Audio Device Driver Programming Guide

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Contents

Figures, Tables, and Listings

Introduction to Audio Device Driver Programming Guide

Note: This document was previously called *Writing Audio Device Drivers*.

This book describes the architecture, services, and mechanisms of the I/O Kit's Audio family, and explains how you use the APIs of the family to write an audio device driver for Mac OS X. It does not cover any aspect of user-space audio programming (MIDI, synthesizers, CD players, and so on) except to discuss the overall composition of the Mac OS X audio system, which includes Core Audio and other audio frameworks.

To gain the most value from reading this book, it helps to be familiar with the I/O Kit and object-oriented programming, preferably C++ programming. The book *I/O Kit Fundamentals* provides a thorough introduction to the I/O Kit; see "Additional Information on the I/O Kit" (page 8) for details on this and other I/O Kit documentation.

Who Should Read This Document?

As with any kernel-level device driver, you should only write a driver if there is no other way to achieve your goals. Many audio devices are supported natively in Mac OS X. If your device complies with USB or FireWire audio standards, you should not need to write a custom driver unless you need to implement features beyond those supported in the relevant audio standards.

In some cases, even if you need to do special device-specific processing, you may be able to do so without writing an entire driver. For example, some USB audio hardware (for USB speakers, for example) may require additional software filtering, such as equalization. Mac OS X provides a mechanism in the kernel for doing this through the AppleUSBAudio plug-in model. For more information, see the *SampleUSBAudioPlugin* example code.

Organization of This Document

This document describes all aspects of creating an audio device driver using the I/O Kit's Audio family. It includes conceptual and procedural information and consists of the following chapters:

- "Audio on Mac OS X" (page 11)—Describes the features, benefits, and architecture of the Mac OS X audio system. It includes an overview of the audio I/O model.

- "Audio Family Design" (page 21)—Presents a comprehensive overview of the Audio family's architecture, classes, object relationships, and primary mechanisms. It also goes into more detail about the workings of the audio I/O model in Mac OS X.
- "Implementing an Audio Driver" (page 39)—Describes the various steps required to design and implement an audio device driver using the Audio family. Most steps are amply illustrated with sample code.

For details of specific methods, structures, and other API elements, consult the reference documentation for the Audio family. See "Additional Information on the I/O Kit" (page 8) for instructions on accessing this documentation.

See Also

Apple offers several other resources to developers of audio software for Mac OS X, including:

- See http://developer.apple.com/audio for a page full of links to audio-related material.
- See *Core Audio* for a description of the Core Audio framework (Audio HAL).
- For information on MIDI frameworks, see *Core MIDI Framework Reference* and *Core MIDI Server Framework Reference*.
- See `/Developer/Examples/Kernel/IOKit/Audio` for some sample audio driver projects and other code examples relevant to audio development.

Additional Information on the I/O Kit

For additional information on the I/O Kit in general, see the following documents:

- Overviews of the Darwin kernel, including *Kernel Programming Guide*.
- The aforementioned *I/O Kit Fundamentals* describes the features, architecture, classes, and general mechanisms of the I/O Kit and includes discussions of driver matching and loading, event handling, memory management, and power management.
- *I/O Kit Device Driver Design Guidelines*, which describes the general steps required to design, code, debug, and build a device driver that will be resident in the kernel.
- *Kernel Extension Concepts*, a collection of tutorials that introduce you to the development tools and take you through the steps required to create, debug, and package kernel extensions and I/O Kit drivers (a type of kernel extension).
- Reference documentation on I/O Kit families and classes.

Of course, you can always browse the header files shipped with the I/O Kit, which are installed in `Kernel.framework/Headers/iokit` (kernel-resident) and `IOKit.framework/Headers` (user-space).

The documentation is in HTML or PDF format. You can access the HTML documentation (and download the PDF) from the Xcode Help menu. To view, click Help > Show Documentation Window. You can then search for specific API or view the entire developer documentation library. You can also access developer documentation on the Apple Developer Connection website at http://developer.apple.com/documentation/index.html.

Other Information on the Web

Apple maintains websites where developers can go for general and technical information on Mac OS X.

- Apple Developer Connection: Mac OS X (http://developer.apple.com/macosx) offers SDKs, release notes, product notes and news, and other resources and information related to Mac OS X.
- Apple Support Area (http://www.apple.com/support/) enables you to locate technical articles on Mac OS X (and other areas) using a natural language search.

Audio on Mac OS X

This chapter gives an overview of audio on Mac OS X, describing its capabilities, its interrelated technologies, and its architecture. Reading this chapter will help you to understand how the I/O Kit's Audio family fits together and interacts with the other pieces of audio software on Mac OS X.

Mac OS X Audio Capabilities

In versions of Macintosh system software prior to Mac OS X, the sound capabilities of a system largely depended on the availability of third-party audio and MIDI protocols and services. Apple has designed the Mac OS X audio system to consolidate, integrate, and standardize these services and protocols, thereby streamlining configuration of audio and MIDI devices and development of future audio and MIDI technologies.

Audio on Mac OS X comprises several audio technologies that, taken together, offer the following capabilities:

- Built-in support for a variety of audio formats, including formats based on pulse code modulation (PCM) and encoded formats such as AC-3 and MP3
- Multi-channel audio I/O that is scalable to a virtually unlimited number of channels
- Variable sample rates
- A remarkably clean signal path requiring little overhead
- Simultaneous access for multiple clients to all of the audio devices attached to the host, no matter how the connection is made (PCI, USB, FireWire, and so on)

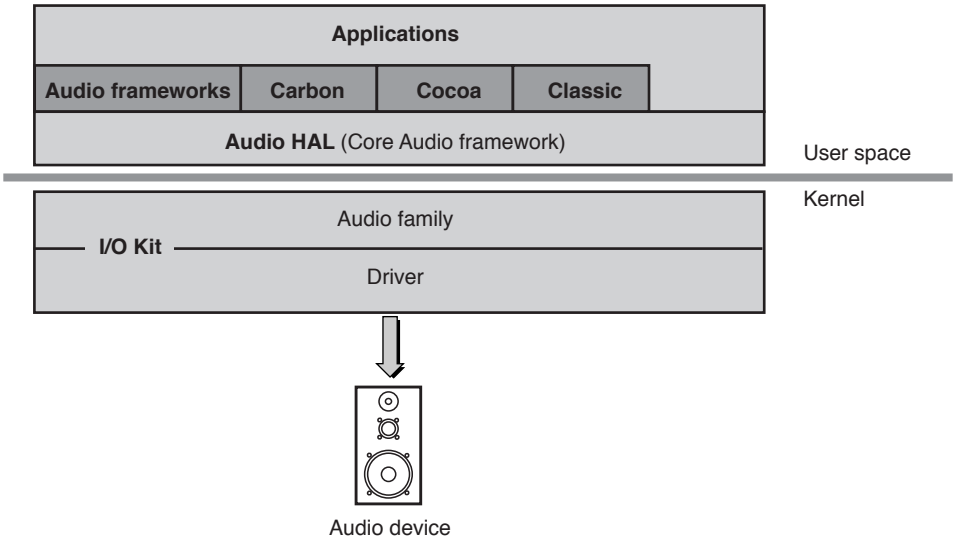
Outside the kernel, Mac OS X represents audio as 32-bit floating point data; this format allows efficient processing of data with today's advanced audio peripherals (for example, those capable of 24-bit, 192 kHz operation) and ensures that the system can scale to future high-resolution formats.

Architecture of Mac OS X Audio

The audio capabilities of Mac OS X arise from several software technologies that are accessible through their public programming interfaces. These technologies are situated at different levels of the operating system where their relationships with each other can be characterized as client and provider. In other words, Mac OS X audio software is layered, with one layer dependent on the layer "under" it and communicating, through defined interfaces, with adjoining layers (see Figure 1-1 (page 12)).

The relative locations of these technologies within the layers of system software suggest their degree of abstraction and their proximity to audio hardware. Some audio technologies in Mac OS X are incorporated into the kernel environment (that is, Darwin) while others are packaged as frameworks for use by application environments, applications, and other user processes.

Figure 1-1 Mac OS X audio layers



At the lowest level of the Mac OS X audio stack is the driver that controls audio hardware. The driver is based on the I/O Kit's audio family, which provides much of the functionality and data structures needed by the driver. For example, the Audio family implements the basic timing mechanisms, provides the user-client objects that communicate with the upper layers, and maintains the sample and mix buffers (which hold audio data for the hardware and the hardware's clients, respectively).

The basic role of the audio driver is to control the process that moves audio data between the hardware and the sample buffer. It is responsible for providing that sample data to the upper layers of the system when necessary, making any necessary format conversions in the process. In addition, an audio driver must make the necessary calls to audio hardware in response to format and control changes (for example, volume and mute).

Immediately above the driver and the I/O Kit's Audio family—and just across the boundary between kernel and user space—is the Audio Hardware Abstraction Layer (HAL). The Audio HAL functions as the device interface for the I/O Kit Audio family and its drivers. For input streams, its job is to make the audio data it receives from drivers accessible to its clients. For output streams, its job is to take the audio data from its clients and pass it to a particular audio driver.

The Audio Units and Audio Toolbox frameworks are two other frameworks that provide specialized audio services. They are both built on top of the Audio HAL, which is implemented in the Core Audio framework.

MIDI System Services, which comprises two other frameworks, is not directly dependent on the Audio HAL. As its name suggests, MIDI System Services makes MIDI services available to applications and presents an API for creating MIDI drivers.

Finally, the ultimate clients of audio on Mac OS X—applications, frameworks, and other user processes—can directly access the Audio HAL or indirectly access it through one of the higher-level audio frameworks. They can also indirectly access the Audio HAL through the audio-related APIs of the application environments they belong to: Sound Manager in Carbon, `NSSound` in Cocoa, and the Java sound APIs.

The following sections examine each of these audio technologies of Mac OS X in more detail.

Audio HAL (Core Audio)

The Audio Hardware Abstraction Layer (HAL) is the layer of the Mac OS X audio system that acts as an intermediary between the I/O Kit drivers controlling audio hardware and the programs and frameworks in user space that are clients of the hardware. More specifically, the Audio HAL is the standardized device interface for the I/O Kit's Audio family. It is implemented in the Core Audio framework (`CoreAudio.framework`) and presents both C-language and Java APIs. In the Audio HAL, all audio data is in 32-bit floating point format.

The API of the Audio HAL includes three main abstractions: audio hardware, audio device, and audio stream.

- The audio hardware API gives clients access to audio entities that exist in the "global" space, such as the list of current devices and the default device.
- The audio device API enables clients to manage and query a specific audio device and the I/O engines that it contains. An audio device in the Audio HAL represents a single I/O cycle, a clock source based on it, and all the buffers that are synchronized to this cycle. The audio device methods permit a client to, among other things, start and stop audio streams, retrieve and translate the time, and get and set properties of the audio device.
- The audio stream API enables a client to control and query an audio stream. Each audio device has one or more audio streams, which encapsulate the buffer of memory used for transferring audio data across the user/kernel boundary. They also specify the format of the audio data.

The abstractions of audio device and audio stream loosely correspond to different I/O Kit Audio family objects in the kernel (see "The Audio Family" (page 16)). For example, the entity referred to as "audio device" in the Audio HAL corresponds to a combination of an `IOAudioDevice` and `IOAudioEngine` in the kernel. For each `IOAudioEngine` the Audio HAL finds in the kernel, it generates an audio-device identifier. However, there is considerable overlap of role among the various Audio family and Audio HAL objects and entities.

A critical part of the APIs for audio hardware, devices, and streams involves audio properties and their associated notifications. These APIs allow clients to get and set properties of audio hardware. The "get" methods are synchronous, but the "set" methods work in an asynchronous manner that makes use of notifications. Clients of the Audio HAL implement "listener procs"—callback functions for properties associated with audio hardware, audio devices, or audio streams. When an audio driver

changes a property of the hardware, either as a result of user manipulation of a physical control or in response to a "set" method, it sends notifications to interested Audio HAL clients. This results in the appropriate "listener procs" being called.

Just as important as the property APIs is the callback prototype (`AudioDeviceIOProc`) that the audio-device subset of the Audio HAL API defines for I/O management. Clients of the Audio HAL must implement a function or method conforming to this prototype to perform I/O transactions for a given device. Through this function, the Audio HAL presents all inputs and outputs simultaneously in an I/O cycle to the client for processing. In this function, a client of the Audio HAL must send audio data to the audio device (for output), or copy and process the audio data received from the audio device (for input).

Secondary Audio Frameworks

Mac OS X has several frameworks other than the Core Audio framework that offer audio-related functionality to applications. Two of these frameworks—Audio Units and Audio Toolbox—are built directly on the Core Audio framework. MIDI System Services (consisting of the Core MIDI and Core MIDI Server frameworks) does not directly depend on the Core Audio framework, but is still a consumer of the services of the audio frameworks.

All of these secondary frameworks are implemented in the C language and present their public programming interfaces in C. Thus, any application or other program in any application environment can take advantage of their capabilities.

Audio Units

The Audio Units framework (`AudioUnits.framework`) provides support for generating, processing, receiving, and manipulating or transforming streams of audio data. This functionality is based on the notion of audio units.

Audio units are one form of a building block called a component. A component is a piece of code that provides a defined set of services to one or more clients. In the case of audio units, these clients can use audio unit components either singly or connected together to form an audio signal graph. To compose an audio signal graph, clients can use the AUGraph API in the Audio Toolbox framework—see "Audio Toolbox" (page 15) for details.

An audio unit can have one or more inputs and outputs. The inputs can accept either encoded audio data or MIDI data. The output is generally a buffer of audio data. Using a "pull I/O" model, an audio unit specifies the number and format of its inputs and outputs through its properties. Each output is in itself a stream of an arbitrary number of interleaved audio channels derived from the audio unit's inputs. Clients also manage the connections between units through properties.

Examples of audio units are DSP processors (such as reverbs, filters, and mixers), format converters (for example, 16-bit integer to floating-point converters), interleavers-deinterleavers, and sample rate converters. In addition to defining the interface for custom audio units in the Audio Units framework, Apple ships a set of audio units. One of these is the MusicDevice component, which presents an API targeted specifically toward software synthesis.

Audio Toolbox

The Audio Toolbox framework (`AudioToolbox.framework`) complements the Audio Units framework with two major abstractions: the AUGraph and the Music Player.

An AUGraph provides a complete description of an audio signal processing network. It is a programmatic entity that represents a set of audio units and the connections (input and output) among them. With the AUGraph APIs, you can construct arbitrary signal paths through which audio can be processed. Audio graphs enact real-time routing changes while audio is being processed, creating and breaking connections between audio units "on the fly," thus maintaining the representation of the graph even when constituent audio units have not been instantiated.

The Music Player APIs use AUGraphs to provide the services of a sequencing toolbox that collects audio events into tracks, which can then be copied, pasted, and looped within a sequence. The APIs themselves consist of a number of related programmatic entities. A Music Player plays a Music Sequence, which can be created from a standard MIDI file. A Music Sequence contains an arbitrary number of tracks (Music Tracks), each of which contains timestamped audio events in ascending temporal order. A Music Sequence usually has an AUGraph associated with it, and a Music Track usually addresses its audio events to a specific Audio Unit within the graph. Events can involve tempo and extended events, as well as regular MIDI events.

The Audio Toolbox framework also includes APIs for converting audio data between different formats.

MIDI System Services

MIDI System Services is a technology that allows applications and MIDI devices to communicate with each other in a single, unified way. It comprises two frameworks: Core MIDI (`CoreMIDI.framework`) and Core MIDI Server (`CoreMIDIServer.framework`).

MIDI System Services gives user processes high-performance access to MIDI hardware. In a manner similar to the Audio HAL, MIDI System Services implements a plug-in interface that enables clients to communicate with a MIDI device driver.

Note: MIDI device drivers are not I/O Kit drivers. The MIDI device driver model is based on the CFPlugIn architecture and typically loads a CFPlugIn bundle from `/System/Library/Extensions` or `Library/Audio/MIDI Drivers`.

For MIDI devices that cannot be directly addressed from a user-space device driver (for example, a MIDI interface built into a PCI card), you must split your driver into two parts: an I/O Kit device driver that matches against the device and a CFPlugIn bundle that manipulates the I/O Kit driver using a user client.

The details of implementing such a mechanism are beyond the scope of this document. For information on user clients, see Device-Interface Development.

Apple provides several default MIDI drivers for interfaces that comply with USB and FireWire MIDI interface standards. Using the Core MIDI Server framework, third-party MIDI manufacturers can create their own driver plug-ins to support additional device-specific features. A MIDI server can then load and manage those drivers.

Applications can communicate with MIDI drivers through the client-side APIs of the Core MIDI framework.

