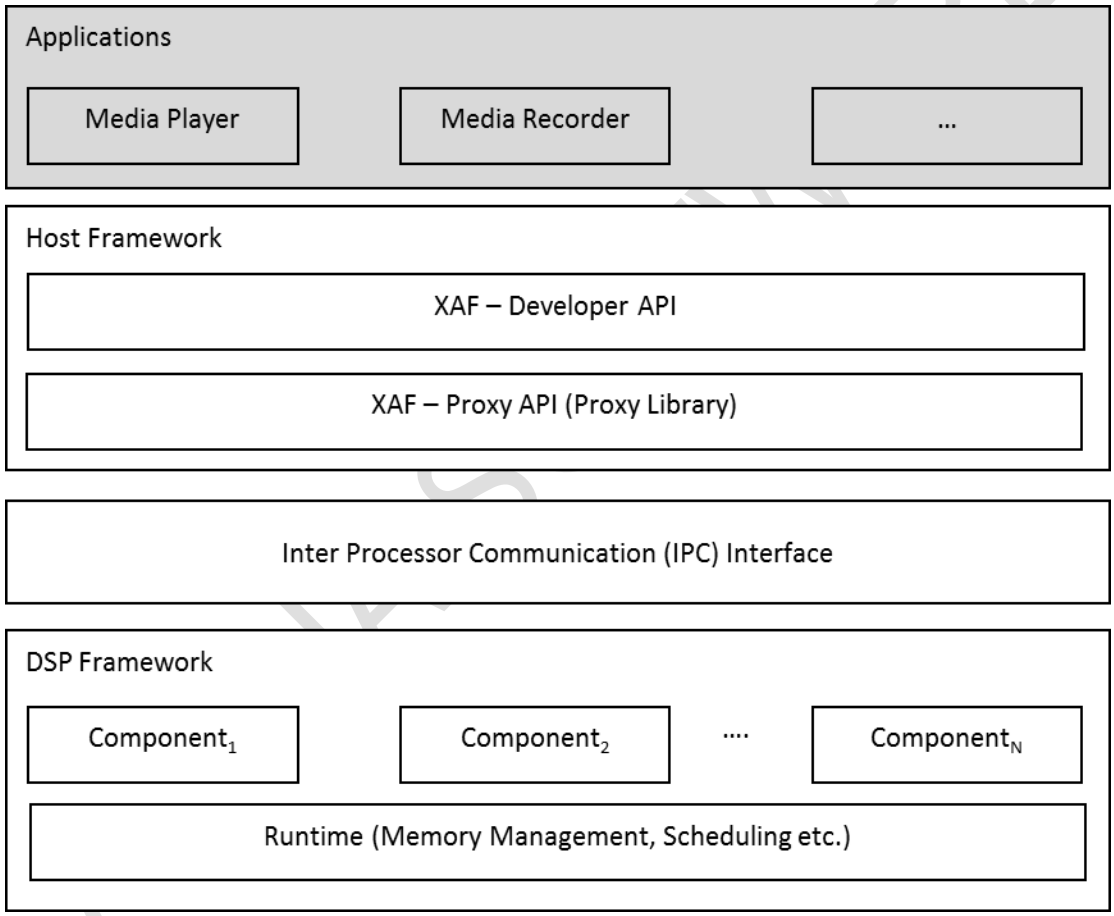


Figure 2-1 Software Stack Diagram

2.1.1 Applications

In this application space, an application developer will leverage the developer API to create a processing chain and Host CPU Operating System (Android) to handle multiple threads. As such, XAF enables chains to be set up and configured.

The developer API is the interface between the Application space and XAF.

Note that XAF allows an unlimited number of components in the audio processing chain — the limitation is only from the system hardware. The system developer must ensure that there is enough memory and CPU bandwidth available on the hardware.

2.1.2 Host Framework

Host framework implements Developer API and underlying Proxy library. Host framework is responsible for sending the commands to DSP framework and receive responses from DSP framework.

2.1.3 DSP Framework

This block interfaces between the host framework and the audio processing components. It also performs all memory management for audio processing chains. It receives commands from applications and sends the responses back.

It also manages the data buffers between each individual audio component. Note that each audio component running in the audio processing chain may consume and produce a different amount of data. The decoder component from Cadence consumes one encoded frame and produces one output PCM frame in one execution call. For example, the MP3 decoder from Cadence may consume 2048 bytes (largest encoded frame size) and produce one output PCM frame of 4608 bytes (1152 samples, stereo channel, 16-bit data). Post process components from Cadence generally consume and produce a variable block size (from a pre-defined set) in one execution call.

2.1.4 IPC Interface

Inter Processor Communication (IPC) interface is hardware specific module and is responsible for communication between Host CPU and HiFi DSP with appropriate interrupts.

