UNIVERSAL SERIAL BUS DEVICE CLASS DEFINITION FOR AUDIO DEVICES

Release 3.0

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SCOPE OF THIS RELEASE

This document is the Release 3.0 of this Device Class Definition.

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1 INTRODUCTION

1.1 SCOPE

The Audio Device Class Definition applies to all devices or functions embedded in composite devices that are used to manipulate audio, voice, and sound-related functionality. This includes both audio data (analog and digital) and the functionality that is used to directly control the audio environment, such as Volume and Tone Control. The Audio Device Class does not include functionality to operate transport mechanisms that are related to the reproduction of audio data, such as tape transport mechanisms or CD-ROM drive control.

1.2 PURPOSE

The purpose of this document is to describe the minimum capabilities and characteristics an audio device shall support to comply with the USB. This document also provides recommendations for optional features.

1.3 RELATED DOCUMENTS

- *Universal Serial Bus Specification*, Revision 2.0 (referred to in this document as the *USB Specification*). In particular, see Chapter 5, “USB Data Flow Model” and Chapter 9, “USB Device Framework.”
- *Universal Serial Bus 3.1 Specification*, Revision 1.0 (referred to in this document as the *USB 3.1 Specification*). This document covers details specific to SuperSpeed and SuperSpeed+ devices.
- *Universal Serial Bus Device Class Definition for Audio Data Formats* (referred to in this document as USB Audio Data Formats).
- *Universal Serial Bus Device Class Definition for Terminal Types* (referred to in this document as USB Audio Terminal Types).
- ANSI S1.11-1986 standard.
- MPEG-1 standard ISO/IEC 111172-3 1993.
- MPEG-2 standard ISO/IEC 13818-3 Feb. 20, 1997.
- Digital Audio Compression Standard (AC-3), ATSC A/52A Aug. 20, 2001. (available from http://www.atsc.org/)
- ANSI/IEEE-754 floating-point standard.
- ISO/IEC 60958 International Standard: *Digital Audio Interface and Annexes.*
- ISO/IEC 61937 standard.

1.4 TERMS AND ABBREVIATIONS

This section defines terms used throughout this document. For additional terms that pertain to the Universal Serial Bus, see Chapter 2, “Terms and Abbreviations,” in the *USB Specification*.

ASRC	Acronym for Asynchronous Sampling Rate Conversion.
(Audio Channel) Cluster	Group of logical audio channels that carry tightly related synchronous audio information. A stereo audio stream is a typical example of a two-channel Cluster.
Audio Control Attribute	Parameter of an Audio Control. Examples are Current, Minimum, Maximum and Resolution attributes of a Volume Control.
Audio Control	Logical object that is used to manipulate a specific audio property. Examples are Volume Control, Mute Control, etc.

Audio data stream	Transport medium that can carry audio information.
AEC	Acronym for Audio Echo Cancellation.
Audio Function	Independent part of a USB device that deals with audio-related functionality.
Audio Interface Association (AIA)	Grouping of a single AudioControl interface, zero or more AudioStreaming interfaces and zero or more MIDIStreaming interfaces that together constitute a complete interface to a particular version (compliance level) of the Audio Function.
AudioControl interface (ACI)	USB interface used to access the Audio Controls inside an Audio Function.
AudioStreaming interface (ASI)	USB interface used to transport audio streams into or out of the Audio Function.
Clock Multiplier (CM)	Multiplies a clock signal by a fractional multiplier.
Clock Selector (CX)	Selects from a number of input clock signals.
Clock Source (CS)	Generates an audio-related clock signal (master clock or sample clock).
CMD	Acronym for Clock Multiplier descriptor.
CSD	Acronym for Clock Source descriptor.
CXD	Acronym for Clock Selector descriptor.
Effect Unit (EU)	Provides advanced audio manipulation on the incoming logical audio channels.
Entity	Addressable logical object inside an Audio Function.
Extension Unit (XU)	Applies an undefined process to a number of logical input channels.
Feature Unit (FU)	Provides basic audio manipulation on the incoming logical audio channels.
FUD	Acronym for Feature Unit descriptor.
High Capability Descriptor	A new representation for class-specific descriptors that allows for potentially large (>256 bytes) and dynamic descriptors.
Input Pin	Logical input connection to an Entity. Carries a single Cluster.
Input Terminal (IT)	Receptacle for audio information flowing into the Audio Function.
ITD	Acronym for Input Terminal descriptor.
Logical Audio Channel	Logical transport medium for a single audio channel. Makes abstraction of the physical properties and formats of the connection. Is usually identified by spatial location. Examples are Left channel, Right Surround channel, etc.

MIDIStreaming interface (MSI)	USB interface that may be used to transport MIDI data streams into or out of the Audio Function.
Mixer Unit (MU)	Mixes a number of logical input channels into a number of logical output channels.
MUD	Acronym for Mixer Unit descriptor.
OTD	Acronym for Output Terminal descriptor.
Output Pin	Logical output connection to an Entity. Carries a single Cluster.
Output Terminal (OT)	An outlet for audio information flowing out of the Audio Function.
Phase Locked Loop	A hardware or software control system that generates an output signal whose phase is related to the phase of an input signal. Typically used in clock recovery applications.
Processing Unit (PU)	Applies a predefined process to a number of logical input channels.
PUD	Acronym for Processing Unit descriptor.
Reserved	Reserved is a keyword indicating reserved bits, bytes, words, fields, and code values that are set-aside for future standardization. Their use and interpretation may be specified by future extensions to this specification and, unless otherwise stated, shall not be utilized or adapted by vendor implementation. A reserved bit, byte, word, or field shall be set to zero by the sender and shall be ignored by the receiver.
RUD	Acronym for Sampling Rate Converter Unit descriptor.
Sampling Rate Converter Unit (RU)	Converts the incoming audio data stream, running at a sampling rate, synchronous to a first master clock, into an outgoing audio data stream that is running at a sampling rate, synchronous to a second master clock, which is free running with respect to the first master clock.
Selector Unit (SU)	Selects from a number of input Clusters.
SUD	Acronym for Selector Unit descriptor.
Terminal	Addressable logical object inside an Audio Function that represents a connection to the Audio Function’s outside world.
Unit	Addressable logical object inside an Audio Function that represents a certain audio sub-functionality.
XUD	Acronym for Extension Unit descriptor.

2 MANAGEMENT OVERVIEW

The USB is very well suited for transport of audio, ranging from low fidelity voice connections to high quality, multi-channel audio streams. The USB has become a ubiquitous connector on modern PC's and is well-understood by most consumers today. As such, it has become the connector of choice for many peripherals and is indeed the simplest and most pervasive digital audio connector available today. Consumers can count on this medium to meet all of their current and future audio needs. Many applications from communications, to entertainment, to music recording and playback, can take advantage of the audio features of the USB.

In principle, a versatile bus specification like the USB provides many ways to propagate and/or control digital audio. For the industry, however, it is very important that audio transport mechanisms be well defined and standardized on the USB. Only in this way can interoperability be guaranteed among the many possible audio devices on the USB. Standardized audio transport mechanisms also help to keep software drivers as generic as possible. The Audio Device Class described in this document satisfies those requirements. It is written and revised by experts in the audio field. Other device classes that address audio in some way should refer to this document for their audio interface specification.

An essential issue in audio is synchronization of the data streams. Indeed, the smallest artifacts are easily detected by the human ear. Therefore, a robust synchronization scheme on isochronous transfers has been developed and incorporated in the *USB Specification* and the *USB 3.1 Specification*. The Audio Device Class definition adheres to these synchronization schemes to transport audio data reliably over the bus.

This document contains all necessary information for a designer to build a USB-compliant device that incorporates Audio Functionality. It specifies the standard and class-specific descriptors that shall be present in each USB Audio Function. It further explains the use of class-specific requests that allow for full Audio Function control. A number of predefined data formats are listed and fully documented. Each format defines a standard way of transporting audio over the USB. Provisions have been made so that vendor-specific audio formats and compression schemes can be handled as well.

Many of the changes introduced in this version of the USB Specification for Audio Devices are inspired by the desire to use USB Audio in modern portable devices. Special attention has been paid to make the Audio Device Class more power-friendly by providing new tools to selectively enable and disable parts of the Audio Function and also by supporting burst mode data transfers for longer sleep times in between data transfers. In addition, the specification supports new codec types and data formats for consumer audio applications, provides numerous clarifications of the original specification and extensions to support various changes in the core specification.

2.1 OVERVIEW OF KEY DIFFERENCES BETWEEN ADC V2.0 AND V3.0

The following list is not an exhaustive list of all changes that have been introduced. For complete information, refer to the full specification.

- Dolby Processing Unit removed
- Encoder and Decoder support removed
- Copy protection (S/PDIF-style) removed
- bmAttributes (MaxPacketsOnly bit) field in class-specific isochronous endpoint descriptor removed
- Type II and Extended Type II Audio Data Formats removed
- New Power Domains
- New Multi-Function Processing Unit
- Additional Type III formats introduced

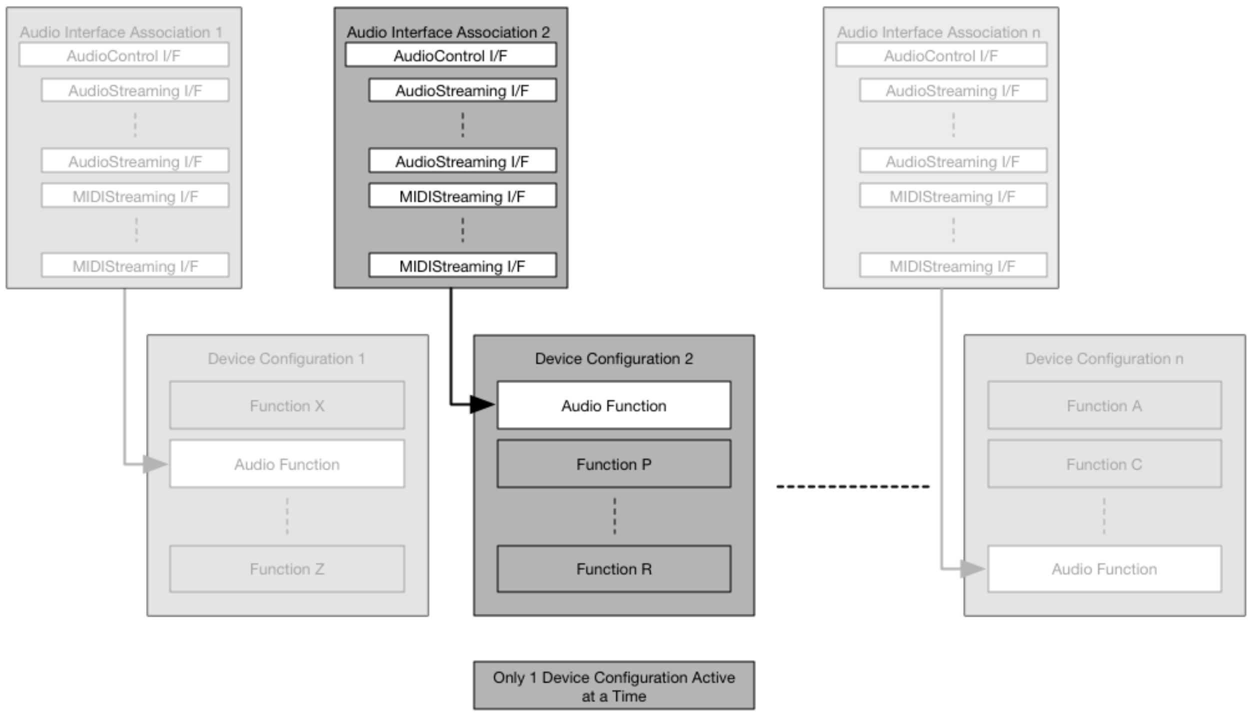
- New Cluster descriptor
- New Extended Terminal descriptor
- New Connectors descriptor
- Layout of all class-specific descriptors has changed
- New class-specific String descriptors
- New High Capability descriptors
- Additional sources for interrupts
- Provides for backward compatibility, using multiple Configuration descriptors
- Support for LPM/L1
- Mandated support for BADD on each ADC3.0 compliant device

3 FUNCTIONAL CHARACTERISTICS

3.1 INTRODUCTION

In many cases, Audio Functionality does not exist as a standalone device. It is one capability that, together with other functions, constitutes a "composite" device. A perfect example of this is a DVD-ROM player, which can incorporate video, audio, data storage, and transport control. The Audio Function is thus located at the interface level in the device class hierarchy. The following figure provides details.

Figure 3-1: Multiple Configurations and their Audio Interface Associations and Interfaces



An Audio Function is considered to be a 'closed box' that has very distinct and well defined interfaces to the outside world. Audio Functions are addressed and accessed through an Audio Interface Association (AIA). The AIA groups all USB interfaces that together provide access to the Audio Function for control and streaming purposes.

A USB Device may express different AIAs representing the same underlying hardware by using multiple USB Device Configuration descriptors. These AIAs may be used to provide access to the Audio Function at different compliance levels. For example, one AIA might provide access to the Audio Function in ADC 1.0 mode, whereas another AIA might provide access to the Audio Function in ADC 3.0 mode, as defined by this document. Note that the Audio Function may take on a different topology and even different functionality when accessed through different AIAs.

Because only one Configuration may be active at a time, only one AIA shall be active at a time for the same Audio Function. Host software should choose the most desirable Configuration and use the Standard SET_CONFIGURATION command to select the Configuration containing the most desired AIA or AIAs (if there are multiple independent Audio Functions residing in the same USB Device).

An AIA shall have a single AudioControl interface and can have zero or more AudioStreaming and zero or more MIDIStreaming interfaces. The AudioControl (AC) interface is used to access the Audio Controls of the function whereas the AudioStreaming (AS) interfaces are used to transport audio streams into and out of the Audio

Function. MIDIStreaming (MS) interfaces can be used to transport MIDI data streams into and out of the Audio Function.

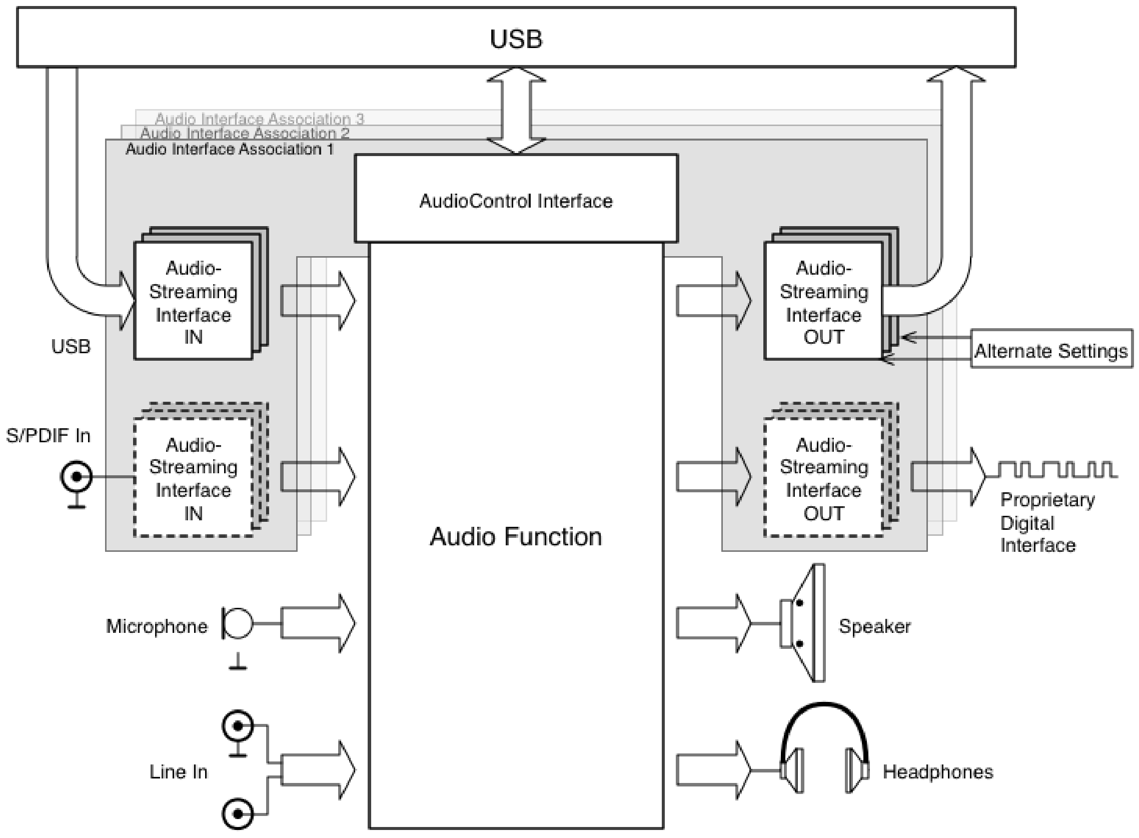
Note: All MIDI-related information is grouped in a separate document, *Universal Serial Bus Device Class Definition for MIDI Devices* that is considered part of this specification. The remainder of this document will therefore not mention MIDIStreaming interfaces and their specifics anymore.

Note that AudioStreaming interfaces do not necessarily all represent USB audio streams. Any audio stream, regardless of its origin or destination that enters or leaves the Audio Function may have an associated AudioStreaming interface, if that interface requires the Host to have access and/or visibility to some interface Controls or descriptors that may have an impact on the operation of the interface from a USB perspective.

A device can have *multiple independent* Audio Functions located in the same composite Device. They are each accessed through their active Audio Interface Association, which must be contained within the currently selected USB Configuration.

The following figure illustrates the concept of an Audio Function and its associated interfaces:

Figure 3-2: Audio Function Global View


USB
Audio Interface Association 3
Audio Interface Association 2
Audio Interface Association 1
AudioControl Interface
Audio-Streaming Interface IN
USB
Audio-Streaming Interface OUT
Alternate Settings
S/PDIF In
Audio-Streaming Interface IN
Audio-Streaming Interface OUT
Proprietary Digital Interface
Audio Function
Microphone
Speaker
Line In
Headphones

All functionality pertaining to controlling parameters that directly influence audio perception (like volume) are located inside the central rectangle and are exclusively controlled through the AudioControl interface. Streaming aspects of the communication to or from the Audio Function are handled through separate AudioStreaming interfaces, as necessary. Each USB audio stream shall be represented by an AudioStreaming interface. A non-USB audio stream may be represented by an AudioStreaming interface if there is a need to convey information between that audio stream's interface and the Host. All control data that is related specifically to the streaming behavior of the interface is conveyed through the AudioStreaming interface.

A physical S/PDIF connection to the Audio Function is a typical example of a non-USB AudioStreaming interface. Although the actual audio data is coming in from the outside world (not through the USB), it might be necessary to control some aspects of the S/PDIF connection. In that case, the S/PDIF connection is represented by an AudioStreaming interface so that it becomes addressable through USB.

Also note that the connection between the AudioStreaming interfaces and the Audio Function is not 'solid'. The reason for this is that when seen from the inside of the Audio Function, each audio stream entering or leaving the Audio Function is represented by a special object, called a Terminal (see further). The Terminal concept abstracts the actual AudioStreaming interface inside the Audio Function and provides a logical view on the connection rather than a physical view. This abstraction allows audio channels within the Audio Function to be treated as 'logical' audio channels that do not have physical characteristics associated with them anymore (analog vs. digital, format, sampling rate, bit resolution, etc.).

3.2 BASIC AUDIO DEVICE DEFINITION

The Basic Audio Device Definition (BADD) exists as a separate document and is based on and fully compatible with this Audio Device Class Definition (ADC 3.0). As such, the BADD defines a subset of functionality commonly found in Headphones, Microphones, and Headsets using the features and tools provided by the ADC but it removes all of the optionality allowed by the ADC and prescribes rigorously how certain features shall be implemented. This allows simple Host drivers to know exactly what to expect when encountering a BADD-compliant device, reducing significantly the work a driver needs to perform parsing the BADD Device's descriptors and interacting with the Device.

3.3 BACKWARDS COMPATIBILITY

This 3.0 version of the *Audio Device Class Definition* (ADC 3.0) as such is not backwards compatible with any of the previous versions of the same specification.

However, to ensure the broadest possible interoperability among new ADC 3.0 Devices and legacy ADC 1.0 or ADC 2.0 Hosts, this specification requires that the ADC 3.0 compliant Device shall do the following:

- The first USB Configuration descriptor (index 0) exposed by the Device shall contain an Audio Interface Association (AIA), which is compliant with ADC 1.0 or ADC 2.0. In other words, if Host software selects the first indexed Device Configuration, it shall expose an Audio Function that is compliant with ADC 1.0 or ADC 2.0. This requirement should allow the Audio Function to interoperate with existing Hosts exactly the same as current legacy ADC 1.0 or ADC 2.0 Functions.
- If the Audio Function contains at least one USB Audio Streaming Interface, the Device shall have another Configuration descriptor that exposes an AIA which is BADD 3.0 compliant. A Device shall only have a single AIA of this type. However, this same BADD compliant AIA may need to be duplicated in more than one Configuration descriptor. This requirement guarantees that the Device will interoperate with new ADC 3.0 Hosts that support the BADD 3.0 specification. To activate this mode of operation for the Audio Function, the Host needs to select a Device Configuration which contains the BADD 3.0 compliant AIA.
- The Device may have one or more Configurations that expose one or more AIAs which shall be compliant with ADC 3.0. This way the Audio Function may provide functionality beyond what is available in BADD 3.0 operating mode. To activate this mode of operation, the Host needs to explicitly select a Configuration that contains the desired AIA(s).

It is strongly recommended that *ALL* ADC 3.0 Hosts support the BADD 3.0 specification.

3.4 AUDIO INTERFACE ASSOCIATION (AIA) AND INTERFACE ASSOCIATION DESCRIPTOR

On the USB, an Audio Function is completely defined by its interfaces. An Audio Function has one or more Audio Interface Associations (AIA) that provide access to it at different compliance levels. Only one AIA can be active at one time. Each AIA shall have one AudioControl interface and can have zero or more AudioStreaming interfaces.

The standard USB Interface Association mechanism is used to describe the Audio Interface Association i.e. to bind those interfaces together. Interface Association is expressed via the standard USB Interface Association descriptor (IAD). Every Interface Association descriptor has a **bFunctionClass**, **bFunctionSubClass** and **bFunctionProtocol** field that together identify the function that is represented by the Association. The following paragraphs define these fields for the Audio Device Class.

3.4.1 AUDIO FUNCTION CLASS

The Audio Function Class is contained in the **bFunctionClass** field of a Standard Interface Association descriptor. This specification requires that the Function Class code be the same as the Audio Interface Class code.

The Audio Function Class code is assigned by the USB-IF. For details, see Appendix A.1, “Audio Function Class Code”.

3.4.2 AUDIO FUNCTION SUBCLASS

The Audio Function Subclass is contained in the **bFunctionSubClass** field of a Standard Interface Association descriptor. Any AIA exposed by a Device that is compliant with this specification shall use this field to further define the device. This field indicates whether the AIA is a full-fledged implementation of the ADC 3.0 specification or is compliant with one of the BADD-defined Profiles.

The assigned codes can be found in A.2, “Audio Function Subclass Codes” of this specification. All other Subclass codes are unused and reserved by this specification for future use.

3.4.3 AUDIO FUNCTION PROTOCOL

The Audio Function Protocol is contained in the **bFunctionProtocol** field of a Standard Interface Association descriptor. The Function Protocol code is used to reflect the compliance level of the Audio Function so that enumeration software can decide which driver version needs to be instantiated. This specification requires that the Function Protocol code be the same as the Audio Interface (both AudioControl and AudioStreaming) Protocol code.

The assigned Protocol codes can be found in Appendix A.3, “Audio Function Protocol Codes” of this specification. All other Protocol codes are unused and reserved by this specification for future use.

3.5 AUDIO INTERFACE CLASS

The Audio Interface Class code is contained in the **bInterfaceClass** field of the Standard Interface descriptor of any Interface contained in an AIA. All USB Interfaces which provide functionality through this specification use the same Audio Interface Class code.

The Audio Interface class code is assigned by the USB. For details, see Appendix A.4, “Audio Interface Class Code”.

3.6 AUDIO INTERFACE SUBCLASS

The Audio Interface Subclass code is contained in the **bInterfaceSubClass** field of the Standard Interface descriptor of any Interface contained in an AIA. All USB interfaces that provide functionality through this specification, as well as any interfaces that provide MIDI streaming functionality, shall use one of the following Audio Interface Subclass codes as provided by this specification:

- AudioControl Interface Subclass
- AudioStreaming Interface Subclass
- MIDIStreaming Interface Subclass

The assigned codes can be found in Appendix A.5, “Audio Interface Subclass Codes” of this specification. All other Subclass codes are unused and reserved by this specification for future use.

3.7 AUDIO INTERFACE PROTOCOL

The Audio Interface Protocol code is contained in the **bInterfaceProtocol** field of the Standard Interface descriptor of any Interface contained in an AIA. The Interface Protocol code is used to reflect the compliance level of the Audio Function.

All Audio Functions compliant with this ADC 3.0 specification shall use Interface Protocol Code IP_VERSION_03_00. The assigned codes can be found in Appendix A.6, “Audio Interface Protocol Codes” of this specification. All other Protocol codes are unused and reserved by this specification for future use.

3.8 AUDIO FUNCTION CATEGORY

The Audio Function Category code is contained in the Class Specific Audio Control Interface descriptor as defined by this specification. This code indicates the primary intended use for the Audio Function. The following Function Categories are currently defined in this specification:

- Desktop Speaker: One or more speakers set up in a small environment to provide audio intended primarily for one person.
- Home Theater: Several speakers set up in a moderately sized environment to provide audio levels significantly louder than a Desktop Speaker setup and intended to be clearly heard by multiple people.
- Microphone: A device set up to record audio from audible sources.
- Headset: A device with at least one speaker and at least one microphone designed to be worn or held by a user to provide personal audio playback and voice input capabilities.
- Telephone: A Headset or handset type device that also connects to a telephone system, (e.g. POTs, PBX, VoIP) capable of making and receiving telephone calls.
- Converter: A device that allows conversion of audio from one electrical or optical format to another electrical or optical format, and/or converting audio data from one encoding format to another (e.g. AC-3 to PCM, etc.).
- Voice/Sound recorder: A device set up with at least one microphone and at least one speaker that is designed to operate, at least some of the time, independently of the Host to record and store audible sources and play back its recorded content.
- IO Box: A device designed to deliver one or more, possibly different, electrical and optical inputs and outputs for connection to other devices.
- Musical Instrument: A musical instrument, e.g. piano, guitar, synthesizer, drum machine, etc.

- Pro-Audio: A device not typically used by consumers of audio, e.g. editing equipment, multitrack recording equipment, etc.
- Audio/Video: The audio from a device that also supplies simultaneous video where the expectation is that the audio is tightly coupled to the video, e.g. a camcorder, a DVD player, a television, etc.
- Control Panel: A device that is used to control the flow of audio through a system of audio devices, such as a mixer panel.
- Other: Any device whose primary purpose is sufficiently different from the above descriptions as to be considered a completely different form of device.

The assigned codes can be found in Appendix A.7, "Audio Function Category Codes" of this specification. All other Category codes are unused and reserved by this specification for future use.

3.9 CLOCK DOMAINS

A Clock Domain is defined as a zone within which all sampling clocks are derived from the same master clock. Therefore, within the same Clock Domain, all sampling clocks are synchronous and their timing relationship is constant. However, the sampling clocks can be at different sampling frequencies. The master clock can be generated in many different ways. An internal crystal could be the master clock; the USB start of frame (SOF) could be used or even an externally supplied clock could serve as a master clock.

Multiple different Clock Domains may exist within the same Audio Function.

3.10 POWER DOMAINS

A Power Domain is defined as a zone within the Audio Function that groups one or more Entities and allows the Host to control power consumption levels for that Domain. This way, the Host can potentially switch parts of the Audio Function to lower power states when these parts are currently not in use, leading to an overall decrease in power consumption. As an example, a USB headset may have two separate Power Domains, one for the output-related functionality (headphone) and one for the input-related functionality (microphones). When the headset is used to simply listen to music, the input-related functionality may be temporarily switched to a low power state (or even completely off) to conserve power. This is especially important if the headset is used in conjunction with a battery-powered device, such as a mobile phone or MP3 player, where extending battery life is important.

Multiple different Power Domains may exist within the same Audio Function. Power Domains are identified by a unique Power Domain ID within the Audio Function.

3.11 AUDIO SYNCHRONIZATION TYPES

Each isochronous audio endpoint used in an AudioStreaming interface belongs to a synchronization type as defined in Section 5 of the *USB Specification*. The following sections briefly describe the possible synchronization types.

3.11.1 ASYNCHRONOUS

Asynchronous isochronous audio endpoints produce or consume data at a rate that is locked either to a clock external to the USB or to a free-running internal clock. These endpoints are not synchronized to a start of frame (SOF) or to any other clock in the USB domain.

3.11.2 SYNCHRONOUS

The clock system of synchronous isochronous audio endpoints can be controlled externally through SOF or Bus Interval synchronization. Such an endpoint shall lock its sample clock to the SOF tick or to the start of a new Bus Interval.

3.11.3 ADAPTIVE

Adaptive isochronous audio endpoints are able to source or sink data at any rate within their operating range. This implies that these endpoints shall run an internal process that allows them to match their natural data rate to the data rate that is imposed at their interface.

3.11.4 IMPLICATIONS OF THE DIFFERENT SYNCHRONIZATION TYPES

The following sections provide some information on the impact the different synchronization types have on both the Device and Host implementation. Several scenarios are considered and for each synchronization type, the implications on Host and Device side are listed.

3.11.4.1 SINGLE DIRECTION SINK ENDPOINT

A typical example for this scenario is a USB speaker, implementing a sink endpoint that receives streaming audio data from the Host.

- Asynchronous
  - **Host**: The Host driver needs to be able to handle an explicit feedback endpoint. From the feedback data, the Host then decides how many samples to send over the data streaming endpoint in subsequent Bus Intervals.
  - **Device**: The Device has its own local, free-running audio sample clock, which determines how many samples are consumed by the Device each Bus Interval. The Device shall implement an explicit feedback endpoint as well as the necessary logic to provide the correct feedback values to send over said endpoint back to the Host. The advantage of this mode of operation is that it is fairly easy to generate a robust, stable, jitter-free, high-quality audio sample clock (derived from a crystal-based master clock, for example).
- Synchronous
  - **Host**: The Host needs to send out a known number of bytes for each packet going to the Device. The Host may need to generate a (fixed) pattern of audio samples to achieve the desired sampling rate. As an example, to generate a sampling rate of 44.1 kHz in a Full-Speed implementation, the Host needs to send a repeating pattern of nine packets containing 44 audio samples, followed by one packet containing 45 audio samples.
  - **Device**: This synchronization type requires the Device to implement either an audio clock PLL or an ASRC.
- Adaptive
  - **Host**: The Host may use any method or means to determine how many samples per Bus Interval to transmit. Effectively operating as a “Synchronous-to-SOF” Source is an easy approach, but not the only one allowed by the USB specification.

    - **Device**: This synchronization type requires the Device to implement either an audio clock PLL or an ASRC to adapt to the average number of samples arriving over a certain period of time.

3.11.4.2 SINGLE DIRECTION SOURCE ENDPOINT

A typical example for this scenario is a USB microphone, implementing a source endpoint that sends streaming audio data to the Host.