The Audio Family

The I/O Kit's Audio family facilitates the creation of drivers for audio hardware. Drivers created through the Audio family can support any hardware on the system, including PCI, USB, and FireWire devices. Essentially, an I/O Kit audio driver transfers audio data between the hardware and the Audio HAL. It provides one or more sample buffers along with a process that moves data between the hardware and those sample buffers. Typically this is done with the audio hardware's DMA engine.

Because the native format of audio data on Mac OS X is 32-bit floating point, the driver must provide routines to convert between the hardware format of the data in the sample buffer and 32-bit floating point. The sequence of steps that a driver follows depends on the direction of the stream. For example, with input audio data, the driver is asked for a block of data. It obtains it from the sample buffer, converts it to the expected client format (32-bit floating point), and returns it. That data is then passed by the family to the Audio HAL through a user-client mechanism.

The interactions between the DMA engine, the driver, and the Audio HAL, are based on the assumption that, in any one direction, the stream of audio data proceeds continuously at the same rate. The Audio family sets up several timers (based on regularly taken timestamps) to synchronize the actions of the agents involved in this transfer of data. These timing mechanisms ensure that the audio data is processed at maximum speed and with minimum latency.

Take again an input stream as an example. Shortly after the DMA engine writes sample frames to the driver's sample buffer, the driver reads that data, converts the integer format to 32-bit floating point, and writes the resulting frames to the mixer buffer, from whence they are passed on to the Audio HAL. Optionally, just before the DMA engine writes new frames to the same location in the sample buffer, an "erase head" zero-initializes the just-processed frames. (By default, however, the erase head only runs on output streams.)

For more on the sample buffer and the timer mechanisms used by the Audio family, see "The Audio I/O Model on Mac OS X" (page 17).

An I/O Kit audio driver consists of a number of objects, the most important of which are derived from the `IOAudioDevice`, `IOAudioEngine`, `IOAudioStream`, and `IOAudioControl` classes. These objects perform the following roles for the driver:

- A single instance of a custom subclass of `IOAudioDevice` represents the audio device itself. The `IOAudioDevice` subclass is the root object of a complete audio driver. It is responsible for mapping all hardware resources from the service-provider's nub and for controlling all access to the hardware (handled automatically through a provided command gate). An `IOAudioDevice` object manages one or more `IOAudioEngine` objects.
- An audio driver must contain one or more instances of a custom subclass of `IOAudioEngine`. This custom subclass manages each audio I/O engine associated with the audio device. Its job is to control the process that transfers data between the hardware and a sample buffer. Typically the I/O process is implemented as a hardware DMA engine (although it doesn't have to be). The sample buffer must be implemented as a ring buffer so that when the I/O process of a running `IOAudioEngine` reaches the end of the buffer, it wraps back around to the beginning and keeps going.

  An `IOAudioEngine` object is also responsible for starting and stopping the engine, and for taking a timestamp each time the sample buffer wraps around to the beginning. It contains one or more `IOAudioStream` objects and can contain any number of `IOAudioControl` objects.

All sample buffers within a single `IOAudioEngine` must be the same size and running at the same rate. If you need to handle more than one buffer size or sampling rate, you must use more than one `IOAudioEngine`.

- An instance of `IOAudioStream` represents a sample buffer, the associated mix buffer, and the direction of the stream. The `IOAudioStream` object also contains a representation of the current format of the sample buffer as well as a list of allowed formats for that buffer.
- An instance of `IOAudioControl` represents any controllable attribute of an audio device, such as volume or mute.

An I/O Kit audio driver uses two user-client objects to communicate with the Audio HAL layer. The Audio HAL communicates with the `IOAudioEngine` and `IOAudioControl` objects through the `IOAudioEngineUserClient` and `IOAudioControlUserClient` objects, respectively. The audio family creates these objects as they are needed. The `IOAudioEngineUserClient` class provides the main linkage to an `IOAudioEngine` subclass; it allows the Audio HAL to control the `IOAudioEngine` and it enables the engine to pass notifications of changes back to the Audio HAL. For each `IOAudioControl` object in the driver, an `IOAudioControlUserClient` object passes notifications of value changes to the Audio HAL.

For more detailed information on the classes and general architecture of the Audio family, see the chapter "Audio Family Design" (page 21).

Apple Audio Drivers

Apple ships several audio device drivers with a standard Mac OS X installation. These drivers are suitable for much of the audio hardware commonly found on Power PC computers. The "onboard driver" kernel extension—`AppleOnboardAudio.kext`—contains almost a half dozen audio drivers packaged as plug-ins. Each of these drivers is based on a specific subclass of `IOAudioDevice` and each uses the code in the AppleDBDMAAudio kernel extension for the `IOAudioEngine` subclass. The I/O Kit, through its matching process, finds and loads the appropriate plug-ins based on existing audio hardware. For USB audio hardware, Apple includes the driver defined in the `AppleUSBAudio.kext` kernel extension.

Important: The set of audio drivers provided by Apple may change at any time. Your drivers should thus avoid depending on the presence or absence of specific drivers.

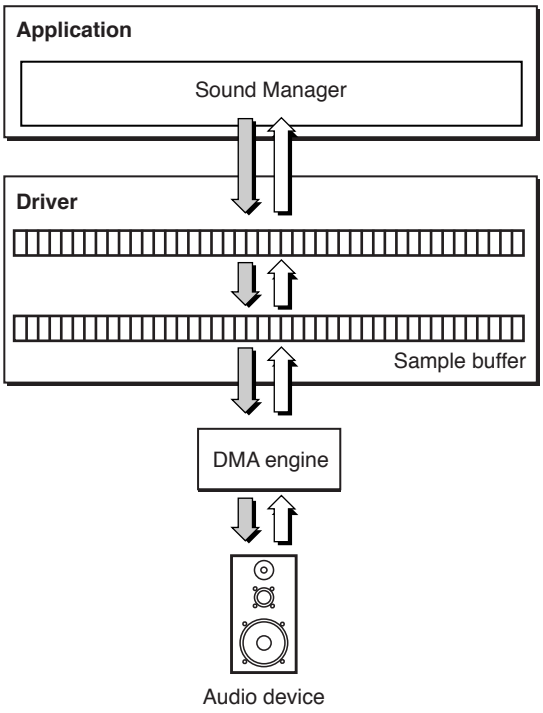
The Audio I/O Model on Mac OS X

Mac OS 9 and Mac OS X perform audio I/O in very different ways. The differences between them are most salient in the lower layers of the audio stack, particularly the audio driver model and the audio access libraries.

In Mac OS 9, an audio driver's DMA engine transfers audio data between a sample buffer, which is provided by the driver, and the hardware. The buffer holds a segment of the audio data containing a sequence of sample frames in temporal order.

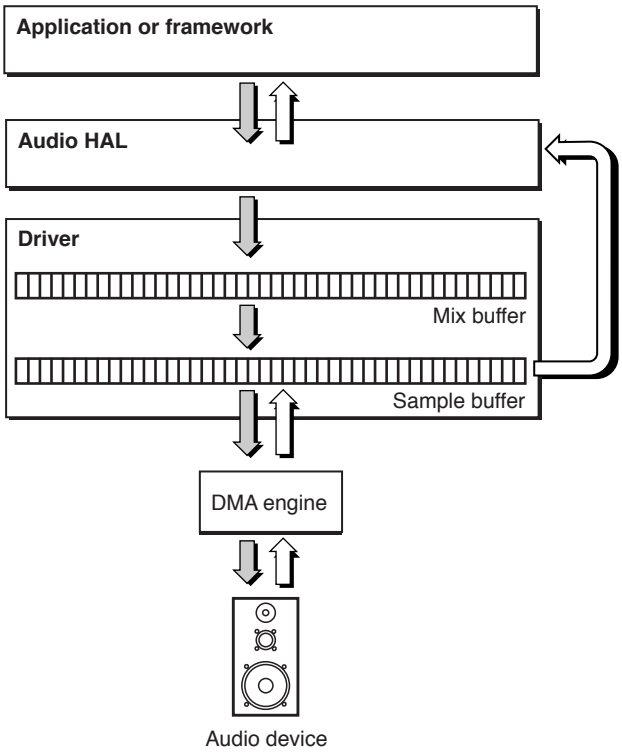
The Mac OS 9 driver model uses double buffering to exchange audio data between the driver and its clients, so there are actually two sample buffers. In the case of audio output, after the driver's clients (using the Sound Manager API) fill one of the buffers, the hardware (usually through its DMA engine) signals the driver (typically through an interrupt) that it is finished playing the other buffer and ready for more data. The driver then gives the hardware the buffer it just filled, receives the just-played buffer from the hardware, and signals the application that it needs more data.

Figure 1-2 Access to the sample buffer on Mac OS 9



The architecture and goals of Mac OS X made this design untenable. With the Mac OS X kernel, an audio driver incurs a greater cost than on Mac OS 9 when it signals an application that more audio data is needed (or that new data is available). Moreover, a major goal of the Mac OS X audio system is to support multiple simultaneous clients, which is not possible with the Mac OS 9 model. A new audio I/O model was needed not only for this goal but also to provide the highest possible performance and the lowest possible latency. Figure 1-3 (page 19) depicts the audio I/O model on Mac OS X.

Figure 1-3 The Mac OS X audio model



The key facet of the Mac OS X audio I/O model involves predictive timing mechanisms. Instead of requiring the driver to message an application directly when an I/O cycle has completed, the timing mechanisms enable the Audio HAL to predict when the cycle will complete. The Audio HAL uses the extremely accurate timing code on Mac OS X to ensure that clients perform their I/O at the proper time, based on the size of their buffers. The audio driver does its part to make this possible by setting up the hardware's sample buffer as a ring buffer and by taking an accurate timestamp every time the I/O engine wraps to the beginning of the buffer.

The Audio HAL keeps track of each timestamp and uses the sequence of timestamps to predict the current location of the audio I/O engine (in terms of sample frame read or written) at any time. Given that information, it can predict when a cycle will complete and sets its wake-up timestamp accordingly. This model, combined with the ability of the I/O Kit Audio family to receive audio data from each client asynchronously, allows any number of clients to provide audio data that gets mixed into the final output. It also allows different client buffer sizes; one client can operate at a very low buffer size (and a correspondingly low latency) while at the same time another client may use a much larger buffer. As long as the timestamps provided by the driver are accurate, the family and the Audio HAL do all of the work to make this possible.

Another important difference between the audio I/O model on Mac OS 9 and the one on Mac OS X is the native format of audio data in the system. In Mac OS 9, because the application (through the Sound Manager) has direct access to the hardware buffer, it has to deal with the native hardware format. Because of this reality, the Mac OS 9 audio libraries only support 16-bit one-channel or two-channel PCM audio data to simplify things.

In Mac OS X, an application cannot directly access the sample buffer. This indirection permits the use of the 32-bit floating point format between the Audio HAL and an audio driver. Consequently, the driver is responsible for providing a routine that can clip and convert that 32-bit floating point output

data into the buffer's native format. It might also have to implement a routine to convert input data into 32-bit floating point. Both routines are called asynchronously as Audio HAL clients pass audio data to the driver and receive data from it.

For detailed information on the Mac OS X audio I/O model, see "The Audio I/O Model Up Close" (page 31).

Audio Family Design

All audio drivers, regardless of platform, must perform the same general actions. For input streams, drivers receive digital audio data from the hardware in a stream of frames consistent with the current sampling rate and audio format. They modify the data, if necessary, to a form acceptable to the clients of the device (say, 32-bit floating point) and make the altered frames accessible to those clients at the current sampling rate. In the reverse (output) direction, the job of the audio driver is essentially the same. It accepts digital audio data from the clients of the device, changes that stream of sample frames to a form required by the hardware (say, 16-bit integer), and gives the data to the device's controller at the current sampling rate.

Drivers must also initially configure the hardware, respond to client requests to change device attributes (for example, volume), and notify clients when some attribute or state of the audio device has changed. They must guard against data corruption in a multithreaded environment, and they must be prepared to respond to systemwide events, such as sleep/wake notifications.

The Audio family provides object-oriented abstractions to help your driver deal with many of these things. The family itself takes care of much of the work for you; you just supply the behavior that is specific to your hardware. To do this, it is useful to know how your code fits together with the family implementation, which is what this chapter is about.

The Classes of the Audio Family

As you can with any object-oriented system, you can come to an understanding of the design of the I/O Kit Audio family by examining the classes of the family. The examination in this section looks at the roles of the family, the properties they encapsulate, the audio entities they represent, and the relationships they have with each other. The relationships considered are not only the static relationships imposed by inheritance but also the dynamic relationships characterized by containment, dependency, and control.

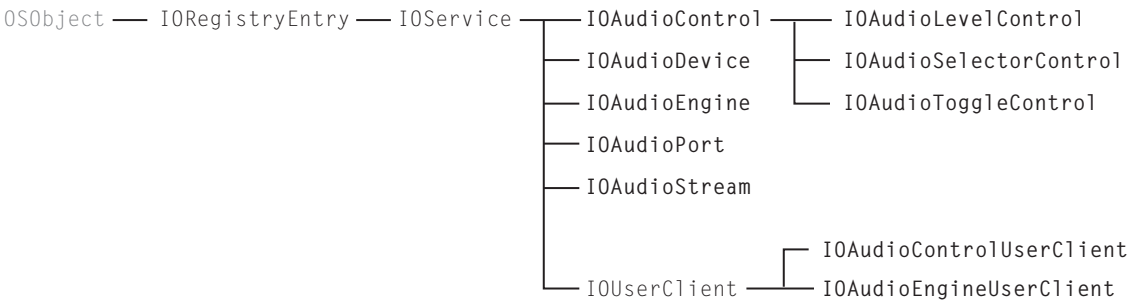
The Audio family consists of about a dozen classes, all having the prefix "IOAudio":

- `IOAudioDevice`
- `IOAudioEngine`
- `IOAudioStream`
- `IOAudioControl`
- `IOAudioPort`

- IOAudioEngineUserClient
- IOAudioControlUserClient
- IOAudioLevelControl
- IOAudioSelectorControl
- IOAudioToggleControl

The inheritance relationships among these classes, as depicted in Figure 2-1 (page 22), are uncomplicated.

Figure 2-1 The Audio family class hierarchy



All classes of the Audio family directly or indirectly inherit from IOService; thus objects of these classes are full-fledged driver objects, with the capability for responding to driver life-cycle messages and for participating in the matching process. In practice, however, an instance of an IOAudioDevice subclass, as root object of the audio driver, usually matches against the provider's nub (the provider being a PCI controller or FireWire or USB device, in most cases). Audio drivers are typically "leaf" objects in the driver stack, and typically their only client is the Audio HAL, in user space. Therefore they do not publish nubs of their own.

Two classes, IOAudioEngineUserClient and IOAudioControlUserClient, inherit from the IOUserClient class. Objects of these classes represent user-client connections that enable the Audio family to communicate with the Audio HAL. Five of the Audio family classes are subclasses of IOAudioControl, providing behavior specific to certain types of audio-device controls (mute switches, volume controls, and so on). For further details on the user-client and control classes of the Audio Family, see "The Roles of Audio Family Objects" (page 25).

An understanding of the static inheritance relationships between classes of the Audio family goes only so far to clarify what instances of those classes do in a typical audio driver. It is more illuminating to consider the dynamic relationships among these objects.

Dynamic Relationships in the Audio Family

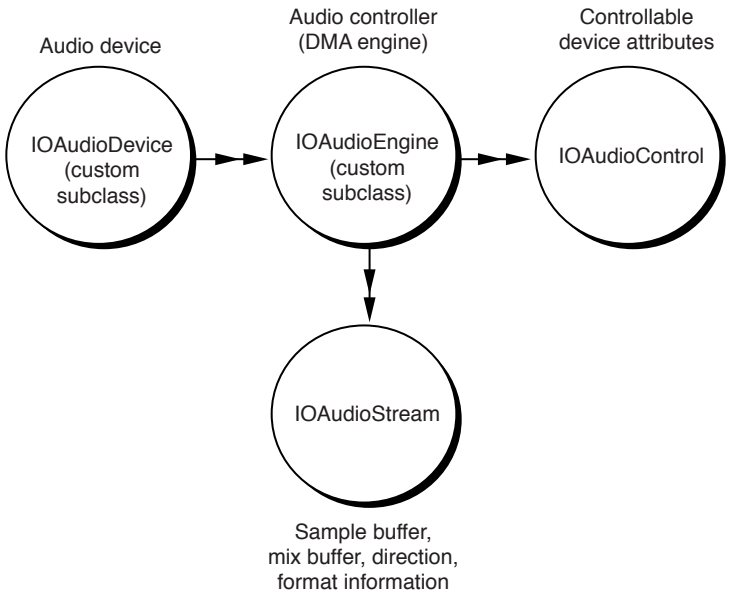
An I/O Kit audio driver consists of a variable number of objects that represent or encapsulate certain aspects of an audio device. Many of these objects own references to other objects. The most significant objects in a "live" audio driver derive from four Audio family classes:

- IOAudioDevice
- IOAudioEngine

- IOAudioStream
- IOAudioControl

Figure 2-2 (page 23) illustrates the dynamic relationships of these objects.