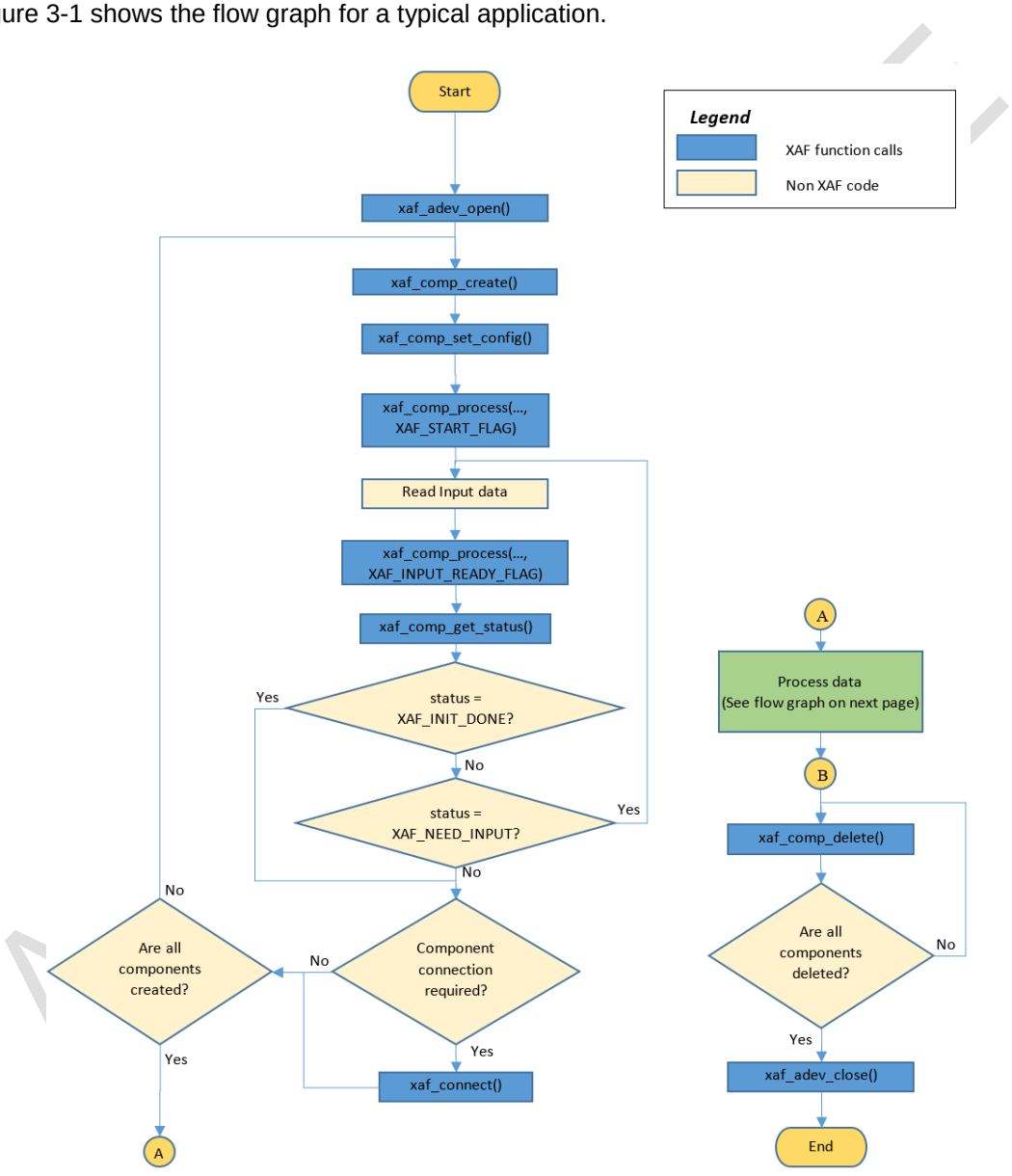
2.1.5 Audio Components

Audio components, used by the Applications, can be connected to form chains with a cascade or parallel interface. Except for the mixer, each audio component type can have one input and one output stream. The Mixer Component type can have up to four input streams and one output stream. Section 5 contains details on how to add a new component.