- Asynchronous
    - **Host**: The Host needs to operate as an Adaptive Sink. Depending on the Host's system design, this may require an ASRC or an audio clock PLL on the Host side.
    - **Device:** The Device has its own local, free-running audio sample clock, which determines how many samples are produced by the Device each Bus Interval. The advantage of this mode of operation is that it is fairly easy to generate a robust, stable, jitter-free, high-quality audio sample clock (derived from a crystal-based master clock, for example).
- Synchronous
    - **Host**: The Host receives a known number of audio samples in each packet coming from the Device. The Host should expect a fixed pattern of audio samples to achieve the desired sampling rate.
    - **Device**: This synchronization type requires the Device to implement either an audio clock PLL to lock on to the SOF or the start of Bus Interval and generate a high-quality audio sample clock directly or use an ASRC as a bridge between the local and USB clock domains. The Device shall generate a fixed packet size pattern as described in the *USB Audio Data Formats Definition* document.
- Adaptive
    - **Host:** This scenario requires the implementation of a feedforward OUT endpoint in the Device that allows the Host to inform the Device how many samples per Bus Interval to send to the Host over the streaming data IN endpoint.
    - **Device:** This synchronization type requires the Device to implement an audio clock PLL or an ASRC to adapt to the sample rate being communicated by the Host via the feedforward OUT endpoint.

3.11.4.3 SOURCE AND SINK ENDPOINTS

A typical example for this scenario is a USB headset, implementing both a sink endpoint that receives streaming audio data from the Host and a source endpoint that sends streaming audio data to the Host.

In general, the endpoints can be treated independently, using the same rules and guidelines as stated above. This includes the ability for the Device to support separate clocks on each endpoint.

However, if the Device has its own clock and both data streams share this clock, the following two special cases can be used to allow implicit feedback:

- Both endpoints are Asynchronous. In this case, the data rate that appears on the Source endpoint can be used by the Host as implicit feedback to adjust the data rate transmitted to the Sink endpoint.

- Both endpoints are Adaptive. In this case, the data rate that appears on the Sink endpoint can be used by the Device to adjust the data rate transmitted by the Source endpoint.

Note: Using either of these two implicit feedback mechanisms would preclude the ability of an ADC 3.0 compliant device from being able to use Power Domains to shut down either the source or the sink, as data must remain flowing in both directions to provide the feedback/feedforward information.

3.12 INTER CHANNEL SYNCHRONIZATION

An important issue when dealing with audio, and 3-D audio in particular, is the phase relationship between different physical audio channels. Indeed, the virtual spatial position of an audio source is directly related to and influenced by the phase differences that are applied to the different physical audio channels used to reproduce the audio source. Therefore, it is imperative that USB Audio Functions respect the phase relationship among all related audio channels. However, the responsibility for maintaining the phase relation is shared among the USB host software, hardware, and all of the audio peripheral devices or functions.

3.13 AUDIO FUNCTION TOPOLOGY

To be able to manipulate the physical properties of an Audio Function, its functionality shall be divided into addressable Entities. Two types of such generic Entities are identified and are called Units and Terminals. In addition, a special type of Entity is defined. These Entities are called Clock Entities and they are used to describe and manipulate the clock signals inside the Audio Function.

Units provide the basic building blocks to fully describe most Audio Functions. Audio Functions are built by connecting together several of these Units. A Unit has one or more Input Pins and a single Output Pin, where each Pin represents a Cluster of logical audio channels inside the Audio Function (see Section 3.13.1, “Cluster”). Units are wired together by connecting their I/O Pins according to the required topology. Note that it is perfectly legal to connect the Output Pin of an Entity to multiple Input Pins residing on different other Entities, effectively creating a one-to-many connection.

In addition, the concept of a Terminal is introduced. There are two types of Terminals. An Input Terminal (IT) is an Entity that represents a starting point for audio channels inside the Audio Function. An Output Terminal (OT) represents an ending point for audio channels inside the Audio Function. From the Audio Function’s perspective, a USB endpoint is a typical example of an Input or Output Terminal. It either provides data streams to the Audio Function (IT) or consumes data streams coming from the Audio Function (OT). Likewise, a Digital to Analog converter, built into the Audio Function is represented as an Output Terminal in the Audio Function’s model. Connection to the Terminal is made through its single Input or Output Pin.

Input Pins of a Unit are numbered starting from one up to the total number of Input Pins on the Unit. The Output Pin number is always one. Input Terminals have only one Output Pin and its number is always one. Output Terminals have only one Input Pin and it is always numbered one.

The information, traveling over I/O Pins is not necessarily of a digital nature. It is perfectly possible to use the Unit model to describe fully analog or even hybrid Audio Functions. The mere fact that I/O Pins are connected together is a guarantee (by construction) that the protocol and format, used over these connections (analog or digital), is compatible on both ends.

Every Unit in the Audio Function is fully described by its associated Unit descriptor (UD). The Unit descriptor contains all necessary fields to identify and describe the Unit. Likewise, there is a Terminal descriptor (TD) for every

Terminal in the Audio Function. In addition, these descriptors provide all necessary information about the topology of the Audio Function. They fully describe how Terminals and Units are interconnected.

This specification describes the following types of standard Units and Terminals that are considered adequate to represent most Audio Functions:

- Input Terminal (IT)
- Output Terminal (OT)
- Mixer Unit (MU)
- Selector Unit (SU)
- Feature Unit (FU)
- Sampling Rate Converter Unit (RU)
- Effect Unit (EU)
- Processing Unit (PU)
- Extension Unit (XU)

Besides Units and Terminals, the concept of a Clock Entity is introduced. Three types of Clock Entities are defined by this specification:

- Clock Source (CS)
- Clock Selector (CX)
- Clock Multiplier (CM)

A Clock Source provides a certain sampling clock frequency to all or part of the Audio Function. A Clock Source can represent an internal sampling frequency generator, but it can also represent an external sampling clock signal input to the Audio Function.

A Clock Source has a single Clock Output Pin that carries the sampling clock signal, represented by the Clock Source. The Clock Output Pin number is always one.

A Clock Selector is used to select between multiple sampling clock signals that might be available inside an Audio Function. It has multiple Clock Input Pins and a single Clock Output pin. Clock Input Pins are numbered starting from one up to the total number of Clock Input Pins on the Clock Selector. The Clock Output Pin number is always one.

A Clock Multiplier is used to derive a new clock signal with a different frequency from the clock signal at its single Clock Input Pin. It does this by multiplying that clock signal frequency by a numerator P and then dividing it by a denominator Q. The new clock signal is guaranteed to be synchronous with the input clock signal. A Clock Multiplier has one Input Pin and one Output Pin and their numbers are always one. An implementation can choose to make the values P and Q programmable, although most Audio Functions will expose P and Q Controls as Read-Only and report any changes to those values via the interrupt mechanism. Such changes may be necessary when the Audio Function detects a change in native sampling rate for incoming encoded content, for example.

By using a combination of Clock Source, Clock Selector, and Clock Multiplier Entities, the most complex clock systems can be represented and exposed to Host software.

Clock Input and Output Pins are fundamentally different from Input and Output Pins defined for Units and Terminals. Clock Pins carry only clock signals and therefore cannot be connected to Unit or Terminal Input and Output Pins. They are only used to express clock circuitry topology.

Each Input and Output Terminal has a single Clock Input Pin that is connected to a Clock Output Pin of a Clock Entity. The clock signal carried by that Clock Output Pin determines at which sampling frequency the hardware represented by the Terminal is operating.

Each Sampling Rate Converter Unit has two Clock Input Pins that are typically connected to the Clock Output Pins of two different Clock Entities. The clock signals carried by those Clock Output Pins determine the sampling frequencies between which the Sampling Rate Converter Unit is converting.

Each Clock Entity is described by a Clock Entity descriptor (CED). The Clock Entity descriptor contains all necessary fields to identify and describe the Clock Entity.

The descriptors are further detailed in Section 4, "Descriptors" of this document.

The ensemble of Unit descriptors, Terminal descriptors and Clock Entity descriptors provide a full description of the Audio Function to the Host. This information is typically retrieved from the device at enumeration time. By parsing the descriptors, a generic audio driver should be able to fully control the Audio Function, except for the functionality represented by Extension Units. Those require vendor-specific extensions to the audio class driver.

Important Note:

> The complete set of Audio Function descriptors provides only a static initial description of the Audio Function. During operation, a number of events can happen that force the Audio Function to change its state. Host software shall be notified of these changes to remain 'in sync' with the Audio Function at all times. An extensive interrupt mechanism is in place to report any and all state changes to Host software.

Figure 3-3, "Inside the Audio Function" illustrates the concepts defined above. Using the iconic symbols defined further, it describes a hypothetical Audio Function that incorporates 15 Entities: three Input Terminals, five Units, three Output Terminals, two Clock Sources, a Clock Selector, and one Clock Multiplier. Each Entity has its unique ID (from 1 to 15) and descriptor that fully describes the functionality of the Entity and also how that particular Entity is connected into the overall topology of the Audio Function.

Input Terminal 1 (IT 1) is the representation of a USB OUT endpoint used to stream audio from the Host to the audio device. IT 2 is the representation of an analog Line-In connector on the audio device whereas IT 3 is an analog Microphone-In connector on the audio device. Selector Unit 4 (SU 4) selects between the audio coming from the Host and the audio present at the Line-In connector. Feature Unit 5 (FU 5) is then used to manipulate the audio (Volume, Bass, Treble …) before it is presented to Output Terminal 9 (OT 9). OT 9 is the representation of a Headphone Out jack on the audio device.

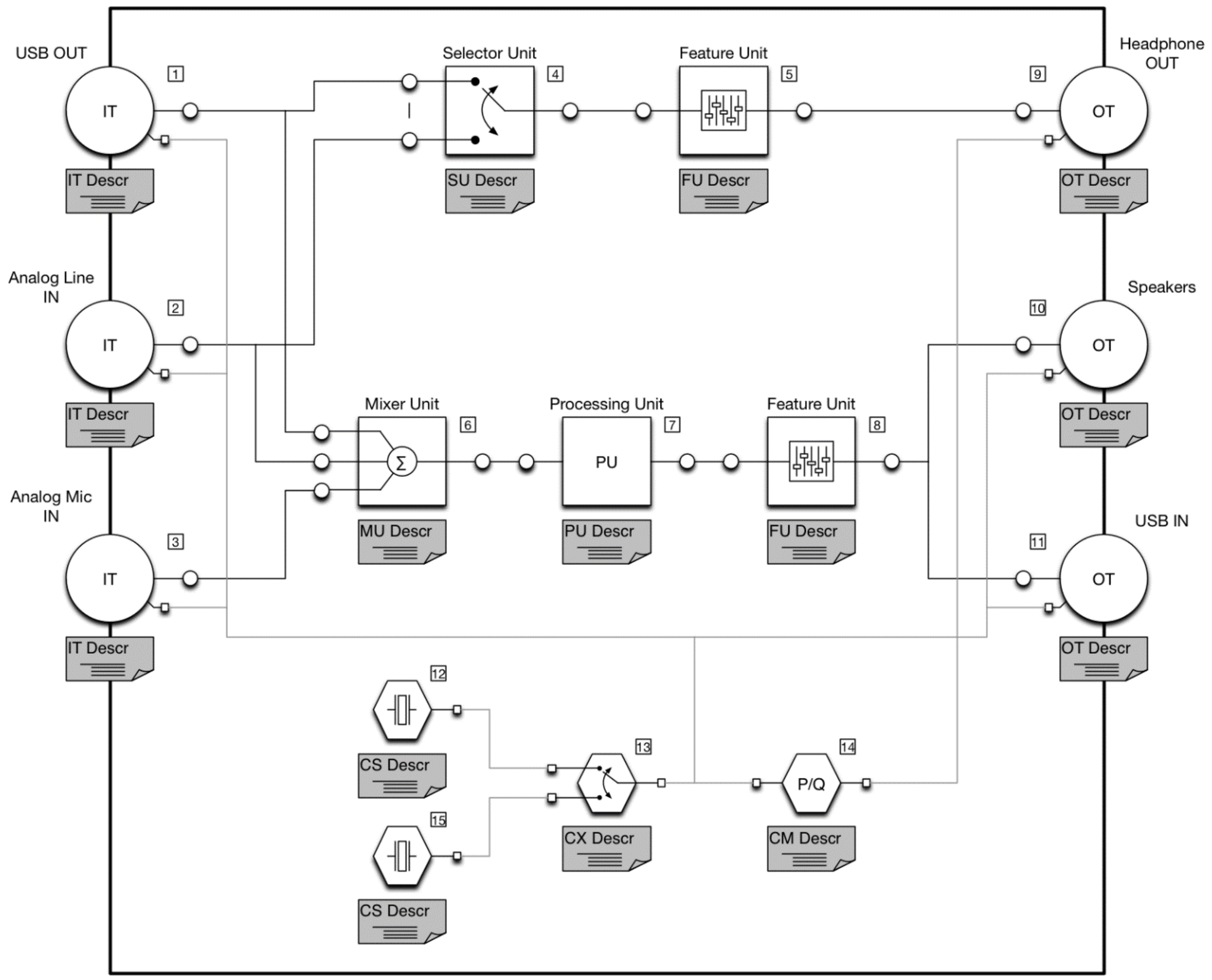
At the same time, all three input sources (USB OUT, Line-In, and Mic-In) are connected to a Mixer Unit 6 (MU 6) that effectively mixes the three sources together. The output of the Mixer is then fed into a Processing Unit 7 (PU 7) that performs some audio processing algorithm(s) on the mix. The result is in turn sent to FU 8 where some final adjustments to the audio (Volume …) are made. FU 8 is connected to OT 10 and OT 11. OT 10 represents speakers incorporated into the audio device and OT 11 represents a USB IN endpoint used to send the processed audio to the Host for recording purposes.

Clock Source 12 (CS 12) represents an internal sampling frequency generator, running at 96 kHz for instance. Clock Source 15 (CS 15) is the representation of an external master sampling clock input that can be used to synchronize the device to an external source. Clock Selector 13 (CS 13) enables selection between the two available Clock Sources. The output of CS 13 provides a sampling frequency to IT 1, IT 2, IT3, OT 10, and OT 11 of 96 kHz. Clock Multiplier CM 14 further multiplies that clock signal by 0.5, providing a sampling frequency of 48 kHz to OT 9 for driving the headphone. Since all sampling frequencies used inside the Audio Function are at all times derived from a single master clock (internal or external), all audio streams in the Audio Function are synchronous.

The descriptors, associated with each Entity clearly indicate to the Host what the exact nature of each Entity is. For instance, the IT 2 descriptor contains a field that indicates to the Host that it represents an external connector on the device, used as an analog Line In. Likewise, the MU 6 descriptor has a field that indicates that its Input Pin 1 is connected to the Output Pin of IT 1, Input Pin 2 is connected to the Output Pin of IT 2, and Input Pin 3 is connected to the Output Pin of IT 3.

For further details on descriptor contents, refer to Section 4, “Descriptors” of this document.

Figure 3-3: Inside the Audio Function



Inside an Entity, functionality is further described through Audio Controls. A Control typically provides access to a specific audio or clock property. Each Control has a set of attributes that can be manipulated or that present additional information on the behavior of the Control. A Control can have the following attributes:

- Current setting attribute
- Range attribute triplet consisting of:
    - Minimum setting attribute

- Maximum setting attribute
- Resolution attribute

As an example, consider a Volume Control inside a Feature Unit. By issuing the appropriate Get requests, the Host software can obtain values for the Volume Control's attributes and, for instance, use them to correctly display the Control on the screen. Setting the Volume Control's current attribute allows the Host software to change the volume setting of the Volume Control.

Additionally, each Entity in an Audio Function can have a memory space attribute. This attribute optionally provides generic access to the internal memory space of the Entity. This could be used to implement vendor-specific control of an Entity through generically provided access.

3.13.1 CLUSTER

A Cluster is a grouping of audio channels that carry tightly related synchronous audio information. Inside the Audio Function, complete abstraction is made of the actual physical representation form of the audio data that travels over the connections among Terminals and Units. Each audio channel in the Cluster is considered to be a logical channel and all the physical attributes of the channel (bit width, bit resolution, etc.) are not specified and considered irrelevant in the context of the Audio Function as seen through the AudioControl interface. The fact that an Input Pin and an Output Pin are connected together in the Audio Function's topology is a guarantee (by construction) that the information flowing over the connection is compatible with both Entities that are connected.

Channel numbering in the Cluster starts with channel one up to the number of channels in the Cluster. The virtual channel zero is used to address a master Control in a Unit (if present), effectively influencing all the channels at once. Note that the master Control (if present) shall be implemented separate from the per-channel Controls. Changing the setting of a master Control does not affect the settings of any of the individual channel Controls. The maximum number of independent channels in a Cluster is limited to 255 (Channel zero is used to reference the master channel).

A Cluster is characterized by a number of attributes:

- A unique identifier for the Cluster
- The number of audio channels in the Cluster
- The purpose of each audio channel, such as Voice, Speech, etc.
- The audio channel relationship (or spatial location) of each audio channel in the Cluster
- An indication whether an audio channel is part of an Ambisonic group

There is a Cluster descriptor (CD) associated with each Cluster that fully describes the Cluster.

There are two types of Clusters used in this specification:

- A logical Cluster describes audio channels within the Audio Function (the closed box) where audio channels are treated as logical concepts.
- A physical Cluster describes physical audio channels, for example when travelling over a connector.

3.13.2 INPUT TERMINAL

The Input Terminal (IT) is used to interface between the Audio Function's 'outside world' and other Units in the Audio Function. It serves as a receptacle for audio information flowing into the Audio Function. Its function is to represent a source of incoming audio data after this data has been properly extracted from the original audio stream into the separate logical channels that are embedded in this stream (the decoding process). The logical channels are grouped into a Cluster and leave the Input Terminal through a single Output Pin.

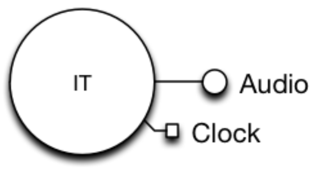
An Input Terminal can represent inputs to the Audio Function other than USB OUT endpoints. A Line-In connector on an audio device is an example of such a non-USB input. However, if the audio stream is entering the Audio Function by means of a USB OUT endpoint, there is a one-to-one relationship between the AudioStreaming interface that contains this endpoint and its associated Input Terminal. The class-specific interface descriptor contains a field that holds a direct reference to this Input Terminal. The Host needs to use both the AudioStreaming interface and endpoint descriptors and the Input Terminal descriptor to get a full understanding of the characteristics and capabilities of the Input Terminal. Stream-related parameters are stored in the AudioStreaming interface descriptors. Control-related parameters are stored in the Terminal descriptor.

The conversion process from incoming, possibly encoded, audio streams to logical audio channels always involves some kind of decoding engine. The decoding types range from rather trivial decoding schemes like converting interleaved stereo 16 bit PCM data into a Left and Right logical channel to very sophisticated schemes like converting an MPEG-2 7.1 encoded audio stream into Front Left, Front Left of Center, Front Center, Front Right of Center, Front Right, Back Left, Back Right and Low Frequency Effects logical channels. The decoding engine is considered part of the Entity that actually receives the encoded audio data streams (like a USB AudioStreaming interface). The type of decoding is therefore implied by the value in the **bmFormats** field, located in the class-specific AudioStreaming interface descriptor. The associated Input Terminal deals with the logical channels after they have been decoded.

An Input Terminal has a single Clock Input Pin. The clock signal present at that Pin is used as the sampling clock for all underlying hardware that is represented by this Input Terminal. There is a field in the Input Terminal descriptor that uniquely identifies the Clock Entity to which the Input Terminal is connected.

The symbol for the Input Terminal is depicted in the following figure:

Figure 3-4: Input Terminal Icon



3.13.3 OUTPUT TERMINAL

The Output Terminal (OT) is used to interface between Units inside the Audio Function and the 'outside world'. It serves as an outlet for audio information, flowing out of the Audio Function. Its function is to represent a sink of outgoing audio data before this data is properly packed from the original separate logical channels into the outgoing audio stream (the encoding process). The Cluster enters the Output Terminal through a single Input Pin.

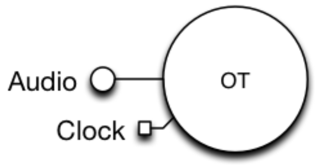
An Output Terminal can represent outputs from the Audio Function other than USB IN endpoints. A speaker built into an audio device or a Line Out connector is an example of such a non-USB output. However, if the audio stream is leaving the Audio Function by means of a USB IN endpoint, there is a one-to-one relationship between the AudioStreaming interface that contains this endpoint and its associated Output Terminal. The class-specific interface descriptor contains a field that holds a direct reference to this Output Terminal. The Host needs to use both the AudioStreaming interface and endpoint descriptors and the Output Terminal descriptor to fully understand the characteristics and capabilities of the Output Terminal. Stream-related parameters are stored in the AudioStreaming interface descriptors. Control-related parameters are stored in the Terminal descriptor.

The conversion process from incoming logical audio channels to possibly encoded audio streams always involves some kind of encoding engine. The encoding engine is considered part of the Entity that actually transmits the encoded audio data streams (like a USB AudioStreaming interface). The type of encoding is therefore implied by the value in the **bmFormats** field, located in the class-specific AudioStreaming interface descriptor. The associated Output Terminal deals with the logical channels before encoding.

An Output Terminal has a single Clock Input Pin. The clock signal present at that Pin is used as the sampling clock for all underlying hardware that is represented by this Output Terminal. There is a field in the Output Terminal descriptor that uniquely identifies the Clock Entity to which the Output Terminal is connected.

The symbol for the Output Terminal is depicted in the following figure:

Figure 3-5: Output Terminal Icon



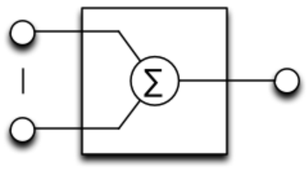
3.13.4 MIXER UNIT

The Mixer Unit (MU) transforms a number of logical input channels into a number of logical output channels. The input channels are grouped into one or more Clusters. Each Cluster enters the Mixer Unit through an Input Pin. The logical output channels are grouped into one Cluster and leave the Mixer Unit through a single Output Pin.

Every input channel can virtually be mixed into all of the output channels. If n is the total number of input channels and m is the number of output channels, then there is a two-dimensional array (n · m) of Mixer Controls in the Mixer Unit. Not all of these Controls have to be physically implemented. Some Controls can have a fixed setting and be non-programmable. The Mixer Unit descriptor reports which Controls are programmable in the **bmControls** bitmap field. Using this model, a permanent connection can be implemented by reporting the Control as non-programmable in the **bmControls** bitmap and by returning a Control setting of 0 dB when requested. Likewise, a missing (muting) connection can be implemented by reporting the Control as non-programmable in the **bmControls** bitmap and by returning a Control setting of –∞ dB. A Mixer Unit SHALL respond to the appropriate Get Request to allow the Host to retrieve the actual settings on each of the (n · m) Mixer Controls.

The symbol for the Mixer Unit can be found in the following figure:

Figure 3-6: Mixer Unit Icon

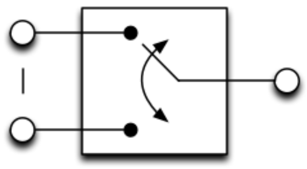


3.13.5 SELECTOR UNIT

The Selector Unit (SU) selects from n Clusters, each containing m logical input channels and routes them unaltered to the single output Cluster, containing m output channels. It represents a multi-channel source selector, capable of selecting between n m-channel sources. It has n Input Pins and a single Output Pin.

The symbol for the Selector Unit can be found in the following figure:

Figure 3-7: Selector Unit Icon



3.13.6 FEATURE UNIT

The Feature Unit (FU) is essentially a multi-channel processing unit that provides basic manipulation of multiple single-parameter Audio Controls on the incoming logical channels. For each logical channel, the Feature Unit optionally provides Audio Controls for the following features:

- Mute
- Volume
- Tone Control (Bass, Mid, Treble)
- Graphic Equalizer
- Automatic Gain Control
- Delay
- Bass Boost
- Loudness
- Input Gain
- Input Gain Pad
- Phase Inverter

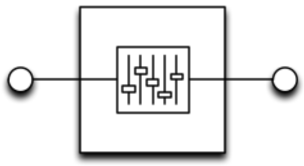
In addition, the Feature Unit optionally provides the above Audio Controls but now influencing all channels of the Cluster at once. In this way, 'master' Controls can be implemented. The master Controls are cascaded after the individual channel Controls. This setup is especially useful in multi-channel systems where the individual channel Controls can be used for channel balancing and the master Controls can be used for overall settings.

The logical channels in the Cluster are numbered from one to the total number of channels in the Cluster. The 'master' channel has channel number zero and is always virtually present.

The Feature Unit descriptor reports what Controls are present for every channel in the Feature Unit and for the 'master' channel. All logical channels in a Feature Unit are fully independent. There exist no cross couplings among channels within the Feature Unit. There are as many logical output channels, as there are input channels. These are grouped into one Cluster that enters the Feature Unit through a single Input Pin and leaves the Unit through a single Output Pin.

The symbol for the Feature Unit is depicted in the following figure:

Figure 3-8: Feature Unit Icon



3.13.7 SAMPLING RATE CONVERTER UNIT

The Sampling Rate Converter (SRC) Unit (RU) is included here as an optional way to indicate where exactly within the Audio Function sampling rate conversion takes place. In many cases, it is unnecessary to indicate this point and any SRC Unit can be omitted from the topology without materially affecting the information presented to the Host. The primary reason to include the SRC Unit is to accurately report any latencies incurred by the Sampling Rate Conversion process.

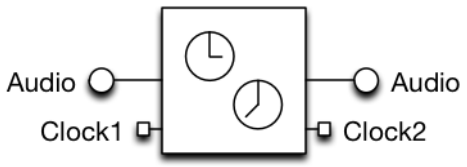
The Sampling Rate Converter Unit provides a bridge function between different clock domains within the Audio Function. The Sampling Rate Converter does not provide any Audio Controls. It takes the audio on all the logical channels in the single input Cluster belonging to a certain clock domain and converts them into the same logical channels in the single output Cluster but now belonging to another clock domain.

There are as many logical output channels, as there are input channels. These are grouped into one Cluster that enters the SRC Unit through a single Input Pin and leaves the Unit through a single Output Pin.

A SRC Unit has two Clock Input Pins. One Clock Input Pin is associated with the single Input Pin of the SRC Unit. The other Clock Input Pin is associated with the single Output Pin of the SRC Unit. The clock signals present at those two Clock Input Pins identify the two clock domains between which the SRC Unit is converting. Note that it is allowed to have both Clock Input Pins connected to clock signals belonging to the same clock domain.

The symbol for the SRC Unit is depicted in the following figure:

Figure 3-9: Sampling Rate Converter Unit Icon



3.13.8 EFFECT UNIT

The Effect Unit (EU) is a multi-channel processing unit that provides advanced manipulation of a multi-parameter Audio Control on the incoming logical channels on a per-channel basis. For each logical channel, the Effect Unit provides one of the following Audio Controls:

- Parametric Equalizer Section
- Reverberation
- Modulation Delay
- Dynamic Range Compressor

In addition, the Effect Unit optionally provides one of the above Audio Controls but now influencing all channels of the Cluster at once. In this way, a 'master' Control can be implemented. The master Control is cascaded after the individual channel Controls. This setup is especially useful in multi-channel systems where the individual channel Controls can be used for channel balancing and the master Control can be used for overall settings.

The logical channels in the Cluster are numbered from one to the total number of channels in the Cluster. The 'master' channel has channel number zero and is always virtually present.

The Effect Unit descriptor reports what Controls are present for every channel in the Effect Unit and for the 'master' channel. All logical channels in an Effect Unit are fully independent. There exist no cross couplings among channels within the Effect Unit. There are as many logical output channels, as there are input channels. These are grouped into one Cluster that enters the Effect Unit through a single Input Pin and leaves the Unit through a single Output Pin.

If the Effect Unit is an explicit member of a Power Domain, then switching the Power Domain to any Power State other than D0 shall render the Unit non-functional and its output is undefined. Explicit bypass functionality may be required to preserve the integrity of the signal path.

3.13.8.1 PARAMETRIC EQUALIZER SECTION EFFECT UNIT

The Parametric Equalizer Section (PEQS) Effect Unit is used to manipulate and equalize the frequency characteristics of the original audio information around a certain center frequency. In order to build a parametric equalizer, a number of these PEQS Effect Units may need to be cascaded to obtain the desired functionality. The parameters that can be manipulated to obtain the desired equalizing effect are:

- Center Frequency: the frequency around which the audio spectrum is manipulated. Expressed in Hz.
- Q Factor: a measure for the range of frequencies around the center frequency that are influenced. Expressed as a ratio.
- Gain: the amount of gain or attenuation at the center frequency. Expressed in dB.

The algorithm to produce the desired equalization effect can be manipulated on a per-channel basis. The master channel concept allows equalization for all channels simultaneously.

The symbol for the PEQS Processing Unit can be found in the following figure:

Figure 3-10: PEQS Effect Unit Icon

3.13.8.2 REVERBERATION EFFECT UNIT

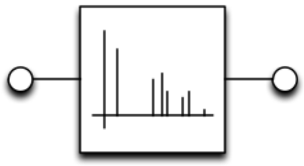
The Reverberation Effect Unit is used to add room acoustics effects to the original audio information. These effects can range from small room reverberation effects to simulation of a large concert hall reverberation. A number of parameters can be manipulated to obtain the desired reverberation effects.

- Reverb Type: Room1, Room2, Room3, Hall1, Hall2, Plate, Delay, and Panning Delay.
- Reverb Level: sets the amount of reverberant sound versus the original sound. Expressed as a ratio.
- Reverb Time: sets the time over which the reverberation will continue. Expressed in s.
- Reverb Delay Feedback: used with Reverb Types Delay and Delay Panning. Sets the way in which delay repeats. Expressed as a ratio.
- Reverb Pre-Delay: sets the delay time between original sound and initial reverb reflection. Expressed in ms.
- Reverb Density: sets the density of the reverb reflections.
- Reverb Hi-Freq Roll-Off: sets the cut-off frequency of a low pass filter on the reflections. Expressed in Hz.

It is entirely left to the designer how a certain reverberation effect is obtained. It is not the intention of this specification to precisely define all the parameters that influence the reverberation experience (for instance in a multi-channel system, it is possible to create very similar reverberation impressions, using different algorithms and parameter settings on all channels).

The symbol for the Reverberation Effect Unit can be found in the following figure:

Figure 3-11: Reverberation Effect Unit Icon



3.13.8.3 MODULATION DELAY EFFECT UNIT

The Modulation Delay Effect Unit is used to add modulation (like chorus) effects to the original audio information. A number of parameters can be manipulated to obtain the desired modulation effects.

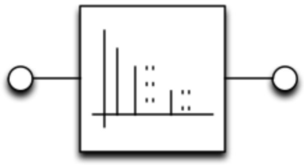
- Modulation Delay Balance: controls the ratio of the original sound to that of the effected sound. Expressed as a ratio.
- Modulation Delay Rate: sets the speed (frequency) of the modulator. Expressed in Hz.
- Modulation Delay Depth: sets the depth at which the sound is modulated. Expressed in ms.

- Modulation Delay Time: sets the delay that is added to the modulated sound before adding it to the original sound. Expressed in ms.
- Modulation Delay Feedback Level: controls the amount of the modulated sound that is routed back to the input of the modulator unit. Expressed as a ratio.

It is entirely left to the designer how a certain modulation effect is obtained.

The symbol for the Modulation Delay Effect Unit can be found in the following figure:

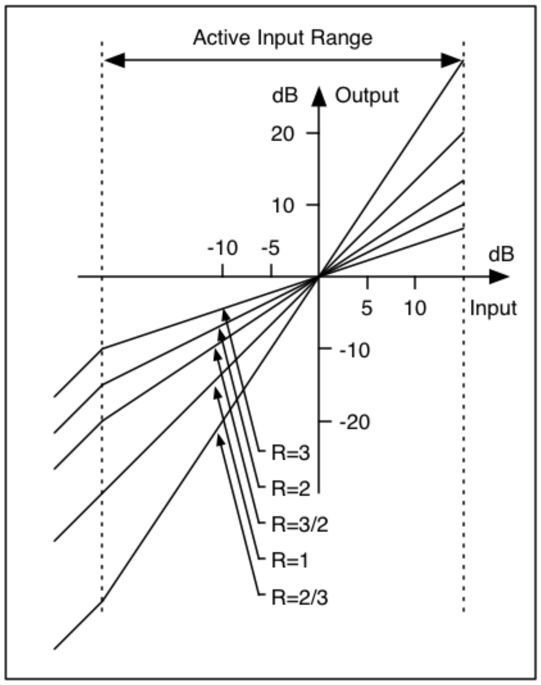
Figure 3-12: Modulation Delay Effect Unit Icon



3.13.8.4 DYNAMIC RANGE COMPRESSOR EFFECT UNIT

The Dynamic Range Compressor Effect Unit is used to intelligently limit the dynamic range of the original audio information. A number of parameters can be manipulated to influence the desired compression.