Figure 2-2 Audio family objects in a typical driver and what they represent



The root object in an audio driver is an instance of a custom subclass of IOAudioDevice. It represents an audio device in a general, overall sense. An IOAudioDevice object is the root object for a couple of reasons: it creates and coordinates many of the other objects in the driver, and it is typically the object that must match against the provider's nub.

The custom subclass of IOAudioDevice adds attributes and implements behavior that are specific to the device. It is responsible for identifying, configuring, and creating all necessary audio-engine objects and attaching those objects to itself. It must map all hardware resources from the provider's nub and, when requested by the system, it must change the values of controls.

Furthermore, an IOAudioDevice object is the power controller and power policy maker for the driver; in coordination with its IOAudioEngine objects, it must properly determine system idleness and deal with power-state transitions (sleep and wake), deactivating and reactivating its audio engines as necessary. (See "Handling Sleep/Wake Notifications" (page 46) for more information.)

A driver's IOAudioDevice object contains one or more IOAudioEngine objects as instance variables. Each of these objects is an instance of a custom subclass of IOAudioEngine. An IOAudioEngine object represents the I/O engine (usually a DMA engine) of the audio device; its job is to transfer audio data to or from one or more sample buffers and the hardware. The object starts and stops the audio I/O engine when requested; once started, it should run continuously, looping through the sample buffers until stopped. While it is running, an IOAudioEngine takes a timestamp and increments a loop count each time it "wraps around" a sample buffer (see "The Audio I/O Model Up Close" (page 31)). The Core Audio framework (Audio HAL) uses this timing information to calculate the exact position of the audio engine at any time.

An audio driver needs only one `IOAudioEngine` object unless it needs to manage sample buffers of different sizes or to have sample frames transferred at different rates. In these cases, it should instantiate and configure the required number of `IOAudioEngine` instances.

An `IOAudioEngine` object itself contains one or more instances of the `IOAudioStream` class. An `IOAudioStream` object primarily represents a sample buffer, which it encapsulates. It also encapsulates the mix buffer for an output audio stream. It describes the direction of the stream as well as the format information that can be applied to the sample buffer. The format information includes such data as number of channels, sampling format, and bit depth. If a sample buffer has multiple channels, the channels are typically interleaved (although separate `IOAudioStream` instances can be used to represent non-interleaved different channels). Often an audio engine has one `IOAudioStream` object for an input stream and another for an output stream.

An `IOAudioEngine` also contains one or more `IOAudioControl` objects as instance variables. Such an object represents a controllable attribute of the audio device, such as mute, volume, or master gain. An `IOAudioControl` is usually associated with a specific channel in a specific stream. However, it can control all channels of an `IOAudioStream` or even all channels of an `IOAudioEngine`. At hardware-initialization time, an `IOAudioEngine` (or perhaps the driver's `IOAudioDevice` object) creates the necessary `IOAudioControl` objects and adds them to the appropriate `IOAudioEngine`.

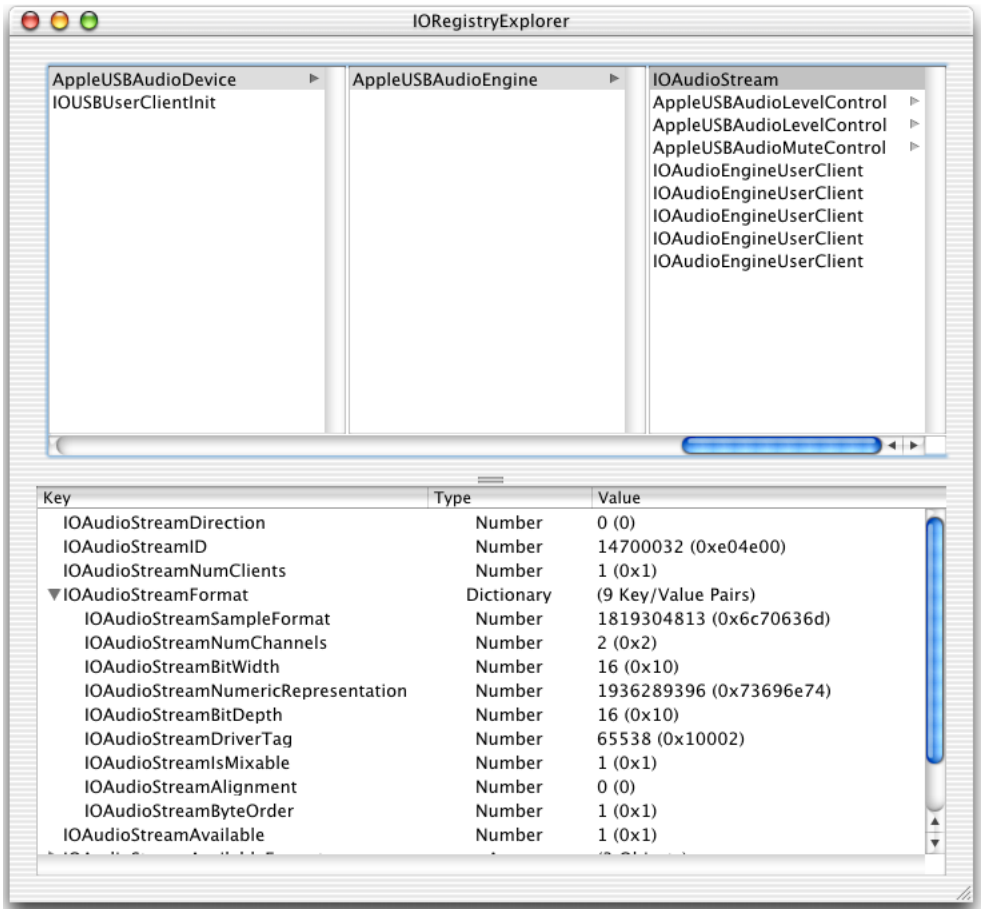
Each `IOAudioControl` is known as a "default control" because the Audio HAL recognizes and uses it based on its attributes. Most value changes to the controls originate with clients of the Audio HAL, which passes them via the user-client interface to the designated value-change handlers in the driver. Notifications of value changes in audio controls can also travel in the other way; for example, if a user turns the volume knob on a speaker, the driver communicates this change to Audio HAL clients.

The Audio Family and the I/O Registry

As it does with all I/O Kit drivers, the I/O Registry captures the client-provider relationships and the properties of audio drivers. By using the I/O Registry Explorer application or the `ioreg` command-line tool, you can view the objects of "live" audio drivers in the I/O Registry and examine the properties of those objects. The visual presentation that these tools provide clarifies the client-provider relationships among Audio-family objects and the relationships between audio objects and other objects in the driver stack. You can use these tools to verify driver status and for debugging problems.

Figure 2-3 (page 25) shows how some of the objects in a USB audio driver appear in the I/O Registry.

Figure 2-3 A USB audio driver displayed in the IORegistryExplorer application



In addition to these programming uses, the I/O Registry also serves a critical function in the architecture of the Mac OS X audio system. The control properties of an audio driver—volume, mute, gain settings—are stored in the I/O Registry and are associated with an `IOAudioControl` object. The Audio HAL looks for, recognizes, and uses the object based on its attributes, which it discovers through the I/O Registry. See "IOAudioControl" (page 28) for further information.

For details on matching properties, device attributes, and other I/O Registry keys, see the header file `Kernel.framework/Headers/IOKit/audio/IOAudioDefines.h` or the associated reference documentation for `IOAudioDefines.h`.

The Roles of Audio Family Objects

The previous section looked in a general way at the major objects in a "live" audio driver, describing what those objects basically do and what their relationships are with one another. This section probes a little deeper and examines the roles of all Audio family classes (and objects) in more detail.

IOAudioDevice

Every audio driver based on the Audio family must have one instance of a custom subclass of `IOAudioDevice`. The driver's `IOAudioDevice` object is the central, coordinating node of the driver's object tree—the "root" object. All other objects ultimately depend on it or are contained by it.

Because of its status as root object, the `IOAudioDevice` object represents the audio hardware generally. This central position gives it several roles:

- It is the object that usually matches against the provider's nub.
- It initializes the device, mapping hardware resources from the provider's nub, and otherwise reads and writes to the device registers as necessary.
- It creates the `IOAudioEngine` objects of the driver and can create the `IOAudioControl` objects used by the driver.
- It usually manages synchronization of values between hardware controls and the associated software controls associated with its audio-engine objects.
- It acts as the power controller and policy maker for the audio hardware; in this role, it must respond to system sleep, system wake, and domain idleness transitions by deactivating and reactivating its audio engines as necessary.

The driver's `IOAudioDevice` object fulfills other functions. It controls access to the audio hardware to ensure, in the driver's multithreaded environment, that the hardware doesn't get into an inconsistent state. Toward this end, the `IOAudioDevice` superclass provides a separate work loop (`IOWorkLoop`) and a command gate (`IOCommandGate`) to synchronize access by all objects in the driver. All other objects in the driver—`IOAudioEngine`, `IOAudioStream`, and `IOAudioControl`—contain references to the `IOAudioDevice`'s work loop and the command gate as instance variables. All Audio family classes take care of executing I/O and hardware-related code on the command gate for all calls into the driver that they know about. Generally, drivers should ensure that all I/O and hardware-specific operations are executed with the command gate closed.

`IOAudioDevice` also offers timer services to the audio driver. These services allow different objects within the driver to receive notifications that are guaranteed to be delivered at the requested timer interval, if not sooner. Different target objects can register for timer callbacks at a specific interval; however, `IOAudioDevice` makes the actual timer interval the smallest of those requested.

The idea behind this design is that there is no harm in having timed events in an audio driver occur sooner than requested. By coalescing the callback intervals, the Audio family obviates the overhead of multiple timers in a single driver.

In some cases, however, this may result in unexpected behavior if you make assumptions based on the amount of time elapsed, such as assuming that the hardware has played a certain number of samples. You should thus always make certain to test to make sure conditions are appropriate before performing such operations.

Your driver itself can have localized strings that are accessible by the Audio HAL. These strings can include such things as name, manufacturer, and input sources. Follow the Mac OS X localization procedure for these strings, putting them in a file named `Localizable.strings` in the locale-specific subdirectories of your bundle. The driver should have a property named `IOAudioDeviceLocalizedBundleKey`, which has a value of the path of the bundle or kernel extension holding the localized strings, relative to `/System/Library/Extensions`. The driver's `IOAudioDevice` object should set this property in its implementation of the `initHardware` method.

IOAudioEngine

An audio engine object represents and manages an audio device's I/O engine. In an audio driver, the object is an instance of a custom subclass of `IOAudioEngine`. The audio engine has two main roles:

- To configure a hardware DMA engine to transfer audio data (in the form of a stream of sample frames) between the device and the sample buffer at a specific sampling rate. (In the absence of a hardware DMA engine, the audio engine may emulate this functionality in software.)
- To move data between the sample buffer and the mix buffer after appropriately converting the data to the format expected by the client or hardware (depending on direction).

You can find more information on this topic in "The Audio I/O Model Up Close" (page 31).)

An instance of `IOAudioStream` (described in "IOAudioStream" (page 28)) represents and encapsulates a sample buffer in a driver (and a mix buffer for output streams). Each `IOAudioEngine` in a driver must create one or more `IOAudioStream` objects for each sample buffer required by the I/O engine. A typical driver has at least an input `IOAudioStream` and an output `IOAudioStream`.

An `IOAudioEngine` object may also create the audio-control objects (`IOAudioControl`) required by the device, although this task can be handled by the driver's `IOAudioDevice`. During the initialization phase, the driver must add all created `IOAudioStream` instances and `IOAudioControl` instances to the `IOAudioEngine` as instance variables using the appropriate `IOAudioEngine` methods. This must happen before it activates the `IOAudioEngine` (with the `activateAudioEngine` method).

In addition to facilitating the transfer of audio data in and out of the sample and mix buffers, an `IOAudioEngine` has a number of functions:

- It must stop and start the I/O engine when requested.
- When the I/O engine is started, the `IOAudioEngine` object must ensure that it runs continuously and, at the end of the sample buffer, loops to the beginning of the buffer. As the engine loops, the `IOAudioEngine` takes a timestamp and increments a loop count.
- It must provide the current sample on demand.
- If the `IOAudioEngine` supports multiple stream formats or sampling rates, it must modify the hardware appropriately when a format or rate changes.

An `IOAudioEngine` has several attributes and structures associated with it. Most important of these is a status buffer (`IOAudioEngineStatus`) that it shares with the Audio HAL. This status buffer is a structure that the `IOAudioEngine` must update each time the I/O engine loops around to the start of the sample buffer. The structure contains four fields, three of which hold critical values:

- The number of times the audio engine has looped to the start of the sample buffer
- The timestamp of the most recent occurrence of this looping
- The current location of the erase head (in terms of sample frame)

It is important that these fields, especially the timestamp field, be as accurate as possible. The Core Audio framework (Audio HAL) uses this timing information to calculate the exact position of the audio engine at any time. The shared status buffer is thus the basis for the timer and synchronization mechanism used by the Mac OS X audio subsystem.

The erase head mentioned in the previous paragraph is another attribute of an IOAudioEngine object. The erase head is a software construct that zeroes out the mix and sample buffers just after the sample frames have been played in an output stream. It is always moving just behind the audio engine to avoid erasing data that has not yet been played. However, it also must remain well ahead of the IOAudioEngine clipping and conversion routines that convert the audio data in the mix buffer to ensure that no stale data from a previous loop iteration is mixed or clipped.

An IOAudioEngine object performs a number of initializations to fine-tune the synchronization mechanism described above. For example, it provides methods for setting the latency of the audio engine and for varying the offset between the Audio HAL and the audio engine's I/O head.

IOAudioStream

An IOAudioStream object represents a single, independently addressable audio input or output stream (which may include multiple channels). It contains the following (as instance variables):

- A sample buffer
- A mix buffer (for output streams)
- Supported format information (sample rate, bit depth, and number of channels)
- The starting channel ID
- The number of current clients
- All IOAudioControl objects that affect the channels of the stream

An IOAudioStream is an instance variable of the IOAudioEngine object that creates it. When the audio engine creates an IOAudioStream object, it must list all supported sample formats as well as all the supported sample rates for each format. The current format must be explicitly set.

If a sample buffer has multiple channels, the channels are typically interleaved on a frame-by-frame basis. If your hardware uses separate buffers for each channel, however, you may use separate IOAudioStream instances for different channels.

The IOAudioStream class defines the AudioIOFunction type for the callbacks (typically implemented by the owning IOAudioEngine) that clip and convert output audio data from the float mix buffer to the sample buffer in the format required by the hardware. See "The Audio I/O Model Up Close" (page 31) for further information.

IOAudioStream includes convenience methods that permit IOAudioStream objects to be created from and saved to OSDictionary objects.

IOAudioControl

An IOAudioControl object represents a controllable attribute of an audio device, such as mute, volume, input/output selector, or master gain. It is usually associated with a specific channel in a specific stream, but can be used to control all channels in an IOAudioStream or even all channels in an IOAudioEngine.

IOAudioControl objects are typically instance variables of the owning IOAudioEngine object. However, IOAudioControl objects associated with a specific stream may also be stored in the relevant IOAudioStream object.

Usually an instance of an `IOAudioEngine` subclass creates its `IOAudioControl` instances when it initializes the hardware (in the `initHardware` method). However, the driver's `IOAudioDevice` object may be the object that creates the necessary `IOAudioControl` objects. In either case, the driver must add the control objects to the appropriate `IOAudioEngine` using the `addDefaultAudioControl` method.

Thus an `IOAudioControl` object is associated with an `IOAudioEngine` object, an `IOAudioStream` object, and a channel of the stream. All of its attributes are stored in the I/O Registry. It is known as a "default control" because the Audio HAL recognizes and uses it based on its attributes, which are discovered through the I/O Registry.

When the `IOAudioEngine` (or `IOAudioDevice`) object creates `IOAudioControl` objects, it must obtain from the audio device the starting channel identifier (an integer) for the audio stream. When the driver creates the first `IOAudioControl` for the stream, it assigns this channel ID to it. When it creates `IOAudioControl` objects for any other channels of the stream (based on the number of channels the stream supports), it simply increments the ID of the channel associated with the control.

For audio devices with more than one audio stream, each stream should start at the next free ID beyond the highest numbered ID that the previous stream could contain. This can be obtained by adding the maximum number of channels in any given stream format to the starting ID.

The Audio family assigns `enum` identifiers to channel IDs in `IOAudioTypes.h`; these include identifiers for left, right, center, left-rear, and right-rear channels, as well as an identifier for all channels.