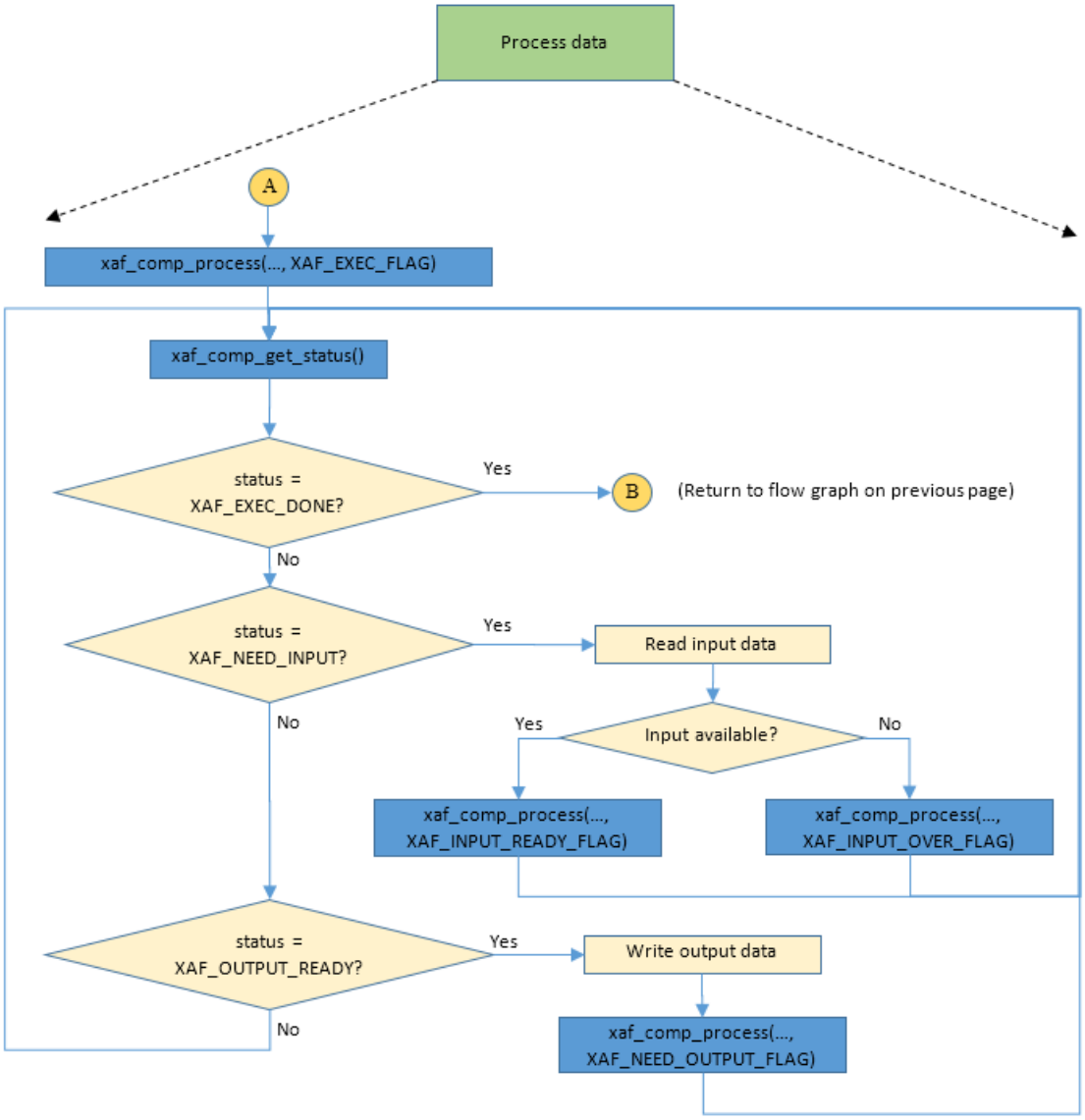
3. Xtensa Audio Framework Developer APIs

This section discusses XAF developer APIs that are available for the application programmer to create, configure and run audio processing chains.

Figure 3-1 shows the flow graph for a typical application.



(a) Flowgraph sequence for API calls of testbench



(b) Flowgraph sequence for API calls for each input and output component in the graph

Figure 3-1 Flowgraph Sequence for API Calls

Following is a brief description of the flowgraph sequence:

Initialize XAF: The XAF is initialized by calling `xaf_adev_open`. The framework memory allocation is performed at this stage.

Create Processing Chain: The various components in the chain are instantiated by calling `xaf_comp_create` for each component. Then, the component configuration parameters (if any) are set using `xaf_comp_set_config`. Finally, the components are connected together using `xaf_connect`.

Process data: Input and output data is passed to the components using `xaf_comp_process`. This must be performed only for components that need to be supplied with input/output data (typically the edge components of the chain). The component status should be queried using `xaf_comp_status`. This stage continues until all the data has been processed.

Delete Processing Chain: The various components of the chain are deleted by calling `xaf_comp_delete`.

Terminate XAF: The XAF is terminated by calling `xaf_adev_close`. The memory allocated by the framework is freed at this stage.

3.1 Files Specific to Developer APIs

Developer API Header File (/include/)

`xaf-api.h`

3.2 Developer API-Specific Error Codes

The errors in this section can result from the developer API layer of the Xtensa Audio Framework. All errors are fatal (unrecoverable) errors. In response to an error, the function `xaf_adev_close(p_adev, 0)` may be called to close the device and release resources used by XAF.

3.2.1 Common API Errors

XAF_PTR_ERROR

This error indicates that a null pointer was passed to the developer API where a valid pointer was expected.

XAF_INVALID_VALUE

This error code indicates that an invalid value (out of valid range) was passed to the developer API.

XAF_API_ERR

This error code indicates a developer API call sequence error, for example, `xaf_comp_create()` is called before `xaf_adev_open()`.

3.2.2 Specific Errors

The following error is specific to some APIs.

XAF_ROUTING_ERROR

This error code indicates that the developer API was unable to connect `(xaf_connect() call)` the two requested components.

3.3 Developer APIs

This section contains tables describing the developer APIs.

Table 3-1 `xaf_adev_open` API

API	`XAF_ERR_CODE xaf_adev_open(pVOID *p_adev,`
	`WORD32 audio_frmwk_buf_size,`
	`WORD32 audio_comp_buf_size,`
	`xaf_mem_malloc_fxn_t mem_malloc,`
	`xaf_mem_free_fxn_t mem_free)`

Description	This API opens and initializes the audio device structure. It also memory maps shared memory for the framework.