Figure 3-13: Dynamic Range Compressor Transfer Characteristic



- Compression ratio R: determines the slope of the static input-to-output transfer characteristic in the compressor's active input range. The compression is defined in terms of the compression ratio R, which is the inverse of the derivative of the output power P_O as a function of the input power P_I when P_O and P_I are expressed in dB.

$$R^{-1} = \frac{\partial Log(\frac{P_O}{P_R})}{\partial Log(P_I/P_R)}$$

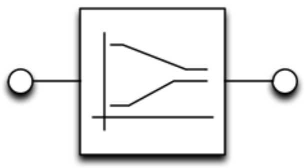
P_R is the reference level and it is made equal to the so-called line level. All levels are expressed relative to the line level (0 dB), which is usually 15-20 dB below the maximum level. Compression is obtained when R > 1, R = 1 does not affect the signal and R < 1 gives rise to expansion.

- Maximum Amplitude: the upper boundary of the active input range, relative to the line level (0 dB). Expressed in dB.
- Threshold level: the lower boundary of the active input level, relative to the line level (0 dB).
- Attack Time: determines the response of the compressor as a function of time to a step in the input level. Expressed in ms.
- Release Time: relates to the recovery time of the gain of the compressor after a loud passage. Expressed in ms.
- Make-up Gain: set to compensate for the gain loss in the compressor. Expressed in dB.

It is entirely left to the designer how a certain dynamic range compression is obtained.

The symbol for the Dynamic Range Compressor Effect Unit can be found in the following figure:

Figure 3-14: Dynamic Range Compressor Effect Unit Icon



3.13.9 PROCESSING UNIT

The Processing Unit (PU) represents a functional block inside the Audio Function that transforms a number of logical input channels, grouped into one or more Clusters into a number of logical output channels, grouped into one Cluster. Therefore, the Processing Unit can have multiple Input Pins and has a single Output Pin. This specification defines several standard transforms (algorithms) that are considered necessary to support additional Audio Functionality; these transforms are not covered by the other Unit types but are commonplace enough to be included in this specification so that a generic driver can provide control for it.

If it is necessary to be able to bypass the functionality incorporated in the Processing Unit, then an explicit bypass topology using a Selector Unit shall be implemented.

If the Processing Unit is an explicit member of a Power Domain, then switching the Power Domain to any Power State other than D0 shall render the Unit non-functional and its output is undefined. Explicit bypass functionality may be required to preserve the integrity of the signal path.

3.13.9.1 UP/DOWN-MIX PROCESSING UNIT

The Up/Down-mix Processing Unit provides facilities to derive *m* output audio channels from *n* input audio channels. The algorithms and transforms applied to accomplish this are not defined by this specification and can be proprietary. The input channels are grouped into one input channel Cluster that enters the Processing Unit over a single Input Pin. Likewise, all output channels are grouped into one output channel Cluster, leaving the Processing Unit over a single Output Pin.

The Up/Down-mix Processing Unit can support multiple modes of operation. The available input audio channels are dictated by the Entity to which the Up/Down-mix Processing Unit is connected. The Up/Down-mix Processing Unit descriptor reports which up/down-mixing modes the Unit supports through its **waClusterDescrID()** array. Each element of the **waClusterDescrID ()** array indicates which output channels in the output Cluster are effectively used in a particular mode. The unused output channels in the output Cluster shall produce muted output. Mode selection is implemented using the Get/Set Mode Select Control request.

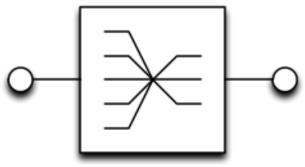
As an example, consider the case where an Up/Down-mix Processing Unit is connected to an Input Terminal, producing Dolby™ AC-3 5.1 decoded audio. The input Cluster to the Up/Down-mix Processing Unit therefore contains Left, Right, Center, Left Surround, Right Surround and LFE logical channels.

Suppose the Audio Function's hardware is limited to reproducing only dual channel audio. Then the Up/Down-mix Processing Unit could use some (sophisticated) algorithms to down-mix the available spatial audio information into two ('enriched') channels so that the maximum spatial effects can be experienced, using only two channels. It is left to the implementation to use the appropriate down-mix algorithm depending on the physical nature of the Output Terminal to which the Up/Down-mix Processing Unit is routed. For instance, a different down-mix algorithm is needed whether the 'enriched' stereo stream is sent to a pair of speakers or to a headphone set. However, this knowledge already resides within the Audio Function and deciding which down-mix algorithm to use does not need Host intervention.

As a second interesting example, suppose the hardware is capable of servicing eight discrete audio channels (for example, a full-fledged MPEG-2 7.1 system). Now the Up/Down-mix Processing Unit could use certain techniques to derive meaningful content for the extra audio channels (Left of Center, Right of Center) that are present in the output Cluster and are missing in the input channel Cluster (AC-3 5.1). This is a typical example of an up-mix situation.

The symbol for the Up/Down-mix Processing Unit is depicted in the following figure:

Figure 3-15: Up/Down-mix Processing Unit Icon

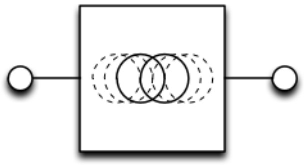


3.13.9.2 STEREO EXTENDER PROCESSING UNIT

The Stereo Extender Processing Unit operates on Left and Right channels only. It processes an existing stereo (two channel) soundtrack to expand the sound field and to make it appear to originate from outside the Front Left/Right speaker locations. Extended stereo effects can be achieved via various methods. The algorithms and transforms applied to accomplish this are not defined by this specification and can be proprietary. The perceived width of the sound field can be controlled using the Get/Set Width Control request.

The symbol for the Stereo Extender Unit is depicted in the following figure:

Figure 3-16: Stereo Extender Processing Unit Icon



3.13.9.3 MULTI-FUNCTION PROCESSING UNIT

The Multi-Function Processing Unit groups together different but related algorithmic blocks that together provide a certain functionality. This specification defines several algorithms that are considered useful and commonplace enough to be included in this specification.

The following algorithms are currently supported:

- Algorithm Undefined
- Beam Forming Algorithm
- Acoustic Echo Cancellation Algorithm
- Active Noise Cancellation Algorithm
- Blind Source Separation Algorithm
- Noise Suppression/Reduction

The Multi-Function Processing Unit descriptor contains a bitmap, indicating which algorithms are supported by the Unit. The exact implementation of these algorithms and how they may interact is implementation-dependent.

Note 1: If there is a need to expose to the Host how the algorithms are interconnected, a designer may choose to model the assembly of algorithms using multiple Multi-Function Processing Units, each containing just one or a subset of algorithms and explicitly connecting them together.

Note 2: Most of the algorithms mentioned above involve some form of signal processing that cannot be assumed to be linear and time invariant.

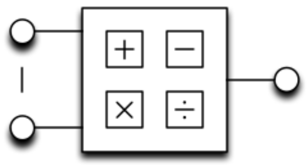
During normal operation, the Multi-Function Processing Unit transforms a number of logical input channels that enter the Unit over one or more Input Pins into a number of logical output channels that leave the Unit over a single Output Pin.

If it is necessary to be able to bypass the functionality incorporated in the Multi-Function Processing Unit, then an explicit bypass topology using a Selector Unit shall be implemented.

It is strongly recommended to implement the bypass functionality so that a generic audio driver that does not understand what functionality is implemented in the Multi-Function Processing Unit will be capable of removing it from the signal path.

The symbol for the Multi-Function Processing Unit is depicted in the following figure:

Figure 3-17 Multi-Function Processing Unit Icon



3.13.10 EXTENSION UNIT

The Extension Unit (XU) is the method provided by this specification to easily add vendor-specific building blocks to the specification. The Extension Unit provides one or more logical input channels, grouped into one or more Clusters and transforms them into a number of logical output channels, grouped into one Cluster. Therefore, the Extension Unit can have multiple Input Pins and has a single Output Pin.

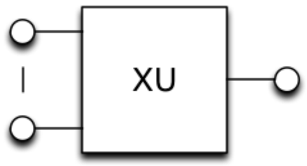
If it is necessary to be able to bypass the functionality incorporated in the Extension Unit, then an explicit bypass topology using a Selector Unit shall be implemented.

It is strongly recommended to implement the bypass functionality so that a generic audio driver that does not understand what functionality is implemented in the Extension Unit will be capable of removing it from the signal path.

If the Extension Unit is an explicit member of a Power Domain, then switching the Power Domain to any Power State other than D0 shall render the Unit non-functional and its output is undefined. Explicit bypass functionality may be required to preserve the integrity of the signal path.

The symbol for the Extension Unit can be found in the following figure:

Figure 3-18: Extension Unit Icon



3.13.11 CLOCK ENTITIES

Clock Entities are special in the sense that they do not directly manipulate logical audio streams. Instead, they provide the functionality needed to manipulate sampling clock signals and clock routing for the different Input and Output Terminals within the Audio Function. A Terminal inside the Audio Function can only be connected to one Clock Entity. The clock signal present at the Clock Input Pin of a Terminal determines the sampling frequency at which the underlying hardware is operating.

3.13.11.1 CLOCK SOURCE

A Clock Source Entity provides an independent sampling clock signal on its single Clock Output Pin. The Clock Source Entity serves as the master clock for a Clock Domain. Several different other synchronous sampling frequencies can be derived by connecting multiple Clock Multiplier Entities to the same Clock Source Entity. By manipulating the Sampling Frequency Control inside the Clock Source Entity, all sampling frequencies in the Clock domain are influenced.

Each independent master sampling clock inside the Audio Function shall be represented by a separate Clock Source Entity. Even if the clock is generated 'inside a Terminal', that clock needs to be represented by a Clock Source Entity. As an example, a sampling clock could be recovered based on the amount of audio samples coming into the Audio Function over a USB OUT adaptive endpoint. Alternatively, a sampling clock may be derived from the S/PDIF signal coming into the Audio Function on an external connector.

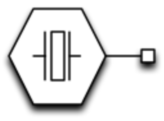
Note: In the case of an adaptive isochronous data endpoint that support only a discrete number of sampling frequencies, the endpoint shall at least tolerate ±1000 PPM inaccuracy on the reported Sampling Frequency Control values to accommodate sample clock inaccuracies.

The Clock Source Entity descriptor contains a field that indicates the origin of the actual clock signal, represented by the Entity. Furthermore, since Input and Output Terminals only have a Clock Input Pin, a clock signal can never be generated from a Terminal directly.

The output of a Clock Source Entity does not have to be valid at all times. For instance, if a Clock Source Entity represents an external sampling clock input on the Audio Function, the output of that Clock Source might not be valid when there is nothing connected to the external clock input. The Clock Source can be queried for the validity of its output signal at all times.

The symbol for the Clock Source Entity can be found in the following figure:

Figure 3-19: Clock Source Icon



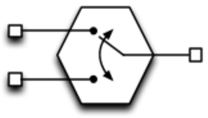
3.13.11.2 CLOCK SELECTOR

A Clock Selector Entity provides the functionality to select among different available sampling clock signals. It can have n Clock Input Pins from which one is routed to the single Clock Output Pin.

Switching between Clock Inputs can be Host controlled (the Clock Selector is programmable via the appropriate Set request) or the Audio Function can switch Clock Inputs due to some external event. A Clock Selector may support both control methods. The Clock Selector can notify the Host of the change by generating the appropriate interrupt.

The symbol for the Clock Selector Entity can be found in the following figure:

Figure 3-20: Clock Selector Icon

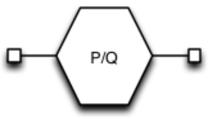


3.13.11.3 CLOCK MULTIPLIER

A Clock Multiplier Entity provides the functionality to derive a new sampling clock signal frequency from the sampling clock signal, present at its single Clock Input Pin. The algorithm used to derive the new clock signal is not defined by this specification. However, there is a requirement that the output clock signal shall be synchronous to the input clock signal so that both clocks belong to the same Clock Domain. A Clock Multiplier Entity contains a multiplier followed by a divider. Both the multiplication factor P and division factor Q can be programmable and in the range [1, 2^{16}-1]. The resulting sampling frequency is obtained by multiplying the input signal frequency by P/Q.

The symbol for the Clock Multiplier Entity can be found in the following figure:

Figure 3-21: Clock Multiplier Icon



3.14 OPERATIONAL MODEL

A device can support multiple configurations. Within each configuration can be multiple interfaces, each possibly having Alternate Settings. These interfaces can pertain to different functions that co-reside in the same composite device. Even several independent Audio Functions can exist in the same device. Interfaces, belonging to the same Audio Function are grouped into an Audio Interface Association. If the device contains multiple independent Audio Functions, there shall be multiple Audio Interface Associations, each providing full access to their associated Audio Function.

As an example of a composite device, consider a PC monitor equipped with a built-in stereo speaker system. Such a device could be configured to have one interface dealing with configuration and control of the monitor part of the device (HID Class), while a Collection of two other interfaces deals with its audio aspects. One of those, the AudioControl interface, is used to control the inner workings of the function (Volume Control etc.) whereas the other, the AudioStreaming interface, handles the data traffic, sent to the monitor’s audio subsystem.

The AudioStreaming interface could be configured to operate in mono mode (Alternate Setting *x*) in which only a single channel data stream is sent to the Audio Function. The receiving Input Terminal could duplicate this audio stream into two logical channels, and those could then be reproduced on both speakers. From an interface point of view, such a setup requires one isochronous endpoint in the AudioStreaming interface to receive the mono audio data stream, in addition to the mandatory control endpoint and optional interrupt endpoint in the AudioControl interface.

The same system could be used to play back stereo audio. In this case, the stereo AudioStreaming interface is selected (Alternate Setting *y*). This interface also consists of a single isochronous endpoint, now receiving a data stream that interleaves Front Left and Front Right channel samples. The receiving Input Terminal now splits the stream into a Front Left and Front Right logical channel. The AudioControl interface Alternate Setting remains unchanged.

If the above AudioStreaming interface were an asynchronous sink, one extra isochronous Feedback endpoint would also be necessary.

As stated earlier, Audio Functionality is located at the interface level in the device class hierarchy. The following sections describe the Audio Interface Association, containing a single AudioControl interface and optional AudioStreaming interfaces, together with their associated endpoints that are used for Audio Function control and for audio data stream transfer.

3.14.1 AUDIOCONTROL INTERFACE

To control the functional behavior of a particular Audio Function, the Host can manipulate the Clock Entities, Units and Terminals inside the Audio Function. To make these objects accessible, the Audio Function shall expose a single AudioControl interface. This interface can contain the following endpoints:

- A control endpoint for manipulating Clock Entity, Unit and Terminal settings and retrieving the state of the Audio Function. This endpoint is mandatory, and the default endpoint 0 is used for this purpose.
- An interrupt endpoint. The endpoint is optional, but shall be implemented if any of the Controls inside the device have the need to interrupt the Host to notify the Host of a change in the Audio Function's behavior.

The AudioControl interface is the single entry point to access the internals of the Audio Function. All requests that are concerned with the manipulation of certain Audio Controls within the Audio Function's Clock Entities, Units or Terminals shall be directed to the AudioControl interface of the Audio Function. Likewise, all descriptors related to the internals of the Audio Function are part of the class-specific AudioControl interface descriptor.

The AudioControl interface of an Audio Function can only support a single Alternate Setting (Alternate Setting 0).

3.14.1.1 CONTROL ENDPOINT

The audio interface class uses endpoint 0 (the default pipe) as the standard way to control the Audio Function using class-specific requests. These requests are always directed to one of the Clock Entities, Units, or Terminals that make up the Audio Function. The format and contents of these requests are detailed further in this document.

3.14.1.2 INTERRUPT ENDPOINT

A USB AudioControl interface can support an optional interrupt endpoint to inform the Host about dynamic changes that occur on the different addressable Entities (Clock Entities, Terminals, Units, interfaces and endpoints) inside the Audio Function. The interrupt endpoint is used by the entire Audio Interface Association to convey change information to the Host. It is considered part of the AudioControl interface because this is the anchor interface for the Collection.

3.14.2 AUDIOSTREAMING INTERFACE

AudioStreaming interfaces are used to interchange digital audio data streams between the Host and the Audio Function. They are optional. An Audio Function can have zero or more AudioStreaming interfaces associated with it, each possibly carrying data of a different nature and format. Each AudioStreaming interface can have at most one isochronous data endpoint. This construction guarantees a one-to-one relationship between the AudioStreaming interface and the single audio data stream, related to the endpoint. In some cases, the isochronous data endpoint is accompanied by an associated isochronous explicit feedback endpoint for synchronization purposes. The isochronous data endpoint and its associated feedback endpoint shall follow the endpoint numbering scheme as set forth in the *USB Specification*.

An AudioStreaming interface can have Alternate Settings that can be used to change certain characteristics of the interface and underlying endpoint. A typical use of Alternate Settings is to provide a way to change the subframe size and/or number of channels on an active AudioStreaming interface. Whenever an AudioStreaming interface requires an isochronous data endpoint, it shall at least provide the default Alternate Setting (Alternate Setting 0) with zero bandwidth requirements (no isochronous data endpoint defined) and an additional Alternate Setting that contains the actual isochronous data endpoint.

The AudioStreaming interface is essentially used to provide an access point for the Host software (drivers) to manipulate the behavior of the physical interface it represents. Therefore, even external connections to the Audio Function (S/PDIF interface, analog input, etc.) can be represented by an AudioStreaming interface so that the Host software can control certain aspects of those connections. This type of AudioStreaming interface has no associated USB endpoints. The related audio data stream is not using USB as a transport medium.

For every isochronous OUT or IN endpoint defined in any of the AudioStreaming interfaces, there shall be a corresponding Input or Output Terminal defined in the Audio Function. For the Host to fully understand the nature and behavior of the connection, it needs to take into account the interface- and endpoint-related descriptors as well as the Terminal-related descriptor.

3.14.2.1 ISOCHRONOUS AUDIO DATA STREAM ENDPOINT

In general, the data streams that are handled by an isochronous audio data endpoint do not necessarily map directly to the logical channels that exist within the Audio Function. As an example, consider the case where multiple logical audio channels are compressed into a single data stream (AC-3, WMA …). The format of such a data stream can be entirely different from the native format of the logical channels (for example, 640 Kbits/s AC-3 5.1 audio as opposed to 6 channel 16 bit 44.1 kHz audio). Therefore, to describe the data transfer at the endpoint level correctly, the notion of logical channel is replaced by the notion of audio data stream. It is the responsibility of the AudioStreaming interface which contains the OUT endpoint to convert between the audio data stream and the embedded logical channels before handing the data over to the Input Terminal. In many cases, this conversion process involves some form of decoding. Likewise, the AudioStreaming interface which contains the IN endpoint shall convert logical channels from the Output Terminal into an audio data stream, often using some form of encoding. If the decoding or encoding process exposes Controls that influence the encoding or decoding, these Controls can be accessed through the AudioStreaming interface.

Requests to control properties that exist within an Audio Function, such as volume or mute cannot be sent to the endpoint in an AudioStreaming interface. An AudioStreaming interface operates on audio data streams and is unaware of the number of logical channels it eventually serves. Instead, these requests shall be directed to the proper Audio Function's Units or Terminals via the AudioControl interface.

As already mentioned, an AudioStreaming interface can have zero or one isochronous audio data endpoint. If multiple synchronous audio channels shall be communicated between Host and Audio Function, they shall be clustered into one physical Cluster by interleaving the individual audio data, and the result can be directed to the single endpoint.

If an Audio Function needs more than one Cluster to operate, each Cluster is directed to the endpoint of a separate AudioStreaming interface, belonging to the same Audio Interface Association (all servicing the same Audio Function).

3.14.2.2 ISOCHRONOUS FEEDBACK ENDPOINT

For adaptive audio source endpoints and asynchronous audio sink endpoints, an explicit synchronization mechanism is needed to maintain synchronization during transfers. For details about synchronization at different endpoint speeds, see the applicable *USB Specification*.

3.14.2.3 AUDIO DATA FORMAT

The format used to transport audio data over the USB is entirely determined by the bits set in the **bmFormats** field of the class-specific interface descriptor. Some additional fields in this descriptor further describe the format. For details about the defined Format Types and associated data formats, see the separate document, *USB Audio Data Formats* that is considered part of this specification. Vendor-specific protocols shall be fully documented by the manufacturer.

3.14.3 CLOCK MODEL

Clock Entities provide a way to accurately describe the use and distribution of sampling clock signals throughout the Audio Function. Sampling frequencies inside the Audio Function can only be influenced by directly interacting with the Sampling Frequency Control inside a Clock Source Entity. The Sampling Frequency Control RANGE attributes provide the necessary information for Host software to determine what sampling frequencies the Control (and thus the associated Clock Domain) supports.

A side effect of changing the sampling frequency could be that certain AudioStreaming interfaces may need to switch to a different Alternate Setting to support the bandwidth needed for the new sampling frequency. This specification does not allow an AudioStreaming interface to switch from one Alternate Setting to another on its own except to change to Alternate Setting zero, which is the idle setting. Instead, when the Audio Function detects that it can no longer support a certain Alternate Setting on an AudioStreaming interface, it shall switch to Alternate Setting zero on that interface and report the change to Host software through the Active Alternate Setting Control interrupt. The Host can then query the interface for new valid Alternate Settings for the interface through the Get Valid Alternate Settings Control request and make an appropriate selection.

> Note: To keep the number of Alternate Settings in an AudioStreaming interface to a minimum, it is not recommended to provide a separate Alternate Setting for every supported sampling frequency. A few Active Alternate Settings (low bandwidth, medium bandwidth, high bandwidth) might be enough to provide reasonable bandwidth control.

Audio streams can be bridged from one clock domain to another through the use of the Sampling Rate Converter Unit.

3.14.4 POWER DOMAINS MODEL

Managed Power Domain support is optional for an Audio Function. If supported, the Audio Function is subdivided into one or more Power Domains, identified by their unique Power Domain ID, that can be individually manipulated by the Host to optimize overall power consumption. If supported, the AudioControl interface contains an array of Power Domain Controls that allows the Host to set Power Domain States for each individual Power Domain inside the Audio Function. The Power Domain Control array has as many elements as there are Power Domains and access to a specific Power Domain Control is based on the Power Domain ID.

A Power Domain shall support three possible Power Domain States D0 to D2. Power Domain State D0 is the fully operational state and shall be the default Power Domain State for all Power Domains. Power Domain State D1 is a non-functional state that shall consume less power than Power Domain State D0, but still has the capability to generate wake-up events (interrupts) for the Controls in its Power Domain. Power Domain State D2 shall consume even less (possibly zero) power than Power Domain State D1, and shall not be able to generate any wake-up events for the Controls in its Power Domain.

An Audio Function is not allowed to change the Power Domain State of any of its Power Domains autonomously. A change in Power Domain State shall always be initiated through an explicit request from the Host. Power Domain States shall be retained across function or device suspend.

Note: Power Domain States operate independently from the function or device suspend state. For example, bringing all Power Domains in an Audio Function into Power Domain State D2 does not automatically put the Audio Function into function suspend.

Actual power consumption levels for each Power Domain State are not advertised. The only requirement is that a higher numbered Power Domain State consume less power than all lower numbered Power Domain States, potentially at the expense of higher recovery times. Recovery time is defined as the approximate time it takes to get from the lower Power Domain State D1 or D2 back to the fully operational Power Domain State D0. The Power Domain descriptor shall indicate the recovery time for the transition from Power Domain State D1 to D0 and from Power Domain State D2 to D0.

The Audio Function is best placed to manage the details of its resources and their power consumption under various conditions. These details are therefore not exposed to the Host. Rather, the Host indicates to the Audio Function which parts of the Audio Function it intends to use at the current time and the Audio Function shall autonomously decide what resources it can safely bring into a lower power state.

With a few exceptions, the Host expresses to the Audio Function how it will use its building blocks by indicating which Input and Output Terminals it will actively use to accommodate a certain usage scenario. From that information, the Audio Function can derive on its own what internal resources (Entities) can be brought into a lower power state or even shut down completely for that usage scenario. Therefore, there is no need to exhaustively enumerate membership to a certain Power Domain for all Entities in the Audio Function. It suffices to indicate Power Domain membership for Input and Output Terminals only. By manipulating a particular Power Domain's State, the Host now indicates to the Audio Function whether it wants to use the inputs and/or outputs associated with that Power Domain in the current usage scenario.

It is possible to have a Power Domain that only includes Entities other than Input and Output Terminals. In this case, an explicit bypass of these Entities should be implemented in the Audio Function's topology via a Selector Unit that resides outside that Power Domain. Setting the Power Domain State of the Power Domain to which such an Entity belongs to anything other than D0, shall render that Entity non-functional and therefore it shall cease to produce meaningful output (the output is undefined by this specification in this case). To preserve the integrity of the signal path, Host software needs to bypass the Entity in question first (by selecting the appropriate input on the Selector Unit) before switching the Power Domain to a lower Power State.

Entities that can be explicitly bypassed may be the sole member of a Power Domain or may be a member of a larger membership, provided all non-Terminal members of the membership can be bypassed either individually or as a group.

3.14.5 ADDITIONAL POWER CONSIDERATIONS AND REQUIREMENTS

An Audio 3.0 compliant bus-powered device shall support LPM/L1 according to the requirements defined in the USB 2.0 LPM/L1 ECN and LPM Errata. The Audio 3.0 device USB subsystem shall implement and enter low-power mode during L1. An Audio 3.0 device shall accept LPM/L1 entry requests (i.e. return an ACK) at all times, even if the device's isochronous RX endpoint buffer has not yet been emptied or the isochronous TX endpoint already contains samples for the next burst. The Host is responsible for and must be trusted to resume the device in time to service the next isochronous burst.

The only exception is when the device needs additional uSOFs tokens in order to resynchronize its internal clock with the USB clock. In such cases the device may deny the request to put the link into an L1 state. It shall do so only once every 64 ms or longer.

A USB Host that supports LPM/L1 is required to follow the requirements defined in the USB 2.0 LPM/L1 ECN and LPM Errata. However, for isochronous endpoints the host shall also be required to send at least one uSOF token before resuming transactions to an endpoint. The USB Host shall be required to request LPM/L1 entry after servicing the device in each service interval. The USB host is additionally required to implement low power state and save power during L1.

3.14.6 BINDING BETWEEN PHYSICAL BUTTONS AND AUDIO CONTROLS

Most devices that contain an Audio Function will also have one or more front panel buttons that are intended to control certain aspects of the Audio Function inside the device. The most obvious example is a Volume Control button on the front of a multimedia speaker. Since an Audio Function can potentially contain many Audio Controls of the same type, there is a need to bind a physical control (button, knob, slider, jog, …) to a particular Audio Control inside the Audio Function.

This specification provides two mutually exclusive methods to provide this binding:

- The physical button is implemented as a HID Control
- The physical button is an integral part of the Audio Control

It is prohibited to implement both methods for the same physical button. However, it is allowed to use the first method for some of the front panel buttons and the second method for the remaining front panel buttons. It is strongly discouraged to implement front panel buttons that use neither of the above mentioned methods, i.e. buttons that are invisible to Host software and have a local effect only.

3.14.6.1 PHYSICAL BUTTON IS A HID CONTROL

In this case, the physical button is completely separate from the Audio Function and is implemented within the device’s HID interface. The Audio Function is not even aware of the button’s existence. Any change of state for the button is communicated to Host software via HID reports. It is then up to Host software to interpret the button state change and derive from there the appropriate action to be taken toward the Audio Function. Therefore, the binding responsibility resides entirely within the application or Operating System software. Although this method provides extensive flexibility, it also puts the burden of providing the correct binding on the software, making it sometimes hard to create generic application or OS software that generates the proper (manufacturer intended) binding.

3.14.6.2 PHYSICAL BUTTON IS INTEGRAL PART OF THE AUDIO CONTROL

In this case, the physical button directly interacts with the actual Audio Control. Button state changes are not reported to Host software. Instead, the change of state of the Audio Control resulting from the button manipulation is reported to Host software through the Audio Control interrupt mechanism. As a consequence, the binding between the physical button and the Audio Control is very direct and entirely dictated by the design of the device. Although less flexible, this method provides a very clear and straightforward way to perform the binding.

4 DESCRIPTORS

The following sections describe the standard and class-specific USB descriptors for the Audio Interface Class.

4.1 STANDARD DESCRIPTORS

Because Audio Functionality is always considered to reside at the interface level, all relevant fields in the standard descriptors shall indicate that class information is to be found at the interface level so that enumeration software looks down at the interface level to determine the Interface Class and to also ensure that IAD-aware enumeration software gets loaded.

Because any Device compliant with this specification is required to contain multiple Configuration descriptors, it is also required that any such device include a Configuration Summary Descriptor as part of the standard BOS descriptor. These Configuration Summary descriptors may be used by Host software to decide which Configuration to use to obtain the desired functionality, not just for any Audio Function, but also for any other function available in the Configuration. The Audio Function portion of the Configuration Summary descriptor shall contain the **bFunctionClass**, **bFunctionSubClass**, and **bFunctionProtocol** fields of each AIA contained in the Configuration being summarized. As mentioned previously in Section 3.3, "Backwards Compatibility", the same AIA may need to be included in multiple Configuration descriptors if the USB Device contains other USB Class Functions with differing versions.

Refer to the BOS Configuration Summary ECR/ECN for more details.

Also note that because Devices that are compliant with this specification are required to contain a BOS descriptor, the minimum **bDeviceProtocol** value in the Standard USB Device descriptor shall be 0x0201.

The standard Interface Association mechanism is used to describe an Audio Interface Association (AIA). All interfaces belonging to the same AIA shall be identified by means of the standard Interface Association descriptor (IAD).

Each AIA shall consist of the mandatory AudioControl interface that shall be the first in the AIA (having the lowest interface number). All AudioStreaming interfaces shall be contiguously numbered and immediately follow the AudioControl interface in the AIA. All MIDIStreaming interfaces shall be contiguously numbered and immediately follow the AudioStreaming interfaces in the AIA.

Note: For more information on Interface Association, refer to *USB Interface Association Descriptor Device Class Code and Use Model White Paper*, available on the USB web site.

4.2 CLASS-SPECIFIC DESCRIPTORS

This specification allows for two different methods to express class-specific descriptors. The first method follows the traditional layout for USB descriptors whereas the second method introduces a new way for expressing and retrieving class-specific descriptors. These descriptors are called High Capability descriptors. For each class-specific descriptor, implementations shall use the method as indicated in this specification. High Capability descriptors are used in the following cases:

- The descriptor length exceeds the 256-byte limit imposed by traditional descriptors
- The descriptor is dynamic in nature as per this specification, i.e. it may change after enumeration (High Capability descriptors have the ability to report changes).

High Capability descriptors are never part of the configuration descriptor hierarchy, returned by the Get Configuration request. Only traditional layout descriptors can be included in this hierarchy. A High Capability descriptor shall always be referenced by a traditional class-specific descriptor that includes the High Capability descriptor's unique ID as one of its fields.

4.2.1 TRADITIONAL CLASS-SPECIFIC DESCRIPTORS

Traditional class-specific descriptors as defined in this specification all follow a common layout. The first three fields of any traditional class-specific descriptor are common to all traditional descriptors and are followed by a layout that is specific to the type and subtype of the descriptor.

The **bLength** field contains the total length of the descriptor, in bytes.