In addition to channel ID, the Audio family uses a multitier classification scheme (defined by enums in `IOAudioTypes.h`) to identify `IOAudioControl` types:

- Audio types: output, input, mixer, pass-through, and processing
- Audio subtypes:
  - For output: internal speaker, external speaker, headphones, line, and S/PDIF
  - For input: internal microphone, external microphone, CD, line, and S/PDIF
- Control types: level and selector
- Control subtypes: volume, mute, input, output, clock services
- Usage type: input, output, and pass-through

Note: For information about creating custom control types beyond those specified, see "Tips, Tricks, and Frequently Asked Questions" (page 65).

When you create an `IOAudioControl` object, you specify control type, control subtype, usage type, and channel name (in addition to channel ID).

The level and selector control types correspond to subclasses of `IOAudioControl`, described in Table 2-1 (page 30).

Table 2-1 Subclasses of IOAudioControl

Subclass	Description
`IOAudioLevelControl`	Implements an audio control based on a minimum and maximum value. A control subtype specifically creates a volume control using minimum and maximum decibels associated with these levels.
`IOAudioSelectorControl`	Implements an audio control based on selection of discrete elements. Control subtypes include those for mute, input/output, and clock services.
`IOAudioToggleControl`	Implements an audio control based on binary values (off and on, start and stop, and so on) such as might pertain to a mute control.

Some objects in an audio driver—typically the `IOAudioDevice` object because of its central role—must implement what is known as "value change handlers." A value change handler is a callback routine that conforms to one of three prototypes defined in `IOAudioControl.h` based on the type of value (integer, `OSObject`, or `void *` data). When invoked, a value change handler should write the change in value to the audio hardware.

Changes to control values that originate with clients of the Audio HAL—for example, a user moving the volume slider in the menu bar—initiate a long series of actions in the Audio HAL and the Audio family:

1. The Audio HAL goes through the I/O Registry to determine the property or properties associated with the value change.
2. Via the `IOAudioEngineUserClient` object, the `IOAudioControl` superclass's implementation of `setProperties` is invoked.
3. Using the dictionary of properties passed into `setProperties`, `IOAudioControl` locates the target control object and calls `setValueAction` on it.
4. The `setValueAction` method calls `setValue` on the driver's work loop while holding the driver's command gate.
5. The `setValue` method first calls `performValueChange`, which does two things:
    a. It calls the value change handler for the `IOAudioControl` (which must conform to the appropriate function prototype for the callback).
    b. It sends a notification of the change to all clients of the `IOAudioControl` (`sendValueChangeNotification`).
6. Finally, `setValue` calls `updateValue` to update the I/O Registry with the new value.

When a change is physically made to audio hardware—for example, a user turns a volume dial on an external speaker—what must be done is much abbreviated. When the driver detects a control-value change in hardware, it simply calls `hardwareValueChanged` on the driver's work loop. This method updates the value in the `IOAudioControl` instance and in the I/O Registry, and then sends a notification to all interested clients.

User Client Classes

The Audio family provides two user-client classes, `IOAudioEngineUserClient` and `IOAudioControlUserClient`. The Audio family automatically instantiates objects of each class for each `IOAudioEngine` object and each `IOAudioControl` in a driver. These objects enable the communication of audio data and notifications between the driver and the Audio HAL. You should not have to do anything explicitly in your code to have the default user-client objects created for, and used by, your driver.

For further details, see "User Client Objects" (page 35).

IOAudioPort

The `IOAudioPort` class instantiates objects that represent a logical or physical port, or a functional unit in an audio device. An `IOAudioPort` object represents an element in the signal chain in the audio device and may contain one or more `IOAudioControl` objects through which different attributes of the port can be represented and adjusted.

The `IOAudioPort` class is deprecated and may eventually be made obsolete. The class is currently public to maintain compatibility. Driver writers are discouraged from using `IOAudioPort` objects in their code.

The Audio I/O Model Up Close

In the previous chapter, the section "The Audio I/O Model on Mac OS X" (page 17) described the Mac OS X audio I/O model from the perspective of how that model compares to the Mac OS 9 model. Because it was a comparative overview, that description left out some important details. The following discussion supplies those details, with the intent that a fuller understand of the audio I/O model is of particular benefit to audio driver writers.

Ring Buffers and Timestamps

In Mac OS X, the driver's audio engine programs the audio device's DMA engine to read from or write to a single (typically large) ring buffer. In a ring buffer, the DMA engine (or a software emulation thereof) wraps around to the start of the buffer when it finishes writing to (or reading from) the end of the buffer. Thus the DMA engine continuously loops through the sample buffer, reading or writing audio data, depending on direction. As it wraps, the DMA engine is expected to fire an interrupt. The driver (in an `IOAudioEngine` object) records the time when this interrupt occurs by calling `takeTimeStamp` in the driver's work loop.

In calling `takeTimeStamp`, the driver writes two critical pieces of data to an area of memory shared between the `IOAudioEngine` and its Audio HAL clients. The first is an extremely accurate timestamp (based on `utime`), and the other is an incremented loop count. The structure defining these (and other) fields is `IOAudioEngineStatus`. The engine's user-client object maps the memory holding the `IOAudioEngineStatus` information into the address spaces of the Audio HAL clients.

For information about handling timestamp approximation, see "Faking Timestamps" (page 66).

The Audio HAL Predicts

The Audio HAL uses the accumulated timestamps and loop counts in a sophisticated calculation that predicts when the I/O engine will be at any location in the sample buffer; as a result, it can also predict when each client of a particular I/O engine should be ready to provide audio data to the hardware or accept audio data from it. This calculation takes into account not only the current sample-frame position of the I/O engine, but also the buffer sizes of the clients, which can vary.

Each client of the Audio HAL has its own I/O thread. The Audio HAL puts this thread to sleep until the time comes for the client to read or write audio data. Then the Audio HAL wakes the client thread. This is a kind of software-simulated interrupt, which involves much less overhead than a hardware interrupt.

Interpolation

Before going further, it is worthwhile to consider some of the theory behind this design. The Mac OS X audio system makes the assumption that a hardware I/O engine, as it processes audio data in the sample buffer, is proceeding continuously at a *more or less* constant rate. The "more or less" qualification is important here because, in reality, there will be slight variations in this rate for various reasons, such as imperfections in clock sources. So the mechanism by which the Audio HAL continually uses timestamps to calculate and predict a wake-up time for each of its client I/O threads can be considered an interpolation engine. It is a highly accurate predictive mechanism that "smooths out" these slight variations in engine rate, building in some leeway so that there is no discernible effect on audio quality.

Client Buffers and I/O Procedures

As noted earlier, each client of the Audio HAL can define the size of its audio buffer. There are no restrictions, except that the buffer can be no larger than the size of the hardware sample buffer. For performance reasons, almost all clients prefer buffer sizes that are considerably smaller. Buffer sizes are typically a power of two. The Audio HAL takes the buffer sizes of its clients into account when it calculates the next I/O cycle for those clients.

Each client of the Audio HAL must also implement a callback function conforming to the type `AudioDeviceIOProc`. When the Audio HAL wakes a sleeping client I/O thread, it calls this function, passing in the buffers (input and output) whose sizes were specified by the client. It is in this implementation of the `AudioDeviceIOProc` routine that the client gives audio data to the hardware or receives it from the hardware.

The following section, "A Walk Through the I/O Model" (page 32), discusses what happens next in detail.

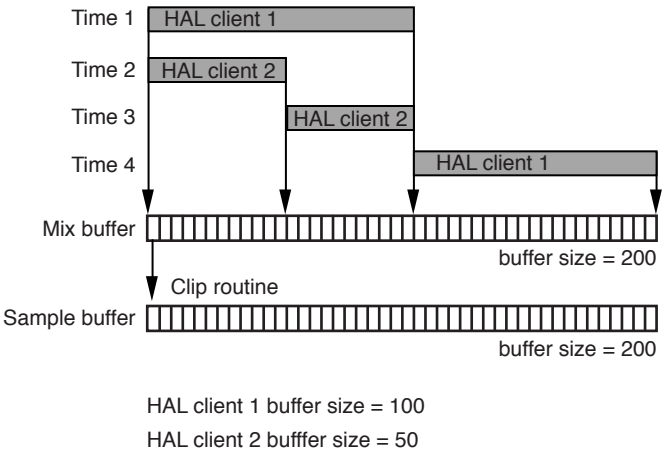
A Walk Through the I/O Model

With the essential timing mechanism used for audio I/O in mind, we can now follow a cycle of that I/O through the audio system to see exactly what happens. What happens is different between input and output audio streams. An output stream involves a more complicated path because each client is contributing, frame by frame, to the final sound played by speakers or recorded by some device.

Output Streams

Let's begin with an output stream. Figure 2-4 (page 33) illustrates the relationship between the buffers of Audio HAL clients and the buffers of the audio driver during an output cycle. Refer to this diagram during the following discussion.

Figure 2-4 Multiple Audio HAL client buffers and the mix buffer (output)



For each of its clients, the Audio HAL calculates intervals that are based on the accumulated timestamps and loop counts associated with an I/O engine as well as client buffer sizes. The Audio HAL sleeps the I/O threads of its clients for these intervals, waking each thread when it's time for the client to give the hardware its data. In waking the thread, it calls the `AudioDeviceIOProc` routine implemented by the client, passing in a number of buffers and timestamps:

- A list of input buffers along with a timestamp that indicates when the data was recorded
- A list of output buffers along with a timestamp that indicates when the data will be played
- A timestamp to be used for "now" rather than the device clock

The input and output timestamps allow the client to make various calculations, such as how much time it has before the data is played. The inclusion of both input and output parameters enables clients that are both producers and consumers of audio data (for example, a recording unit with playback capabilities) to process both streams at the same time. In this case, the client first takes the data in the list of input buffers before filling the output buffers with 32-bit floating-point samples.

When the client returns in its `AudioDeviceIOProc` routine, the Audio HAL puts the I/O thread to sleep until the next time data is required from the client. The Audio HAL gives the samples in the output buffer to the associated `IOAudioEngineUserClient` object, which calls the appropriate `IOAudioStream` object to have the samples moved from the client buffer to the appropriate frames in the engine's mix buffer. Other clients can also deposit data in the same locations in the mix buffer. If another client already has deposited data in those frames, the new client's floating-point values are simply added to the existing values.

Clients can contribute output data to a frame almost until the I/O engine is ready for that data. The `IOAudioEngine` object containing the mix buffer knows how many clients it has and when each has contributed its share of data to any one frame of the mix buffer (for the current loop through it). In addition, the driver (through the Audio family) maintains a "watchdog" timer that tracks the current

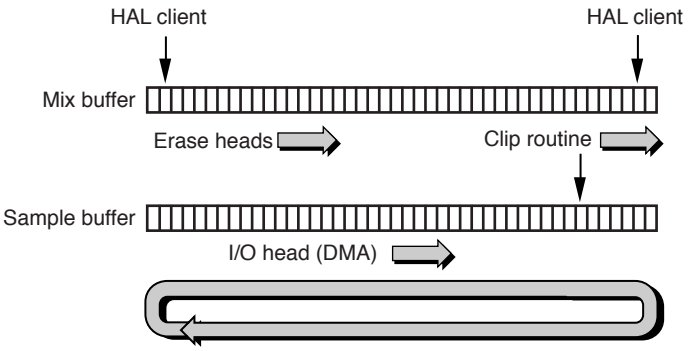
location of each client relative to the I/O engine. If a client has not provided audio data by the time the I/O engine needs to accesses it, the watchdog timer fires and clips all of the currently mixed samples into the sample buffer.

Because some time is needed to perform this clip operation, the watchdog actually fires a short amount of time before the data is needed. It is possible that a "late" client could attempt to put data in the location of the mix buffer after the watchdog has fired but before the I/O engine has processed the data. To accommodate this situation, the driver backs up and remixes and clips the data in an attempt to get the "late" samples to the I/O engine in time.

Next, the driver's clip routine, `clipOutputSamples`, is invoked. In its implementation of this method, the driver must clip any excess floating-point values under –1.0 and over 1.0 —which can happen when multiple clients are adding their values to existing values in the same frame—and then convert these values to whatever format is required by the hardware. When `clipOutputSamples` returns, the converted values have been written to the corresponding locations in the sample buffer. The DMA engine grabs the frames as it progresses through the sample buffer and the hardware plays them as sound.

Since a picture is worth a thousand words, the interaction of these processes is described in Figure 2-5 (page 34).

Figure 2-5 Interplay of the I/O engine, erase heads, and clip routine (output)



Erase Heads and Timer Services

The Audio family includes a further refinement to the synchronized actions described in the preceding paragraphs. Between the I/O engine and the clipping and converting done by the driver, it runs parallel "erase heads" in both the mix and sample buffers. These erase heads simply zero-fill the corresponding frames at the same time. This precaution reduces the possibility that any frame could become polluted with leftover bits.

The erase heads are run in a separate thread and have their own timer. They are programmed to run four times per sample-buffer cycle. They do not erase the entire range of frames between the current locations of the DMA engine and the driver's clip routine, allowing a little space for the remixing of data from tardy clients.

The erase head's timer is run using `IOAudioDevice`'s timer services. Its interval is, of course, closely tied to the rate of the I/O engine and the timestamps taken by the `IOAudioEngine`.

Input Streams

With input audio streams, the picture is much simpler. There is no mix buffer and there are no erase heads. The Audio HAL clients are consumers of the data in this case, and not producers of it, so there is no need for these things.

Neither is there any need for a clip routine. The driver has to convert the integer data coming from the hardware to the 32-bit floating point required by the Audio HAL. But in the input direction, the driver is in a position to fit the converted data within the –1.0 to 1.0 floating-point maximum range.

So the simplified sequence is this: shortly after the I/O engine writes the input data into the sample buffer, the driver—in its implementation of the `IOAudioEngine` method `convertInputSamples`—converts that data to 32-bit floating point. Then the data is given, via the `IOAudioEngineUserClient` interface, to each Audio HAL client in that client's `AudioDeviceIOProc` callback routine.

Interfaces With the Audio HAL

Audio drivers communicate with the Audio HAL and its clients using two mechanisms. The principal mechanism uses user-client objects to pass audio data, control value changes, and notifications across the kernel-user space boundary. The other mechanism allows driver writers to export custom device properties to Audio HAL clients.

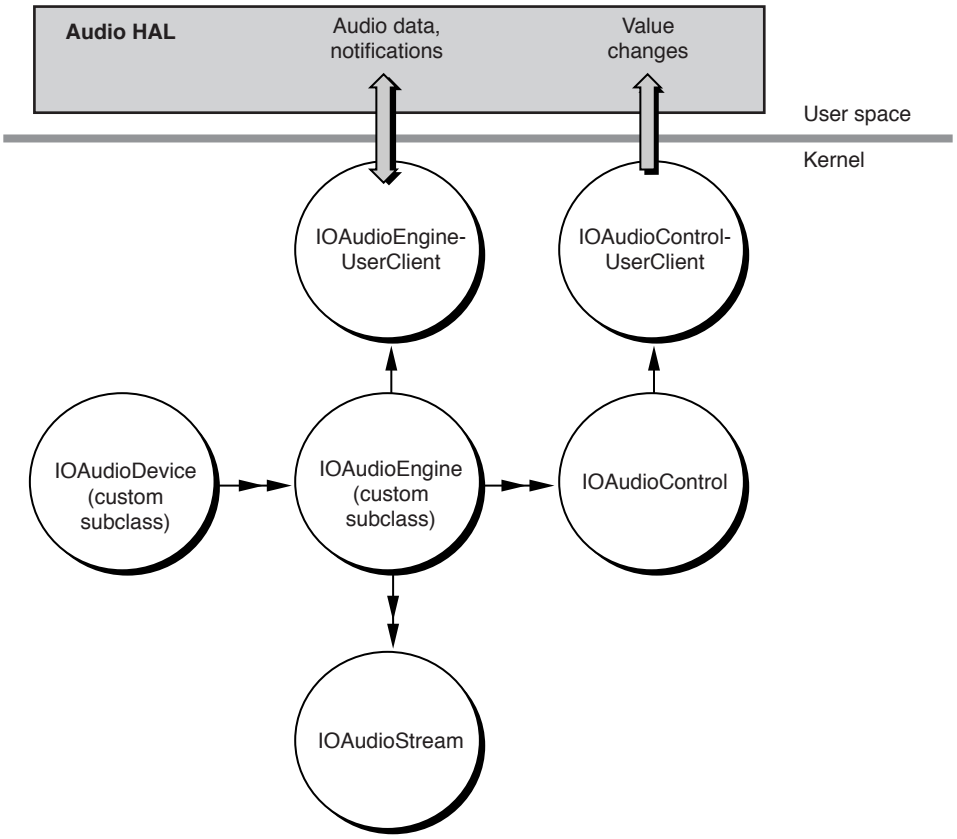
User Client Objects

As described earlier in "User Client Classes" (page 31), the Audio family automatically configures and creates the appropriate number of user-client objects for a driver. These objects enable the communication of audio data and notifications between the driver and the Audio HAL. The objects typically are instances of either the `IOAudioEngineUserClient` class or the `IOAudioControlUserClient` class. One `IOAudioEngineUserClient` object is created for each `IOAudioEngine` in a driver, and one `IOAudioControlUserClient` object is created for each `IOAudioControl`.