Actual Parameters	`p_adev`
	Pointer to audio device handle which will be returned with the valid pointer to audio device structure.
	
	`audio_frmwk_buf_size`
	Unused in this release.
	
	`audio_comp_buf_size`
	Unused in this release.
	`mem_malloc`
	Function pointer to testbench implementation of memory allocation. The 'id' indicates whether the memory is allocated for device (`DEV_ID`) or for component (`COMP_ID`).
	`pVOID mem_malloc(WORD32 size, WORD32 id);`
	
	Note: XAF expects that `mem_malloc` should return a 4-byte aligned address.
	
	`mem_free`
	Function Pointer to testbench implementation of memory free
	`VOID mem_free(pVOID ptr, WORD32 id);`

Restrictions	Only one instance of XAF can run at a time.

Example

```
ret = xaf_adev_open(&p_adev,
                    0,
```

```
0,
&mem_malloc,
&mem_free);
```

Errors

Common API Errors

Table 3-2 `xaf_adev_close` API

API	`XAF_ERR_CODE xaf_adev_close(pVOID p_adev, xaf_comp_flag flag)`
Description	This API closes the Audio Device and frees up allocated memory.

Actual Parameters	`p_adev`
	Pointer to the audio device
	`flag`
	Unused in this release.

Restrictions	Should not be called before `xaf_adev_open` API. All components must be deleted before closing the audio device.

Example

```
ret = xaf_adev_close(p_adev, 0);
```

Errors

Common API Errors

Table 3-3 `xaf_comp_create` API

API	`XAF_ERR_CODE xaf_comp_create(pVOID p_adev,`
	`pVOID *p_comp,`
	`xf_id_t comp_id,`
	`UWORD32 ninbuf,`
	`UWORD32 noutbuf,`
	`pVOID pp_inbuf[],`
	`xaf_comp_type comp_type)`

Description	This API creates the Audio Component. The component is identified by `comp_id` and `comp_type`. You can specify the number of input and output buffers for the component. The IO buffer requirement is dependent upon the position of the component in the audio processing chain – see the parameter description below for details.

Actual Parameters	`p_adev`
	Pointer to the audio device structure
	
	`p_comp`
	Pointer to the audio component structure (should be one of the available audio components). This pointer will be allocated and returned with a valid pointer to allocated component structure.
	
	`comp_id`
	Component Identifier string. e.g. “mixer”, “audio-decoder/mp3”, etc. It should match with `class_id`’s defined under the constant definition of `xf_component_id` in `xa-factory.c` file. (Refer to Section 5.2, Step 6)
	
	`ninbuf`
	Unsigned integer containing the number of input buffers. This is the number of buffers that the testbench needs to pass to the component. For components connected in the chain where it receives input from other components, this must be configured as zero (0).
	Valid values: 0, 1, 2.
	
	`noutbuf`
	Unsigned integer containing the number of output buffers. This is the number of buffers that the component passes to the testbench as output. For components connected in the chain where the output is passed to another component, this must be configured as zero (0).
	Valid values: 0, 1.
	
	`pp_inbuf`
	Pointer to the array to hold `ninbuf` input buffer addresses that have been allocated within XAF. If the pointer is NULL, the input buffer addresses will not be returned.
	
	`type`
	Type of audio component
	Following are valid values:
	Type — **Description**
	`XAF_DECODER:` — Decoder component
	`XAF_ENCODER:` — Encoder component
	`XAF_MIXER:` — Mixer component
	`XAF_PRE_PROC:` — Preprocessing component
	`XAF_POST_PROC:` — Post processing component

Restrictions	Should not be called before `xaf_adev_open`

Example

```
ret = xaf_comp_create(p_adev,
                      &p_audioComp,
                      "comp_id",
                      N_INP_BUFF,
                      N_OUT_BUFF,
                      & inbuf[0],
                      XAF_POST_PROC);
```

Errors

Common API Errors

Table 3-4 `xaf_comp_delete` API

API	`XAF_ERR_CODE xaf_comp_delete(pVOID p_comp)`
Description	This API deletes the Audio Component and frees the memory associated with it.
Actual Parameters	`p_comp` Pointer to the audio component structure
Restrictions	Should not be called before `xaf_comp_create`

Example

```
ret = xaf_comp_delete(p_audioComp);
```

Errors

Common API Errors

Table 3-5 `xaf_comp_set_config` API

API	`XAF_ERR_CODE xaf_comp_set_config(`
	`pVOID p_comp,`
	`WORD32 num_param,`
	`pWORD32 p_param)`

Description	This API sets (writes) configuration parameters to the Audio Component.`num_param` provides the number of configuration parameters to be set.`p_param` points to an array containing ID/value pairs for all `num_param` parameters. For example, for two parameters, `p_param` will contain ID1, VAL1, ID2, VAL2.

Actual Parameters	`p_comp`
	Pointer to the audio component structure
	
	`num_param`
	Integer containing the number of parameters to be set. The maximum limit is 16.
	