The **bDescriptorType** field in part follows the bit allocation scheme of the bmRequestType field identifies the descriptor as being a class-specific descriptor. Bit D7 of this field is reserved. Bits D6..5 are used to indicate that this is a class-specific descriptor (D6..5 = 0b01). Bits D4..0 are used to encode the descriptor type.

The **bDescriptorSubtype** field further qualifies the exact nature of the descriptor.

Table 4-1: Traditional Class-Specific Descriptor Layout

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 3+n.
1	bDescriptorType	1	0b001xxxxx	Descriptor type.
2	bDescriptorSubtype	1	Constant	Descriptor subtype.
3	---	n	---	Descriptor type- and subtype-specific descriptor layout.

4.2.1.1 COMMON FIELDS IN SOME CLASS-SPECIFIC DESCRIPTORS

Some fields appear in more than one class-specific descriptor. Instead of repeating the description of that field for all descriptor instances, the description is indicated once in this Section and applies to all descriptor instances in which the field appears.

4.2.1.1.1 BMCONTROLS FIELD

The **bmControls** field contains a set of bit pairs, indicating which Controls are present in the Entity and what their capabilities are. If a certain Control is not present, then the bit pair shall be set to 0b00. If a Control is present but Read-Only, the bit pair shall be set to 0b01. If a Control is also Host programmable, the bit pair shall be set to 0b11 (Read/Write). The value 0b10 is not allowed.

4.2.2 HIGH CAPABILITY CLASS-SPECIFIC DESCRIPTORS

This specification defines a new type of class-specific descriptors, called High Capability descriptors. A High Capability descriptor is not retrieved from the Audio Function during enumeration time. Rather, it can only be retrieved using the Get High Capability Descriptor request as defined in Section 5.2.3.3, "High Capability Descriptor Request".

A High Capability descriptor is always referenced by another descriptor via its unique ID.

A High Capability class-specific descriptor can be up to 64K bytes in length. Also, it can generate an interrupt of source type HIGH_CAPABILITY_DESCRIPTOR whenever the Audio Function changes the descriptor during normal operation (dynamic descriptor).

High Capability descriptors all follow a common layout, very similar to the traditional class-specific descriptors. The first three fields of any High Capability descriptor are common to all High Capability descriptors and are followed by a layout that is specific to the type and subtype of the descriptor.

The **wLength** field contains the total length of the descriptor, in bytes. Descriptor lengths up to a maximum of 64K bytes are supported.

The **bDescriptorType** field in part follows the bit allocation scheme of the **bmRequestType** field identifies the descriptor as being a class-specific descriptor. Bit D7 of this field is reserved. Bits D6..5 are used to indicate that this is a class-specific descriptor (D6..5 = 0b01). Bits D4..0 are used to encode the descriptor type.

The **bDescriptorSubtype** field further qualifies the exact nature of the descriptor. **bDescriptorSubtype** field bits D6..0 are used to encode the descriptor subtype.

The **wDescriptorID** field contains a value that uniquely identifies the High Capability descriptor within an interface or endpoint.

Table 4-2: High Capability Class-Specific Descriptor Layout

Offset	Field	Size	Value	Description
0	wLength	2	Number	Size of this descriptor, in bytes: 6+n.
2	bDescriptorType	1	0b001xxxxx	Descriptor type.
3	bDescriptorSubtype	1	Constant	Descriptor subtype.
4	wDescriptorID	2	Number	Unique ID for this High Capability class-specific descriptor.
6	---	n	---	Descriptor type- and subtype-specific descriptor layout.

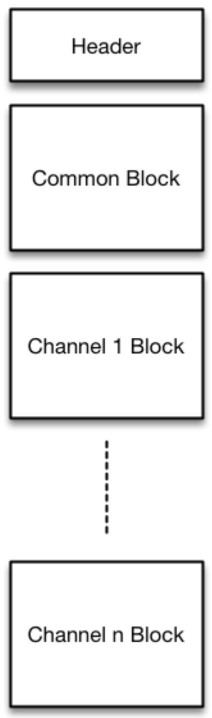
4.3 CLUSTER DESCRIPTOR

A Cluster is a grouping of audio channels that share the same characteristics like sampling frequency, bit resolution, etc. To characterize a Cluster, a Cluster descriptor is introduced.

This specification introduces a Cluster descriptor that is significantly different from earlier Cluster descriptor definitions. The goal is to be much more flexible and extensible and at the same time describe the Cluster more accurately and with richer content.

The Cluster descriptor always uses the High Capability representation. At the highest level, it is structured as follows:

Figure 4-1: Cluster Descriptor



The descriptor consists of a fixed Header, followed by an optional Common Block, followed by as many Channel Blocks as there are channels in the Cluster.

4.3.1 CLUSTER DESCRIPTOR HEADER

The Header starts with the **wLength** field that contains the total number of bytes in the entire descriptor.

The **wDescriptorID** field contains an ID number that uniquely identifies the descriptor within the Audio Function. The value zero is reserved and shall not be used as a valid Cluster descriptor ID.

The **bNrChannels** field indicates the number of audio channels present in the Cluster.

Table 4-3: Cluster Descriptor Header

Offset	Field	Size	Value	Description
0	wLength	2	Number	Total length of the Cluster descriptor, in bytes.
2	bDescriptorType	1	Constant	CS_CLUSTER descriptor type.
3	bDescriptorSubtype	1	Constant	SUBTYPE_UNDEFINED descriptor subtype.
4	wDescriptorID	2	Number	Unique ID of this Cluster descriptor.
6	bNrChannels	1	Number	Number of channels present in the Cluster: n.

4.3.2 CLUSTER DESCRIPTOR BLOCK

The Cluster descriptor Header is followed by a number of Cluster descriptor Blocks. There is an optional Common Block, followed by as many Channel Blocks as there are channels in the Cluster. Each Block consists of a number of Segments.

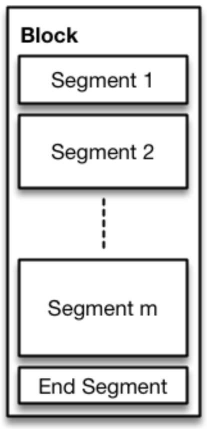
The Common Block consists of Segments that contain relevant information about characteristics of the Cluster as a whole.

A Channel Block consists of Segments that contain relevant information about that channel's characteristics. At a minimum, each Channel Block shall contain either an Information Segment or an Ambisonic Segment and a single End Segment. All other Segments are optional. It is highly recommended that the same layout is used for each Channel Block, i.e. the same Segments appear in the same order in each Channel Block.

However, if for any reason, there is currently no Cluster information available, then the Cluster descriptor shall only contain a Header with the **bNrChannels** field set to zero. This is called the Empty Cluster descriptor.

Figure 4-2 further illustrates the above concepts.

Figure 4-2: Cluster Descriptor Block



4.3.2.1 SEGMENTS

There are two types of Segments. Common Block Segments contain pertinent information about the Cluster as a whole. Channel Block Segments contain pertinent information about certain aspects of a particular channel. Both Segment types share the same layout.

The **bSegmentType** field describes the Segment Type (Common Block or Channel Block) and also the type of content contained in the Segment.

The layout for a Segment is always as follows:

Table 4-4: Cluster Descriptor Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Segment, in bytes: 3+n.
2	bSegmentType	1	Constant	Describes the Segment Type and the type of content in the Segment.

Offset	Field	Size	Value	Description
3	Segment-specific	n		Segment-specific content.

4.3.2.1.1 END SEGMENT

Each Block is terminated by an End Segment. The End Segment marks the end of the variable length Block. The End Segment does not have a Segment-specific section and is structured as follows:

Table 4-5: End Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the End Segment, in bytes: 3.
2	bSegmentType	1	Constant	END_SEGMENT.

4.3.2.1.2 COMMON BLOCK SEGMENTS

The following Common Block Segment types are defined:

- Cluster Description
- Vendor-specific

Values for the Common Block Segment types can be found in Appendix A.10, "Cluster Descriptor Segment types."

4.3.2.1.2.1 CLUSTER DESCRIPTION SEGMENT

The Cluster Description Segment contains the ID of a String descriptor in the **wCDDescrStr** field that provides more information about the Cluster.

Table 4-6: Cluster Description Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Segment, in bytes: 5.
2	bSegmentType	1	Constant	CLUSTER_DESCRIPTION.
3	wCDDescrStr	2	wStrDescrID	ID of a String descriptor that further describes the Cluster.

4.3.2.1.2.2 VENDOR-DEFINED SEGMENT

Vendors are allowed to add vendor-defined Segments to the Common Block to convey additional, proprietary Cluster information. The vendor-defined Segment shall use the following layout:

Table 4-7: Vendor-defined Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the End Segment, in bytes: 3+n.
2	bSegmentType	1	Constant	CLUSTER_VENDOR_DEFINED.
3	Vendor-defined	n	N/A	Vendor-defined extension of the Segment.

4.3.2.1.3 CHANNEL BLOCK SEGMENTS

The following Channel Block Segment types are defined:

- Information
- Ambisonic
- Channel Description
- Vendor-specific

Values for the Channel Block Segment types can be found in Appendix A.10, “Cluster Descriptor Segment types.”

4.3.2.1.3.1 INFORMATION SEGMENT

The Information Segment contains relevant information for a particular channel in the Cluster. It is mutually exclusive with the Ambisonic Segment for that same channel.

The **bChPurpose** field indicates the purpose of the channel. Currently defined purposes for a channel are:

- Generic Audio: contains audio primarily used for direct capture or reproduction.
- Voice: intended to be interpreted by humans.
- Speech: intended to be interpreted or generated by machine.
- Ambient: contains audio other than the primary channels.
- Reference: contains final processed audio. For example, a reference for AEC processing.
- Ultrasonic: contains signals with spectral content above audible limits (typically >20 kHz).
- Vibrokinetic: contains very low frequency information, typically used to actuate vibrators or motion actuators.
- Non-Audio: indicates that the channel carries non-audio information. Examples of non-audio information are real-time pressure sensing data or amplifier gain feedback data, etc.

The values for the purposes listed above can be found in Appendix A.11, “Channel Purpose Definitions.”

Note that great care should be taken when indicating purposes for an audio channel. Only when the channel has been specifically tailored to serve a purpose should the appropriate bit(s) be set. For example, if raw audio is processed specifically to be used by applications that use human voice as input, such a channel may be marked Voice.

The **wChRelationship** field describes the relationship of this channel with respect to the other channels in the Cluster. Currently defined relationships for a channel are described in the table below.

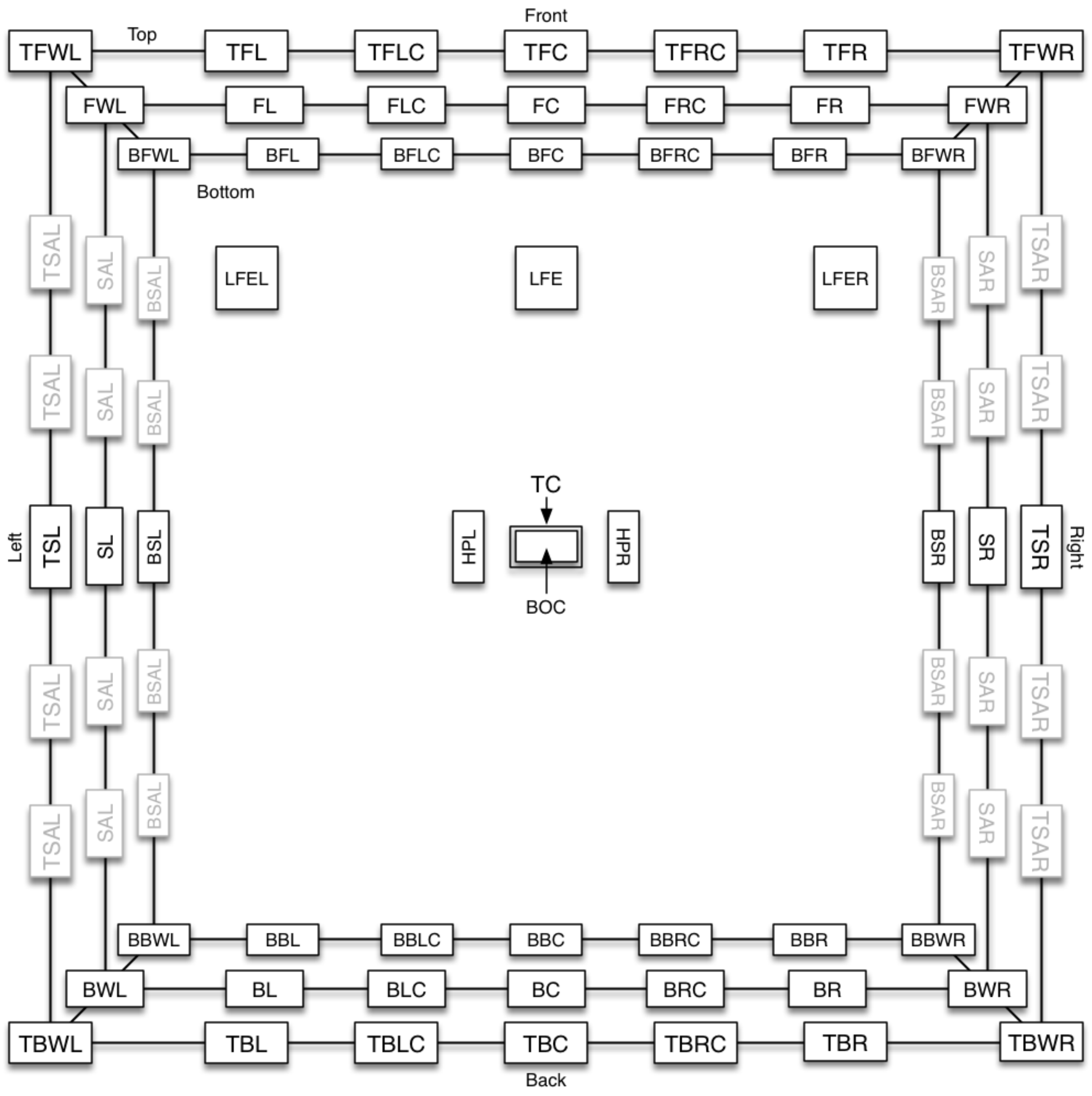
Table 4-8: Channel Relationships

	Channel Relationship	Acronym	Description
Generic	RELATIONSHIP_UNDEFINED	UND	Relationship undefined or unknown
	MONO	M	A mono channel
	LEFT	L	A generic Left channel
	RIGHT	R	A generic Right channel
	ARRAY	AR	A channel that is part of an array configuration

	Channel Relationship	Acronym	Description
Input-Related	PATTERN_X	PX	The X channel of an X/Y microphone
	PATTERN_Y	PY	The Y channel of an X/Y microphone
	PATTERN_A	PA	The A channel of an A/B microphone
	PATTERN_B	PB	The B channel of an A/B microphone
	PATTERN_M	PM	The M channel of an M/S microphone
	PATTERN_S	PS	The S channel of an M/S microphone
Output-Related	Front Left	FL	Refer to Figure 4-3
	Front Right	FR	Refer to Figure 4-3
	Front Center	FC	Refer to Figure 4-3
	Front Left of Center	FLC	Refer to Figure 4-3
	Front Right of Center	FRC	Refer to Figure 4-3
	Front Wide Left	FWL	Refer to Figure 4-3
	Front Wide Right	FWR	Refer to Figure 4-3
	Side Left	SL	Refer to Figure 4-3
	Side Right	SR	Refer to Figure 4-3
	Surround Array Left	SAL	Refer to Figure 4-3
	Surround Array Right	SAR	Refer to Figure 4-3
	Back Left	BL	Refer to Figure 4-3
	Back Right	BR	Refer to Figure 4-3
	Back Center	BC	Refer to Figure 4-3
	Back Left of Center	BLC	Refer to Figure 4-3
	Back Right of Center	BRC	Refer to Figure 4-3
	Back Wide Left	BWL	Refer to Figure 4-3
	Back Wide Right	BWR	Refer to Figure 4-3
	Top Center	TC	Refer to Figure 4-3
	Top Front Left	TFL	Refer to Figure 4-3
	Top Front Right	TFR	Refer to Figure 4-3
	Top Front Center	TFC	Refer to Figure 4-3
	Top Front Left of Center	TFLC	Refer to Figure 4-3
	Top Front Right of Center	TFRC	Refer to Figure 4-3
	Top Front Wide Left	TFWL	Refer to Figure 4-3
	Top Front Wide Right	TFWR	Refer to Figure 4-3
	Top Side Left	TSL	Refer to Figure 4-3
	Top Side Right	TSR	Refer to Figure 4-3
	Top Surround Array Left	TSAL	Refer to Figure 4-3

	Channel Relationship	Acronym	Description
	Top Surround Array Right	TSAR	Refer to Figure 4-3
	Top Back Left	TBL	Refer to Figure 4-3
	Top Back Right	TBR	Refer to Figure 4-3
	Top Back Center	TBC	Refer to Figure 4-3
	Top Back Left Of Center	TBLC	Refer to Figure 4-3
	Top Back Right Of Center	TBRC	Refer to Figure 4-3
	Top Back Wide Left	TBWL	Refer to Figure 4-3
	Top Back Wide Right	TBWR	Refer to Figure 4-3
	Bottom Center	BC	Refer to Figure 4-3
	Bottom Front Left	BFL	Refer to Figure 4-3
	Bottom Front Right	BFR	Refer to Figure 4-3
	Bottom Front Center	BFC	Refer to Figure 4-3
	Bottom Front Left Of Center	BFLC	Refer to Figure 4-3
	Bottom Front Right Of Center	BFRC	Refer to Figure 4-3
	Bottom Front Wide Left	BFWL	Refer to Figure 4-3
	Bottom Front Wide Right	BFWR	Refer to Figure 4-3
	Bottom Side Left	BSL	Refer to Figure 4-3
	Bottom Side Right	BSR	Refer to Figure 4-3
	Bottom Surround Array Left	BSAL	Refer to Figure 4-3
	Bottom Surround Array Right	BSAR	Refer to Figure 4-3
	Bottom Back Left	BBL	Refer to Figure 4-3
	Bottom Back Right	BBR	Refer to Figure 4-3
	Bottom Back Center	BBC	Refer to Figure 4-3
	Bottom Back Left Of Center	BBLC	Refer to Figure 4-3
	Bottom Back Right Of Center	BBRC	Refer to Figure 4-3
	Bottom Back Wide Left	BBWL	Refer to Figure 4-3
	Bottom Back Wide Right	BBWR	Refer to Figure 4-3
	Low Frequency Effects	LFE	Refer to Figure 4-3
	Low Frequency Effects Left	LFEL	Refer to Figure 4-3
	Low Frequency Effects Right	LFER	Refer to Figure 4-3
	Headphone Left	HPL	Refer to Figure 4-3
	Headphone Right	HPR	Refer to Figure 4-3

Figure 4-3: 3D Representation of the Channel Relationships



Constant names, acronyms, and their associated values are listed in Appendix A.12, “Channel Relationship Definitions.”

The **bChGroupID** field is used to create a link among a subset of channels in the Cluster by specifying the same value in the **bChGroupID** field for those channels. This field is useful when multiple sets of related channels are present in the same Cluster. For example, a single Cluster may carry a Group of 5.1 channels and another Group of microphone channels. The 5.1 channels would share the same Group ID value, indicating that they belong to a (5.1) Group, and the microphone channels would share another Group ID value, indicating they belong to a different (microphone) Group. The scope of the **bChGroupID** field is limited to the Cluster to which the channels belong. Note that specifying a value of 0 in this field for all channels in the Cluster effectively creates a Group with ID=0 to which all channels belong.

Table 4-9: Information Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Channel Information Segment, in bytes: 6.
2	bSegmentType	1	Constant	CHANNEL_INFORMATION.
3	bChPurpose	1	Number	Intended purpose of the channel.
4	bChRelationship	1	Number	Describes the relationship of this channel with the other channels. Refer to Appendix A.12, "Channel Relationship Definitions."
5	bChGroupID	1	Number	ID used to group channels together.

4.3.2.1.3.2 AMBISONIC SEGMENT

The Ambisonic Segment contains relevant information for a particular Ambisonic channel in the Cluster. It is mutually exclusive with the Information Segment for that same channel.

The **bCompOrdering** field indicates the convention used for ordering of the spherical harmonics. See Appendix A.13, "Ambisonic Component Ordering Convention Types" for the supported component ordering conventions. All channels in a Group (see below) shall indicate the same component ordering convention.

The **bACN** field contains the Ambisonic Channel Number.

The **bAmbNorm** field indicates the type of normalization used for the channel. See Appendix A.14, "Ambisonic Normalization Types" for the supported normalization types. All channels in a Group (see below) shall indicate the same normalization type.

The **bChGroupID** field is used to create a link among a subset of channels in the Cluster by specifying the same value in the **bChGroupID** field for those channels. This field is useful when multiple sets of related channels are present in the same Cluster. For example, a single Cluster may carry a Group of 5.1 channels and another Group of microphone channels. The 5.1 channels would share the same Group ID value, indicating that they belong to a (5.1) Group, and the microphone channels would share another Group ID value, indicating they belong to a different (microphone) Group. The scope of the **bChGroupID** field is limited to the Cluster to which the channels belong. Note that specifying a value of 0 in this field for all channels in the Cluster effectively creates a Group with ID=0 to which all channels belong.

Table 4-10: Ambisonic Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Ambisonic Channel Information Segment, in bytes: 7.
2	bSegmentType	1	Constant	CHANNEL_AMBISONIC.
3	bCompOrdering	1	Number	The Component Ordering convention for the Ambisonic Spherical Harmonics. Refer to Appendix A.13, "Ambisonic Component Ordering Convention Types."
4	bACN	1	Number	Ambisonic Channel Number.
5	bAmbNorm	1	Number	Applied channel normalization. Refer to Appendix A.14, "Ambisonic Normalization Types."

Offset	Field	Size	Value	Description
6	bChGroupID	1	Number	ID used to group channels together.

4.3.2.1.3.3 CHANNEL DESCRIPTION SEGMENT

The Channel Description Segment contains the ID of a String descriptor in the **wChDescrStr** field that provides more information about the channel.

Table 4-11: Channel Description Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Segment, in bytes: 5.
2	bSegmentType	1	Constant	CHANNEL_DESCRIPTION.
3	wChDescrStr	2	wStrDescrID	ID of a String descriptor that further describes the channel.

4.3.2.1.3.4 VENDOR-DEFINED SEGMENT

Vendors are allowed to add vendor-defined Segments to the Channel Block to convey additional, proprietary channel information. The vendor-defined Segment shall use the following layout:

Table 4-12: Vendor-defined Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the End Segment, in bytes: 3+n.
2	bSegmentType	1	Constant	CHANNEL_VENDOR_DEFINED.
3	Vendor-defined	n	N/A	Vendor-defined extension of the Segment.

4.3.3 EXAMPLE CLUSTER DESCRIPTOR

A Cluster that carries 5.1 discrete channels may have the following Cluster descriptor:

Table 4-13: Cluster Descriptor Example

	Offset	Field	Size	Value	Description
Header	0	wLength	2	0x3E	Total length of the Cluster descriptor, in bytes: 62.
	2	bDescriptorType	1	0x26	CS_CLUSTER descriptor type.
	3	bDescriptorSubtype	1	0x00	SUBTYPE_UNDEFINED descriptor subtype.
	4	wDescriptorID	2	Implementation-dependent	Unique ID of this Cluster descriptor.
	6	bNrChannels	1	6	Number of channels present in the Cluster.
Information Segment	8	wLength	2	0x06	Length of the Information Segment, in bytes.
	10	bSegmentType	1	0x20	CHANNEL_INFORMATION.
	11	bChPurpose	1	0x00	Generic Audio.
	12	bChRelationship	1	0x80	Front Left.
	13	bChGroupID	1	0x01	ID used to group channels together.

	Offset	Field	Size	Value	Description
End	14	wLength	2	0x03	Length of the End Segment, in bytes.
	16	bSegmentType	1	0xFF	END_SEGMENT.
Information	17	wLength	2	0x06	Length of the Information Segment, in bytes.
	19	bSegmentType	1	0x20	CHANNEL_INFORMATION.
	20	bChPurpose	1	0x00	Generic Audio.
	21	bChRelationship	1	0x81	Front Right.
	22	bChGroupID	1	0x01	ID used to group channels together.
End	23	wLength	2	0x03	Length of the End Segment, in bytes.
	25	bSegmentType	1	0xFF	END_SEGMENT.
Information	26	wLength	2	0x06	Length of the Information Segment.
	28	bSegmentType	1	0x20	CHANNEL_INFORMATION.
	29	bChPurpose	1	0x00	Generic Audio.
	30	bChRelationship	1	0x82	Front Center.
	31	bChGroupID	1	0x01	ID used to group channels together.
End	32	wLength	2	0x03	Length of the End Segment, in bytes.
	34	bSegmentType	1	0xFF	END_SEGMENT.
Information	35	wLength	2	0x06	Length of the Information Segment.
	37	bSegmentType	1	0x20	CHANNEL_INFORMATION.
	38	bChPurpose	1	0x00	Generic Audio.
	39	bChRelationship	1	0x89	Surround Array Left.
	40	bChGroupID	1	0x01	ID used to group channels together.
End	41	wLength	2	0x03	Length of the End Segment.
	43	bSegmentType	1	0xFF	END_SEGMENT.
Information	44	wLength	2	0x06	Length of the Information Segment.
	46	bSegmentType	1	0x20	CHANNEL_INFORMATION.
	47	bChPurpose	1	0x00	Generic Audio.
	48	bChRelationship	1	0x8A	Surround Array Right.
	49	bChGroupID	1	0x01	ID used to group channels together.
End	50	wLength	2	0x03	Length of the End Segment.
	52	bSegmentType	1	0xFF	END_SEGMENT.
Information	53	wLength	2	0x06	Length of the Information Segment.
	55	bSegmentType	1	0x20	CHANNEL_INFORMATION.
	56	bChPurpose	1	0x00	Generic Audio.
	57	bChRelationship	1	0xB8	Low Frequency Effects.
	58	bChGroupID	1	0x01	ID used to group channels together.

	Offset	Field	Size	Value	Description
End	59	wLength	2	0x03	Length of the End Segment.
	61	bSegmentType	1	0xFF	END_SEGMENT.

4.3.4 CEA-861.2 CHANNEL MAPPING

The Advanced Audio Extensions specification CEA-861.2 allocates speaker spatial locations in a different order than what is defined in this USB Audio Definition. Also, there are channel relationships defined that do not appear in the USB Audio Definition. To provide consistency in implementations that use both the CEA-861.2 allocations and the USB Audio channel relationships, a mapping scheme between the two is included here to which such implementations shall adhere.

The mapping between the USB-defined channel relationships and the CEA speaker allocations is included in the table in Appendix A.12, "Channel Relationship Definitions."

4.4 PHYSICAL VERSUS LOGICAL CLUSTER

This specification makes a distinction between a logical and physical Cluster. Hence, there are also two types of Cluster descriptors defined:

- Logical Cluster descriptor
- Physical Cluster descriptor

The layout of these descriptors is identical and shall always use the High Capability representation as described in Section 4.2.2, "High Capability Class-Specific Descriptors." Both the logical and physical Cluster descriptors are not independent descriptors as such. They are always referenced by other descriptors. The referencing descriptors always include a **wClusterDescrID** field that contains the unique ID of the Cluster descriptor they reference.

The logical Cluster descriptor is referenced by one of the following descriptors:

- Input Terminal descriptor
- Mixer Unit descriptor
- Processing Unit descriptor
- Extension Unit descriptor

The physical Cluster descriptor is referenced by the class-specific AudioStreaming interface descriptor in each Alternate Setting of an AudioStreaming interface (except for Alternate Setting 0).

The physical Cluster descriptor is also referenced by the Connectors descriptor to provide information about the physical channels that travel over the various connectors, associated with a Terminal.

4.4.1 MAPPING BETWEEN PHYSICAL AND LOGICAL CLUSTERS

Audio streams always enter or leave the Audio Function via an AudioStreaming – Terminal pair. The following sections describe in more detail the possible cases and the internal operations that take place.

4.4.1.1 AUDIOSTREAMING OUT INTERFACE – INPUT TERMINAL

For Type I Audio streams, the physical Cluster descriptor referenced by the class-specific AudioStreaming interface descriptor of the active Alternate Setting of that interface describes the audio channels as they enter the interface.

For Type III/IV Audio streams, the physical Cluster describes the audio channels after they have been decoded into discrete audio channels.

The audio samples (Audio Subslots) shall be ordered in the exact same order as they are listed in the physical Cluster descriptor. The Cluster then enters an implicit Up/Down-mix process to produce an output Cluster as described by the Cluster descriptor that is referenced by the associated Input Terminal descriptor.

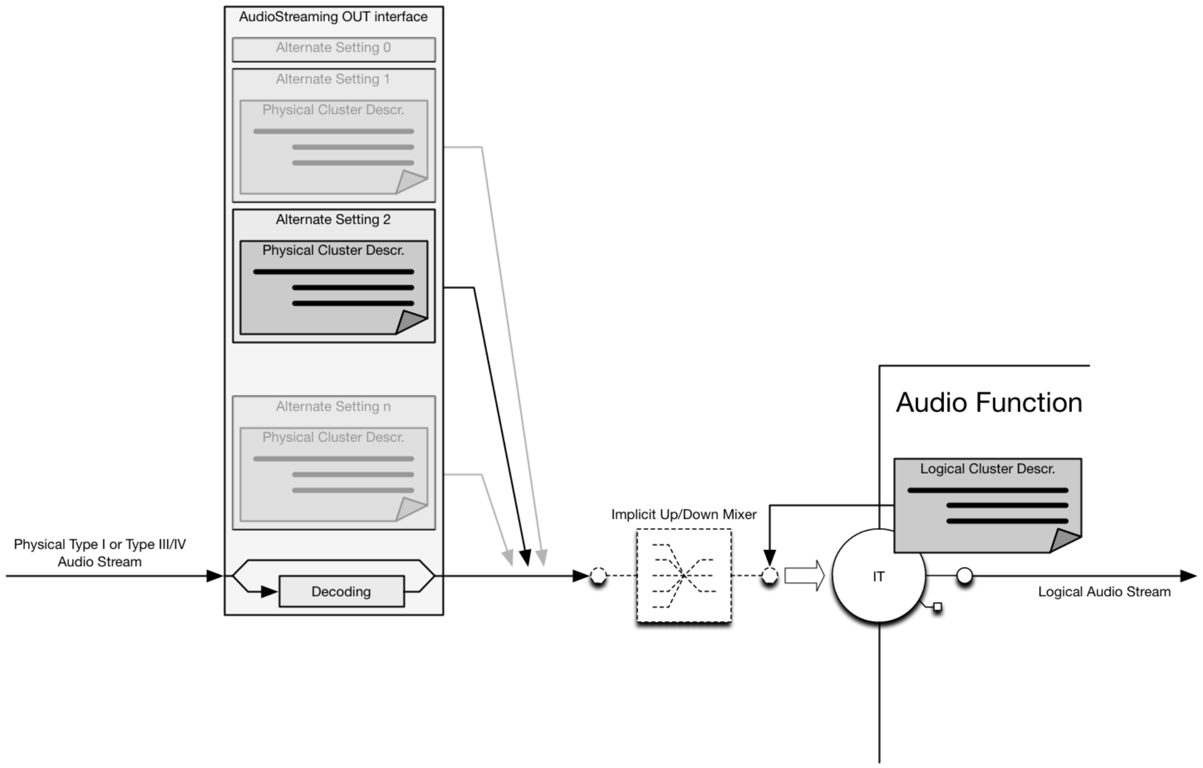
Finally, that Cluster then enters the Audio Function via the Output Pin of the Input Terminal.

This construct allows for a complete decoupling between the physical Cluster configuration as selected by the active Alternate Setting of the interface, and the logical Cluster configuration as described by the Input Terminal's Cluster descriptor. In other words, the logical Cluster configuration is independent of the chosen Alternate Setting of the AudioStreaming interface.