An `IOAudioEngineUserClient` object acts as the conduit through which audio data is passed between the audio driver and the Audio HAL. It is also the agent that maps the buffer maintained by an `IOAudioEngine` into memory shared with the associated Audio HAL device. (As you may recall, this buffer contains the timestamp and count of the most recent "wrap" of the sample buffer by the I/O engine.) Finally, the `IOAudioEngineUserClient` responds to requests by Audio HAL clients to get and set the properties of the hardware. An `IOAudioControlUserClient` object has a more limited role compared to a `IOAudioEngineUserClient` object. It merely sends a notification to interested clients of the Audio HAL when the value of a control (represented by `IOAudioControl`) changes.

The interaction of these user clients with the Audio HAL and other parts of your driver is shown in Figure 2-6 (page 36).

Figure 2-6 The Audio family's user clients



You do not have to do anything explicitly in your code to have the Audio family create the default user-client objects for your driver, nor do you have to write any code to enable your driver to use these objects. It all happens automatically for your driver. But it could happen that you want custom behavior from your user clients; for example, you might want the user client to perform hardware mixing, writing the combined samples into a new buffer used by a single client. When you want custom user-client behavior, you can subclass the `IOAudioEngineUserClient` class or the `IOAudioControlUserClient` class. These classes are described in `IOAppleEngineUserClient.h` and `IOAppleControlUserClient.h`.

Custom Core Audio Properties

Sometimes you might have an audio device with properties that are not covered by what the Audio HAL specifies (in the Core Audio framework's `AudioHardware.h`). For these situations, you can create a bundle that contains code implementing these device-specific properties on behalf of the Audio HAL. Then you can put this bundle in a location where it can be exported to user space. The bundle must be accessible by Core Foundation Bundle Services APIs (CFBundle).

To give the Audio HAL access to your device-property code, the bundle must also present the programmatic interface defined in the Core Audio header file `AudioDriverPlugIn.h`. These routines allow the Audio HAL clients to open and close the device and to get and set the device properties.

When the driver changes a property, it calls one of two notification callbacks implemented by clients (one defined for Audio HAL device properties and the other for stream properties) to notify them of the change.

When you have created a bundle conforming to the interface in `AudioDriverPlugIn.h`, you usually install it inside your driver's kernel extension in `/System/Library/Extensions`. Because it is a bundle, it can also contain localizations of strings relevant to the new properties. The Audio HAL finds the bundle by looking in the I/O Registry for two keys: `kIOAudioEngineCoreAudioPlugInKey` and `kIOAudioDeviceLocalizedBundleKey`.

Implementing an Audio Driver

As discussed in the chapter "Audio Family Design" (page 21), writing an audio driver using the Audio family requires, in object-oriented terms, that you do some certain things in your code:

- Create a subclass of `IOAudioDevice` which, among other things, initializes the hardware and registers for sleep/wake notifications.
- Create a subclass of `IOAudioEngine` which, among other things, initializes the I/O engine and stops and starts it.
- Create, configure, and attach to the `IOAudioEngine` object the number of `IOAudioStream` and `IOAudioControl` objects appropriate to your driver.
- Respond to value changes in the `IOAudioControl` objects.
- In a separate code module (but as part of the `IOAudioEngine` subclass implementation), implement the driver's clipping and converting routines.

This chapter will guide you through these implementation steps. It uses as a code source the SamplePCIAudioDriver example project (located in `/Developer/Examples/Kernel/IOKit/Audio/Templates` when you install the Developer package). In the interest of brevity, this chapter does not use all the code found in that project and strips the comments from the code. Refer to the SamplePCIAudioDriver project for the full range of code and comments on it.

Setting Up the Project

Even before you create a project for your audio driver, you should consider some elemental facets of design. Examine the audio hardware and decide which Audio-family objects are required to support it. Of course, your driver must have one `IOAudioDevice` object (instantiated from a custom subclass), but how many `IOAudioEngine`, `IOAudioStream`, and `IOAudioControl` objects should you create?

Table 3-1 (page 40) provides a decision matrix for determining how many Audio-family objects of each kind that you need.

Table 3-1 Deciding which Audio family objects to create (and other design decisions)

Question	What to create
Are there sample buffers of different sizes?	Create a custom `IOAudioEngine` object for each sample buffer.
How many I/O or DMA engines are there on the device?	Create a custom `IOAudioEngine` object for each I/O or DMA engine.
How many separate or interleaved sample buffers are there?	Create an `IOAudioStream` object for each buffer (both input and output).
How many controllable attributes are there (volume, gain, mute, and so on)?	Create an `IOAudioControl` object for each attribute.

The SamplePCIAudioDriver project requires one custom `IOAudioEngine` subclass object, two `IOAudioStream` objects (input and output), and six `IOAudioControl` objects (left and right output volume, left and right input gain, and input and output mute).

You also should decide what properties your driver must have to match against your provider's nub and specify those properties in your driver's `IOKitPersonalities` dictionary. In the SamplePCIAudioDriver personality (see Figure 3-1 (page 40)), the provider is the PCI family and the nub class is `IOPCIDevice`. In addition, a PCI audio driver would usually specify the vendor and device ID registers (primary or subsystem) as the value of the `IOPCIMatch` key. (Note that in the SamplePCIAudioDriver example, the vendor and device ID registers are specified as zeros; for your driver, you would substitute the appropriate values.) Finally, for your `IOClass` property, append the name of your `IOAudioDevice` subclass to the standard reverse-DNS construction `com_`*company*`_driver_`; in the case of the SamplePCIAudioDriver project, the `IOClass` value is `com_MyCompany_driver_SamplePCIAudioDevice`.

Figure 3-1 Bundle settings of the sample PCI audio driver

Property List	Class	Value
CFBundleDevelopmentRegion	String	English
CFBundleExecutable	String	SamplePCIAudioDriver
CFBundleIconFile	String	
CFBundleIdentifier	String	com.MyCompany.driver.SamplePCIAudioDriver
CFBundleInfoDictionaryVersion	String	6.0
CFBundlePackageType	String	KEXT
CFBundleSignature	String	????
CFBundleVersion	String	1.0.0d1
▼IOKitPersonalities	Dictionary	1 key/value pairs
▼SamplePCIAudioDriver	Dictionary	4 key/value pairs
CFBundleIdentifier	String	com.MyCompany.driver.SamplePCIAudioDriver
IOClass	String	com_MyCompany_driver_SamplePCIAudioDevice
IOPCIMatch	String	0x00000000
IOProviderClass	String	IOPCIDevice
▼OSBundleLibraries	Dictionary	2 key/value pairs
com.apple.iokit.IOAudioFamily	String	1.1fc6
com.apple.iokit.IOPCIDevice	String	1.0

Note: For more on PCI device matching, see the document Writing PCI Drivers.

Of course, if your driver's provider is different (say, USB or FireWire), the matching properties that you would specify in an `IOKitPersonalities` dictionary would be different.

As Figure 3-1 (page 40) suggests, also make sure that you specify other necessary properties in your driver's `Info.plist` file, including the versioning and dependency information in the `OSBundleLibraries` dictionary.

Warning: Use of floating point is generally discouraged in the kernel to avoid a performance penalty during function calls. Threads running in the kernel only keep floating point registers in their stack if floating point has already been used in that thread.

For threads called from user space (including the threads that call `clipOutputSamples` and `convertInputSamples`), however, floating point has already been used, and thus no additional penalty is incurred by using hardware floating point.

However, there is a catch. By default, most I/O Kit code is compiled with floating-point emulation (the `-msoft-float` compiler flag). You must be very careful to ensure that code that needs to interact with actual floating-point samples does *not* get compiled with this compiler flag. This is described in more detail in the relevant sections.

Implementing an IOAudioDevice Subclass

Every I/O Kit audio driver must implement a subclass of `IOAudioDevice`. One instance of this class is created when the driver is loaded. An `IOAudioDevice` object is the central, coordinating object of the driver; it represents the audio hardware in an overall sense.

Despite its central role, an `IOAudioDevice` subclass generally does not do as much as an `IOAudioEngine` subclass. It merely initializes the hardware at startup and creates the custom `IOAudioEngine` objects required by the driver. It may also create the `IOAudioControl` objects used by the driver and respond to requests to change the values of these controls, but the `IOAudioEngine` subclass could do these tasks instead. In the example used for this chapter (SamplePCIAudioDriver), the `IOAudioDevice` subclass creates and manages the device's controls.

Begin by adding a header file and an implementation file for the `IOAudioDevice` superclass you are going to implement. In the header file, specify `IOAudioDevice` as the superclass and provide the necessary declarations.

Listing 3-1 (page 41) shows the beginning of `SamplePCIAudioDevice.h`.

Listing 3-1 Partial class declaration of the IOAudioDevice subclass

```
#include <IOKit/audio/IOAudioDevice.h>

typedef struct SamplePCIAudioDeviceRegisters {
    UInt32 reg1;
    UInt32 reg2;
    UInt32 reg3;
    UInt32 reg4;
} SamplePCIAudioDeviceRegisters;
```

```
class IOPCIDevice;
class IOMemoryMap;

#define SamplePCIAudioDevice com_MyCompany_driver_SamplePCIAudioDevice

class SamplePCIAudioDevice : public IOAudioDevice
{
    friend class SampleAudioEngine;

    OSDeclareDefaultStructors(SamplePCIAudioDevice)

    IOPCIDevice     *pciDevice;
    IOMemoryMap     *deviceMap;

    SamplePCIAudioDeviceRegisters *deviceRegisters;
// ...
};
```

Hardware Initialization

I/O Kit audio drivers do not need to override the `IOService::start` method. Instead, the default `IOAudioDevice` implementation of `start` first invokes the superclass implementation and then calls the `initHardware` method of the subclass. Your `IOAudioDevice` subclass must override the `initHardware` method.

Your implementation of `initHardware` must do two general things:

- It must perform any necessary hardware-specific initializations (on both the provider and the audio sides), such as mapping resources and setting the hardware to a known state. It also involves creating and initializing the necessary Audio family objects.
- It must set the names by which the driver is to be known to the Audio HAL and its clients.

If the `initHardware` call succeeds, the `IOAudioDevice` superclass (in the `start` method) sets up power management if the family is supposed to manage power and then calls `registerService` to make the `IOAudioDevice` object visible in the I/O Registry.

Listing 3-2 (page 42) shows how the SamplePCIAudioDevice class implements the `initHardware` method.

Listing 3-2 Implementing the initHardware method

```
bool SamplePCIAudioDevice::initHardware(IOService *provider)
{
    bool result = false;

    IOLog("SamplePCIAudioDevice[%p]::initHardware(%p)\n",  this, provider);

    if (!super::initHardware(provider)) {
        goto Done;
    }

    pciDevice = OSDynamicCast(IOPCIDevice, provider);
    if (!pciDevice) {
```

```
        goto Done;
    }

    deviceMap = pciDevice->mapDeviceMemoryWithRegister(kIOPCIConfigBaseAddress0);
    if (!deviceMap) {
        goto Done;
    }

    deviceRegisters = (SamplePCIAudioDeviceRegisters
*)deviceMap->getVirtualAddress();
    if (!deviceRegisters) {
        goto Done;
    }

    pciDevice->setMemoryEnable(true);

    setDeviceName("Sample PCI Audio Device");
    setDeviceShortName("PCIAudio");
    setManufacturerName("My Company");

#error Put your own hardware initialization code here...and in other  routines!!

    if (!createAudioEngine()) {
        goto Done;
    }

    result = true;

Done:

    if (!result) {
        if (deviceMap) {
            deviceMap->release();
            deviceMap = NULL;
        }
    }

    return result;
}
```

The first part of this method does some provider-specific initializations. The implementation gets the provider, an `IOPCIDevice` object, and with it, configures a map for the PCI configuration space base registers. With this map, it gets the virtual address for the registers. Then it enables PCI memory access by calling `setMemoryEnable`.

Next, the SamplePCIAudioDevice implementation sets the full and short name of the device as well as the manufacturer's name, making this information available to the Audio HAL.

The last significant call in this implementation is a call to `createAudioEngine`. This method creates the driver's `IOAudioEngine` and `IOAudioControl` objects (and, indirectly, the driver's `IOAudioStream` objects).

Creating the IOAudioEngine Objects

In the `initHardware` method, create an instance of your driver's `IOAudioEngine` subclass for each I/O engine on the device. After it's created and initialized, call `activateAudioEngine` to signal to the Audio HAL that the engine is ready to begin vending audio services.

The SamplePCIAudioDevice subclass creates its sole `IOAudioEngine` object in a subroutine of `initHardware` named `createAudioEngine` (see Listing 3-3 (page 44)).

Listing 3-3 Creating an IOAudioEngine object

```
bool SamplePCIAudioDevice::createAudioEngine()
{
    bool result = false;
    SamplePCIAudioEngine *audioEngine = NULL;
    IOAudioControl *control;

    audioEngine = new SamplePCIAudioEngine;
    if (!audioEngine) {
        goto Done;
    }
    if (!audioEngine->init(deviceRegisters)) {
        goto Done;
    }
     // example code skipped...
     // Here create the driver's IOAudioControl objects
     // (see next section)...

    activateAudioEngine(audioEngine);

    audioEngine->release();
    result = true;

Done:
    if (!result && (audioEngine != NULL)) {
        audioEngine->release();
    }
    return result;
}
```

Note: In the interest of concision, this listing excludes the code that creates the driver's `IOAudioControl` objects; see "Creating and Adding the IOAudioControl Objects" (page 45) for this step.

In this example, the `IOAudioDevice` subclass creates a raw instance of the driver's subclass of `IOAudioEngine` (SamplePCIAudioEngine) and then initializes it, passing in the device registers so the object can access those registers. You can define your `init` method to take any number of parameters.

Next, the `IOAudioDevice` implementation activates the audio engine (`activateAudioEngine`); this causes the newly created `IOAudioEngine` object's `start` and `initHardware` methods to be invoked. When `activateAudioEngine` returns, the `IOAudioEngine` is ready to begin vending audio services to the system. Because the `IOAudioDevice` superclass retains the driver's `IOAudioEngine` objects, be sure to release each `IOAudioEngine` object so that it is freed when the driver is terminated.

Creating and Adding the IOAudioControl Objects

A typical I/O Kit audio driver must instantiate several `IOAudioControl` objects to help it manage the controllable attributes of the audio hardware. These attributes include such things as volume, mute, and input/output selection. You can create and manage these control objects in your `IOAudioEngine` subclass or in your `IOAudioDevice` subclass; it doesn't matter which.

As summarized in Table 3-2 (page 45), the Audio family provides three subclasses of `IOAudioControl` that implement behavior specific to three functional types of control. Instantiate a control from the subclass that is appropriate to a controllable attribute of the device.

Table 3-2 Subclasses of IOAudioControl

Subclass	Purpose
`IOAudioLevelControl`	For controls such as volume, where a range of measurable values (such as decibels) is associated with an integer range.
`IOAudioToggleControl`	For controls such as mute, where the state is either off or on.
`IOAudioSelectorControl`	For controls that select a discrete attribute, such as input gain.

Each subclass (or control type) has a `create` method and a convenience method specific to a subtype of control. The `IOAudioTypes.h` header file, which defines constants for control type and subtype, also defines other constants intended to be supplied as parameters in the control-creation methods. Table 3-3 (page 45) summarizes the categories that these constants fall into.

Table 3-3 Categories of audio-control constants in IOAudioTypes.h

Category	Purpose	Examples and comments
Type	General function of control	Level, toggle, or selector (each corresponding to an `IOAudioControl` subclass).
Subtype	Purpose of the control	Volume, mute, or input/output; subclass convenience methods assume a subtype.
Channel ID	Common defaults for channels	Default right channel, default center channel, default sub woofer, all channels.
Usage	How the control is to be used	Output, input, or pass-through.

See `IOAudioTypes.h` for the complete set of audio-control constants.

After you create an `IOAudioControl` object you must take two further steps:

- Set the value-change handler for the control.

  The value-change handler is a callback routine that is invoked when a client of the Audio HAL requests a change in a controllable attribute. See "Implementing Control Value-Change Handlers" (page 47) for more on these routines.

- Add the `IOAudioControl` to the `IOAudioEngine` object they are associated with.