	`p_param`
	Pointer to an integer array containing ID/Value pairs – i.e., parameter ID followed by parameter value.

Restrictions	Should not be called before `xaf_comp_create`

Example

```
ret = xaf_comp_set_config(p_comp,
                          N_PARAMS,
                          &param[0]);
```

Errors

Common API Errors

Table 3-6 `xaf_comp_get_config` API

API	`XAF_ERR_CODE xaf_comp_get_config(`
	`pVOID p_comp,`
	`WORD32 num_param,`
	`pWORD32 p_param)`

Description	This API gets (reads) configuration parameters from the Audio Component.`num_param` provides the number of configuration parameters to get.`p_param` points to an array containing ID/value pairs for all `num_param` parameters. For example, for two parameters, `p_param` will contain ID1, VAL1, ID2, VAL2. VAL1 and VAL2 can contain any arbitrary value, as they will be over-written when the function returns. Upon successful execution of this API, the value field of the ID/value pair will be set to the correct value.

Actual Parameters	`p_comp`
	Pointer to the audio component structure
	
	`num_param`
	Integer containing the number of parameters to get. The maximum limit is 16.
	
	`p_param`
	Pointer to an integer array containing ID/Value pairs – i.e., parameter ID followed by parameter value.

Restrictions	Should not be called before `xaf_comp_create`

Example

```
ret = xaf_comp_get_config(p_comp,
                          N_PARAMS,
                          &param[0]);
```

Errors

Common API Errors

Table 3-7 `xaf_connect` API

<table>
<tr><td>API</td><td>

```
XAF_ERR_CODE xaf_connect(pVOID p_src,
                         pVOID p_dest,
                         WORD32 num_buf)
```

</td></tr>
<tr><td>Description</td><td>This API connects the output port of audio component <code>p_src</code> to the input port of audio component <code>p_dest</code> with <code>num_buf</code> buffers between them. The size of these buffers will be equal to the size of the output buffer of <code>p_src</code> .

This API will fail if there are no free input or output ports on Audio Components. Audio Components have input and output ports as follows:
<table>
<tr><th>Component Type</th><th>Input Ports</th><th>Output Ports</th></tr>
<tr><td><code>XAF_DECODER</code> or
<code>XAF_ENCODER</code> or
<code>XAF_PRE_PROC</code> or
<code>XAF_POST_PROC</code></td><td>1</td><td>1</td></tr>
<tr><td><code>XAF_MIXER</code></td><td>4</td><td>1</td></tr>
</table>
</td></tr>
<tr><td>Actual Parameters</td><td><code>p_src</code>
Pointer to the source audio component structure

<code>p_dest</code>
Pointer to the destination audio component structure

<code>num_buf</code>
Number of buffers to be added between components
Valid values: 2, 3, 4</td></tr>
<tr><td>Restrictions</td><td>Should not be called before at least two audio components are created using <code>xaf_comp_create</code>.</td></tr>
</table>

Example

```
ret = xaf_connect(p_audioComp1,
                  p_audioComp2,
                  N_BUFFS);
```

Errors

Common API Errors

XAF_ROUTING_ERROR

- Indicates that the API was unable to connect the two requested components

Table 3-8 `xaf_comp_process` API

API	`XAF_ERR_CODE xaf_comp_process(pVOID p_adev,`
	`pVOID p_comp,`
	`pVOID p_buf,`
	`UWORD32 length,`
	`xaf_comp_flag flag)`

Description	This API is the main process function for the audio component; it will do Audio Component start, initialization, execution, and wrap-up based on the process `flag` provided to it. This API needs to be called only for components that need to be supplied with input/output data, typically the edge components of the chain. After processing has started, this API should be called until end of stream, alternatively along with `xaf_comp_get_status` API. The value to be set for the parameter '`flag`' depends on the status returned by the `xaf_comp_get_status` API. **Note**: This API is asynchronous, i.e., it delivers the process command to the audio component and returns. The audio component will process this request when all required resources (IO buffers, CPU, etc.) from the processing chain are available. The status of this process command can be probed by the API described in Table 3-9. **Note**: The pointer to an audio device (`p_adev`) is not required and can be passed as NULL during the execution phase of the audio component (after the component is initialized).

<table>
<tr><td>Actual Parameters</td><td>
<code>p_adev</code>
Pointer to the audio device structure

<code>p_comp</code>
Pointer to the audio component structure

<code>p_buf</code>
Pointer to the input buffer with the input data or output buffer to be filled