The process used to perform the Up/Down-mix operation is implementation-specific and not specified here.

Figure 4-4 illustrates the concept for AudioStreaming OUT interfaces.

Figure 4-4: Physical to Logical Cluster Mapping



4.4.1.2 AUDIOSTREAMING IN INTERFACE – OUTPUT TERMINAL

For Type I Audio streams, the physical Cluster descriptor referenced by the class-specific AudioStreaming interface descriptor of the active Alternate Setting of that AudioStreaming OUT interface describes the audio channels as they enter the interface.

For Type III/IV Audio streams, the physical Cluster describes the audio channels after they have been decoded into discrete audio channels.

An upstream Cluster descriptor describes the logical Cluster entering the Input Pin of the Output Terminal.

The Cluster then enters an implicit Up/Down-mix process to produce an output Cluster as described by the physical Cluster descriptor referenced by the class-specific AudioStreaming interface descriptor of the active Alternate Setting of the interface. The audio samples (Audio Subslots) shall be ordered in the exact same order as they are listed in the physical Cluster descriptor.

For Type I Audio streams, that Cluster then leaves the AudioStreaming interface.

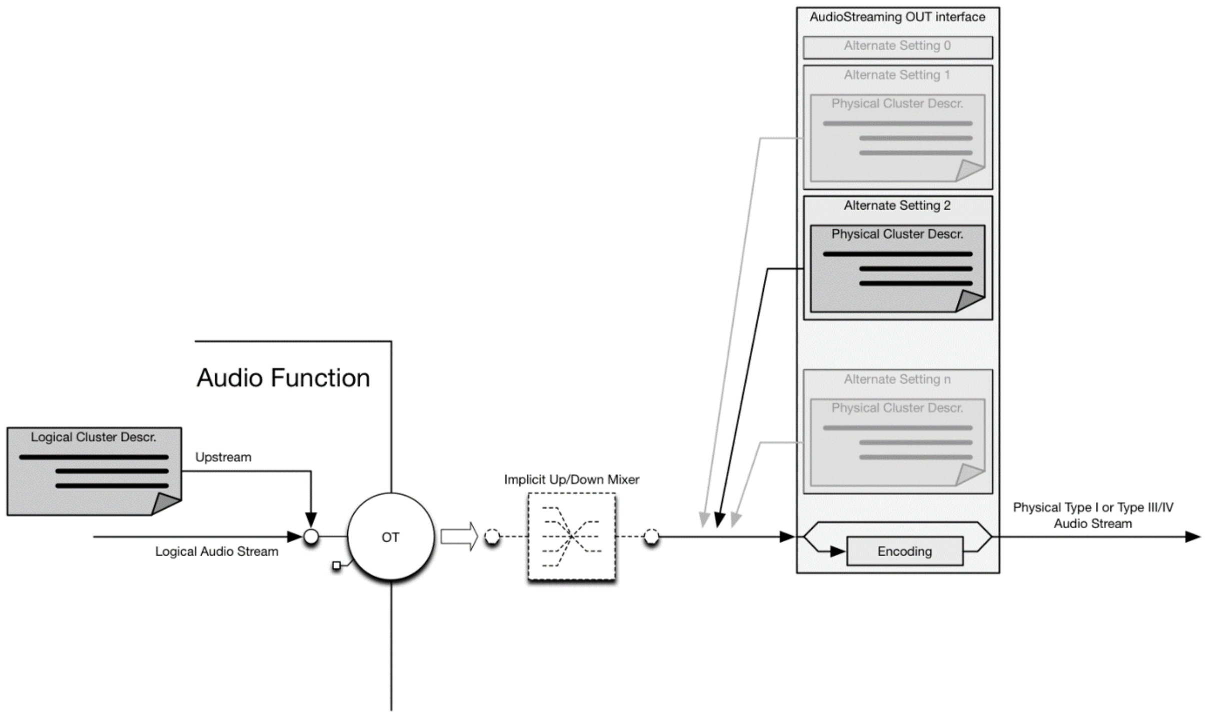
For Type III/IV Audio streams, the Cluster is encoded before leaving the AudioStreaming interface.

This construct allows for a complete decoupling between the logical Cluster configuration as described by the upstream Cluster descriptor and the physical Cluster configuration as selected by the active Alternate Setting of the interface. In other words, the physical Cluster configuration is independent of the upstream Cluster configuration and solely determined by the chosen Alternate Setting of the AudioStreaming interface.

The process used to perform the Up/Down-mix operation is implementation-specific and not specified here.

Figure 4-4 illustrates the concept for AudioStreaming IN interfaces.

Figure 4-5: Logical to Physical Cluster Mapping



4.5 AUDIOCONTROL INTERFACE DESCRIPTORS

The AudioControl (AC) interface descriptors contain all relevant information to fully characterize the corresponding Audio Function. The standard interface descriptor characterizes the interface itself, whereas the class-specific interface descriptor provides pertinent information concerning the internals of the Audio Function. It specifies revision level information and lists the capabilities of each Unit and Terminal.

4.5.1 STANDARD AC INTERFACE DESCRIPTOR

The standard AC interface descriptor is identical to the standard interface descriptor defined in the *USB Specification*, except that some fields have now dedicated values.

Table 4-14: Standard AC Interface Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 9.
1	bDescriptorType	1	Constant	INTERFACE descriptor type.
2	bInterfaceNumber	1	Number	Number of interface. A zero-based value identifying the index in the array of concurrent interfaces supported by this configuration.
3	bAlternateSetting	1	Number	Value used to select an Alternate Setting for the interface identified in the prior field. Shall be set to 0.
4	bNumEndpoints	1	Number	Number of endpoints used by this interface (excluding endpoint 0). This number is either 0 or 1 if the optional interrupt endpoint is present.
5	bInterfaceClass	1	Class	AUDIO. Audio Interface Class code (assigned by the USB). See Appendix A.4, "Audio Interface Class Code."
6	bInterfaceSubClass	1	Subclass	AUDIOCONTROL. Audio Interface Subclass code. Assigned by this specification. See Appendix A.5, "Audio Interface Subclass Codes."
7	bInterfaceProtocol	1	Protocol	IP_VERSION_03_00 Interface Protocol code. Indicates the current version of the specification. See Appendix A.6, "Audio Interface Protocol Codes"
8	iInterface	1	Index	Index of a String descriptor that describes this interface.

4.5.2 CLASS-SPECIFIC AC INTERFACE DESCRIPTOR

The class-specific AC interface descriptor is a concatenation of all the descriptors that are used to fully describe the Audio Function, i.e. all Clock descriptors (CDs), all Unit descriptors (UDs), all Terminal descriptors (TDs), and all Power Domain descriptors (PDD).

The total length of the class-specific AC interface descriptor depends on the number of Clock Entities, Units, Terminals, and Power Domains in the Audio Function. Therefore, the descriptor starts with a header that reflects the total length in bytes of the entire class-specific AC interface descriptor in the **wTotalLength** field. The **bCategory** field contains a constant that indicates what the primary use of this Audio Function is as intended by the manufacturer.

The order in which the Clock Entity, Unit, Terminal, and Power Domain descriptors are reported is not important because every descriptor can be identified through its **bDescriptorType** and **bDescriptorSubtype** field.

The following table defines the class-specific AC interface header descriptor.

Table 4-15: Class-Specific AC Interface Header Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 10.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	HEADER descriptor subtype.
3	bCategory	1	Constant	Constant, indicating the primary use of this Audio Function, as intended by the manufacturer. See Appendix A.7, "Audio Function Category Codes."
4	wTotalLength	2	Number	Total number of bytes returned for the class-specific AudioControl interface descriptor. Includes the combined length of this descriptor header and all Clock Source, Unit, Terminal, and Power Domain descriptors.
6	bmControls	4	Bitmap	D1..0: Latency Control. D31..2: Reserved.

The Clock Entity descriptors, Unit descriptors, Terminal descriptors, and Power Domain descriptors appear next, in any order. The layout of the Entity descriptors depends on the type of Clock Entity, Unit, or Terminal they represent. The Power Domain descriptor has a fixed layout. There are three types of Clock Entity descriptors, seven types of Unit descriptors, two types of Terminal descriptors, and a single type of Power Domain descriptor. The Entity descriptors are summarized in the following sections.

The first three fields are common for all descriptors. They contain the Descriptor Length, Descriptor Type, and Descriptor Subtype. Entity descriptors have an additional common field that contains the Clock Entity ID, Unit ID, Terminal ID, or Power Domain ID for the Entity.

Each Entity (Clock, Unit, Terminal, and Power Domain) within the Audio Function is assigned a unique identification number, the Clock Entity ID (CID), Unit ID (UID), Terminal ID (TID), or Power Domain ID (PID), contained in the **bClockID**, **bUnitID**, **bTerminalID, bPowerDomainID** field of the descriptor. The value 0x00 is reserved for undefined ID, effectively restricting the total number of addressable Entities in the Audio Function (Clock Entities, Units, Terminals, and Power Domains) to 255.

Besides uniquely identifying all addressable Entities in an Audio Function, the IDs (except for the Power Domain ID) also serve to describe the topology of the Audio Function; i.e. the **bSourceID** field of a Unit or Terminal descriptor indicates to which other Unit or Terminal this Unit or Terminal is connected. Likewise, the **bCSourceID** field in a Terminal descriptor indicates to which Clock Entity this Terminal is connected. Furthermore, the Entity IDs are also used to indicate to which Power Domain each Entity belongs.

4.5.2.1 INPUT TERMINAL DESCRIPTOR

The Input Terminal descriptor (ITD) provides information to the Host that is related to the functional aspects of the Input Terminal.

The Input Terminal is uniquely identified by the value in the **bTerminalID** field. No other Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Terminal.

The **wTerminalType** field provides pertinent information about the physical entity that the Input Terminal represents. This could be a USB OUT endpoint, an external Line In connection, a microphone, etc. A complete list of Terminal Type codes is provided in a separate document, *USB Audio Terminal Types* that is considered part of this specification.

The **bAssocTerminal** field is used to associate an Output Terminal to this Input Terminal, effectively implementing a bi-directional Terminal pair. If no association exists, the **bAssocTerminal** field shall be set to zero.

The Host software can treat the associated Terminals as being physically related. In many cases, one Terminal cannot exist without the other. A typical example of such a Terminal pair is an Input Terminal, which represents the microphone, and an Output Terminal, which represents the earpiece of a headset.

The **bCSourceID** contains a constant indicating to which Clock Entity the Clock Input Pin of this Input Terminal is connected.

The **wClusterDescrID** field contains the unique ID of the Cluster descriptor that characterizes the logical Cluster that leaves the Input Terminal over the single Output Pin (‘downstream’ connection). For a detailed description of the Cluster descriptor, see Section 4.3, “Cluster Descriptor”.

The **wExTerminalDescrID** field contains the unique ID of the Extended Terminal descriptor that is associated with this Terminal. For a detailed description of the Extended Terminal descriptor, see Section 4.5.2.3, “Extended Terminal Descriptor.”

The **wConnectorsDescrID** field contains the unique ID of the Connectors descriptor that is associated with this Terminal. For a detailed description of the Connectors descriptor, see Section 4.5.2.4, “Connectors Descriptor.”

An ID of a String descriptor is provided to further describe the Input Terminal.

The following table presents an outline of the Input Terminal descriptor.

Table 4-16: Input Terminal Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 20.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	INPUT_TERMINAL descriptor subtype.
3	bTerminalID	1	Number	Value uniquely identifying the Terminal within the Audio Function. This value is used in all requests to address this Terminal.
4	wTerminalType	2	Constant	Constant characterizing the type of Terminal. See *USB Audio Terminal Types*.
6	bAssocTerminal	1	Number	ID of the Output Terminal to which this Input Terminal is associated.
7	bCSourceID	1	Number	ID of the Clock Entity to which this Input Terminal is connected.
8	bmControls	4	Bitmap	D1..0: Insertion Control. D3..2: Overload Control. D5..4: Underflow Control. D7..6: Overflow Control. D31..8: Reserved.

Offset	Field	Size	Value	Description
12	wClusterDescrID	2	Number	ID of the Cluster descriptor for this Input Terminal.
14	wExTerminalDescrID	2	Number	ID of the Extended Terminal descriptor for this Input Terminal. Shall be set to zero if no Extended Terminal descriptor is present.
16	wConnectorsDescrID	2	Number	ID of the Connectors descriptor for this Input Terminal. Shall be set to zero if no Connectors descriptor is present.
18	wTerminalDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing the Input Terminal.

4.5.2.2 OUTPUT TERMINAL DESCRIPTOR

The Output Terminal descriptor (OTD) provides information to the Host that is related to the functional aspects of the Output Terminal.

The Output Terminal is uniquely identified by the value in the **bTerminalID** field. No other Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Terminal.

The **wTerminalType** field provides pertinent information about the physical entity the Output Terminal represents. This could be a USB IN endpoint, an external Line Out connection, a speaker system etc. A complete list of Terminal Type codes is provided in a separate document, *USB Audio Terminal Types* that is considered part of this specification.

The **bAssocTerminal** field is used to associate an Input Terminal to this Output Terminal, effectively implementing a bi-directional Terminal pair. If no association exists, the **bAssocTerminal** field shall be set to zero.

The Host software can treat the associated Terminals as being physically related. In many cases, one Terminal cannot exist without the other. A typical example of such a Terminal pair is an Input Terminal, which represents the microphone, and an Output Terminal, which represents the earpiece of a headset.

The **bSourceID** field is used to describe the connectivity for this Terminal. It contains the ID of the Unit or Terminal to which this Output Terminal is connected via its Input Pin. The Cluster descriptor, describing the logical channels entering the Output Terminal is not repeated here. It is up to the Host software to trace the connection ‘upstream’ to locate the Cluster descriptor pertaining to this Cluster.

The **bCSourceID** contains a constant indicating to which Clock Entity the Clock Input Pin of this Output Terminal is connected.

The **wExTerminalDescrID** field contains the unique ID of the Extended Terminal descriptor that is associated with this Terminal. For a detailed description of the Extended Terminal descriptor, see Section 4.5.2.3, “Extended Terminal Descriptor.”

The **wConnectorsDescrID** field contains the unique ID of the Connectors descriptor that is associated with this Terminal. For a detailed description of the Connectors descriptor, see Section 4.5.2.4, “Connectors Descriptor.”

An ID of a String descriptor is provided to further describe the Output Terminal.

The following table presents an outline of the Output Terminal descriptor.

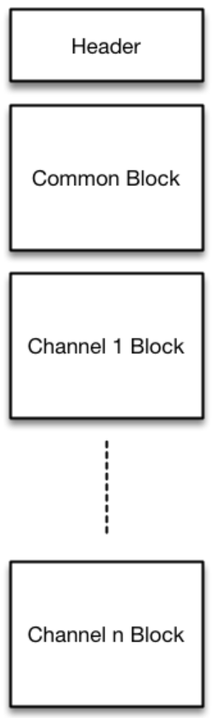
Table 4-17: Output Terminal Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 19.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	OUTPUT_TERMINAL descriptor subtype.
3	bTerminalID	1	Number	Value uniquely identifying the Terminal within the Audio Function. This value is used in all requests to address this Terminal.
4	wTerminalType	2	Constant	Constant characterizing the type of Terminal. See *USB Audio Terminal Types*.
6	bAssocTerminal	1	Number	Value identifying the Input Terminal to which this Output Terminal is associated.
7	bSourceID	1	Number	ID of the Unit or Terminal to which this Terminal is connected.
8	bCSourceID	1	Number	ID of the Clock Entity to which this Output Terminal is connected.
9	bmControls	4	Bitmap	D1..0: Insertion Control. D3..2: Overload Control. D5..4: Underflow Control. D7..6: Overflow Control. D31..8: Reserved.
13	wExTerminalDescrID	2	Number	ID of the Extended Terminal descriptor for this Output Terminal. Shall be set to zero if no Extended Terminal descriptor is present.
15	wConnectorsDescrID	2	Number	ID of the Connectors descriptor for this Output Terminal. Shall be set to zero if no Connectors descriptor is present.
17	wTerminalDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing the Output Terminal.

4.5.2.3 EXTENDED TERMINAL DESCRIPTOR

The Extended Terminal descriptor optionally returns additional physical information about the channels that enter or leave the Terminal. It always uses the High Capability representation.

Figure 4-6: Extended Terminal Descriptor



The descriptor consists of a fixed Header, followed by an optional Common Block, followed by as many Channel Blocks as there are channels in the Cluster of the Terminal.

4.5.2.3.1 EXTENDED TERMINAL DESCRIPTOR HEADER

The Header starts with the **wLength** field that contains the total number of bytes in the entire descriptor.

The **wDescriptorID** field contains an ID number that uniquely identifies the descriptor within the Audio Function. The value zero is reserved and shall not be used as a valid Extended Terminal descriptor ID.

The **bNrChannels** field indicates the number of audio channels present in the logical Cluster of the Terminal.

Table 4-18: Extended Terminal Descriptor Header

Offset	Field	Size	Value	Description
0	wLength	2	Number	Total length of the Cluster descriptor, in bytes.
2	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
3	bDescriptorSubtype	1	Constant	EXTENDED_TERMINAL descriptor subtype.
4	wDescriptorID	2	Number	Unique ID of this Extended Terminal descriptor.
6	bNrChannels	1	Number	Number of channels present in the Cluster of the Terminal: n.

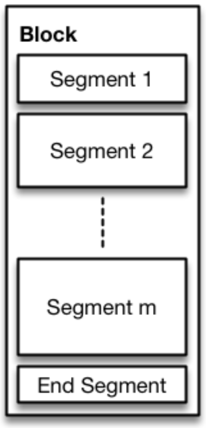
4.5.2.3.2 EXTENDED TERMINAL DESCRIPTOR BLOCK

The Extended Terminal descriptor Header is followed by a number of Extended Terminal descriptor Blocks. There is an optional Common Block, followed by as many Channel Blocks as there are channels in the Cluster of the Terminal. Each Block consists of a number of Segments.

The Common Block consists of Segments that contain relevant information about characteristics of the Terminal as a whole.

A Channel Block consists of Segments that contain relevant information about that channel's physical characteristics. All Segments are optional. It is highly recommended that the same layout is used for each Channel Block, i.e. the same Segments appear in the same order in each Channel Block. Figure 4-7 further illustrates the above concepts.

Figure 4-7: Extended Terminal Channel Block



4.5.2.3.3 SEGMENTS

There are two types of Segments. Common Block Segments contain pertinent information about the Terminal as a whole. Channel Block Segments contain pertinent information about certain aspects of a particular channel in the Cluster of the Terminal. Both Segment types share the same layout.

The **bSegmentType** field describes the Segment Type (Common Block or Channel Block) and also the type of content contained in the Segment.

The layout for a Segment is always as follows:

Table 4-19: Cluster Descriptor Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Segment, in bytes: 3+n.
2	bSegmentType	1	Constant	Describes the Segment Type and the type of content in the Segment.
3	Segment-specific	n		Segment-specific content.

4.5.2.3.3.1 END SEGMENT

Each Block is terminated by an End Segment. The End Segment marks the end of the variable length Block. The End Segment does not have a Segment-specific section and is structured as follows:

Table 4-20: End Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the End Segment, in bytes: 3.
2	bSegmentType	1	Constant	END_SEGMENT.

4.5.2.3.3.2 COMMON BLOCK SEGMENTS

The following Common Block Segment types are defined:

- Vendor-specific

Values for the Common Block Segment types can be found in Appendix A.18, “Extended Terminal Segment Types.”

4.5.2.3.3.2.1 VENDOR-DEFINED SEGMENT

Vendors are allowed to add vendor-defined Segments to the Common Block to convey additional, proprietary Terminal information. The vendor-defined Segment shall use the following layout:

Table 4-21: Vendor-defined Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the End Segment, in bytes: 3+n.
2	bSegmentType	1	Constant	TERMINAL_VENDOR_DEFINED.
3	Vendor-defined	n	N/A	Vendor-defined extension of the Segment.

4.5.2.3.3.3 CHANNEL BLOCK SEGMENTS

The following Channel Block Segment types are defined:

- Bandwidth
- Magnitude Response
- Magnitude/Phase Response
- Position
- Vendor-specific

Values for the Channel Block Segment types can be found in Appendix A.18, “Extended Terminal Segment Types.”

4.5.2.3.3.3.1 BANDWIDTH SEGMENT

The Bandwidth Segment contains basic information about the audio bandwidth in the channel. The bandwidth is specified by providing the lower and upper -3 dB frequency points, in Hz, of the available band in the **dMinFreq** and **dMaxFreq** fields.

Table 4-22: Bandwidth Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Bandwidth Segment, in bytes: 11.
2	bSegmentType	1	Constant	CHANNEL_BANDWIDTH.
3	dMinFreq	4	Number	Lower -3 dB frequency point.
7	dMaxFreq	4	Number	Upper -3 dB frequency point.

4.5.2.3.3.3.2 MAGNITUDE RESPONSE

The Magnitude Response Segment contains detailed information about the magnitude of the transfer function (frequency response) of the channel. The magnitude is specified as an array of [frequency point, magnitude] pairs. The frequency values are specified in Hz and the magnitude values can range from +127.9961 dB (0x7FFF) down to -127.9961 dB (0x8001) in steps of 1/256 dB or 0.00390625 dB (0x0001). In addition, code 0x8000, representing silence (i.e., -∞ dB), may be used as well.

Table 4-23: Magnitude Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Magnitude Response Segment, in bytes: 3+n*6.
2	bSegmentType	1	Constant	CHANNEL_MAGNITUDE_RESPONSE.
3	dFreq(1)	4	Number	First frequency point.
7	wMagnitude(1)	2	Number	First magnitude value.
…	…	…	…	
3+(n-1)*6	dFreq(n)	4	Number	Last frequency point.
7+(n-1)*6	wMagnitude(n)	2	Number	Last magnitude value.

4.5.2.3.3.3.3 MAGNITUDE/PHASE RESPONSE

The Magnitude/Phase Response Segment contains detailed information about the magnitude and phase of the transfer function (frequency response) of the channel. The magnitude/phase is specified as an array of [frequency point, magnitude, phase] triplets. The frequency values are specified in Hz. The magnitude values can range from +127.9961 dB (0x7FFF) down to -127.9961 dB (0x8001) in steps of 1/256 dB or 0.00390625 dB (0x0001). In addition, code 0x8000, representing silence (i.e., -∞ dB), may be used as well. The phase values can range from +0.99996948242 * π down to -π (0x8000) in steps of 1/32768 * π (0x0001).

Table 4-24: Magnitude/Phase Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Magnitude Response Segment, in bytes: 3+n*8.
2	bSegmentType	1	Constant	CHANNEL_MAGNITUDE/PHASE_RESPONSE.
3	dFreq(1)	4	Number	First frequency point.
7	wMagnitude(1)	2	Number	First magnitude value.
9	wPhase(1)	2	Number	First phase value.

Offset	Field	Size	Value	Description
…	…	…	…	
3+(n-1)*8	dFreq(n)	4	Number	Last frequency point.
7+(n-1)*8	wMagnitude(n)	2	Number	Last magnitude value.
9+(n-1)*8	wPhase(n)	2	Number	Last phase value.

4.5.2.3.3.3.4 POSITION_XYZ SEGMENT

The Position_XYZ Segment contains the (X, Y, Z) Cartesian coordinates of the source or sink associated with the channel. The X, Y, and Z values are expressed in micrometers (µm) and are relative to an unspecified origin at (0, 0, 0). The Audio Function may have out-of-band means to indicate to the Host where the actual origin of the coordinate system is located on the device. The value 0xFFFFFFFF has special meaning. It is used to indicate that the coordinate in a particular dimension is unknown.

Table 4-25: Position Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Segment, in bytes: 15.
2	bSegmentType	1	Constant	CHANNEL_POSITION_XYZ.
3	dX	4	Number	X-coordinate of the audio source or sink, associated with the channel.
7	dY	4	Number	Y-coordinate of the audio source or sink, associated with the channel.
11	dZ	4	Number	Z-coordinate of the audio source or sink, associated with the channel.

4.5.2.3.3.3.5 POSITION_RΘΦ SEGMENT

The Position_ RΘΦ Segment contains the (R, Θ, Φ) spherical coordinates of the source or sink associated with the channel. The R, Θ, and Φ values are expressed in µm, µrad and µrad respectively, and are relative to an unspecified origin at (0, 0, 0). The Audio Function may have out-of-band means to indicate to the Host where the actual origin of the coordinate system is located on the device. The value 0xFFFFFFFF has special meaning. It is used to indicate that the coordinate in a particular dimension is unknown.

Table 4-26: Position Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the Segment, in bytes: 15.
2	bSegmentType	1	Constant	CHANNEL_POSITION_ RΘΦ.
3	dR	4	Number	R-coordinate of the audio source or sink, associated with the channel.
7	dΘ	4	Number	Θ-coordinate of the audio source or sink, associated with the channel.
11	dΦ	4	Number	Φ-coordinate of the audio source or sink, associated with the channel.

4.5.2.3.3.3.6 VENDOR-DEFINED SEGMENT

Vendors are allowed to add vendor-defined Segments to the Channel Block to convey additional, proprietary channel information. The vendor-defined Segment shall use the following layout:

Table 4-27: Vendor-defined Segment

Offset	Field	Size	Value	Description
0	wLength	2	Number	Length of the End Segment, in bytes: 3+n.
2	bSegmentType	1	Constant	CHANNEL_VENDOR_DEFINED.
3	Vendor-defined	n	N/A	Vendor-defined extension of the Segment.

4.5.2.4 CONNECTORS DESCRIPTOR

The optional Connectors descriptor returns information about the physical attributes of all Connectors, associated with either an Input or an Output Terminal. The Connectors descriptor always uses the High Capability representation.

The **wDescriptorID** field contains an ID number that uniquely identifies the descriptor within the Audio Function. The value zero is reserved and shall not be used as a valid Connectors descriptor ID.

The **bNrConnectors** field indicates how many distinct physical Connectors are associated with this Terminal. Connectors are numbered from 1 to the total number of Connectors, associated with this Terminal. For each Connector, the descriptor contains a group of fields that further describe the Connector.

The **baConID(i)** field contains a unique identifier for Connector(i). The primary use for this is to indicate that the same Connector is associated with multiple Terminals. For example, one headset Connector incorporates the signals for the stereo headphone of the headset and also the signal for the mono microphone. This Connector would therefore be part of the Output Terminal that represents the stereo headphone and also be part of the Input Terminal that represents the microphone. This Connector would then be listed in both the Input Terminal and Output Terminal Connectors descriptor, using the same **baConID(i)** value in both descriptors to indicate the binding. Note that the **baConID(i)** value is never used as an identifier for an addressable Entity and therefore, **baConID(i)** values may overlap with Entity ID values.

The **waClusterDescrID(i)** contains the unique ID of a physical Cluster descriptor, describing the actual physical channels that are carried over Connector(i). Note that in most cases, the union of all the physical Cluster descriptors, referenced by the Connectors descriptor, will be identical to the logical Cluster descriptor, associated with the Pin of the Terminal. It is allowed for channels to appear in more than one physical Cluster descriptor to accommodate the situation where the same physical channels are available on different Connectors. For example, the same channel may be available on a balanced XLR connector as well as on an unbalanced BNC connector. For a detailed description of the Cluster descriptor, see Section 4.3, “Cluster Descriptor”.

The **baConType(i)** field contains a value that identifies the physical appearance of Connector(i). The constant definitions for the **baConTyp(i)** field can be found in Appendix A.24, “Connector Types”.

The **bmaConAttributes(i)** field contains a bitmap that identifies the gender of Connector(i) (D1..0) and also indicates whether this Control is able to detect insertion and removal. If this bit is set for Connector(i), then bit D_{i-1} in the bitmap returned by the Insertion Control request indicates whether this Connector is inserted ($D_{i-1} = 1$) or not ($D_{i-1} = 0$). If this bit is clear for Connector(i), then bit D_{i-1} in the bitmap returned by the Insertion Control request has no meaning and shall be ignored.

The **waConDescrStr(i)** field is the ID of a String descriptor that contains a human-readable identifier for Connector(i). It is recommended that this string makes it easy to uniquely identify the Connector on the device's chassis. This could be achieved by explicitly labeling the Connector on the device's enclosure with the **waConDescrStr(i)** string content.

The **daConColor(i)** field contains either 0x00 in the upper byte and the RGB-coded color of Connector(i) in the lower 3 bytes or 0x01 in the upper byte and 0x000000 in the lower 3 bytes to indicate color unspecified.

Table 4-28: Connectors Descriptor

Offset	Field	Size	Value	Description
0	wLength	2	Number	Size of this descriptor, in bytes: 7+n*11.
2	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
3	bDescriptorSubtype	1	Constant	CONNECTORS descriptor subtype.
4	wDescriptorID	2	Number	Unique ID of this Connectors descriptor.
6	bNrConnectors	1	Number	The number of Connectors associated with this Terminal: n
7	baConID(1)	1	Number	ID for Connector(1). Can be used to indicate that the same Connector is associated with multiple Terminals.
8	waClusterDescrID(1)	2	Number	ID of the physical Cluster descriptor transported over Connector(1).
10	baConType(1)	1	Constant	The Connector Type of Connector(1).
11	bmaConAttributes(1)	1	Bitmap	D1..0: Gender: 00: Gender Neutral. 01: Male. 10: Female. 11: Reserved. D2: Insertion/Removal Reporting: 0: No. 1: Yes. D7..3: Reserved.
12	waConDescrStr(1)	2	wStrDescrID	ID of a String descriptor that provides a physical description of Connector(1).
14	daConColor(1)	4	Number	The Connector color of Connector(1).
…	…	…	…	…
7+(n-1)*11	baConID(n)			Unique identifier for Connector(n). Can be used to indicate that the same Connector is associated with multiple Terminals.
8+(n-1)*11	waClusterDescrID(n)	2	Number	The unique ID of the physical Cluster descriptor transported over Connector(n).
10+(n-1)*11	baConType(n)	1	Constant	The Connector Type of Connector(n).

11+(n-1)*11	bmaConAttributes(n)	1	Bitmap	D1..0: Gender: 00: Gender Neutral. 01: Male. 10: Female. 11: Reserved. D2: Insertion/Removal Reporting: 0: No. 1: Yes. D7..3: Reserved.
12+(n-1)*11	waConDescrStr(n)	2	wStrDescrID	ID of a String descriptor that provides a physical description of Connector(n).
14+(n-1)*11	daConColor(n)	4	Number	The Connector color of Connector(n).

4.5.2.5 MIXER UNIT DESCRIPTOR

The Mixer Unit is uniquely identified by the value in the **bUnitID** field of the Mixer Unit descriptor (MUD). No other Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Mixer Unit.

The **bNrInPins** field contains the number of Input Pins (*p*) of the Mixer Unit. This evidently equals the number of Clusters that enter the Mixer Unit. The connectivity of the Input Pins is described via the **baSourceID()** array, containing p elements. The index *I* into the array is one-based and directly related to the Input Pin numbers. **baSourceID(i)** contains the ID of the Unit or Terminal to which Input Pin I is connected. The Cluster descriptors, describing the logical channels entering the Mixer Unit are not repeated here. It is up to the Host software to trace the connections 'upstream' to locate the Cluster descriptors pertaining to the Clusters.

Because a Mixer Unit can redefine the spatial locations of the logical output channels, contained in its output Cluster, there is a need for a Mixer output Cluster descriptor.

The **wClusterDescrID** contains the unique ID of the Cluster descriptor that characterizes the Cluster that leaves the Mixer Unit over the single Output Pin ('downstream' connection). For a detailed description of the Cluster descriptor, see Section 4.3, "Cluster Descriptor".

As mentioned before, every input channel can virtually be mixed into all of the output channels. If *n* is the total number of logical input channels, contained in all the Clusters that are entering the Mixer Unit:

$$n = \sum_{i=1}^{number\ of\ clusters} (number\ of\ logical\ channels\ in\ cluster\ i)$$

and *m* is the number of logical output channels, then there are (*n* · *m*) Mixer Controls in the Mixer Unit, some of which may not be programmable.