In the SamplePCIAudioDriver example, the `IOAudioDevice` subclass creates and initializes the driver's `IOAudioControl` objects. This happens in the `createAudioEngine` method; Listing 3-4 (page 46) shows the creation and initialization of one control.

Listing 3-4 Creating an IOAudioControl object and adding it to the IOAudioEngine object

```
    // ... from createAudioEngine()
    control = IOAudioLevelControl::createVolumeControl(
          65535,     // initial value
          0,     // min value
          65535,     // max value
          (-22 << 16) + (32768),     // -22.5 in IOFixed (16.16)
          0,     // max 0.0 in IOFixed
          kIOAudioControlChannelIDDefaultLeft,
          kIOAudioControlChannelNameLeft,
          0,     // control ID - driver-defined
          kIOAudioControlUsageOutput);
    if (!control) {
        goto Done;
    }
    control->setValueChangeHandler((IOAudioControl::IntValueChangeHandler)
                                   volumeChangeHandler, this );
    audioEngine->addDefaultAudioControl(control);
    control->release();

/* Here create more IOAudioControl objects for right output channel,
** output mute,left and right input gain, and input mute. For each,  set
** value change handler and add to the IOAudioEngine
*/
// ...
```

Note: This code fragment shows the creation of only one `IOAudioControl` object. What must be done for the other controls—right output volume, output mute, left and right input gain, and input mute—is similar.

In this example, the `IOAudioDevice` subclass creates a left output volume control with an integer range from 0 to 65535 and a corresponding decibel range from –22.5 to 0.0. A channel must always be associated with an `IOAudioControl` object. You do this when you create the object by specifying constants (defined in `IOAudioDefines.h`) for both channel ID and channel name. You must also specify a "usage" constant that indicates how the `IOAudioControl` will be used (input, output, or pass-through).

Once you have added an `IOAudioControl` to its `IOAudioEngine`, you should release it so that it is properly freed when the `IOAudioEngine` object is done with it.

Handling Sleep/Wake Notifications

As the power controller for your device, it is necessary to register for sleep/wake notifications. At a minimum, your handlers should stop and restart any audio engines under their control. Depending on the device, this may not be sufficient, however.

In general—and particularly for PCI devices—device power will be cycled during sleep, but the device will not disappear from the device tree. This means that your driver will not be torn down and reinitialized. Thus, for these devices, it is crucial that you register for sleep/wake notifications and reinitialize your device registers to a known state on wake. Otherwise, unexpected behavior may result.

For information about how to register for sleep/wake notifications, see the Power Management chapter of *I/O Kit Fundamentals*.

Implementing Control Value-Change Handlers

For each `IOAudioControl` object that your driver creates, it must implement what is known as a value-change handler for it. (This doesn't imply that you need you need to create a separate handler for each control; one handler could be used to manage multiple controls of similar type.) The value-change handler is a callback routine that is invoked when the controllable device attribute associated with an `IOAudioControl` object needs to be changed.

The header file `IOAudioControl.h` defines three prototypes for control value-change handlers:

```
typedef IOReturn (*IntValueChangeHandler)(OSObject *target,
    IOAudioControl *audioControl, SInt32 oldValue, SInt32 newValue);
typedef IOReturn (*DataValueChangeHandler)(OSObject *target,
    IOAudioControl *audioControl, const void *oldData, UInt32
    oldDataSize, const void *newData, UInt32 newDataSize);
typedef IOReturn (*ObjectValueChangeHandler)(OSObject *target,
    IOAudioControl *audioControl, OSObject *oldValue,
    OSObject *newValue);
```

Each prototype is intended for a different kind of control value: integer, pointer to raw data (`void *`), and (`libkern`) object. For most cases, the integer handler should be sufficient. All of the existing `IOAudioControl` subclasses pass integer values to the `IntValueChangeHandler` object.

The essential task of the value-change handler is to update the proper attribute of the audio hardware to the new control value. Listing 3-5 (page 47) shows how one might implement a value-change handler (excluding the actual attribute-setting code).

Listing 3-5 Implementing a control value-change handler

```
IOReturn SamplePCIAudioDevice::volumeChangeHandler(IOService *target,
        IOAudioControl *volumeControl, SInt32 oldValue, SInt32 newValue)
{
    IOReturn result = kIOReturnBadArgument;
    SamplePCIAudioDevice *audioDevice;

    audioDevice = (SamplePCIAudioDevice *)target;
    if (audioDevice) {
        result = audioDevice->volumeChanged(volumeControl, oldValue,
                    newValue);
    }

    return result;
}

IOReturn SamplePCIAudioDevice::volumeChanged(IOAudioControl *volumeControl,
                            SInt32 oldValue, SInt32 newValue)
```

```
{
    IOLog("SamplePCIAudioDevice[%p]::volumeChanged(%p, %ld,  %ld)\n", this,
            volumeControl, oldValue, newValue);

    if (volumeControl) {
        IOLog("\t-> Channel %ld\n", volumeControl->getChannelID());
    }

    // Add hardware volume code change

    return kIOReturnSuccess;
}
```

The reason for the nested implementation in this example is that the value-change callback itself must be a straight C-language function (in this case, it's a static member function). The static function simply forwards the message to the actual target for processing.

Implementing an IOAudioEngine Subclass

In addition to implementing a subclass of `IOAudioDevice`, writers of audio drivers must also implement a subclass of `IOAudioEngine`. This subclass should define the attributes and behavior of the driver that are specific to the hardware's I/O engine. These include specifying the size and characteristics of the sample and mix buffers, getting the current sample frame on demand, handling interrupts to take a timestamp, handling format changes, and starting and stopping the I/O engine upon request.

Start by defining the interface of your `IOAudioEngine` subclass in a header file. Listing 3-6 (page 48) shows the main contents of the `SamplePCIAudioEngine.h` file.

Listing 3-6 Interface definition of the SamplePCIAudioEngine class

```
class SamplePCIAudioEngine : public IOAudioEngine
{
    OSDeclareDefaultStructors(SamplePCIAudioEngine)

    SamplePCIAudioDeviceRegisters   *deviceRegisters;

    SInt16                          *outputBuffer;
    SInt16                          *inputBuffer;

    IOFilterInterruptEventSource     *interruptEventSource;

public:

    virtual bool init(SamplePCIAudioDeviceRegisters *regs);
    virtual void free();

    virtual bool initHardware(IOService *provider);
    virtual void stop(IOService *provider);

    virtual IOAudioStream *createNewAudioStream(IOAudioStreamDirection
            direction, void *sampleBuffer, UInt32 sampleBufferSize);

    virtual IOReturn performAudioEngineStart();
    virtual IOReturn performAudioEngineStop();
```

```
    virtual UInt32 getCurrentSampleFrame();

    virtual IOReturn performFormatChange(IOAudioStream *audioStream,
            const IOAudioStreamFormat *newFormat, const IOAudioSampleRate
            *newSampleRate);

    virtual IOReturn clipOutputSamples(const void *mixBuf, void  *sampleBuf,
            UInt32 firstSampleFrame, UInt32 numSampleFrames, const
            IOAudioStreamFormat *streamFormat, IOAudioStream *audioStream);
    virtual IOReturn convertInputSamples(const void *sampleBuf,  void *destBuf,

            UInt32 firstSampleFrame, UInt32 numSampleFrames, const
            IOAudioStreamFormat *streamFormat, IOAudioStream *audioStream);

    static void interruptHandler(OSObject *owner, IOInterruptEventSource
            *source, int count);
    static bool interruptFilter(OSObject *owner, IOFilterInterruptEventSource
            *source);
    virtual void filterInterrupt(int index);
};
```

Most of the methods and types declared here are explained in the following sections—including (for example) why there is a cluster of interrupt-related methods.

Note: The `clipOutputSamples` and `convertInputSamples` methods, although they are declared in the class interface-definition file (in this case, `SamplePCIAudioEngine.h`), are implemented in a separate source module. The reason for that, and the procedures for implementing these methods, are found in "Clipping and Converting Samples" (page 56).

Hardware Initialization

As you did in your `IOAudioDevice` subclass, you must implement the `initHardware` method in your `IOAudioEngine` subclass to perform certain hardware initializations. The `IOAudioEngine` `initHardware` method is invoked indirectly when the `IOAudioDevice` object calls `activateAudioEngine` on an `IOAudioEngine` object.

In your implementation of `initHardware`, you should accomplish two general tasks: configure the I/O engine and create the `IOAudioStream` objects used by the engine. As part of initialization, you should also implement the `init` method if anything special should happen prior to the invocation of `initHardware`; in the case of the SamplePCIAudioEngine class, the `init` method calls the superclass implementation and then assigns the passed-in device registers to an instance variable.

Configuring the I/O Engine

Configuring the audio hardware's I/O engine involves the completion of many recommended tasks:

- Determine the current sample rate and set the initial sample rate using `setSampleRate`.
- Call `setNumSampleFramesPerBuffer` to specify the number of sample frames in each buffer serviced by this I/O engine.
- Call `setDescription` to make the name of the I/O engine available to Audio HAL clients.

- Call `setOutputSampleLatency` or `setInputSampleLatency` (or both methods, if appropriate) to indicate how much latency exists on the input and output streams. The Audio family makes this information available to the Audio HAL so it can pass it along to its clients for synchronization purposes.
- Call `setSampleOffset` to make sure that the Audio HAL stays at least the specified number of samples away from the I/O engine's head. This setting is useful for block-transfer devices.
- Create the `IOAudioStream` objects to be used by the I/O engine and add them to the `IOAudioEngine`. See "Creating IOAudioStream Objects" (page 52) for details.
- Add a handler to your command gate for the interrupt fired by the I/O engine when it wraps to the beginning of the sample buffer. (This assumes a "traditional" interrupt.)
- Perform any necessary engine-specific initializations.

Listing 3-7 (page 50) illustrates how the SamplePCIAudioEngine class does some of these steps. Note that some initial values, such as `INITIAL_SAMPLE_RATE`, have been defined earlier using `#define` preprocessor commands.

Listing 3-7 Configuring the I/O engine

```
bool SamplePCIAudioEngine::initHardware(IOService *provider)
{
    bool result = false;
    IOAudioSampleRate initialSampleRate;
    IOAudioStream *audioStream;
    IOWorkLoop *workLoop;

    if (!super::initHardware(provider)) {
        goto Done;
    }
    initialSampleRate.whole = INITIAL_SAMPLE_RATE;
    initialSampleRate.fraction = 0;
    setSampleRate(&initialSampleRate);
    setDescription("Sample PCI Audio Engine");
    setNumSampleFramesPerBuffer(NUM_SAMPLE_FRAMES);

    workLoop = getWorkLoop();
    if (!workLoop) {
        goto Done;
    }

    interruptEventSource =
IOFilterInterruptEventSource::filterInterruptEventSource(this,
            OSMemberFunctionCast(IOInterruptEventAction, this,
                    &SamplePCIAudioEngine::interruptHandler),
            OSMemberFunctionCast(Filter, this,
                    &SamplePCIAudioEngine::interruptFilter),
            audioDevice->getProvider());
    if (!interruptEventSource) {
        goto Done;
    }
    workLoop->addEventSource(interruptEventSource);

    outputBuffer = (SInt16 *)IOMalloc(BUFFER_SIZE);
    if (!outputBuffer) {
        goto Done;
```

```
    }
    inputBuffer = (SInt16 *)IOMalloc(BUFFER_SIZE);
    if (!inputBuffer) {
        goto Done;
    }

    audioStream = createNewAudioStream(kIOAudioStreamDirectionOutput,
                outputBuffer, BUFFER_SIZE);
    if (!audioStream) {
        goto Done;
    }
    addAudioStream(audioStream);
    audioStream->release();

    audioStream = createNewAudioStream(kIOAudioStreamDirectionInput,
                inputBuffer, BUFFER_SIZE);
    if (!audioStream) {
        goto Done;
    }
    addAudioStream(audioStream);
    audioStream->release();
    result = true;
Done:
    return result;
}
```

The following section, "Creating IOAudioStream Objects" (page 52), describes the implementation of `createNewAudioStream`, which this method calls. A couple of other things in this method merit a bit more discussion.

First, in the middle of the method are a few lines of code that create a filter interrupt event source and add it to the work loop. Through this event source, an event handler specified by the driver will receive interrupts fired by the I/O engine. In the case of SamplePCIAudioEngine, the driver wants the interrupt at primary interrupt time instead of secondary interrupt time because of the better periodic accuracy. To do this, it creates an `IOFilterInterruptEventSource` object that makes a filtering call to the primary interrupt handler (`interruptFilter`); the usual purpose of this callback is to determine which secondary interrupt handler should be called, if any. The SamplePCIAudioEngine in the `interruptFilter` routine (as you'll see in "Taking a Timestamp" (page 54)) calls the method that actually takes the timestamp and always returns `false` to indicate that the secondary handler should not be called. For the driver to receive interrupts, the event source must be enabled. This is typically done when the I/O engine is started.

Second, this method allocates input and output sample buffers in preparation for the creation of `IOAudioStream` objects in the two calls to `createNewAudioStream`. The method of allocation in this example is rather rudimentary and would be more robust in a real driver. Also note that `BUFFER_SIZE` is defined earlier as:

```
NUM_SAMPLE_FRAMES * NUM_CHANNELS * BIT_DEPTH / 8
```

In other words, compute the byte size of your sample buffers by multiplying the number of sample frames in the buffer by the number of the channels in the audio stream; then multiply that amount by the bit depth and divide the resulting amount by 8 (bit size of one byte).

Creating IOAudioStream Objects

Your `IOAudioEngine` subclass should also create its `IOAudioStream` objects when it initializes the I/O engine (`initHardware`). You should have one `IOAudioStream` instance for each sample buffer serviced by the I/O engine. In the process of creating an object, make sure that you do the following things:

- Initialize it with the `IOAudioEngine` object that uses it (in this case, your `IOAudioEngine` subclass instance).
- Initialize the fields of a `IOAudioStreamFormat` structure with the values specific to a particular format.
- Call `setSampleBuffer` to pass the actual hardware sample buffer to the stream. If the sample buffer resides in main memory, it should be allocated before you make this call.

  The `SamplePCIAudioEngine` subclass allocates the sample buffers (input and output) in `initHardware` before it calls `createNewAudioStream`.
- Call `addAvailableFormat` for each format to which the stream can be set. As part of the `addAvailableFormat` call, specify the minimum and maximum sample rates for that format.
- Once you have added all supported formats to an `IOAudioStream`, call `setFormat` to specify the initial format for the hardware. Currently, `performFormatChange` is invoked as a result of the `setFormat` call.

Listing 3-8 (page 52) shows how the SamplePCIAudioEngine subclass creates and initializes an `IOAudioStream` object.

Listing 3-8 Creating and initializing an IOAudioStream object

```
IOAudioStream *SamplePCIAudioEngine::createNewAudioStream(IOAudioStreamDirection

                direction, void *sampleBuffer, UInt32 sampleBufferSize)
{
    IOAudioStream *audioStream;

    audioStream = new IOAudioStream;
    if (audioStream) {
        if (!audioStream->initWithAudioEngine(this, direction,  1)) {
            audioStream->release();
        } else {
            IOAudioSampleRate rate;
            IOAudioStreamFormat format = {
                2,      // number of channels
                kIOAudioStreamSampleFormatLinearPCM, // sample format
                kIOAudioStreamNumericRepresentationSignedInt,
                BIT_DEPTH,      // bit depth
                BIT_DEPTH,      // bit width
                kIOAudioStreamAlignmentHighByte,  // high byte aligned
                kIOAudioStreamByteOrderBigEndian, // big endian
                true,       // format is mixable
                0       // driver-defined tag - unused by this driver
            };
            audioStream->setSampleBuffer(sampleBuffer, sampleBufferSize);

            rate.fraction = 0;
            rate.whole = 44100;
```

```
            audioStream->addAvailableFormat(&format, &rate,  &rate);
            rate.whole = 48000;
            audioStream->addAvailableFormat(&format, &rate,  &rate);
            audioStream->setFormat(&format);
        }
    }

    return audioStream;
}
```

Starting and Stopping the I/O Engine

Your `IOAudioEngine` subclass must implement `performAudioEngineStart` and `performAudioEngineStop` to start and stop the I/O engine. When you start the engine, make sure it starts at the beginning of the sample buffer. Before starting the I/O engine, your implementation should do two things:

- Enable the interrupt event source to allow the I/O engine to fire interrupts as it wraps from the end to the beginning of the sample buffer; in its interrupt handler, the `IOAudioEngine` instance can continually take timestamps.
- Take an initial timestamp to mark the moment the audio engine started, but do so without incrementing the loop count.

By default, the method `takeTimeStamp` automatically increments the current loop count as it takes the current timestamp. But because you are starting a new run of the I/O engine and are not looping, you don't want the loop count to be incremented. To indicate that, pass `false` into `takeTimeStamp`.