<code>length</code>
Unsigned integer containing the length of buffer in bytes

<code>process_flag</code> – Process flag
Following are valid values:
<table>
<tr><th>Flag</th><th>Description</th></tr>
<tr><td><code>XAF_START_FLAG</code></td><td>Initiates processing, to be called only once for each component, during initialization.</td></tr>
<tr><td><code>XAF_EXEC_FLAG</code></td><td>Executes, to be called only once for each component to start processing.</td></tr>
<tr><td><code>XAF_INPUT_OVER_FLAG</code></td><td>Indicates input is complete, when <code>xaf_comp_get_status</code> returns <code>XAF_NEED_INPUT</code>, and input stream is exhausted.</td></tr>
<tr><td><code>XAF_INPUT_READY_FLAG</code></td><td>Indicates input buffer availability, when <code>xaf_comp_get_status</code> returns <code>XAF_NEED_INPUT</code>, and input data is available.</td></tr>
<tr><td><code>XAF_NEED_OUTPUT_FLAG</code></td><td>Request for output, when <code>xaf_comp_get_status</code> returns <code>XAF_OUTPUT_READY</code>, and size returned in <code>xaf_info_t</code> structure is non-zero.</td></tr>
</table>
</td></tr>
<tr><td>Restrictions</td><td>Should not be called before <code>xaf_comp_create</code></td></tr>
</table>

Example

```
ret = xaf_comp_process( p_adev,
                        p_audioComp,
                        &Buff,
                        length,
                        compFlag);
```

Errors

Common API Errors

Table 3-9 `xaf_comp_get_status` API

API	`XAF_ERR_CODE xaf_comp_get_status(`
	`pVOID p_adev,`
	`pVOID p_comp,`
	`xaf_comp_status *p_status,`
	`xaf_info_t *p_info)`

Description	This API returns the status of the audio component and associated information. `p_adev` and `p_comp` should point to the valid audio device and audio component structures respectively. This API will return one of following status and associated information. **Note**: This API is a blocking API, i.e., it may block for status from the DSP thread for a previously issued process command.

Actual Parameters	`p_adev`
	Pointer to the audio device structure
	
	`p_comp`
	Pointer to the audio component structure
	
	`p_status`
	Pointer to get the audio component status
	Valid values are:
	Flag / **Description** / **p_info**
	`XAF_STARTING` / Started /
	`XAF_INIT_DONE` / Initialization complete /
	`XAF_NEED_INPUT` / Component needs data / Buffer pointer, size in bytes
	`XAF_OUTPUT_READY` / Component has generated output / Buffer pointer, size in bytes
	`XAF_EXEC_DONE` / Execution done /
	
	`p_info`
	Pointer to component information structure to get information from the audio component associated with its status. When the p_status returned is `XAF_STARTING` or `XAF_INIT_DONE`, this buffer is not updated.

Restrictions	Should not be called before `xaf_comp_create`

Example

```
ret = xaf_comp_get_status(p_adev,
                          p_audioComp,
                          &compStatus,
                          &Info[0]);
```

Errors

Common API Errors

Table 3-10 `xaf_get_verinfo` API

API	`XAF_ERR_CODE xaf_get_verinfo (pUWORD8 ver_info[3])`
Description	This API gets the version information from the XAF library. It returns an array of the following three strings. `ver_info[0]` Library name `ver_info[1]` Library version `ver_info[2]` API version
Actual Parameters	`ver_info` Pointer to array of three strings
Restrictions	NOT IMPLEMENTED IN ALPHA RELEASE

Example

```
ret = xaf_get_verinfo(&versionInfo[0]);
```

Errors

Common API Errors

4. Xtensa Audio Framework Sample Applications

Two sample applications (testbenches) are provided, which implement two different audio processing chains as described below. Audio components and links are shown in blue in the diagrams below.

Testbench 1 (`xaf_dec_test`) applies gain to PCM streams when '`.pcm`' file is fed as input. The output is written to `dec-out.pcm` file.

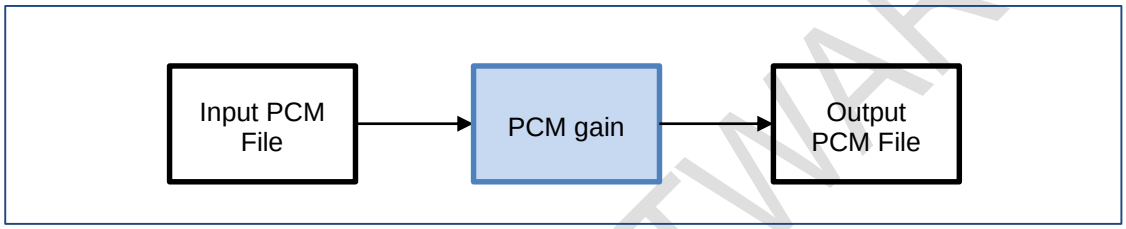


Figure 4-1 Testbench 1 (pcm-gain) Block Diagram

Testbench 1 also (`xaf_dec_test`) decodes Ogg Vorbis streams when '`.ogg`' file is fed as input. The output is written to `dec-out.pcm` file.

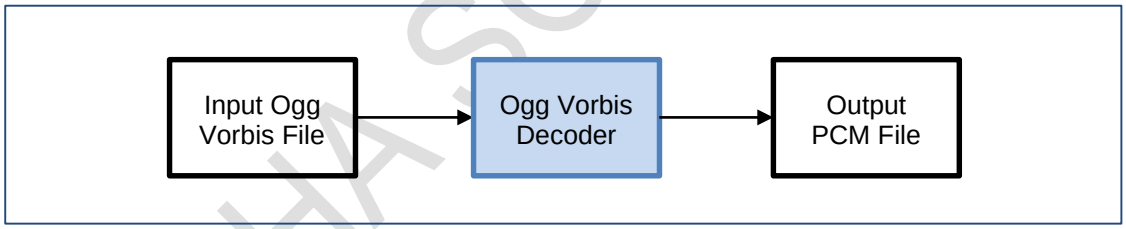


Figure 4-2 Testbench 1 (decoder) Block Diagram

Testbench 2 (`xaf-dec-mix-test`) decodes two Ogg Vorbis streams and mixes the output. The output is written to `dec-mix-out.pcm` file.