Note: (*n* · *m*) shall be limited to 256.

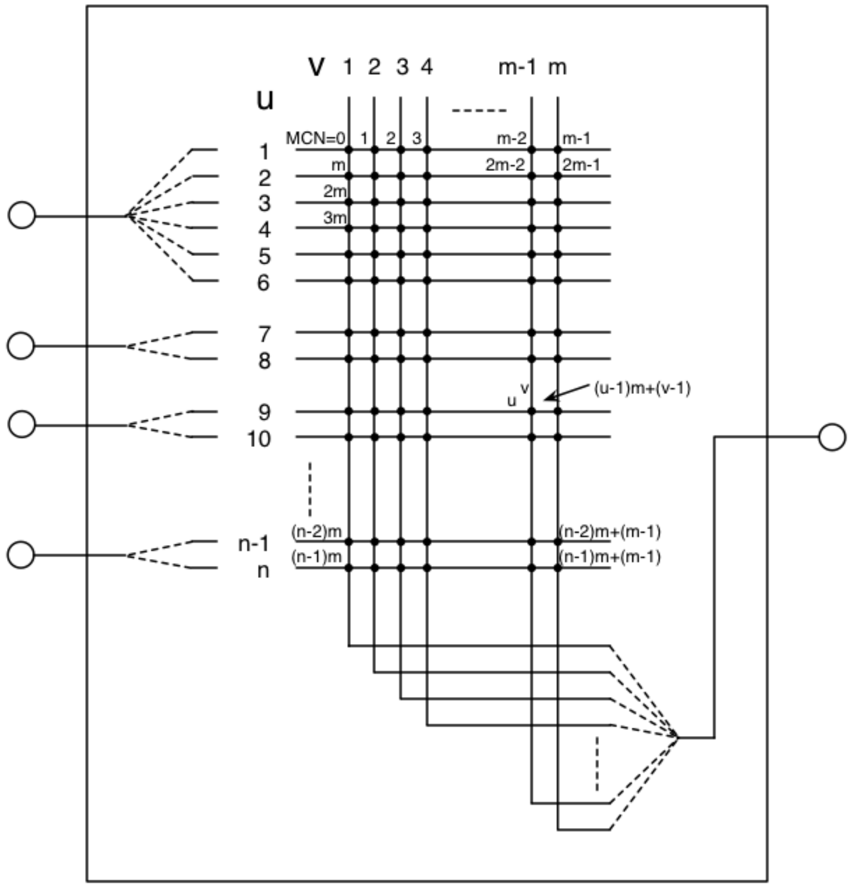
The Mixer Unit descriptor reports which Controls are programmable in the **bmMixerControls** bitmap field. This bitmap shall be interpreted as a two-dimensional bit array that has a row for each logical input channel and a column for each logical output channel. If a bit at position [*u*, *v*] is set to one, this means that the Mixer Unit contains a programmable Mixer Control that connects input channel *u* to output channel *v*. If bit [*u*, *v*] is set to zero, this indicates that the connection between input channel *u* and output channel *v* is non-programmable. The valid range for *u* is from one to *n*. The valid range for *v* is from one to *m*.

Each Mixer Control is assigned a unique Mixer Control Number (MCN). This number is used to address a particular Mixer Control in a Get/Set Mixer Control request. The MCN is calculated as follows:

$$MCN = (u-1).m + (v-1)$$

The following figure presents a more graphical explanation.

Figure 4-8: Mixer internals



The current setting of the Mixer Control (both programmable and fixed) at any position in the matrix can always be retrieved through the appropriate request. Therefore, the Mixer Unit shall always implement the Get request with the CUR attribute for each node in the matrix. See Section 5.2.1.7, “Mixer Unit Control Request” for further details.

The **bmMixerControls** field stores the bit array row after row where the MSb of the first (highest) byte corresponds to the connection between input channel 1 and output channel 1. If ($n \cdot m$) is not an integer multiple of 8, the bit array is padded with zeroes until an integer number of bytes is occupied. The number of bytes used to store the bit array, *N*, can be calculated as follows:

```
IF ((n * m) MOD 8) <> 0 THEN
  N = ((n * m) DIV 8) + 1
ELSE
  N = ((n * m) DIV 8)
```

An ID of a String descriptor is provided to further describe the Mixer Unit.

The following table details the structure of the Mixer Unit descriptor.

Table 4-29: Mixer Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 13+p+N.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	MIXER_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	bNrInPins	1	Number	Number of Input Pins of this Unit: p
5	baSourceID(1)	1	Number	ID of the Unit or Terminal to which the first Input Pin of this Mixer Unit is connected.
…	…	…	…	…
5+(p-1)	baSourceID (p)	1	Number	ID of the Unit or Terminal to which the last Input Pin of this Mixer Unit is connected.
5+p	wClusterDescrID	2	Number	ID of the Cluster descriptor for this Mixer Unit.
7+p	bmMixerControls	N	Number	Bitmap indicating which Mixer Controls are programmable.
7+p+N	bmControls	4	Bitmap	D1..0: Underflow Control. D3..2: Overflow Control. D31..4: Reserved.
11+p+N	wMixerDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing the Mixer Unit.

4.5.2.6 SELECTOR UNIT DESCRIPTOR

The Selector Unit is uniquely identified by the value in the **bUnitID** field of the Selector Unit descriptor (SUD). No other Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Selector Unit.

The **bNrInPins** field contains the number of Input Pins (*p*) of the Selector Unit. The connectivity of the Input Pins is described via the **baSourceID()** array that contains *p* elements. The index *I* into the array is one-based and directly related to the Input Pin numbers. **baSourceID(i)** contains the ID of the Unit or Terminal to which Input Pin *I* is connected.

The Cluster descriptors, describing the logical channels that enter the Selector Unit are not repeated here. In order for a Selector Unit to be legally connected, *all* of the Clusters that enter the Selector Unit shall have exactly the same Cluster descriptor content.

An ID of a String descriptor is provided to further describe the Selector Unit.

The following table details the structure of the Selector Unit descriptor.

Table 4-30: Selector Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 11+p.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	SELECTOR_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	bNrInPins	1	Number	Number of Input Pins of this Unit: p
5	baSourceID(1)	1	Number	ID of the Unit or Terminal to which the first Input Pin of this Selector Unit is connected.
…	…	…	…	…
5+(p-1)	baSourceID (p)	1	Number	ID of the Unit or Terminal to which the last Input Pin of this Selector Unit is connected.
5+p	bmControls	4	Bitmap	D1..0: Selector Control. D31..2: Reserved.
9+p	wSelectorDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing the Selector Unit.

4.5.2.7 FEATURE UNIT DESCRIPTOR

The Feature Unit is uniquely identified by the value in the **bUnitID** field of the Feature Unit descriptor (FUD). No other Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Feature Unit.

The **bSourceID** field is used to describe the connectivity for this Feature Unit. It contains the ID of the Unit or Terminal to which this Feature Unit is connected via its Input Pin. The Cluster descriptor, describing the logical channels entering the Feature Unit is not repeated here. It is up to the Host software to trace the connection 'upstream' to locate the Cluster descriptor pertaining to this Cluster.

An ID of a String descriptor is provided to further describe the Feature Unit.

The layout of the Feature Unit descriptor is detailed in the following table.

Table 4-31: Feature Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 7+(ch+1)*4
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	FEATURE_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	bSourceID	1	Number	ID of the Unit or Terminal to which this Feature Unit is connected.

Offset	Field	Size	Value	Description
5	bmaControls(0)	4	Bitmap	The Controls bitmap for master channel 0: D1..0: Mute Control. D3..2: Volume Control. D5..4: Bass Control. D7..6: Mid Control. D9..8: Treble Control. D11..10: Graphic Equalizer Control. D13..12: Automatic Gain Control. D15..14: Delay Control. D17..16: Bass Boost Control. D19..18: Loudness Control. D21..20: Input Gain Control. D23..22: Input Gain Pad Control. D25..24: Phase Inverter Control. D27..26: Underflow Control. D29..28: Overfow Control. D31..30: Reserved.
5+(1*4)	bmaControls(1)	4	Bitmap	The Controls bitmap for logical channel 1.
…	…	…	…	…
5+(ch*4)	bmaControls(ch)	4	Bitmap	The Controls bitmap for logical channel ch.
5+(ch+1)*4	wFeatureDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Feature Unit.

4.5.2.8 SAMPLING RATE CONVERTER UNIT DESCRIPTOR

The Sampling Rate Converter Unit descriptor (RUD) provides information to the Host that is related to the functional aspects of the SRC Unit.

The SRC Unit is uniquely identified by the value in the **bUnitID** field. No other Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Feature Unit.

The **bSourceID** field is used to describe the connectivity for this SRC Unit. It contains the ID of the Unit or Terminal to which this SRC Unit is connected via its Input Pin. The Cluster descriptor, describing the logical channels entering the SRC Unit is not repeated here. It is up to the Host software to trace the connection 'upstream' to locate the Cluster descriptor pertaining to this Cluster.

The **bCSourceInID** contains a constant indicating to which Clock Entity the Clock Input Pin associated with the audio Input Pin is connected.

The **bCSourceOutID** contains a constant indicating to which Clock Entity the Clock Input Pin associated with the audio Output Pin is connected.

An ID of a String descriptor is provided to further describe the SRC Unit.

The following table presents an outline of the SRC Unit descriptor.

Table 4-32: Sampling Rate Converter Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 9.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	SAMPLE_RATE_CONVERTER descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	bSourceID	1	Number	ID of the Unit or Terminal to which this SRC Unit is connected.
5	bCSourceInID	1	Number	ID of the Clock Entity to which this SRC Unit input section is connected.
6	bCSourceOutID	1	Number	ID of the Clock Entity to which this SRC Unit output section is connected.
7	wSRCDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing the SRC Unit.

4.5.2.9 EFFECT UNIT DESCRIPTOR

The Effect Unit is uniquely identified by the value in the **bUnitID** field of the Effect Unit descriptor (EUD). No other Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Effect Unit.

The **wEffectType** field contains a value that fully identifies the Effect Unit. For a list of all supported Effect Unit Types, see Appendix A.19, “Effect Unit Effect Types.”

The **bSourceID** field is used to describe the connectivity for this Effect Unit. It contains the ID of the Unit or Terminal to which this Effect Unit is connected via its Input Pin. The Cluster descriptor, describing the logical channels entering the Effect Unit is not repeated here. It is up to the Host software to trace the connection ‘upstream’ to locate the Cluster descriptor pertaining to this Cluster.

bmaControls() is a (ch+1)-element array of 4-byte bitmaps, each following the same semantics as the **bmControls** field in other descriptors.

An ID of a String descriptor is provided to further describe the Effect Unit.

The following table outlines the Effect Unit descriptor.

Table 4-33: Effect Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 17+(ch*4).
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	EFFECT_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.

Offset	Field	Size	Value	Description
4	wEffectType	2	Constant	Constant identifying the type of effect this Unit is performing.
6	bSourceID	1	Number	ID of the Unit or Terminal to which this Effect Unit is connected.
7	bmaControls(0)	4	Bitmap	The Controls bitmap for master channel 0: D31..0: Effect-specific allocation.
11	bmaControls(1)	4	Bitmap	The Controls bitmap for channel 1: D31..0: Effect-specific allocation.
…	…	…	…	…
11+(ch*4)	bmaControls(ch)	4	Bitmap	The Controls bitmap for channel ch: D31..0: Effect-specific allocation.
15+(ch*4)	wEffectsDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Effect Unit.

4.5.2.9.1 PARAMETRIC EQUALIZER SECTION EFFECT UNIT DESCRIPTOR

The **wEffectType** field of the common Effect Unit descriptor contains the value PARAM_EQ_SECTION_EFFECT. (See Appendix A.19, "Effect Unit Effect Types."

The following table outlines the PEQS Effect Unit descriptor. It is identical to the common Effect Unit descriptor, except for some field values. It is repeated here for clarity.

The Enable Control shall always be present and implemented as Read-Write. It is therefore not represented in the **bmControls** field.

Table 4-34: Parametric Equalizer Section Effect Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 17+(ch*4)
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	EFFECT_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	wEffectType	2	Constant	PARAM_EQ_SECTION_EFFECT effect type.
6	bSourceID	1	Number	ID of the Unit or Terminal to which this Effect Unit is connected.
7	bmaControls(0)	4	Bitmap	The Controls bitmap for master channel 0: D1..0: Center Frequency Control. D3..2: Q Factor Control. D5..4: Gain Control. D7..6: Underflow Control. D9..8: Overflow Control. D31..10: Reserved.
11	bmaControls(1)	4	Bitmap	The Controls bitmap for logical channel 1.

Offset	Field	Size	Value	Description
…	…	…	…	…
11+(ch*4)	bmaControls(ch)	4	Bitmap	The Controls bitmap for logical channel ch.
15+(ch*4)	wEffectsDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Effect Unit.

4.5.2.9.2 REVERBERATION EFFECT UNIT DESCRIPTOR

The **wEffectType** field of the common Effect Unit descriptor contains the value REVERBERATION_EFFECT. (see Appendix A.19, "Effect Unit Effect Types."

The following table outlines the Reverberation Effect Unit descriptor. It is identical to the common Effect Unit descriptor, except for some field values. It is repeated here for clarity.

The Enable Control shall always be present and implemented as Read-Write. It is therefore not represented in the **bmControls** field.

Table 4-35: Reverberation Effect Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 17+(ch*4).
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	EFFECT_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	wEffectType	2	Constant	REVERBERATION_EFFECT effect type.
6	bSourceID	1	Number	ID of the Unit or Terminal to which this Effect Unit is connected.
7	bmaControls(0)	4	Bitmap	The Controls bitmap for master channel 0: D1..0: Type Control. D3..2: Level Control. D5..4: Time Control. D7..6: Delay Feedback Control. D9..8: Pre-Delay Control. D11..10: Density Control. D13..12: Hi-Freq Roll-Off Control. D15..14: Underflow Control. D17..16: Overflow Control. D31..18: Reserved.
11	bmaControls(1)	4	Bitmap	The Controls bitmap for logical channel 1.
…	…	…	…	…
11+(ch*4)	bmaControls(ch)	4	Bitmap	The Controls bitmap for logical channel ch.
15+(ch*4)	wEffectsDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Effect Unit.

4.5.2.9.3 MODULATION DELAY EFFECT UNIT DESCRIPTOR

The **wEffectType** field of the common Effect Unit descriptor contains the value MOD_DELAY_EFFECT. (see Appendix A.19, "Effect Unit Effect Types."

The following table outlines the Modulation Delay Effect Unit descriptor. It is identical to the common Effect Unit descriptor, except for some field values. It is repeated here for clarity.

Table 4-36: Modulation Delay Effect Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 17+(ch*4).
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	EFFECT_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	wEffectType	2	Constant	MOD_DELAY_EFFECT effect type.
6	bSourceID	1	Number	ID of the Unit or Terminal to which this Effect Unit is connected.
7	bmaControls(0)	4	Bitmap	The Controls bitmap for master channel 0: D1..0: Enable Control. D3..2: Balance Control. D5..4: Rate Control. D7..6: Depth Control. D9..8: Time Control. D11..10: Feedback Level Control. D13..12: Underflow Control. D15..14: Overflow Control. D31..16: Reserved.
11	bmaControls(1)	4	Bitmap	The Controls bitmap for logical channel 1.
…	…	…	…	…
11+(ch*4)	bmaControls(ch)	4	Bitmap	The Controls bitmap for logical channel ch.
15+(ch*4)	wEffectsDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Effect Unit.

4.5.2.9.4 DYNAMIC RANGE COMPRESSOR EFFECT UNIT DESCRIPTOR

The **wEffectType** field of the common Effect Unit descriptor contains the value DYN_RANGE_COMP_EFFECT. (see Appendix A.19, "Effect Unit Effect Types."

The following table outlines the Dynamic Range Compressor Effect Unit descriptor. It is identical to the common Effect Unit descriptor, except for some field values. It is repeated here for clarity.

Table 4-37: Dynamic Range Compressor Effect Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 17+(ch*4).

Offset	Field	Size	Value	Description
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	EFFECT_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	wEffectType	2	Constant	DYN_RANGE_COMP_EFFECT effect type.
6	bSourceID	1	Number	ID of the Unit or Terminal to which this Effect Unit is connected.
7	bmaControls(0)	4	Bitmap	The Controls bitmap for master channel 0: D1..0: Enable Control. D3..2: Compression Ratio Control. D5..4: MaxAmpl Control. D7..6: Threshold Control. D9..8: Attack Time Control. D11..10: Release Time Control. D13..12: Underflow Control. D15..14: Overflow Control. D31..16: Reserved.
11	bmaControls(1)	4	Bitmap	The Controls bitmap for logical channel 1.
…	…	…	…	…
11+(ch*4)	bmaControls(ch)	4	Bitmap	The Controls bitmap for logical channel ch.
15+(ch*4)	wEffectsDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Effect Unit.

4.5.2.10 PROCESSING UNIT DESCRIPTOR

The Processing Unit is uniquely identified by the value in the **bUnitID** field of the Processing Unit descriptor (PUD). No other Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Processing Unit.

The **wProcessType** field contains a value that fully identifies the Processing Unit. For a list of all supported Processing Unit Types, see Appendix A.20, "Processing Unit Process Types."

The **bNrInPins** field contains the number of Input Pins (*p*) of the Processing Unit. The connectivity of the Input Pins is described via the **baSourceID()** array that contains *p* elements. The index *I* into the array is one-based and directly related to the Input Pin numbers. **baSourceID(i)** contains the ID of the Unit or Terminal to which Input Pin *i* is connected. The Cluster descriptors, describing the logical channels entering the Processing Unit are not repeated here. It is up to the Host software to trace the connections 'upstream' to locate the Cluster descriptors pertaining to the Clusters.

Because a Processing Unit can freely redefine the spatial locations of the logical output channels, contained in its output Cluster, there is a need for an output Cluster descriptor. The **wClusterDescrID** field contains the unique ID of the Cluster descriptor that characterizes the Cluster that leaves the Processing Unit over the single Output Pin ('downstream' connection). For a detailed description of the Cluster descriptor, see Section 4.3, "Cluster Descriptor".

A Processing Unit shall not support an Enable Control. If the functionality of the Audio Function requires that the Processing Unit can be bypassed in certain scenarios, then the topology of the Function shall make this explicit through the use of a Selector Unit where one Input Pin of the Selector Unit is connected to the Output Pin of the Processing Unit and the other Input Pin of the Selector Unit is connected to an Entity that provides an output channel Cluster that is compatible with the output Cluster of the Processing Unit. This may be the same Output Pin to which the Input Pin of the Processing Unit is connected (direct bypass) if the Cluster configuration is not altered by the Processing Unit.

In general, all Controls are optional. However, some Processing Types may define certain Controls as mandatory.

An ID of a String descriptor is provided to further describe the Processing Unit.

The previous fields are common to all Processing Units. However, depending on the value in the **wProcessType** field, a process-specific part is added to the descriptor. The following paragraphs describe these process-specific parts.

The following table outlines the common part of the Processing Unit descriptor.

Table 4-38: Common Part of the Processing Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 9+p+x.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	PROCESSING_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	wProcessType	2	Constant	Constant identifying the type of processing this Unit is performing.
6	bNrInPins	1	Number	Number of Input Pins of this Unit: p
7	baSourceID(1)	1	Number	ID of the Unit or Terminal to which the first Input Pin of this Processing Unit is connected.
…	…	…	…	…
7+(p-1)	baSourceID (p)	1	Number	ID of the Unit or Terminal to which the last Input Pin of this Processing Unit is connected.
7+p	wProcessingDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Processing Unit.
9+p	Process-specific	x	N/A	A process-specific descriptor is appended to the common descriptor. See the following paragraphs.

4.5.2.10.1 UP/DOWN-MIX PROCESSING UNIT DESCRIPTOR

The **wProcessType** field of the common Processing Unit descriptor contains the value UP/DOWNMIX_PROCESS. (See Appendix A.20, “Processing Unit Process Types”)

The Up/Down-mix Processing Unit has a single Input Pin. Therefore, the **bNrInputs** field shall contain the value 1.

The Mode Select Control (D1..0) is used to change the behavior of the Processing Unit by selecting different modes of operation.

The process-specific descriptor of the Up/Down-mix Processing Unit describes the supported modes of operation of the Processing Unit. Selecting a mode of operation is done by issuing the Set Mode Select Control Request. The number of supported modes (*m*) is contained in the **bNrModes** field. This field is followed by an array of Cluster descriptor ID fields, **waClusterDescrID()**. The index *i* into this array is one-based and directly related to the number of the mode described by entry **waClusterDescrID(i)**. It is the value *i* that shall be used as a parameter for the Set Mode request to select the mode *i*.

Each **waClusterDescrID(i)** field contains the unique ID of a Cluster descriptor that describes the output Cluster configuration when mode i is selected. The Host changing the mode via a Set Mode Select Control request results in a change of output Cluster descriptor. However, this shall not lead to a High Capability Descriptor interrupt as the Host itself has initiated the change.

The following table outlines the combination of the common and process-specific Up/Down-mix Processing Unit descriptors.

Table 4-39: Up/Down-mix Processing Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 15+2*m.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	PROCESSING_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	wProcessType	2	Constant	UP/DOWNMIX_PROCESS process type.
6	bNrInPins	1	Number	Number of Input Pins of this Unit: 1
7	baSourceID(1)	1	Number	ID of the Unit or Terminal to which the Input Pin of this Processing Unit is connected.
8	wProcessingDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Processing Unit.
10	bmControls	4	Bitmap	D1..0: Mode Select Control. D3..2: Underflow Control. D5..4: Overflow Control. D31..6: Reserved.
14	bNrModes	1	Number	Number of modes, supported by this Processing Unit: m
15	waClusterDescrID(1)	2	Number	Unique ID of the Cluster descriptor for mode(1).
…	…	…	…	…
15+2*(m-1)	waClusterDescrID(m)	2	Number	Unique ID of the Cluster descriptor for mode(m).

4.5.2.10.2 STEREO EXTENDER PROCESSING UNIT DESCRIPTOR

The **wProcessType** field of the common Processing Unit descriptor contains the value STEREO_EXTENDER_PROCESS. (See Appendix A.20, "Processing Unit Process Types")

The Stereo Extender Processing Unit has a single Input Pin. Therefore, the **bNrInputs** field shall contain the value 1.

The input Cluster to the Stereo Extender Processing Unit shall contain only the Left and Right logical input channels. The output Cluster shall therefore also only contain the Left and Right logical channels. As a consequence, there is no need for an output Cluster descriptor for this Processing Unit.

There is no process-specific descriptor for the Stereo Extender Processing Unit.

The following table outlines the Stereo Extender Processing Unit descriptor.

Table 4-40: Stereo Extender Processing Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 14.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	PROCESSING_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	wProcessType	2	Constant	STEREO_EXTENDER_PROCESS process type.
6	bNrInPins	1	Number	Number of Input Pins of this Unit: 1
7	baSourceID(1)	1	Number	ID of the Unit or Terminal to which the Input Pin of this Processing Unit is connected.
8	wProcessingDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Processing Unit.
10	bmControls	4	Bitmap	D1..0: Width Control. D3..2: Underflow Control. D5..4: Overflow Control. D31..6: Reserved.

4.5.2.10.3 MULTI-FUNCTION PROCESSING UNIT DESCRIPTOR

The **wProcessType** field of the Processing Unit descriptor contains the value MULTI_FUNCTION_PROCESS. (See Appendix A.20, "Processing Unit Process Types")

The Multi-Function Processing Unit may have multiple Input Pins Input Pins as indicated in the **bNrInputs** field.

Because a Multi-Function Processing Unit can freely redefine its output Cluster configuration, there is a need for an output Cluster descriptor. The **wClusterDescrID** field **contains the** unique ID of the Cluster descriptor that characterizes the Cluster that leaves the Processing Unit over its single Output Pin ('downstream' connection). For a detailed description of the Cluster descriptor, see Section 4.3, "Cluster Descriptor".

The **bmAlgorithms** field is a bitmap that indicates what types of algorithms are performed inside the Multi-Function Processing Unit. Multiple bits may be set simultaneously.

The following table outlines the Multi-Function Processing Unit descriptor.

Table 4-41: Multi-Function Processing Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 19+p.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	PROCESSING_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	wProcessType	2	Constant	MULTI_FUNCTION_PROCESS process type.
6	bNrInPins	1	Number	Number of Input Pins of this Unit: 1
7	baSourceID(1)	1	Number	ID of the Unit or Terminal to which Input Pin 1 of this Processing Unit is connected.
…	…	…	…	…
7+(p-1)	baSourceID (p)	1	Number	ID of the Unit or Terminal to which the last Input Pin of this Processing Unit is connected.
7+p	wProcessingDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Processing Unit.
9+p	bmControls	4	Bitmap	D1..0: Underflow Control. D3..2: Overflow Control. D31..4: Reserved.
13+p	wClusterDescrID	2	Number	Unique ID of the Cluster descriptor for this Processing Unit.
15+p	bmAlgorithms	4	bitmap	D0: Algorithm Undefined. D1: Beam Forming. D2: Acoustic Echo Cancellation. D3: Active Noise Cancellation. D4: Blind Source Separation. D5: Noise Suppression/Reduction. D31..6: Reserved.

4.5.2.11 EXTENSION UNIT DESCRIPTOR

The Extension Unit is uniquely identified by the value in the **bUnitID** field of the Extension Unit descriptor (XUD). No other Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Extension Unit.

The Extension Unit descriptor provides minimal information about the Extension Unit for a generic driver at least to notice the presence of vendor-specific components within the Audio Function. The **wExtensionCode** field may contain a vendor-specific code that further identifies the Extension Unit. If it is not used, it should be set to zero.

The **bNrInPins** field contains the number of Input Pins (*p*) of the Extension Unit. The connectivity of the Input Pins is described via the **baSourceID()** array that contains *p* elements. The index *I* into the array is one-based and directly related to the Input Pin numbers. **baSourceID(i)** contains the ID of the Unit or Terminal to which Input Pin *I* is connected. The Cluster descriptors that describe the logical channels that enter the Extension Unit are not repeated here. It is up to the Host software to trace the connections 'upstream' to locate the Cluster descriptors pertaining to the Clusters.

Because an Extension Unit can freely redefine its output Cluster configuration, there is a need for an output Cluster descriptor. The **wClusterDescrID** field **contains the** unique ID of the Cluster descriptor that characterizes the Cluster that leaves the Extension Unit over its single Output Pin ('downstream' connection). For a detailed description of the Cluster descriptor, see Section 4.3, "Cluster Descriptor".

An ID of a String descriptor is provided to further describe the Extension Unit.

The following table outlines the Extension Unit descriptor.

Table 4-42: Extension Unit Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 15+p.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	EXTENSION_UNIT descriptor subtype.
3	bUnitID	1	Number	Value uniquely identifying the Unit within the Audio Function. This value is used in all requests to address this Unit.
4	wExtensionCode	2	Constant	Vendor-specific code identifying the Extension Unit.
6	bNrInPins	1	Number	Number of Input Pins of this Unit: p
7	baSourceID(1)	1	Number	ID of the Unit or Terminal to which the first Input Pin of this Extension Unit is connected.
…	…	…	…	…
7+(p-1)	baSourceID (p)	1	Number	ID of the Unit or Terminal to which the last Input Pin of this Extension Unit is connected.
7+p	wExtensionDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Extension Unit.
9+p	bmControls	4	Bitmap	D1..0: Underflow Control. D3..2: Overflow Control. D31..4: Reserved.
13+p	wClusterDescrID	2	Number	Unique ID of the Cluster descriptor for this Extension Unit.

4.5.2.12 CLOCK SOURCE DESCRIPTOR

The Clock Source Entity is uniquely identified by the value in the **bClockID** field of the Clock Source Entity descriptor (CSD). No other Clock Entity, Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Clock Source Entity.

The **bmAttributes** field contains a Clock Type bit field (D0) that indicates whether the Clock Source represents an external clock (D0 = 0b0) or an internal clock (D0=0b1). Further characteristics of the Clock Source can be derived from the value of the Clock Frequency Control bit pair D1..0 in the **bmControls** field. Since the Clock Frequency Control shall always be present, the only allowed values are D1..0 = 0b01 (Read-Only) or D1..0 = 0b11 (Read-

Write). The supported clock frequencies can be derived from the Range attribute of the Clock Frequency Control (fixed rate vs. variable rate). Note that even a Clock Source of Type External can be Read-Write if the Audio Function has the ability to influence that external clock through means outside of USB. The actual sampling frequency of the Clock Source can be manipulated through the Clock Frequency request. In addition, the Clock Source can be queried for the validity of its current sampling clock signal through a Get Clock Validity request.

Bit D1 in the **bmAttributes** field indicates whether an internal clock is free running (D1 = 0b0) or synchronized to the Start of Frame (D1 = 0b1). If D0 = 0b0, then D1 shall also be set to 0b0.

The **bReferenceTerminal** field contains a reference to a Terminal from which the Clock Source is derived. This is useful for instance when a Clock Source's clock signal is derived from the input signal on an S/PDIF connector, which is represented by an Input Terminal. If the Clock Source is free running or derived from USB SOF (not derived from a Terminal), this field shall be set to zero.

An ID of a String descriptor is provided to further describe the Clock Source Unit.

The following table presents an outline of the Clock Source descriptor.

Table 4-43: Clock Source Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 12.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	CLOCK_SOURCE descriptor subtype.
3	bClockID	1	Number	Value uniquely identifying the Clock Source Entity within the Audio Function. This value is used in all requests to address this Entity.
4	bmAttributes	1	Bitmap	D0: Clock Type: 0: External Clock. 1: Internal Clock. D1: Synchronization Type: 0: Asynchronous. 1: Clock synchronized to SOF. D7..2: Reserved.
5	bmControls	4	Bitmap	D1..0: Clock Frequency Control. D3..2: Clock Validity Control. D31..4: Reserved.
9	bReferenceTerminal	1	Number	Terminal ID of the Terminal from which this Clock Source is derived.
10	wClockSourceStr	2	wStrDescrID	ID of a class-specific String descriptor, describing the Clock Source Entity.

4.5.2.13 CLOCK SELECTOR DESCRIPTOR

The Clock Selector Entity is uniquely identified by the value in the **bClockID** field of the Clock Selector Entity descriptor (CXD). No other Clock Entity, Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Clock Selector Entity.

The **bNrInPins** field contains the number of Clock Input Pins (*p*) of the Clock Selector Entity. The connectivity of the Input Pins is described via the **baCSourceID()** array that contains *p* elements. The index *I* into the array is one-based and directly related to the Clock Input Pin numbers. **baCSourceID(i)** contains the ID of the Clock Entity to which Clock Input Pin *I* is connected.

An ID of a String descriptor is provided to further describe the Clock Selector Entity.

The following table presents an outline of the Clock Selector descriptor.

Table 4-44: Clock Selector Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 11+p.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	CLOCK_SELECTOR descriptor subtype.
3	bClockID	1	Number	Value uniquely identifying the Clock Selector Entity within the Audio Function. This value is used in all requests to address this Entity.
4	bNrInPins	1	Number	Number of Input Pins of this Unit: p
5	baCSourceID(1)	1	Number	ID of the Clock Entity to which the first Clock Input Pin of this Clock Selector Entity is connected.
…	…	…	…	…
5+(p-1)	baCSourceID (p)	1	Number	ID of the Clock Entity to which the last Clock Input Pin of this Clock Selector Entity is connected.
5+p	bmControls	4	Bitmap	D1..0: Clock Selector Control. D31..2: Reserved.
9+p	wCSelectorDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing the Clock Selector Entity.

4.5.2.14 CLOCK MULTIPLIER DESCRIPTOR

The Clock Multiplier Entity is uniquely identified by the value in the **bClockID** field of the Clock Multiplier Entity descriptor (CMD). No other Clock Entity, Unit or Terminal within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Clock Multiplier Entity.