Listing 3-9 (page 53) shows how the SamplePCIAudioEngine class implements the `performAudioEngineStart` method; the actual hardware-related code that starts the engine is not supplied.

Listing 3-9 Starting the I/O engine

```
IOReturn SamplePCIAudioEngine::performAudioEngineStart()
{
    IOLog("SamplePCIAudioEngine[%p]::performAudioEngineStart()\n",  this);


    assert(interruptEventSource);
    interruptEventSource->enable();

    takeTimeStamp(false);

    // Add audio - I/O start code here

#error performAudioEngineStart() - add engine-start code here; driver  will
                                not work without it

    return kIOReturnSuccess;
}
```

In `performAudioEngineStop`, be sure to disable the interrupt event source before you stop the I/O engine.

Taking a Timestamp

A major responsibility of your `IOAudioEngine` subclass is to take a timestamp each time the I/O engine loops from the end of the sample buffer to the beginning of the sample buffer. Typically, you program the hardware to throw the interrupt when this looping occurs. You must also set up an interrupt handler to receive and process the interrupt. In the interrupt handler, simply call `takeTimeStamp` with no parameters; this method does the following:

- It gets the current (machine) time and sets it as the loop timestamp in the `IOAudioEngineStatus`-defined area of memory shared with Audio clients.
- It increments the loop count in the same `IOAudioEngineStatus`-defined area of shared memory.

The Audio HAL requires both pieces of updated information so that it can track where the I/O engine currently is and predict where it will be in the immediate future.

The SamplePCIAudioEngine subclass uses an `IOFilterInterruptEventSource` object in its interrupt-handling mechanism. As "Hardware Initialization" (page 49) describes, when the subclass creates this object, it specifies both an interrupt-filter routine and an interrupt-handler routine. The interrupt-handler routine, however, is never called; instead, the interrupt-filter routine calls another routine directly (`filterInterrupt`), which calls `takeTimeStamp`. Listing 3-10 (page 54) shows this code.

Listing 3-10 The SamplePCIAudioEngine interrupt filter and handler

```
bool SamplePCIAudioEngine::interruptFilter(OSObject *owner,
                            IOFilterInterruptEventSource *source)
{
    SamplePCIAudioEngine *audioEngine = OSDynamicCast(SamplePCIAudioEngine,
            owner);

    if (audioEngine) {
        audioEngine->filterInterrupt(source->getIntIndex());
    }
    return false;
}

void SamplePCIAudioEngine::filterInterrupt(int index)
{

    takeTimeStamp();
}
```

Note that you can specify your own timestamp in place of the system's by calling `takeTimeStamp` with an `AbsoluteTime` parameter (see Technical Q&A QA1398 and the "Using Kernel Time Abstractions" section of *Kernel Programming Guide* for information on `AbsoluteTime`). This alternative typically isn't necessary but may be used in cases where the looping isn't detectable until some time after the actual loop time. In that case, the delay can be subtracted from the current time to indicate when the loop occurred in the past.

Warning: It is crucial to provide a reasonable timestamp, particularly with USB devices. The CoreAudio HAL does additional filtering of timestamps to compensate for deficiencies in USB. If your timestamps are not reasonably close to the expected timestamp (and in particular, if your timestamps are too far in the future), they will be ignored. This can result in various audio glitches and artifacts.

If you are experiencing pops and other glitches, try setting your transport type to something other than `kIOAudioDeviceTransportTypeUSB`. If this significantly improves the situation, you likely have something wrong with your timestamps. (Note, however, that lack of improvement does not necessarily vindicate your timestamps.)

If you are experiencing unexplained glitches in audio playback, the timestamps are the most likely cause. You should use the HALLab tool (available in the CoreAudio SDK, or preinstalled as part of Xcode) to help you determine what is causing the failure.

Providing a Playback Frame Position

An `IOAudioEngine` subclass must implement the `getCurrentSampleFrame` to return the playback hardware's current frame to the caller. This value (as you can see in Figure 2-5 (page 34)) tells the caller where playback is occurring relative to the start of the buffer.

Note: A common mistake is to report the number of frames played since the start of `IOAudioEngine` processing. The value the `getCurrentSampleFrame` method returns should be equal to that value modulo the number of frames per buffer.

The erase-head process uses this value; it erases (zeroes out) frames in the sample and mix buffers up to, but not including, the sample frame returned by this method. Thus, although the sample counter value returned doesn't have to be exact, it should never be larger than the actual sample counter. If it is larger, audio data may be erased by the erase head before the hardware has a chance to play it.

Implementing Format and Rate Changes

If an audio driver supports multiple audio formats or sample rates, it must implement the `performFormatChange` method to make these changes in the hardware when clients request them. The method has parameters for a new format and for a new sample rate; if either of these parameters is `NULL`, the `IOAudioEngine` subclass should change only the item that isn't `NULL`.

Although the SamplePCIAudioDriver driver deals with only one audio format, it is capable of two sample rates, 44.1 kilohertz and 48 kilohertz. Listing 3-11 (page 55) illustrates how `performFormatChange` is implemented to change a sample rate upon request.

Listing 3-11 Changing the sample rate

```
IOReturn SamplePCIAudioEngine::performFormatChange(IOAudioStream
        *audioStream, const IOAudioStreamFormat *newFormat,
        const IOAudioSampleRate *newSampleRate)
{
    IOLog("SamplePCIAudioEngine[%p]::peformFormatChange(%p,  %p, %p)\n", this,

        audioStream, newFormat, newSampleRate);
```

```
    if (newSampleRate) {
        switch (newSampleRate->whole) {
            case 44100:
                IOLog("/t-> 44.1kHz selected\n");

                // Add code to switch hardware to 44.1khz
                break;
            case 48000:
                IOLog("/t-> 48kHz selected\n");

                // Add code to switch hardware to 48kHz
                break;
            default:
                IOLog("/t Internal Error - unknown sample rate  selected.\n");
                break;
        }
    }
    return kIOReturnSuccess;
}
```

Clipping and Converting Samples

Arguably, the most important work that an audio device driver does is converting audio samples between the format expected by the hardware and the format expected by the clients of the hardware. In Mac OS X, the default format of audio data in the kernel as well as in the Audio HAL and all of its clients is 32-bit floating point. However, audio hardware typically requires audio data to be in an integer format.

To perform these conversions, your audio driver must implement at least one of two methods, depending on the directions of the audio streams supported:

- Implement `clipOutputSamples` if your driver has an output `IOAudioStream` object.
- Implement `convertInputSamples` if your driver has an input `IOAudioStream` object.

In addition to performing clipping and conversion, these methods are also a good place to add device-specific input and output filtering code. For example, a particular model of USB speakers might sound better with a slight high frequency roll-off. (Note that if this is the only reason for writing a driver, you should generally use an `AppleUSBAudio` plug-in instead, as described in the *SampleUSBAudioPlugin* example code.)

Because these methods execute floating-point code, you cannot include them in the same source file as the other `IOAudioEngine` methods you implement. The compiler, by default, enables floating-point emulation to prevent floating-point instructions from being generated. To get around this, create a separate library that contains the floating-point code and compile and link this library into the resulting kernel module. The separate library for the SamplePCIAudioDriver project is `libAudioFloatLib`.

A common mistake that people make when developing an audio driver is either failing to write these methods or failing to include this additional library when linking the KEXT. When this occurs, you will execute the `clipOutputSamples` and `convertInputSamples` methods that are built into the base

class. These methods are just stubs that return `kIOReturnUnsupported` (`0xe00002c7`, or `-536870201`). If you see this error returned by one of these methods, you should make sure you are linking your KEXT together correctly.

Important: See the project configuration for SamplePCIAudioDriver (or any other example audio-driver project) to find out how to generate this separate static library and include it in your project. Pay particular attention to the required compiler options.

The `clipOutputSamples` method is passed six parameters:

- A pointer to the start of the source (mix) buffer
- A pointer to the start of the destination (sample) buffer
- The index of the first sample frame in the buffers to clip and convert
- The number of sample frames to clip and convert
- A pointer to the current format (structure `IOAudioStreamFormat`) of the audio stream
- A pointer to the `IOAudioStream` object this method is working on

Your implementation must first clip any floating-point samples in the mix buffer that fall outside the range –1.0 to 1.0 and then convert the floating-point value to the comparable value in the format expected by the hardware. Then copy that value to the corresponding positions in the sample buffer. Listing 3-12 (page 57) illustrates how the SamplePCIAudioDriver implements the `clipOutputSamples` method.

Listing 3-12 Clipping and converting output samples

```
IOReturn SamplePCIAudioEngine::clipOutputSamples(const void *mixBuf,
        void *sampleBuf, UInt32 firstSampleFrame, UInt32 numSampleFrames,
        const IOAudioStreamFormat *streamFormat, IOAudioStream *audioStream)
{
    UInt32 sampleIndex, maxSampleIndex;
    float *floatMixBuf;
    SInt16 *outputBuf;

    floatMixBuf = (float *)mixBuf;
    outputBuf = (SInt16 *)sampleBuf;

    maxSampleIndex = (firstSampleFrame + numSampleFrames) *
                        streamFormat->fNumChannels;

    for (sampleIndex = (firstSampleFrame * streamFormat->fNumChannels);
                        sampleIndex < maxSampleIndex; sampleIndex++)  {
        float inSample;
        inSample = floatMixBuf[sampleIndex];

        // Note: A softer clipping operation could be done here
        if (inSample > 1.0) {
            inSample = 1.0;
        } else if (inSample < -1.0) {
            inSample = -1.0;
        }
        if (inSample >= 0) {
            outputBuf[sampleIndex] = (SInt16) (inSample * 32767.0);
```

```
        } else {
            outputBuf[sampleIndex] = (SInt16) (inSample * 32768.0);
        }
    }
    return kIOReturnSuccess;
}
```

Here are a few comments on this specific example:

1. It starts by casting the `void *` buffers to `float *` for the mix buffer and `SInt16 *` for the sample buffer; in this project, the hardware uses signed 16-bit integers for its samples while the mix buffer is always `float *`.
2. Next, it calculates the upper limit on the sample index for the upcoming clipping and converting loop.
3. The method loops through the mix and sample buffers and performs the clip and conversion operations on one sample at a time.
    a. It fetches the floating-point sample from the mix buffer and clips it (if necessary) to a range between -1.0 and 1.0.
    b. It scales and converts the floating-point value to the appropriate signed 16-bit integer sample and writes it to the corresponding location in the sample buffer.

The parameters passed into the `convertInputSamples` method are *almost* the same as those for the `clipOutputSamples` method. The only difference is that, instead of a pointer to the mix buffer, a pointer to a floating-point destination buffer is passed; this is the buffer that the Audio HAL uses. In your driver's implementation of this method, do the opposite of the `clipOutputSamples`: convert from the hardware format to the system 32-bit floating point format. No clipping is necessary because your conversion process can control the bounds of the floating-point values.

Note: The `convertInputSamples` method should begin writing at the *beginning* of the destination buffer, unlike `clipOuputSamples`, which writes at an offset based on the index passed in.

Listing 3-13 (page 58) shows how the SamplePCIAudioDriver project implements this method.

Listing 3-13 Converting input samples.

```
IOReturn SamplePCIAudioEngine::convertInputSamples(const void *sampleBuf,
        void *destBuf, UInt32 firstSampleFrame, UInt32 numSampleFrames,
        const IOAudioStreamFormat *streamFormat, IOAudioStream
        *audioStream)
{
    UInt32 numSamplesLeft;
    float *floatDestBuf;
    SInt16 *inputBuf;

    // Note: Source is offset by firstSampleFrame
    inputBuf = &(((SInt16 *)sampleBuf)[firstSampleFrame *
                            streamFormat->fNumChannels]);

    // Note: Destination is not.
    floatDestBuf = (float *)destBuf;
```

```
    numSamplesLeft = numSampleFrames * streamFormat->fNumChannels;

    while (numSamplesLeft > 0) {
        SInt16 inputSample;
        inputSample = *inputBuf;

        if (inputSample >= 0) {
            *floatDestBuf = inputSample / 32767.0;
        } else {
            *floatDestBuf = inputSample / 32768.0;
        }

        ++inputBuf;
        ++floatDestBuf;
        --numSamplesLeft;
    }

    return kIOReturnSuccess;
}
```

The code in Listing 3-13 does the following things:

1. It starts by casting the destination buffer to a `float *`.
2. It casts the sample buffer to a signed 16-bit integer and determines the starting point within this input buffer for conversion.
3. It calculates the number of actual samples to convert.
4. It loops through the samples, scaling each to within a range of –1.0 to 1.0 (thus converting it to a float) and storing it in the destination buffer at the proper location.

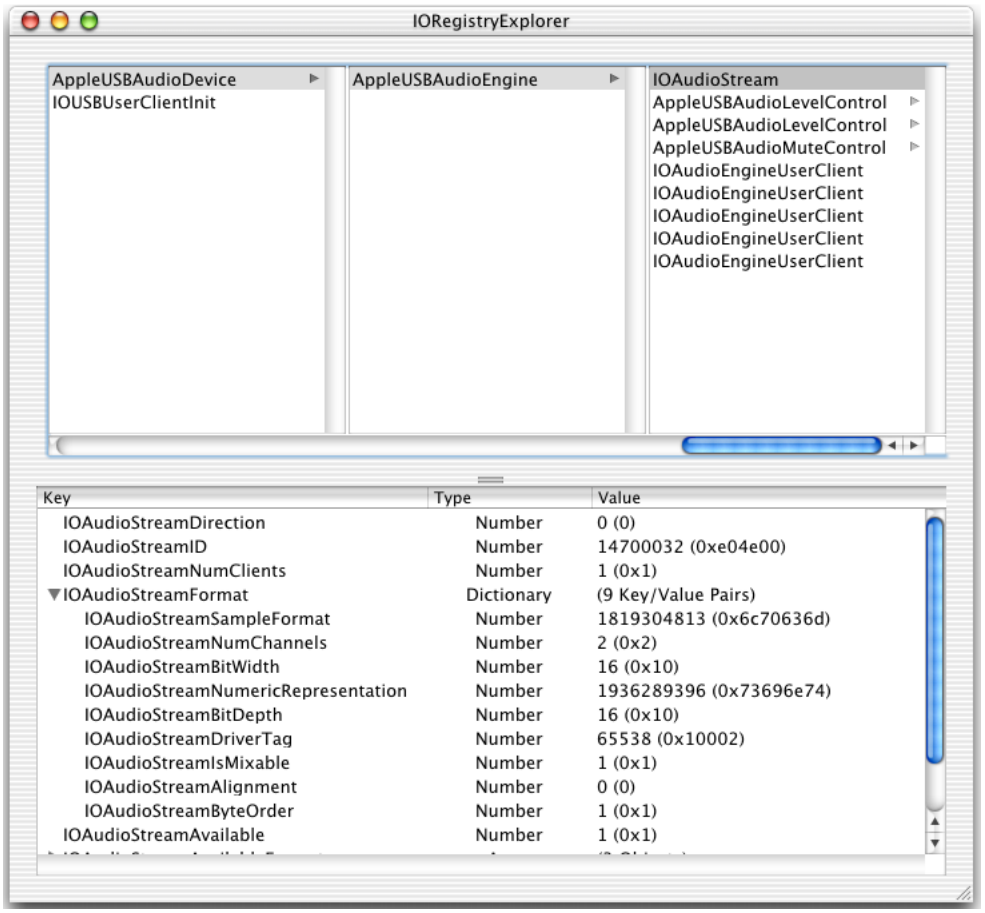
Debugging and Testing the Driver

Many of the techniques you would use in debugging and testing an audio driver are the same ones you'd use with other types of device drivers. After all, any I/O Kit driver has a structure and a behavior that are similar to any other I/O Kit driver, regardless of family.

For example, it's always a good idea when a driver is under development to make `IOLog` calls at critical points in your code, such as before and after an I/O transfer. The `IOLog` function writes a message to the console (accessible through the Console application) and to `/var/log/system.log`. You can format the message string with variable data in the style of `printf`.

Similarly, you can examine the I/O Registry with the I/O Registry Explorer application or the `ioreg` command-line utility. The I/O Registry will show the position of your driver's objects in the driver stack, the client-provider relationships among them, and the attributes of those driver objects. In Figure 3-2 (page 60), the I/O Registry Explorer shows part of the objects and their attributes in a USB audio device driver.

Figure 3-2 The I/O Registry (via I/O Registry Explorer)



However, as "Custom Debugging Information in the I/O Registry" (page 63) explains, your driver can insert information in the I/O Registry to assist the testing and debugging of your driver.

Tools for Testing Audio Drivers

The Mac OS X Developer package provides two applications that are helpful when you're testing audio driver software. These items are not shipped as executables, but are instead included as example-code projects installed in `/Developer/Examples/CoreAudio/HAL`. The two projects that are of interest are HALLab and MillionMonkeys. To obtain the executables, copy the project folders to your home directory (or any file-system location where you have write access) and build the projects.