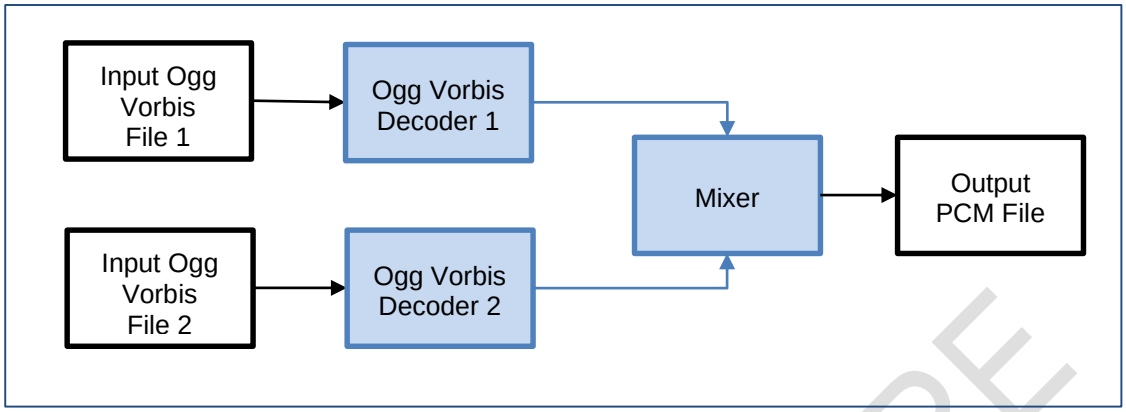


Figure 4-3 Testbench 2 (dec-mix) Block Diagram

Testbench specific source files (`/host-apf/utest/`)

`xaf-dec-test.c`

`xaf-dec-mix-test.c`

Common testbench source files (`/host-apf/utest/`)

.

`xaf-mem-test.c` – Memory allocation functions.

`xaf-utils-test.c` – Other shared utility functions.

4.1 Build and Execute using makefile

To build and execute sample application, please refer to companion document [1] describing how to run the XAF on the HiKey960 Board.

5. Integration of New Audio Components with XAF

This section describes how to create an application with a new audio component in addition to the existing example audio components. The source code pertaining to new audio component and library needs to be added in DSP framework.

5.1 Component Modification

The new component must be modified as follows:

1. Change the component interface to conform to the HiFi Audio/Speech Codec Application Programming Interface [2][3]. The interface (API) is a C-callable API that is exposed by all the HiFi based Audio/Speech Codecs developed by Cadence. An “audio codec” is a generic term for any audio processing component and is not restricted to encoders and decoders.
2. XAF requires all components to support the following configuration parameters for the PCM data ports.
3. XA_CODEC_CONFIG_PARAM_CHANNELS: Number of channels.
4. XA_CODEC_CONFIG_PARAM_SAMPLE_RATE: Sampling rate.
5. XA_CODEC_CONFIG_PARAM_PCM_WIDTH: PCM width.
6. Build the audio component using the Xtensa tools to create a library targeted at the appropriate HiFi core.

5.2 Component Integration

The steps given below need to be followed to integrate the component in XAF. For each step, the corresponding step for the MP3 decoder library is also provided as an example, marked by **MP3_DEC_EG**.

Integration Step 1: Add component files

Three files have to be added to the XAF library to enable support for a new component.

Header file containing the library API definition.

Library file implementing the library.

Wrapper file that “glues” the library to the XAF.

The detailed steps are as follows:

On Host Framework:

1. Copy the API header file for the audio component to the `host-apf/include/audio` folder. This header file must contain the library entry point declaration and all associated structures and constants.

   MP3_DEC_EG: `host-apf/include/audio/xa_mp3_dec_api.h`

On DSP Framework:

2. Create a separate folder under `/hifi-dpf/plugins/` for the new component.

   MP3_DEC_EG: /`hifi-dpf/plugins/cadence/mp3_dec`

3. Copy the component library for the appropriate core(s) to that folder

   MP3_DEC_EG: /`hifi-dpf/plugins/cadence/mp3_dec/lib/xa_mp3_dec.a`

4. Create a wrapper file for the new component in the `/hifi-dpf/plugins/` folder. The wrapper file connects the component library to XAF.

   MP3_DEC_EG: `/hifi-dpf/plugins/cadence/mp3_dec/xa-mp3-decoder.c`

Integration Step 2: Update the DSP Framework to include the component

The DSP Framework must be updated to include references to the new component. The detailed steps are as follows:

5. In the `/hifi-dpf/app/xa-factory.c` file, add the audio component entry point API function extern declaration.

   MP3_DEC_EG: The line below in `xa_factory.c`

   ```
   extern XA_ERRORCODE xa_mp3_decoder(xa_codec_handle_t, WORD32, WORD32, pVOID);
   ```

6. In the constant definition of `xf_component_id` (in `xa_factory.c`), add the registration information for the new audio component.