The bCSourceID field contains the ID of the Clock Entity to which the Clock Multiplier's Clock Input Pin is connected.

An ID of a String descriptor is provided to further describe the Clock Multiplier Entity.

The following table presents an outline of the Clock Multiplier descriptor.

Table 4-45: Clock Multiplier Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 11.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.

Offset	Field	Size	Value	Description
2	bDescriptorSubtype	1	Constant	CLOCK_MULTIPLIER descriptor subtype.
3	bClockID	1	Number	Value uniquely identifying the Clock Multiplier Entity within the Audio Function. This value is used in all requests to address this Entity.
4	bCSourceID	1	Number	ID of the Clock Entity to which the last Clock Input Pin of this Clock Selector Entity is connected.
5	bmControls	4	Bitmap	D1..0: Clock Numerator Control. D3..2: Clock Denominator Control. D31..4: Reserved.
9	wCMultiplierDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing the Clock Multiplier Entity.

4.5.2.15 POWER DOMAIN DESCRIPTOR

The Power Domain descriptor (PDD) provides information to the Host regarding the existence of one or more Power Domains within the Audio Function and lists the Entities, through their respective Entity ID, that are explicit members of a particular Power Domain. Only Input Terminals, Output Terminals, Effect Units, Processing Units, and Extension Units shall be explicit members of a Power Domain.

There is a Power Domain descriptor for each Power Domain in the Audio Function. Therefore, the number of Power Domain descriptors is an indicator for the number of separately managed Power Domains in the Audio Function. Each eligible Entity can only be member of a single Power Domain, i.e. Power Domains never overlap.

The Power Domain is uniquely identified by the value in the **bPowerDomainID** field of the Power Domain descriptor (PDD). No other Entity within the AudioControl interface may have the same ID. This value shall be passed in the Entity ID field (part of the **wIndex** field) of each request that is directed to the Power Domain.

The **waRecoveryTime()** array contains the approximate recovery time for Power Domain State D1 and D2. The **bNrEntities** field contains the number of Entities in this Power Domain. The **baEntityID()** array contains the Entity IDs of all the explicit member Entities.

An ID of a String descriptor is provided to further describe the Power Domain.

Table 4-46: Power Domain Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 11+p.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	POWER_DOMAIN descriptor subtype.
3	bPowerDomainID	1	Number	Value uniquely identifying the Power Domain within the Audio Function. This value is used in all requests to address this Power Domain.

4	waRecoveryTime(1)	2	Number	Time to recover from D1 to D0. Expressed in 50 µs increments. For example, a typical value for this is 600, which indicates a recovery time of 30 ms.
6	waRecoveryTime(2)	2	Number	Time to recover from D2 to D0. Expressed in 50 µs increments. For example, a typical value for this is 6000, which indicates a recovery time of 300 ms.
8	bNrEntities	1	Number	Number of Entities belonging to this Power Domain: p.
9	baEntityID(1)	1	Number	ID of the first Entity that belongs to this Power Domain.
…	…	…	…	
8+p	baEntityID(p)	1	Number	ID of the last Entity that belongs to this Power Domain.
9+p	wPDomainDescrStr	2	wStrDescrID	ID of a class-specific String descriptor, describing this Power Domain.

4.6 AUDIOCONTROL ENDPOINT DESCRIPTORS

The following sections describe all possible endpoint-related descriptors for the AudioControl interface.

4.6.1 AC CONTROL ENDPOINT DESCRIPTORS

4.6.1.1 STANDARD AC CONTROL ENDPOINT DESCRIPTOR

The AudioControl interface uses the default endpoint 0. Therefore, there is no dedicated standard control endpoint descriptor.

4.6.1.2 CLASS-SPECIFIC AC CONTROL ENDPOINT DESCRIPTOR

There is no dedicated class-specific control endpoint descriptor.

4.6.2 AC INTERRUPT ENDPOINT DESCRIPTORS

4.6.2.1 STANDARD AC INTERRUPT ENDPOINT DESCRIPTOR

The interrupt endpoint descriptor is identical to the standard endpoint descriptor defined in the *USB Specification*. Its fields are set to reflect the interrupt type of the endpoint. This endpoint is optional.

The following table outlines the standard AC Interrupt Endpoint descriptor.

Table 4-47: Standard AC Interrupt Endpoint Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 7
1	bDescriptorType	1	Constant	ENDPOINT descriptor type

Offset	Field	Size	Value	Description
2	bEndpointAddress	1	Endpoint	The address of the endpoint on the USB device described by this descriptor. The address is encoded as follows: D7: Direction. 1 = IN endpoint. D6..4: Reserved. D3..0: The endpoint number, determined by the designer.
3	bmAttributes	1	Bitmap	D1..0: Transfer Type 11 = Interrupt. All other bits are reserved.
4	wMaxPacketSize	2	Number	Maximum packet size this endpoint is capable of sending or receiving when this configuration is selected. Used here to pass 6-byte interrupt information.
6	bInterval	1	Number	Interval for polling the Interrupt endpoint.

4.6.2.2 CLASS-SPECIFIC AC INTERRUPT ENDPOINT DESCRIPTOR

There is no class-specific AudioControl interrupt endpoint descriptor.

4.7 AUDIOSTREAMING INTERFACE DESCRIPTORS

The AudioStreaming (AS) interface descriptors contain all relevant information to characterize the AudioStreaming interface in full.

4.7.1 STANDARD AS INTERFACE DESCRIPTOR

The standard AS interface descriptor is identical to the standard interface descriptor defined in the *USB Specification*, except that some fields now have dedicated values.

Table 4-48: Standard AS Interface Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 9.
1	bDescriptorType	1	Constant	INTERFACE descriptor type.
2	bInterfaceNumber	1	Number	Number of the interface. A zero-based value identifying the index in the array of concurrent interfaces supported by this configuration.
3	bAlternateSetting	1	Number	Value used to select an Alternate Setting for the interface identified in the prior field.
4	bNumEndpoints	1	Number	Number of endpoints used by this interface (excluding endpoint 0). Shall be either 0 (no data endpoint), 1 (data endpoint) or 2 (data and explicit feedback endpoint).
5	bInterfaceClass	1	Class	AUDIO Audio Interface Class code (assigned by the USB). See Appendix A.4, “Audio Interface Class Code”

Offset	Field	Size	Value	Description
6	bInterfaceSubClass	1	Subclass	AUDIO_STREAMING Audio Interface Subclass code. Assigned by this specification. See Appendix A.5, "Audio Interface Subclass Codes."
7	bInterfaceProtocol	1	Protocol	IP_VERSION_03_00 Interface Protocol code. Indicates the current version of the specification. See Appendix A.6, "Audio Interface Protocol Codes"
8	iInterface	1	Index	Index of a String descriptor that describes this interface.

4.7.2 CLASS-SPECIFIC AS INTERFACE DESCRIPTOR

The **bTerminalLink** field contains the unique Terminal ID of the Input or Output Terminal to which this interface is associated.

The **wClusterDescrID** field contains the unique ID of the cluster descriptor that characterizes the physical cluster associated with an Alternate Setting of this interface. For a detailed description of the cluster descriptor, see Section 4.3, "Cluster Descriptor".

The remaining fields in this descriptor are described in detail in a separate document, *USB Audio Data Formats* that is considered part of this specification.

An Alternate Setting of an interface is allowed to support multiple Audio Data Formats at the same time, even if they belong to different Format Types. The **bmFormats** bitmap has a bit set for each Audio Data Format that can be used when communicating with this Alternate Setting of the interface. It is up to the implementation to be able to accurately distinguish among the different Audio Data Formats and invoke the correct encoding or decoding processes for the current Audio Data Format. Alternatively, support for the different Format Types and/or Audio Data Formats can be separated out into different Alternate Settings if the interface is not able to accurately distinguish among the different Format Types and/or Audio Data Formats.

Table 4-49: Class-Specific AS Interface Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor in bytes: 23.
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	AS_GENERAL descriptor subtype.
3	bTerminalLink	1	Number	The Terminal ID of the Terminal to which this interface is connected.
4	bmControls	4	Bitmap	D1..0: Active Alternate Setting Control. D3..2: Valid Alternate Settings Control. D5..4: Audio Data Format Control. D31..6: Reserved.
8	wClusterDescrID	2	Number	ID of the cluster descriptor of the AS Interface.

Offset	Field	Size	Value	Description
10	bmFormats	8	Bitmap	The Audio Data Format(s) that can be used to communicate with this interface. See the *USB Audio Data Formats* document for further details.
18	bSubslotSize	1	Number	The number of bytes occupied by one audio subslot.
19	bBitResolution	1	Number	The number of effectively used bits from the available bits in an audio subslot.
20	bmAuxProtocols	2	Bitmap	Bitmap, indicating which Auxiliary Protocols are required.
22	bControlSize	1	Number	Size of the Control Channel Words, in bytes.

4.7.3 CLASS-SPECIFIC AS VALID FREQUENCY RANGE DESCRIPTOR

The AS Valid Frequency Range descriptor provides information to the Host about what sampling frequency ranges are supported by this Alternate Setting of the AudioStreaming interface. An Audio Function shall provide this descriptor to indicate that this Alternate Setting of the interface is only valid if the selected sampling frequency is in the range [dMin..dMax]. The values of dMin and dMax are expressed in Hz. If the Alternate Setting of the interface is valid for any available clock frequency supported by the Audio Function, the descriptor may be omitted. Multiple instances of this descriptor are allowed in the same Alternate Setting of the AudioStreaming interface to describe disjoint frequency ranges.

Table 4-50: Class-Specific AS Valid Frequency Range Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor in bytes: 11
1	bDescriptorType	1	Constant	CS_INTERFACE descriptor type.
2	bDescriptorSubtype	1	Constant	AS_VALID_FREQ_RANGE descriptor subtype.
3	dMin	4	Number	The minimum sampling frequency at which this Alternate Setting of the AudioStreaming interface is valid.
7	dMax	4	Number	The maximum sampling frequency at which this Alternate Setting of the AudioStreaming interface is valid.

4.8 AUDIOSTREAMING ENDPOINT DESCRIPTORS

The following sections describe all possible endpoint-related descriptors for the AudioStreaming interface.

4.8.1 AS ISOCHRONOUS AUDIO DATA ENDPOINT DESCRIPTORS

The standard and class-specific audio data endpoint descriptors provide pertinent information on how audio data streams are communicated to the Audio Function. In addition, specific endpoint capabilities and properties are reported.

4.8.1.1 STANDARD AS ISOCHRONOUS AUDIO DATA ENDPOINT DESCRIPTOR

The standard AS isochronous audio data endpoint descriptor is identical to the standard endpoint descriptor defined in the *USB Specification*. D7 of the **bEndpointAddress** field indicates whether the endpoint is an audio source (D7 = 0b1) or an audio sink (D7 = 0b0). The **bmAttributes** Field bits are set to reflect the isochronous type of the endpoint. The synchronization type is indicated by D3..2 and shall be set to Asynchronous, Adaptive or Synchronous. For further details, refer to the *USB Specification*.

Table 4-51: Standard AS Isochronous Audio Data Endpoint Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 7
1	bDescriptorType	1	Constant	ENDPOINT descriptor type
2	bEndpointAddress	1	Endpoint	The address of the endpoint on the USB device described by this descriptor. The address is encoded as follows: D3..0: The endpoint number, determined by the designer. D6..4: Reserved. D7: Direction: 0 = OUT endpoint. 1 = IN endpoint.
3	bmAttributes	1	Bitmap	D1..0: Transfer type: 01 = Isochronous. D3..2: Synchronization Type: 01 = Asynchronous. 10 = Adaptive. 11 = Synchronous. D5..4: Usage Type: 00 = Data endpoint or 10 = Implicit feedback Data endpoint. All other bits are reserved.
4	wMaxPacketSize	2	Number	(Speed-dependent.) Indicates the maximum packet size and potentially any bursting on this endpoint. This is determined by the audio bandwidth constraints of the endpoint.
6	bInterval	1	Number	Interval for polling endpoint for data transfers.

Note: For SuperSpeed and SuperSpeedPlus endpoints, the SuperSpeed Endpoint Companion and SuperSpeedPlus Endpoint Companion descriptors would follow the standard endpoint descriptor. See the *USB 3.1 specification* for details.

4.8.1.2 CLASS-SPECIFIC AS ISOCHRONOUS AUDIO DATA ENDPOINT DESCRIPTOR

The **bLockDelayUnits** and **wLockDelay** fields are used to indicate to the Host how long it takes for the clock recovery circuitry of this endpoint to lock and reliably produce or consume the audio data stream. This information can be used by the Host to take appropriate action so that no meaningful data gets lost during the locking period. (For instance, sending digital silence during lock period)

Depending on the implementation, the locking period can be a fixed amount of time or can be proportional to the sampling frequency. In this case, it usually takes a fixed amount of samples to become locked. To accommodate both cases, the **bLockDelayUnits** field indicates whether the **wLockDelay** field is expressed in time (milliseconds) or number of samples.

Note: Some implementations may use locking strategies that do not have either a fixed time or a fixed number of samples before locking. In this case, a worst case value can be reported back to the Host.

The **bLockDelayUnits** and **wLockDelay** fields are only applicable for synchronous and adaptive endpoints. For asynchronous endpoints, the clock is generated internally in the Audio Function and is completely independent. In this case, **bLockDelayUnits** and **wLockDelay** shall be set to zero.

Table 4-52: Class-Specific AS Isochronous Audio Data Endpoint Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 10.
1	bDescriptorType	1	Constant	CS_ENDPOINT descriptor type.
2	bDescriptorSubtype	1	Constant	EP_GENERAL descriptor subtype.
3	bmControls	4	Bitmap	D1..0: Pitch Control. D3..2: Data Overrun Control. D5..4: Data Underrun Control. D31..6: Reserved.
7	bLockDelayUnits	1	Number	Indicates the units used for the wLockDelay field: 0: Undefined. 1: Milliseconds. 2: Decoded PCM samples. 3..255: Reserved.
8	wLockDelay	2	Number	Indicates the time it takes this endpoint to reliably lock its internal clock recovery circuitry. Units used depend on the value of the bLockDelayUnits field.

4.8.2 AS ISOCHRONOUS FEEDBACK ENDPOINT DESCRIPTOR

This descriptor is present only when one or more isochronous audio data endpoints of the adaptive source type or the asynchronous sink type are implemented.

4.8.2.1 STANDARD AS ISOCHRONOUS FEEDBACK ENDPOINT DESCRIPTOR

The isochronous feedback endpoint descriptor is identical to the standard endpoint descriptor defined in the *USB Specification*. The **bmAttributes** field bits are set to reflect the isochronous type and synchronization type of the endpoint.

Table 4-53: Standard AS Isochronous Feedback Endpoint Descriptor

Offset	Field	Size	Value	Description
0	bLength	1	Number	Size of this descriptor, in bytes: 7
1	bDescriptorType	1	Constant	ENDPOINT descriptor type.
2	bEndpointAddress	1	Endpoint	The address of the endpoint on the USB device described by this descriptor. The address is encoded as follows: D3..0: The endpoint number, determined by the designer. D6..4: Reserved. D7: Direction: 0 = OUT endpoint. 1 = IN endpoint.
3	bmAttributes	1	Bitmap	D1..0: Transfer type: 01 = Isochronous. D3..2: Synchronization Type: 00 = No Synchronization. D5..4: Usage Type: 01 = Feedback endpoint. All other bits are reserved.
4	wMaxPacketSize	2	Number	Maximum packet size this endpoint is capable of sending or receiving when this configuration is selected.
6	bInterval	1	Number	Interval for polling endpoint for data transfers.

Note: For SuperSpeed and SuperSpeedPlus endpoints, the SuperSpeed Endpoint Companion and SuperSpeedPlus Endpoint Companion descriptors would follow the standard endpoint descriptor. See the *USB 3.1 specification* for details.

4.8.2.2 CLASS-SPECIFIC AS ISOCHRONOUS FEEDBACK ENDPOINT DESCRIPTOR

There is no class-specific AS isochronous feedback endpoint descriptor.

4.9 CLASS-SPECIFIC STRING DESCRIPTORS

This specification defines a new type of class-specific String descriptor. Class-specific String descriptors are retrieved through a class-specific Get String request. See Section 5.2.3.2, “Class-specific String Request” for details. All class-specific strings in the Audio Function shall use this new methodology. Strings that are part of the standard descriptor set shall use the standard string methodology.

Class-specific strings can be dynamic in nature, i.e. change during normal operation and inform the Host of such a change by generating an interrupt with source type set to STRING.

The **wLength** field contains the length of the class-specific String descriptor. Class-specific strings can be up to 65,528 bytes in length.

The **bDescriptorType** field shall be set to CS_STRING.

The **bDescriptorSubtype** field indicates the descriptor subtype for the String descriptor. (Currently, only the value SUBTYPE_UNDEFINED is defined.)

The **wStrDescrID** field contains a unique identifier for the class-specific String descriptor in the range [256..65,535].

The **iLangID** field contains a zero-based index into the LANGID code array as returned by the device. A device can at most support 126 different languages since the LANGID code array is restricted to 254 bytes and each LANGID code takes up 2 bytes. The range of the **bLangID** is therefore from 0 to 125 maximum.

The String field contains the actual Unicode encoded string as outlined in the *USB Specification*.

Table 4-54: Class-specific String Descriptor

Offset	Field	Size	Value	Description
0	wLength	2	Number	Size of the String descriptor in bytes: 7+N.
2	bDescriptorType	1	Number	CS_STRING. Type of this descriptor.
3	bDescriptorSubtype	1	Constant	Descriptor Subtype.
4	wStrDescrID	2	Number	Unique ID for this class-specific String descriptor.
6	iLangID	1	Number	Zero-based index into the LANGID code array as returned by the device.
7	String	N	Number	Unicode encoded string. Follows the definitions outlined in the *USB Specification*.

5 REQUESTS

5.1 STANDARD REQUESTS

The Audio Device Class supports the standard requests described in Section 9, "USB Device Framework," of the *USB Specification*. The Audio Device Class places no specific requirements on the values for the standard requests.

5.2 CLASS-SPECIFIC REQUESTS

Class-specific requests are used to set and get audio related Controls. These Controls fall into two main groups: those that manipulate the Audio Function's Controls, such as volume, tone, selector position, etc. and those that influence data transfer over an isochronous endpoint, such as the current sampling frequency.

- **AudioControl Requests**. Control of an Audio Function is performed through the manipulation of the attributes of individual Controls that are embedded in the Entities of the Audio Function. The class-specific AudioControl interface descriptor contains a collection of Entity descriptors, each indicating which Controls are present in the Entity. AudioControl requests are always directed to the single AudioControl interface of the Audio Function. The request contains enough information (Entity ID, Control Selector, and Channel Number) for the Audio Function to decide to where a specific request shall be routed. The same request layout can be used for vendor-specific requests to Extension Units. However, they are not covered by this specification.
- **AudioStreaming Requests**. Control of the class-specific behavior of an AudioStreaming interface is performed through manipulation of either interface Controls or endpoint Controls. These can be either class-specific (as defined in this specification) or vendor-specific. In either case, the same request layout can be used. AudioStreaming requests are directed to the recipient where the Control resides. This can be either the interface or its associated isochronous endpoint.

The Audio Device Class supports three additional class-specific requests:

- **Memory Requests**. Every addressable Entity in the Audio Function (Clock Entity, Terminal, Unit, Power Domain, Interface, and Endpoint) can expose a memory-mapped interface that provides the means to generically manipulate the Entity. Vendor-specific Control implementations could be based on this type of request.
- **String Requests**. This type of request provides a class-specific method to retrieve String descriptors from the Audio Function. This request is introduced to overcome the USB core specification limitation that only provides for 255 device-wide String descriptors.
- **Descriptor Requests**. This type of request provides a class-specific method to retrieve descriptors from the Audio Function outside the standard descriptor retrieval during enumeration. This request is introduced to overcome the USB core specification limitation that only provides for descriptors that are a maximum of 256 bytes long. It also enables dynamically changing descriptors (after enumeration).

In principle, all Controls and their associated requests are optional. If an Audio Function does not support a certain request, it shall indicate this by stalling the control pipe when that request is issued to the function. However, if a certain Set request is supported, the associated Get request shall also be supported. Get requests may be supported without the associated Set request being supported. If interrupts are supported, then all necessary Get requests shall be implemented that are required to retrieve the appropriate information from the Audio Function in response to these interrupts.

The remainder of this section describes the class-specific requests and their characteristics used to manipulate the incorporated Controls, memory locations, class-specific strings and descriptors. Unless explicitly stated otherwise, all Controls are optional and shall be Read/Write, if present.

5.2.1 AUDIOCONTROL REQUESTS

The following sections describe the possible requests that can be used to manipulate the Audio Controls an Audio Function exposes through its Entities. The same layout of the parameter blocks is used for both the Set and Get requests.

5.2.1.1 CONTROL ATTRIBUTES

Each Control within an Entity can have one or more attributes associated with it. Currently defined attributes for a Control are its:

• Current setting attribute	(CUR)
• Range attribute	(RANGE)
• Interrupt Enable attribute	(INTEN)

The CUR attribute is used to manipulate the current actual setting of a Control. The RANGE attribute provides information about the limitations the Control imposes on the allowed settings of the CUR attribute. The RANGE attribute actually consists of an array of sub-attributes. The sub-attributes are Minimum (MIN), Maximum (MAX), and Resolution (RES). They are always manipulated in triplets of the form [MIN, MAX, RES] and cannot be accessed or modified individually. The RANGE attribute supports an array of these triplets so that discontinuous multiple subranges of a Control can be accurately reported. The first element in the array contains the number of subranges the Control supports. Subsequent triplet elements in the array correspond to each of the subranges. The subranges shall be ordered in ascending order (from lower values to higher values). Individual subranges cannot overlap (i.e. the MAX value of the previous subrange cannot be equal to the MIN value of the next subrange). If a subrange consists of only a single value, the corresponding triplet shall contain that value for both its MIN and MAX sub-attribute and the RES sub-attribute shall be set to zero.

As an example, consider a (hypothetical) Volume Control that can take the following values for its CUR attribute:

- -∞ dB
- -70 dB to -40 dB in steps of 3 dB
- -40 dB to -20 dB in steps of 2 dB
- -20 dB to 0 dB in steps of 1 dB

One possible layout of the RANGE attribute is then:

RANGE(0) = 3
RANGE(1) = [-70, -40, 3]
RANGE(2) = [-38, -20, 2]
RANGE(3) = [-19, 0, 1]

Another way of representing the same Control is as follows:

RANGE(0) = 3
RANGE(1) = [-70, -43, 3]
RANGE(2) = [-40, -22, 2]
RANGE(3) = [-20, 0, 1]

It is left to the designer to choose a suitable representation.

The Interrupt Enable attribute is used to manipulate the Control's ability to generate an interrupt when any of its other attributes change other than through Host manipulation. The Interrupt Enable attribute shall be supported for all Controls that are able to generate an interrupt. When supported, the Interrupt Enable attribute is always Read-Write and therefore, both the Get and Set request shall be supported. The Interrupt Enable attribute can take only two values: 0 (False, Disabled) or 1 (True, Enabled). All Interrupt Enable attributes shall have a default value of 1 (Enabled).

5.2.1.2 CONTROL REQUEST LAYOUT

The Audio Device Class-defined request layout closely follows the standard request layout as defined in the *USB Specification*. The request is used to set or get an attribute of a Control inside an Entity of the Audio Function. The following table details the request layout.

Table 5-1: Request Layout

bmRequest Type	bRequest	wValue	wIndex	wLength	Data
00100001B 10100001B	CUR RANGE INTEN	CS and CN or MCN	Entity ID and Interface	Length of parameter block	Parameter block
00100010B 10100010B			Zero and Endpoint		

Bit D7 of the **bmRequestType** field specifies whether this is a Set request (D7 = 0b0) or a Get request (D7 = 0b1). It is a class-specific request (D6..5 = 0b01), directed to either an interface (AudioControl or AudioStreaming) of the Audio Function (D4..0 = 0b00001) or the isochronous endpoint of an AudioStreaming interface (D4..0 = 0b00010).

The **bRequest** field contains a constant, identifying which attribute of the addressed Control is to be manipulated. Possible attributes for a Control are its:

- Current setting attribute (CUR)
- Range attribute (RANGE)
- Interrupt Enable attribute (INTEN)

If the addressed Control does not support modification of a certain attribute, the control pipe shall indicate a stall when an attempt is made to modify that attribute. In most cases, only the CUR attribute will be supported for the Set request. However, this specification does not prevent a designer from making the RANGE attribute programmable or having the Audio Function adjust a RANGE attribute due to an external event. For the list of Request constants, refer to Appendix A.22, "Audio Class-Specific Request Codes."

> Note: Support for the INTEN attribute is not repeated for each Control in the following sections. It is assumed that support is provided as appropriate for the Control in a particular Audio Function implementation. Issuing a Get INTEN Request will result in a stall on the control pipe when the INTEN attribute is not supported for that Control.

As a general rule, when an attribute value is set, a Control will automatically adjust the passed value to the closest available valid value. This value can be retrieved through a subsequent Get Control request.

The **wValue** field specifies the Control Selector (CS) in the high byte and the Channel Number (CN) in the low byte. The Control Selector indicates which type of Control this request is manipulating. The Channel Number (CN) indicates which logical channel of the Cluster is to be influenced. If a Control is channel independent, then the Control is considered to be a master Control and the virtual channel zero is used to address it (CN = 0). If the request specifies an unknown or unsupported CS or CN to that Unit, the control pipe shall indicate a stall.

There is an exception to the above. If the Mixer Unit Control request wants to address a Mixer Control, it specifies CS = MU_MIXER_CONTROL as the Control Selector in the high byte and the Mixer Control Number (MCN) in the low byte.

When the request addresses an Entity in an interface (**bmRequestType** = 0b00100001 or 10100001), the **wIndex** field specifies the interface in the low byte and the Entity ID (Clock Entity ID, Unit ID, Terminal ID, or Power Domain ID). For addressing the interface itself, an Entity ID of zero shall be specified in the high byte.

When the request addresses an endpoint (**bmRequestType** = 0b00100010 or 10100010), the **wIndex** field specifies the endpoint to be addressed in the low byte and zero in the high byte.

The values in **wIndex** shall be appropriate to the recipient. Only existing Entities in the Audio Function or in the AudioStreaming interfaces can be addressed and only appropriate interface or endpoint numbers may be used. If the request specifies an unknown or non-Entity ID or an unknown interface or endpoint number, the control pipe shall indicate a stall.

The actual parameter(s) for the Set request are passed in the data stage of the control transfer. The length of the parameter block is indicated in the **wLength** field of the request. The layout of the parameter block is qualified by both the **bRequest** and **wIndex** fields. Refer to the following sections for a detailed description of the parameter block layout for all possible Entities.

The actual parameter(s) for the Get request are returned in the data stage of the control transfer. The length of the parameter block to return is indicated in the **wLength** field of the request. If the parameter block is longer than what is indicated in the **wLength** field, only the initial bytes of the parameter block are returned. If the parameter block is shorter than what is indicated in the **wLength** field, the device indicates the end of the control transfer by sending a short packet when further data is requested. The layout of the parameter block is qualified by both the **bRequest** and **wIndex** fields. Refer to the following sections for a detailed description of the parameter block layout for all possible Entities.

5.2.1.3 CONTROL REQUEST PARAMETER BLOCK LAYOUT

With a few exceptions, almost all Control requests manipulate a single Control parameter during a Set or Get request. For those requests, the possible parameter block layouts can be divided into three categories, depending on the byte size of the Control's CUR attribute. A CUR attribute's size can be either a single byte, a word (2 bytes) or a double word (4 bytes). The following paragraphs specify the layout of the CUR and RANGE parameter blocks for the three categories. The layout of the parameter block for the INTEN attribute is the same for all Controls and is specified in Section 5.2.1.3.4, "INTEN Parameter Block".

For those requests that use a deviating parameter block layout, the actual layout is explicitly defined in the relevant sections.

5.2.1.3.1 LAYOUT 1 PARAMETER BLOCK

The parameter block for a 1-byte sized CUR attribute of a Control is as follows:

Table 5-2: 1-byte Control CUR Parameter Block

wLength		1		
Offset	Field	Size	Value	Description
0	bCUR	1	Number	The setting for the CUR attribute of the addressed Control

The associated parameter block for the RANGE attribute of that Control is as follows:

Table 5-3: 1-byte Control RANGE Parameter Block

wLength		2+3*n		
Offset	Field	Size	Value	Description
0	wNumSubRanges	2	Number	The number of subranges of the addressed Control: n
2	bMIN(1)	1	Number	The setting for the MIN attribute of the first subrange of the addressed Control
3	bMAX(1)	1	Number	The setting for the MAX attribute of the first subrange of the addressed Control
4	bRES(1)	1	Number	The setting for the RES attribute of the first subrange of the addressed Control
…	…	…	…	…
2+3*(n-1)	bMIN(n)	1	Number	The setting for the MIN attribute of the last subrange of the addressed Control
3+3*(n-1)	bMAX(n)	1	Number	The setting for the MAX attribute of the last subrange of the addressed Control
4+3*(n-1)	bRES(n)	1	Number	The setting for the RES attribute of the last subrange of the addressed Control

5.2.1.3.2 LAYOUT 2 PARAMETER BLOCK

The parameter block for a 2-byte sized CUR attribute of a Control is as follows:

Table 5-4: 2-byte Control CUR Parameter Block

wLength		2		
Offset	Field	Size	Value	Description
0	wCUR	2	Number	The setting for the CUR attribute of the addressed Control

The associated parameter block for the RANGE attribute of that Control is as follows:

Table 5-5: 2-byte Control RANGE Parameter Block

wLength		2+6*n		
Offset	Field	Size	Value	Description
0	wNumSubRanges	2	Number	The number of subranges of the addressed Control: n

2	wMIN(1)	2	Number	The setting for the MIN attribute of the first subrange of the addressed Control
4	wMAX(1)	2	Number	The setting for the MAX attribute of the first subrange of the addressed Control
6	wRES(1)	2	Number	The setting for the RES attribute of the first subrange of the addressed Control
…	…	…	…	…
2+6*(n-1)	wMIN(n)	2	Number	The setting for the MIN attribute of the last subrange of the addressed Control
4+6*(n-1)	wMAX(n)	2	Number	The setting for the MAX attribute of the last subrange of the addressed Control
6+6*(n-1)	wRES(n)	2	Number	The setting for the RES attribute of the last subrange of the addressed Control

5.2.1.3.3 LAYOUT 3 PARAMETER BLOCK

The parameter block for a 4-byte sized CUR attribute of a Control is as follows:

Table 5-6: 4-byte Control CUR Parameter Block

wLength		4		
Offset	Field	Size	Value	Description
0	dCUR	4	Number	The setting for the CUR attribute of the addressed Control

The associated parameter block for the RANGE attribute of that Control is as follows:

Table 5-7: 4-byte Control RANGE Parameter Block

wLength		2+12*n		
Offset	Field	Size	Value	Description
0	wNumSubRanges	2	Number	The number of subranges of the addressed Control: n
2	dMIN (1)	4	Number	The setting for the MIN attribute of the first subrange of the addressed Control
6	dMAX (1)	4	Number	The setting for the MAX attribute of the first subrange of the addressed Control
10	dRES (1)	4	Number	The setting for the RES attribute of the first subrange of the addressed Control
…	…	…	…	…
2+12*(n-1)	dMIN(n)	4	Number	The setting for the MIN attribute of the last subrange of the addressed Control
6+12*(n-1)	dMAX(n)	4	Number	The setting for the MAX attribute of the last subrange of the addressed Control
10+12*(n-1)	dRES(n)	4	Number	The setting for the RES attribute of the last subrange of the addressed Control

5.2.1.3.4 INTEN PARAMETER BLOCK

The parameter block for the INTEN attribute of a Control is as follows:

Table 5-8: INTEN Parameter Block

wLength		1		
Offset	Field	Size	Value	Description
0	bINTEN	1	Boolean	The setting for the INTEN attribute of the addressed Control. Only allowed values are 0 (False) and 1 (True).