The HALLab application helps you verify the controls and other attributes of a loaded audio driver. With it, you can play back sound files to any channel of a device, check whether muting and volume changes work for every channel, test input operation, enable soft play through, view the device object hierarchy, and do various other tests.

Figure 3-3 (page 61) and Figure 3-4 (page 61) show you what two of the HALLab windows look like.

Figure 3-3 The HALLab System window

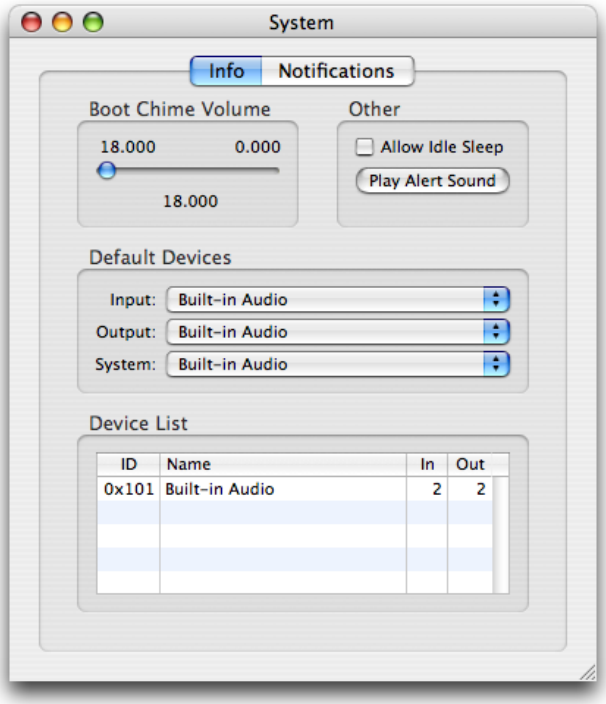
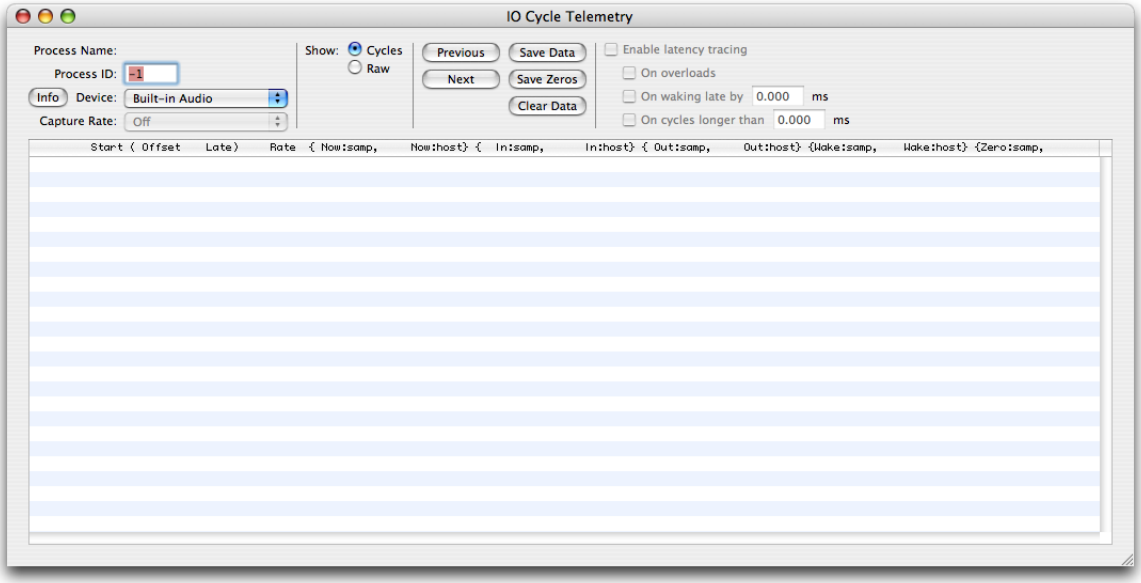


Figure 3-4 The HALLab IO Cycle Telemetry window



The MillionMonkeys application was designed for performance profiling of your driver. In particular, it allows you to determine latency at various steps of audio processing while the system is under load. This can aid in tracking down performance-related issues with audio drivers. Figure 3-5 (page 62) and Figure 3-6 (page 63) show the two panes of the MillionMonkeys application window.

Figure 3-5 The MillionMonkeys Device & Workload pane

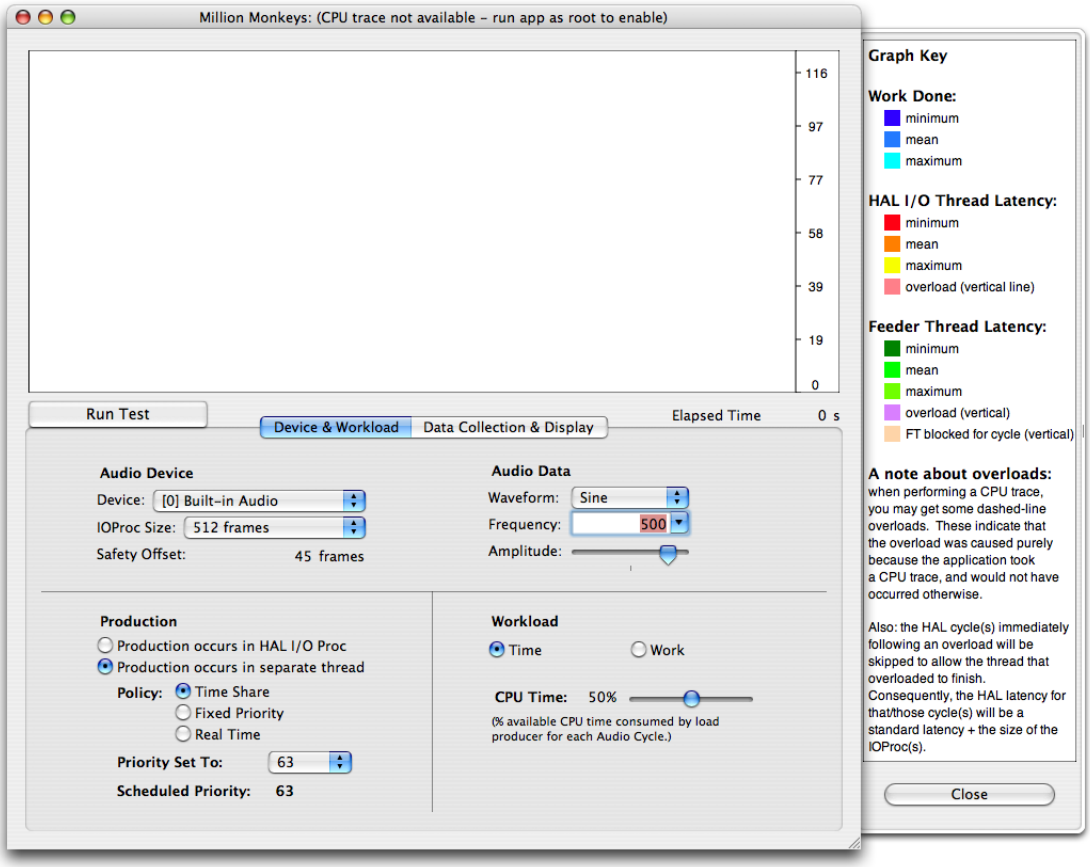
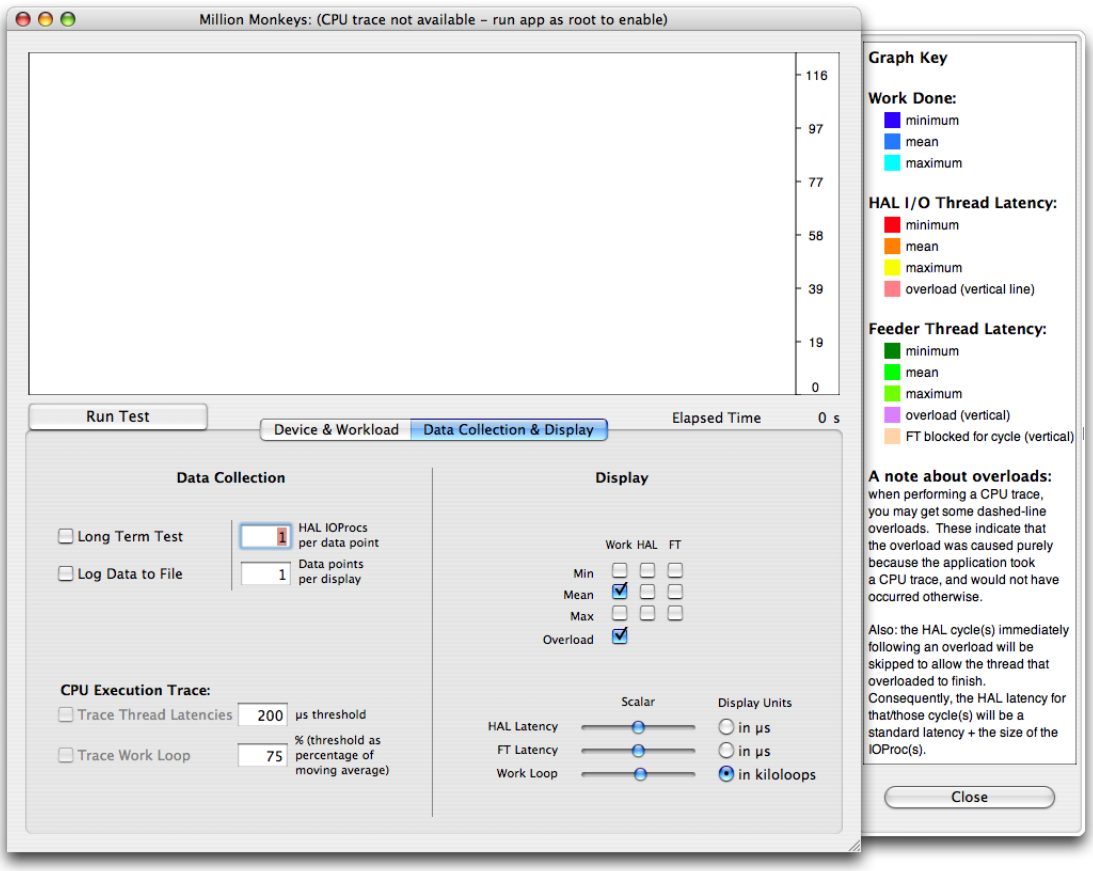


Figure 3-6 The MillionMonkeys Data Collection & Display pane



Custom Debugging Information in the I/O Registry

Another way you can test and debug your audio device driver is to write custom properties to the I/O Registry. For example, you may want to track hardware register state or internal driver state (if the driver has any). Whenever your driver makes a change to the hardware state, it could read the hardware register values and call `setProperty` with the current value. Then, when testing the driver, run the I/O Registry Explorer application and note what the I/O Registry shows this value to be. This technique allows you to easily determine if the driver is putting the hardware in the correct state.

Tips, Tricks, and Frequently Asked Questions

This chapter contains various tips on general concepts, sample buffers, and custom controls.

General Issues

What is the effect of aggregate devices from a driver programming perspective?

Aggregate devices cause multiple devices to behave as a single device. In the process, Core Audio does some extra work to smooth out timing inconsistencies.

The process should be transparent to driver writers, provided that your timestamps are reasonably correct.

Should I create a "whole device" stream containing all outputs from my device, or just a stream for each pair of inputs/outputs?

That's entirely up to you. Aggregate devices make this largely a non-issue. However, it may be convenient to provide a "whole device" stream to better support audio applications in versions of Mac OS X prior to version 10.4.

How do drivers interact with Audio/MIDI Setup?

Audio/MIDI Setup presents the standard controls for an audio device, along with stream selection capabilities. There's no magic here. However, this question often comes up in conjunction with the issue of custom controls. In that case, some additional work is needed. This process is described further in "Creating Custom Controls" (page 67).

Sample Buffer Issues

What is the minimum (practical) size of a sample buffer, and what happens if a driver's buffer is too small?

The size of a sample buffer is limited by a number of factors. For one, the sample offset (*not* sample latency) must be taken into account. If the audio engine is set to read 1000 samples behind the hardware (for example), there had better be room for more than 1000 samples in

the buffer. In fact, there should be at least two additional frames—the one in which the hardware is writing and the frame being erased ahead of it.

If your buffer is hopelessly too small, a good indicator is a continuous stream of errors indicating that the data has already been clipped. If the buffer is only slightly too small, you will merely experience a large number of glitches as the audio engine fails to keep up with the hardware.

What is the difference between sample latency and sample offset?

Sample latency refers to the amount of time the audio hardware requires to reproduce a sound. This includes all delays in the input or output chain. For example, a device might take a few milliseconds between when it posts an interrupt indicating it read the start of the buffer and when the sound is actually played.

Sample offset is a feature designed for audio devices based on block I/O. Consider an output device as an example. If the audio device transfers data in a 32-sample block transaction, it must have at least 32 samples available when the audio engine wakes up. Otherwise, the engine won't be able to queue up a block transfer, and will end up slipping a cycle, potentially resulting in a glitch. To solve this problem, you can specify a sample offset to guarantee that the higher levels stay a certain distance ahead of the I/O head.

I'm having significant performance problems when doing custom input/output filtering in my driver. How can I improve performance?

A common cause of poor performance is using a separate thread for such audio filters. You can get a significant performance gain by doing this processing in your clipping or conversion routines instead.

Another possible performance problem is forgetting to turn off floating point emulation. Software floating point is significantly slower than hardware floating point and should generally be avoided in the critical path for audio data.

I'm not doing any custom filtering, but I'm still having performance problems (dropouts, stuttering, and so on). Any ideas?

The most common cause of audio glitches is bad timestamping. See "Taking a Timestamp" (page 54) for detailed suggestions. If you are using block devices or other devices where the timestamp can't be taken precisely when the buffer wraps around, you may also find the code example in "Faking Timestamps" (page 66) helpful.

Faking Timestamps

One common problem that many audio device driver writers face is working around a transport layer that does not provide a timestamp when each audio packet is sent. If you take a timestamp based on receiving a packet that is larger than the remaining space in the buffer (where wrapping occurs mid-packet), your timestamp will not be particularly accurate.

The following code snippet shows a simple example of how to work around this problem:

```
void set_timestamp_adjusted(int current_bufpos)
{
    static int sec=0, usec=0, lastsec, lastusec=0, lastpos=0;
    int len, stampsec, stampusec;
    uint64_t curtm, lasttm, stampoff, stamptm
```

```
    clock_get_system_microtime(&sec, &usec);
    if (!lastsec && !lastusec) {
        // Engine just started. Initialize values.
        lastsec = sec;
        lastusec = sec;
    }

    curtm = (sec * 1000000UL) + usec; // usec since startup.
    lasttm = (lastsec * 1000000UL) + lastusec;
    stampoff = ((lasttm - curtm) * (uint64_t)(BUFFER_SIZE - lastpos))  /
            (uint64_t)len;
    stamptm = lasttm + stampoff;

    stampsec = (int)(stamptm / 1000000ULL);
    stampusec = (int)(stamptm % 1000000ULL);
    lastpos = current_bufpos;

    // set timestamp here.
}
```

Note that, if at all possible, you should attempt to take a time stamp (ideally at primary interrupt time for maximum accuracy) when the device wraps around to the start of the buffer. If it is possible to obtain a stamp precisely when the device wraps around, these sorts of calculations should not be necessary.

Creating Custom Controls

For most common purposes, the standard audio controls are sufficient. However, in some cases, you may need to create a custom control type.

Note: Apple reserves all-lowercase control types for its own use. If you create a custom control type, you must use at least one capital letter in its type.

The first step in creating a custom audio control is to subclass either the `IOAudioControl` or `IOAudioLevelControl` class. In general, most typical controls express a continuous floating-point value across a particular range. For those controls, subclassing `IOAudioLevelControl` is more appropriate. The more general `IOAudioControl` class is more appropriate for creating toggles and other controls that express noncontinuous values.

The second step is to write a `setValue` method. This method must interpret what those values mean and set appropriate instance variables accordingly, performing any range conversion calculations as needed.

The final step is to implement an application for managing these controls. Nonstandard controls can be manipulated using the same mechanisms as any other controls, but most applications won't do anything with them because they don't know to look for them (or what to do with them when they find them).

Document Revision History

This table describes the changes to *Audio Device Driver Programming Guide*.

Date	Notes
2006-01-10	Corrected a typographical error in code sample.
2005-11-09	Major content revision; changed title from "Writing Audio Device Drivers."
2004-03-25	Added installed location of SamplePCIAudioDriver project.
2001-12-15	First version.

Document Revision History