   MP3_DEC_EG: The line below in `xa_factory.c`

   ```
   {"audio-decoder/mp3", xa_audio_codec_factory, xa_mp3_decoder},
   ```

   The required fields are:

   a. `class_id` (string identifier): This defines the class name and the component name. The different class names are defined in the `comp_id` array.

      MP3_DEC_EG: "audio-decoder/mp3"

   b. `class_constructor` - predefined by XAF and can be either of:
      - `xa_audio_codec_factory` (for components with a single input buffer and a single output buffer), or
      - `xa_mixer_factory` (for components with multiple input buffers and a single output buffer),

      MP3_DEC_EG: xa_audio_codec_factory

c. The function name for the audio component entry point, as defined in the wrapper file which connects the component library to XAF.

MP3_DEC_EG: `xa_mp3_decoder`

Integration Step 3: Create the application to use the component

7. Update makefile in DSP Framework to include new component source files and library.

8. Update application in Host Framework to include and use new component in the audio processing chain.

5.3 Component Integration – Example

One example component is provided that can be used as starting point for the development of new components. This is described in Table 5-1. The table does not include the mixer and PCM Gain component which are considered to be part of XAF. The component folder is under `/hifi-dpf/plugins/cadence` and the application is in `/host-apf/utest` folder.

Table 5-1 Example components

Component Name	API	Description	References
Cadence Ogg Vorbis decoder	Audio	Decodes Ogg Vorbis data	Folder: `vorbis_dec` Application: `xaf-dec-test.c`

6. Known Issues

1. The XAF has only been tested with Version 7.0.5 of the Xtensa tool chain.
2. Fatal errors from XAF component (e.g. Ogg Vorbis Decoder) are ignored and may cause XAF Host application to hang. This issue will be addressed in next XAF release.

7. Appendix: Memory Guidelines

XAF manages the allocation of memory for all created components. Most of the memory is allocated for XAF audio component buffer on DSP (used for audio components allocations) and XAF shared memory buffer (used for IPC between AP and DSP). This section provides guidelines to the application developer to compute these memory sizes.

Notation: Consider a chain of N components, where the n^{th} component has A_n input ports and B_n output ports and requires P_n, S_n, I_n, and O_n KB for persistent, scratch, input, and output buffers respectively. Assume that the n^{th} component is created (`xaf_comp_create`) with X_n input buffers and Y_n output buffers. Note that X_n would be zero except for the components that need to receive data from the application and Y_n would be zero except for the components that need to send data to application. Furthermore, assume that the n^{th} component is connected (`xaf_comp_connect`) to another component with Z_n buffers (to be counted only if the n^{th} component is not the last component in the chain sending data to application).

Guideline for XAF audio component buffer size on DSP:

All memory required by the components is allocated by DSP from this local buffer pool – this includes persistent, scratch, input, and output buffers required by the component. The persistent, scratch, input, and output buffer sizes for a component are typically mentioned in the programmer's guide for that particular component.

Then the total memory required by all components in the chain would be given by the formula:

$$T = T_1 + T_2, \qquad T_1 = \sum_{n=1}^{N} (P_n + A_n I_n + B_n O_n Z_n), \qquad T_2 = \max_n S_n$$

T_1 is the sum of the persistent, input and output sizes required by the components. T_2 is the maximum scratch memory required by the components, as the scratch memory is shared across components. In this version of XAF, T_2 is fixed at 56 KB, via the compile time constant `XF_CFG_CODEC_SCRATCHMEM_SIZE`. Furthermore, some memory is required by XAF itself. The size of the memory required by XAF is (N + 16) KB, where N is the number of components.

Thus, the XAF audio component buffer size on DSP should be set to a value greater than (T1 + 56 + N + 16) KB.

Guideline for XAF shared memory buffer size between AP and DSP:

All buffers exchanged between audio components on DSP and the AP application are allocated from this buffer. The number of buffers exchanged are defined in the `xaf_comp_create` call for each component.

Then the total memory required by all components in the chain would be given by the formula:

$$S = \sum_{n=1}^{N} (8A_n X_n + O_n B_n Y_n),$$

In this version of XAF, the inter-component input buffer size is fixed at 8 KB, via the compile time constant `XAF_INBUF_SIZE`. Furthermore, some memory is also required by XAF itself. The size of the memory required by XAF is 16 KB, independent of the number of components.

Thus, the XAF shared memory buffer between AP and DSP should be set to a value greater than (S + 16) KB.

8. References

[1] *HiKey960 HiFi3 Android SDK Guide*, Ver 0.6. This document is provided as part of the XAF package

[2] *HiFi Audio Codec Application Programming Interface (API) Definition*, Ver 1.0. This document is provided as part of the XAF package.

[3] *HiFi Speech Codec Application Programming Interface (API) Definition*, Ver 1.0. This document is provided as part of the XAF package.