5.2.1.4 COMMON CONTROLS

The following sections describe a number of Controls that can appear in several Entity types. They are described here only once and a reference to these Control descriptions is provided for all those Entities that can incorporate any of these Controls.

5.2.1.4.1 ENABLE CONTROL

The Enable Control is used to either enable the functionality of an Entity or bypass the Entity entirely. In the latter case, the input Cluster is routed unaltered to the output of the Entity. The Enable Control shall have only the CUR attribute. The value of an Enable Control CUR attribute can be either TRUE or FALSE.

The Control Selector field shall be set to XX_ENABLE_CONTROL (where XX shall be replaced by the appropriate two-letter abbreviation for the particular Entity) and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.4.2 UNDERFLOW CONTROL

The Underflow Control is used to indicate the occurrence of a calculation underflow condition within an Entity *since the last Get Underflow request*. Calculation underflow occurs when an attempt is made to assign a negative value to an unsigned variable. If implemented, this Control shall be Read-Only. Responding to the Get request returns the CUR attribute, and then clears its value. An Underflow Control shall have only the CUR attribute. The value of an Underflow Control CUR attribute shall be either TRUE (underflow condition occurred) or FALSE (normal).

The Control Selector field shall be set to XX_UNDERFLOW_CONTROL (where XX shall be replaced by the appropriate two-letter abbreviation for the particular Entity) and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.4.3 OVERFLOW CONTROL

The Overflow Control is used to indicate the occurrence of a calculation overflow condition within an Entity *since the last Get Overflow request*. Calculation overflow occurs when a value is too positive or too negative to be represented after a signed calculation and when it is too positive after an unsigned calculation. If implemented, this Control shall be Read-Only. Responding to the Get request returns the CUR attribute, and then clears its value. An Overflow Control shall have only the CUR attribute. The value of an Overflow Control CUR attribute shall be either TRUE (overflow condition occurred) or FALSE (normal).

The Control Selector field shall be set to XX_OVERFLOW_CONTROL (where XX shall be replaced by the appropriate two-letter abbreviation for the particular Entity) and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.4.4 POWER DOMAIN CONTROL

This Control shall be supported as Read-Write whenever the Audio Function advertises the existence of at least one Power Domain through the presence of at least one Power Domain descriptor. (It is therefore not necessary to advertise the existence of the Control via the regular **bmControls** mechanism.) Otherwise, it shall not be supported.

The Power Domain Control is used to selectively bring parts of the Audio Function (a Power Domain) into different Power Domain States. The Power Domain Control is effectively an array of individual Controls, one for each defined Power Domain. There shall be a Control present for each Power Domain and each Control shall support the CUR attribute. The RANGE(MIN, MAX, RES) attributes shall not be supported. The CUR attribute can only have positive values ranging from 0 (0x00) to 2 (0x02), indicating Power Domain State D0 to D2.

The Control Selector field shall be set to AC_POWER_DOMAIN_CONTROL and the Channel Number field shall be set to zero (master Control). The Entity ID field shall be set to the Power Domain ID of the desired Power Domain.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.5 INTERFACE CONTROL REQUESTS

These requests are used to manipulate the Controls inside an AudioControl interface of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, “Control Request Parameter Block Layout.”

The following paragraphs present a detailed description of all possible Controls an AudioControl Interface can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.1, “AudioControl Interface Control Selectors.”

5.2.1.5.1 LATENCY CONTROL

An Audio Function shall either not support this Control (D1..0 = 0b00 in the **bmControls** field of the class-specific AudioControl Interface descriptor) or support this Read-Only Control for *every* Terminal and Unit within the Audio Function (D1..0 = 0b01 in the **bmControls** field of the class-specific AudioControl Interface descriptor). Terminal latencies shall include all latencies incurred by A/D or D/A converters, encoders, decoders, etc.

Note: The presence of the Latency Controls is advertised in the class-specific AudioControl Interface descriptor and not repeated in every Terminal and Unit descriptor. Its functionality is described here, although the AudioControl interface by itself does not contain a Latency Control.

The Latency Control is used to accurately report the latency, expressed in nanoseconds, incurred by the addressed Entity. If implemented, this Control shall be Read-Only. A Latency Control shall have only the CUR attribute. The settings for the CUR attribute can range from 0 ns (0x00000000) to 4,294,967,295 ns (0xFFFFFFFF) in steps of 1 ns (0x00000001).

The Control Selector field shall be set to XX_LATENCY_CONTROL (where XX shall be replaced by the appropriate two-letter abbreviation for the particular Entity) and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 3 (See Section 5.2.1.3.3, "Layout 3 Parameter Block.")

5.2.1.6 TERMINAL CONTROL REQUEST

This request is used to manipulate the Controls inside a Terminal of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, "Control Request Parameter Block Layout."

The following paragraphs present a detailed description of all possible Controls a Terminal can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.5, "Terminal Control Selectors."

5.2.1.6.1 INSERTION CONTROL

The Insertion Control is used to examine the insertion state of Connectors that are associated with the Terminal. If implemented, this Control shall be Read-Only. An Insertion Control shall have only the CUR attribute. The CUR attribute returns the **bmaConInserted** field that contains a bitmap where each bit represents the insertion state of one of the Connectors, associated with the Terminal. For this purpose, Connectors are identified by their ordinal number. If bit D_{i-1} is set, then Connector(i) is inserted. If bit D_{i-1} is reset, then Connector(i) is not inserted. Note that bit D_{i-1} only has meaning when bit D2 in the corresponding **bmaConAttributes(i)** field of the Connectors descriptor is set, indicating that Connector(i) is able to report meaningful insertion/removal states. For all Connectors, associated with the Terminal, that are not able to report insertion/removal state, the corresponding bit in the returned **bmaConInserted** bitmap shall always be set to zero. Likewise, all unused padding bits in the **bmConInserted** bitmap shall always be set to zero. The **bSize** field indicates the length in bytes of the **bmConInserted** field.

Note: The Audio Function is responsible to perform insertion detection and perform proper de-bounce so that only stable insertion and removal events are reported to the Host.

The Control Selector field shall be set to TE_INSERTION_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for the CUR attribute of the Insertion Control is as follows:

Table 5-9: Insertion Control CUR Parameter Block

wLength		1+n		
Offset	Field	Size	Value	Description
0	bSize	1	Number	Indicates the number of bytes in the bmConInserted bitmap: n
1	bmConInserted	n	Bitmap	Each set bit represents a currently inserted connector.

5.2.1.6.2 OVERLOAD CONTROL

The Overload Control is used to indicate the existence of an overload condition within a Terminal. (E.g. overvoltage at the analog line-in connector; thermal overload in powered speakers; etc.) If implemented, this Control shall be Read-Only. An Overload Control shall have only the CUR attribute. The value of an Overload Control CUR attribute shall be either TRUE (overload condition exists) or FALSE (normal).

The Control Selector field shall be set to TE_OVERLOAD_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.6.3 UNDERFLOW CONTROL

See Section 5.2.1.4.2, “Underflow Control” for a detailed description. XX shall be replaced by TE.

5.2.1.6.4 OVERFLOW CONTROL

See Section 5.2.1.4.3, “Overflow Control” for a detailed description. XX shall be replaced by TE.

5.2.1.7 MIXER UNIT CONTROL REQUEST

This request is used to manipulate the Controls inside a Mixer Unit of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, “Control Request Parameter Block Layout.”

The **wValue** field specifies the Control Selector (CS) in the high byte and the Mixer Control Number (MCN) or the Channel Number (CN) in the low byte. If the CS value indicates that a Mixer Control is addressed (CS = MIXER_CONTROL), then the low byte contains the MCN. The MCN is derived according to the rules established in Section 4.5.2.5, “Mixer Unit Descriptor.”

The following paragraphs present a detailed description of all possible Controls a Mixer Unit can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.6, “Mixer Control Selectors.”

5.2.1.7.1 MIXER CONTROL

A Mixer Unit consists of a number of N Mixer Controls, either programmable or fixed. At a minimum, all N Mixer Controls shall be present and implemented as Read-Only and support the CUR attribute. If a Mixer Control is implemented as Read-Write (programmable), the RANGE(MIN, MAX, RES) attributes shall also be supported. The settings for the CUR, MIN, and MAX attributes can range from +127.9961 dB (0x7FFF) down to -127.9961 dB (0x8001) in steps of 1/256 dB or 0.00390625 dB (0x0001). The settings for the RES attribute can only have positive values and range from 1/256 dB (0x0001) to +127.9961 dB (0x7FFF).

In addition, code 0x8000, representing silence (i.e., -∞ dB), shall always be implemented. However, it shall never be reported as the MIN attribute value.

The Control Selector field shall be set to MU_MIXER_CONTROL and the Mixer Control Number field shall be set to the MCN of the particular Mixer Control that needs to be addressed.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, “Layout 2 Parameter Block.”)

5.2.1.7.2 UNDERFLOW CONTROL

See Section 5.2.1.4.2, “Underflow Control” for a detailed description. XX shall be replaced by MU.

5.2.1.7.3 OVERFLOW CONTROL

See Section 5.2.1.4.3, “Overflow Control” for a detailed description. XX shall be replaced by MU.

5.2.1.8 SELECTOR UNIT CONTROL REQUEST

This request is used to manipulate the Controls inside a Selector Unit of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, "Control Request Parameter Block Layout."

The following paragraphs present a detailed description of all possible Controls a Selector Unit can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.7, "Selector Control Selectors."

5.2.1.8.1 SELECTOR CONTROL

A Selector Unit represents a multi-channel source selector, capable of selecting among a number of identically configured Clusters. The Selector Control shall be present and at least be implemented as Read-Only. (Means external to USB may change this Control.) A Selector Control shall have only the CUR attribute. The valid range for the CUR attribute is from one up to the number of Input Pins of the Selector Unit. This value can be found in the **bNrInPins** field of the Selector Unit descriptor.

The Control Selector field shall be set to SU_SELECTOR_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.9 FEATURE UNIT CONTROL REQUEST

This request is used to manipulate the Controls inside a Feature Unit of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, "Control Request Parameter Block Layout."

The following paragraphs present a detailed description of all possible Controls a Feature Unit can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.8, "Feature Unit Control Selectors."

5.2.1.9.1 MUTE CONTROL

The Mute Control is one of the building blocks of a Feature Unit. A Mute Control shall have only the CUR attribute. The value of a Mute Control CUR attribute shall be either TRUE (muted) or FALSE (not muted).

A particular Mute Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_MUTE_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.9.2 VOLUME CONTROL

The Volume Control is one of the building blocks of a Feature Unit. A Volume Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, and MAX attributes can range from +127.9961 dB (0x7FFF) down to -127.9961 dB (0x8001) in steps of 1/256 dB or 0.00390625 dB (0x0001). The settings for the RES attribute can only have positive values and range from 1/256 dB (0x0001) to +127.9961 dB (0x7FFF).

In addition, code 0x8000, representing silence (i.e., -∞ dB), shall always be implemented. However, it shall never be reported as the MIN attribute value.

A particular Volume Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_VOLUME_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, "Layout 2 Parameter Block.")

5.2.1.9.3 BASS CONTROL

The Bass Control is one of the building blocks of a Feature Unit. The Bass Control influences the general Bass behavior of the Feature Unit. A Bass Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, and MAX attributes can range from +31.75 dB (0x7F) down to –32.00 dB (0x80) in steps of 0.25 dB (0x01). The settings for the RES attribute can only have positive values and range from 0.25 dB (0x01) to +31.75 dB (0x7F). Other parameters that also influence the behavior of the Bass Control, such as cut-off frequency, cannot be altered through this request.

A particular Bass Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_BASS_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.9.4 MID CONTROL

The Mid Control is one of the building blocks of a Feature Unit. The Mid Control influences the general Mid behavior of the Feature Unit. A Mid Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, and MAX attributes can range from +31.75 dB (0x7F) down to –32.00 dB (0x80) in steps of 0.25 dB (0x01). The settings for the RES attribute can only have positive values and range from 0.25 dB (0x01) to +31.75 dB (0x7F). Other parameters that also influence the behavior of the Mid Control, such as cut-off frequency, cannot be altered through this request.

A particular Mid Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_MID_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.9.5 TREBLE CONTROL

The Treble Control is one of the building blocks of a Feature Unit. The Treble Control influences the general Treble behavior of the Feature Unit. A Treble Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, and MAX attributes can range from +31.75 dB (0x7F) down to –32.00 dB (0x80) in steps of 0.25 dB (0x01). The settings for the RES attribute can only have positive values and range from 0.25 dB (0x01) to +31.75 dB (0x7F). Other parameters that also influence the behavior of the Treble Control, such as cut-off frequency, cannot be altered through this request.

A particular Treble Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_TREBLE_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.9.6 GRAPHIC EQUALIZER CONTROL

The Graphic Equalizer Control is one of the optional building blocks of a Feature Unit. The Audio Device Class definition provides for standard support of a third octave graphic equalizer. The bands are defined according to the ANSI S1.11-1986 standard. Bands are numbered from 14 (center frequency of 25 Hz) up to 43 (center frequency of 20,000 Hz), making a total of 30 possible bands. The following table lists the band numbers and their center frequencies

Table 5-10: Band Numbers and Center Frequencies (ANSI S1.11-1986 Standard)

Band Nr.	Center Freq.	Band Nr.	Center Freq.	Band Nr.	Center Freq.
14	25 Hz	24*	250 Hz	34	2500 Hz
15*	31.5 Hz	25	315 Hz	35	3150 Hz
16	40 Hz	26	400 Hz	36*	4000 Hz
17	50 Hz	27*	500 Hz	37	5000 Hz
18*	63 Hz	28	630 Hz	38	6300 Hz
19	80 Hz	29	800 Hz	39*	8000 Hz
20	100 Hz	30*	1000 Hz	40	10000 Hz
21*	125 Hz	31	1250 Hz	41	12500 Hz
22	160 Hz	32	1600 Hz	42*	16000 Hz
23	200 Hz	33*	2000 Hz	43	20000 Hz

Note: Bands marked with an asterisk (*) are those present in an octave equalizer.

A Feature Unit that supports the Graphic Equalizer Control is not required to implement the full set of filters. A subset (for example, octave bands) may be implemented. During a Get Control request, the **bmBandsPresent** field in the parameter block is a bitmap indicating which bands are effectively implemented and thus reported back in the returned parameter block. Consequently, the number of bits set in this field determines the total length of the returned parameter block. During a Set Control request, a bit set in the **bmBandsPresent** field indicates there is a new setting for that band in the parameter block that follows. The new values shall be in ascending order. If the number of bits set in the **bmBandsPresent** field does not match the number of parameters specified in the following block, the control pipe shall indicate a stall.

A Graphic Equalizer Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, and MAX attributes can range from +31.75 dB (0x7F) down to –32.00 dB (0x80) in steps of 0.25 dB (0x01). The settings for the RES attribute can only have positive values and range from 0.25 dB (0x01) to +31.75 dB (0x7F).

A particular Graphic Equalizer Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_GRAPHIC_EQUALIZER_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for the CUR attribute of the Graphic Equalizer Control is as follows:

Table 5-11: Graphic Equalizer Control CUR Parameter Block

wLength		4+(number of bits set in **bmBandsPresent** : NrBits)		
Offset	Field	Size	Value	Description
0	bmBandsPresent	4	Bitmap	A bit set indicates the band is present: D0: Band 14 is present. D1: Band 15 is present. … D29: Band 43 is present. D30: Reserved. D31: Reserved.
4	bCUR(Lowest)	1	Number	The setting for the CUR attribute of the lowest band present: 0x7F: +31.75 dB. 0x7E: +31.50 dB. … 0x00: 0.00 dB. … 0x82: -31.50 dB. 0x81: -31.75 dB. 0x80: -32.00 dB.
…	…	…	…	…
4+(NrBits-1)	bCUR(Highest)	1	Number	The setting for the CUR attribute of the highest band present.

The parameter block for the RANGE attribute of the Graphic Equalizer Control is as follows:

Table 5-12: Graphic Equalizer Control RANGE Parameter Block

wLength		2+3*n		
Offset	Field	Size	Value	Description
0	wNumSubRanges	2	Number	The number of subranges of the Graphic Equalizer Control: n

2	bMIN(1)	1	Number	The setting for the MIN attribute of the first subrange of the Graphic Equalizer Control
3	bMAX(1)	1	Number	The setting for the MAX attribute of the first subrange of the Graphic Equalizer Control
4	bRES(1)	1	Number	The setting for the RES attribute of the first subrange of the Graphic Equalizer Control
…	…	…	…	…
2+3*(n-1)	bMIN(n)	1	Number	The setting for the MIN attribute of the last subrange of the Graphic Equalizer Control
3+3*(n-1)	bMAX(n)	1	Number	The setting for the MAX attribute of the last subrange of the Graphic Equalizer Control
4+3*(n-1)	bRES(n)	1	Number	The setting for the RES attribute of the last subrange of the Graphic Equalizer Control

5.2.1.9.7 AUTOMATIC GAIN CONTROL

The Automatic Gain Control (AGC) is one of the building blocks of a Feature Unit. An Automatic Gain Control shall have only the CUR attribute. The value of an Automatic Gain Control CUR attribute shall be either TRUE or FALSE.

A particular Automatic Gain Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_AUTOMATIC_GAIN _CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.9.8 DELAY CONTROL

The Delay Control is one of the building blocks of a Feature Unit. A Delay Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX, and RES attributes are expressed in seconds using a 10.22 format (32 bits). Therefore, they range from zero (0x00000000) to 1023.999999761581 s (0xFFFFFFFF) in steps of 1/4194304 s (0x00000001).

A particular Delay Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_DELAY_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 3 (See Section 5.2.1.3.3, "Layout 3 Parameter Block.")

5.2.1.9.9 BASS BOOST CONTROL

The Bass Boost Control is one of the building blocks of a Feature Unit. A Bass Boost Control shall have only the CUR attribute. The position of a Bass Boost Control CUR attribute shall be either TRUE or FALSE.

A particular Bass Boost Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_BASS_BOOST_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.9.10 LOUDNESS CONTROL

The Loudness Control is one of the building blocks of a Feature Unit. A Loudness Control shall have only the CUR attribute. The value of a Loudness Control CUR attribute shall be either TRUE or FALSE.

A particular Loudness Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_LOUDNESS_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.9.11 INPUT GAIN CONTROL

The Input Gain Control is one of the building blocks of a Feature Unit. An Input Gain Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, and MAX attributes can range from +127.9961 dB (0x7FFF) down to -127.9961 dB (0x8001) in steps of 1/256 dB or 0.00390625 dB (0x0001). The settings for the RES attribute can only have positive values and range from 1/256 dB (0x0001) to +127.9961 dB (0x7FFF).

A particular Input Gain Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_INPUT_GAIN_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, "Layout 2 Parameter Block.")

5.2.1.9.12 INPUT GAIN PAD CONTROL

The Input Gain Pad Control is one of the building blocks of a Feature Unit. An Input Gain Pad Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, and MAX attributes can range from +127.9961 dB (0x7FFF) down to -127.9961 dB (0x8001) in steps of 1/256 dB or 0.00390625 dB (0x0001). The settings for the RES attribute can only have positive values and range from 1/256 dB (0x0001) to +127.9961 dB (0x7FFF).

A particular Input Gain Pad Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the 'master' channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_INPUT_GAIN_PAD_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, “Layout 2 Parameter Block.”)

5.2.1.9.13 PHASE INVERTER CONTROL

The Phase Inverter Control is one of the building blocks of a Feature Unit. A Phase Inverter Control shall have only the CUR attribute. The value of a Phase Inverter Control CUR attribute shall be either TRUE or FALSE.

A particular Phase Inverter Control within a Feature Unit is addressed through the Unit ID and Channel Number fields of the Set/Get Feature Unit Control request. The valid range for the Channel Number field is from zero (the ‘master’ channel) up to the number of logical channels in the Cluster.

The Control Selector field shall be set to FU_PHASE_INVERTER_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.9.14 UNDERFLOW CONTROL

See Section 5.2.1.4.2, “Underflow Control” for a detailed description. XX shall be replaced by FU.

5.2.1.9.15 OVERFLOW CONTROL

See Section 5.2.1.4.3, “Overflow Control” for a detailed description. XX shall be replaced by FU.

5.2.1.10 EFFECT UNIT CONTROL REQUEST

This request is used to manipulate the Controls inside an Effect Unit of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, “Control Request Parameter Block Layout.”

The following paragraphs present a detailed description of all possible Controls an Effect Unit can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.9, “Effect Unit Control Selectors.”

5.2.1.10.1 PARAMETRIC EQUALIZER SECTION EFFECT UNIT

5.2.1.10.1.1 ENABLE CONTROL

This Control shall be present and implemented as Read-Write. See Section 5.2.1.4.1, “Enable Control” for a detailed description. XX shall be replaced by PE.

5.2.1.10.1.2 CENTER FREQUENCY CONTROL

The Center Frequency Control is used to manipulate the actual center frequency of the PEQS Effect Unit.

The Center Frequency Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX and RES attributes can range from 0 Hz (0x00000000) to 4,294,967,295 Hz (0xFFFFFFFF) in steps of 1 Hz (0x00000001).

Note: A discrete list of supported center frequencies can be expressed using the method as explained in Section 5.2.1.1, “Control Attributes.”

The Control Selector field shall be set to PE_CENTER_FREQ_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 3 (See Section 5.2.1.3.3, “Layout 3 Parameter Block.”)

5.2.1.10.1.3 QFACTOR CONTROL

The Qfactor Control is used to manipulate the range of frequencies affected around the center frequency of the PEQS Effect Unit.

The Qfactor Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX and RES attributes can range from 0 (0x000.00000 – 12.20 format) to 4,095.999999046326 (0xFFF.FFFFF) in steps of 0.000000953674316406 (0x000.00001).

The Control Selector field shall be set to PE_QFACTOR_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 3 (See Section 5.2.1.3.3, “Layout 3 Parameter Block.”)

Note: The Q-factor of a filter is defined as the ratio of the center frequency to the bandwidth measured at the -3 dB point. The result of a Q setting of 10 for a filter set to 1000 Hz is a bandwidth of 100 Hz. Likewise, a center frequency of 5325 Hz and a Q setting of 7.25 results in a bandwidth of 734.48275862 Hz.

5.2.1.10.1.4 GAIN CONTROL

The Gain Control is used to manipulate the amount of gain or attenuation at the center frequency of the PEQS Effect Unit. A Gain Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, and MAX attributes can range from +127.9961 dB (0x7FFF) down to -127.9961 dB (0x8001) in steps of 1/256 dB or 0.00390625 dB (0x0001). The settings for the RES attribute can only have positive values and range from 1/256 dB (0x0001) to +127.9961 dB (0x7FFF).

The Control Selector field shall be set to PE_GAIN_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, “Layout 2 Parameter Block.”)

5.2.1.10.1.5 UNDERFLOW CONTROL

See Section 5.2.1.4.2, “Underflow Control” for a detailed description. XX shall be replaced by PE.

5.2.1.10.1.6 OVERFLOW CONTROL

See Section 5.2.1.4.3, “Overflow Control” for a detailed description. XX shall be replaced by PE.

5.2.1.10.2 REVERBERATION EFFECT UNIT

5.2.1.10.2.1 ENABLE CONTROL

This Control shall be present and implemented as Read-Write. See Section 5.2.1.4.1, “Enable Control” for a detailed description. XX shall be replaced by RV.

5.2.1.10.2.2 TYPE CONTROL

The Type Control is a macro parameter that allows global settings of reverb parameters within the Reverberation Effect Unit. When a certain reverb type is selected, each reverb parameter will be set to the most suitable value.

The Type Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The valid range for the CUR, MIN, and MAX attributes is from 0 to 255. The RES attribute can only have a value of 1.

The CUR attribute subrange from 0 to 7 has predefined behavior:

0:	Room 1	– simulates the reverberation of a small room.
1:	Room 2	– simulates the reverberation of a medium room.
2:	Room 3	– simulates the reverberation of a large room.
3:	Hall 1	– simulates the reverberation of a medium concert hall.
4:	Hall 2	– simulates the reverberation of a large concert hall.
5:	Plate	– simulates a plate reverberation (a studio device using a metal plate).
6:	Delay	– conventional delay that produces echo effects.
7:	Panning Delay	– special delay in which the delayed sounds move left and right.

The Control Selector field shall be set to RV_TYPE_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.10.2.3 LEVEL CONTROL

The Level Control is used to set the amount of reverberant sound introduced by the Reverberation Effect Unit. The Level Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The valid range for the CUR, MIN, MAX, and RES attributes is from 0 % to 255 %, compared to the level of the original signal.

The Control Selector field shall be set to RV_LEVEL_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.10.2.4 TIME CONTROL

The Time Control is used to set the time over which the reverberation, introduced by the Reverberation Effect Unit, will continue. The Time Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX, and RES attributes can range from 0 s (0x0000) to 255.9961 s (0xFFFF) in steps of 1/256 s or 0.00390625 s (0x0001).

The Control Selector field shall be set to RV_TIME_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, “Layout 2 Parameter Block.”)

5.2.1.10.2.5 DELAY FEEDBACK CONTROL

The Delay Feedback Control is used when the reverb type is set to Type 6 (Delay) or Type 7 (Panning Delay). It sets the way in which delay repeats. The Delay Feedback Control range shall support the CUR and RANGE(MIN, MAX, RES) attributes. The valid range for the CUR, MIN, MAX, and RES attributes is from 0 % to 255 %. Higher values result in more delay repeats.

Note: In practice, the delay feedback amount should be limited to 75 % to avoid unexpected feedback distortion and continuous delay loop.

The Control Selector field shall be set to RV_FEEDBACK_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.10.2.6 PRE-DELAY CONTROL

The Pre-Delay Control is used to set the delay time between the original source and the initial reflection in the reverberation, introduced by the Reverberation Effect Unit. The Pre-Delay Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX, and RES attributes can range from 0 ms (0x0000) to 65535 ms (0xFFFF) in steps of 1 ms (0x0001).

The Control Selector field shall be set to RV_PREDELAY_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, “Layout 2 Parameter Block.”)

5.2.1.10.2.7 DENSITY CONTROL

The Density Control is used to set the density of reflections in the reverberant sound introduced by the Reverberation Effect Unit. The Density Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The valid range for the CUR, MIN, MAX, and RES attributes is from 0 (0x00) to 100 (0x64) in steps of 1 (0x01).

The Control Selector field shall be set to RV_DENSITY_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.10.2.8 HI-FREQ ROLL-OFF CONTROL

The Hi-Freq Roll-Off Control is used to set the frequency of a low pass filter on the reverberant sound introduced by the Reverberation Effect Unit. The Hi-Freq Roll-Off Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX and RES attributes can range from 0 Hz (0x00000000) to 4,294,967,295 Hz (0xFFFFFFFF) in steps of 1 Hz (0x00000001).

The Control Selector field shall be set to RV_HIFREQ_ROLLOFF_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 3 (See Section 5.2.1.3.3, “Layout 3 Parameter Block.”)

5.2.1.10.2.9 UNDERFLOW CONTROL

See Section 5.2.1.4.2, “Underflow Control” for a detailed description. XX shall be replaced by RV.

5.2.1.10.2.10 OVERFLOW CONTROL

See Section 5.2.1.4.3, “Overflow Control” for a detailed description. XX shall be replaced by RV.

5.2.1.10.3 MODULATION DELAY EFFECT UNIT

5.2.1.10.3.1 ENABLE CONTROL

See Section 5.2.1.4.1, “Enable Control” for a detailed description. XX shall be replaced by MD.

5.2.1.10.3.2 BALANCE CONTROL

The Balance Control is used to set the amount of effect sound introduced by the Modulation Delay Effect Unit. The Balance Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The valid range for the CUR, MIN, MAX attributes is from -100 % to +100 %. A setting of -100 % results in 100 % of original signal and 0 % of effected signal. A setting of 0 % results in equal amounts of original signal and effected signal. A setting of +100 % results in 0 % of original signal and 100 % of effected signal. The settings for the RES attribute can only have positive values and range from 1 % (0x01) to 100 % (0x64).

The Control Selector field shall be set to MD_BALANCE_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.10.3.3 RATE CONTROL

The Rate Control is used to set the speed (frequency) of the modulator of the delay time introduced by the Modulation Delay Effect Unit. Higher values result in faster modulation. The Rate Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX, and RES attributes can range from 0 Hz (0x0000) to 255.9961 Hz (0xFFFF) in steps of 1/256 Hz or 0.00390625 Hz (0x0001).

The Control Selector field shall be set to MD_RATE_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, “Layout 2 Parameter Block.”)

5.2.1.10.3.4 DEPTH CONTROL

The Depth Control is used to set the depth at which the effect sound introduced by the Modulation Delay Effect Unit is modulated. Higher values result in deeper modulation. The Depth Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX, and RES attributes can range from 0 ms (0x0000) to 255.9961 ms (0xFFFF) in steps of 1/256 ms or 0.00390625 ms (0x0001).

The Control Selector field shall be set to MD_DEPTH_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, “Layout 2 Parameter Block.”)

5.2.1.10.3.5 TIME CONTROL

The Time Control is used to set the length of the delay time introduced by the Modulation Delay Efect Unit. Higher values result in longer times. The Time Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX, and RES attributes can range from 0 ms (0x0000) to 255.9961 ms (0xFFFF) in steps of 1/256 ms or 0.00390625 ms (0x0001).

The Control Selector field shall be set to MD_TIME_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, “Layout 2 Parameter Block.”)

5.2.1.10.3.6 FEEDBACK LEVEL CONTROL

The Feedback Level Control is used to set the level at which the effected (delayed) sound introduced by the Modulation Delay Effect Unit is mixed back into to its own input. Higher values result in higher level of feedback.

The Feedback Level Control range shall support the CUR and RANGE(MIN, MAX, RES) attributes. The valid range for the CUR, MIN, MAX, and RES attributes is from 0 % to 255 %. Higher values result in more delay repeats.

Note: In practice, the modulation delay feedback amount should be limited to 75 % to avoid unexpected feedback distortion and continuous delay loop.

The Control Selector field shall be set to MD_FEEDBACK_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.10.3.7 UNDERFLOW CONTROL

See Section 5.2.1.4.2, "Underflow Control" for a detailed description. XX shall be replaced by MD.

5.2.1.10.3.8 OVERFLOW CONTROL

See Section 5.2.1.4.3, "Overflow Control" for a detailed description. XX shall be replaced by MD.

5.2.1.10.4 DYNAMIC RANGE COMPRESSOR EFFECT UNIT

5.2.1.10.4.1 ENABLE CONTROL

See Section 5.2.1.4.1, "Enable Control" for a detailed description. XX shall be replaced by DR.

5.2.1.10.4.2 COMPRESSION RATIO CONTROL

The Compression Ratio Control is used to influence the slope of the active part of the static input-to-output transfer characteristic of the Dynamic Range Compressor Processing Unit. The Compression Ratio Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The valid range for the CUR, MIN, MAX, and RES attributes is from 0 (0x0000) to 255.9961 (0xFFFF) in steps of 1/256 or 0.00390625 (0x0001).

The Control Selector field shall be set to DR_COMPRESSION_RATIO_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, "Layout 2 Parameter Block.")

5.2.1.10.4.3 MAXAMPL CONTROL

The MaxAmpl Control is used to set the upper boundary of the active input range of the compressor. The MaxAmpl Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, and MAX attributes can range from -128.0000 dB (0x8000) to +127.9961 dB (0x7FFF) in steps of 1/256 dB or 0.00390625 dB (0x0001). The settings for the RES attribute can only have positive values and range from 1/256 dB (0x0001) to +127.9961 dB (0x7FFF).

The Control Selector field shall be set to DR_MAXAMPL_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, "Layout 2 Parameter Block.")

5.2.1.10.4.4 THRESHOLD CONTROL

The Threshold Control is used to set the lower boundary of the active input range of the compressor. The Threshold Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, and

MAX attributes can range from -128.0000 dB (0x8000) to +127.9961 dB (0x7FFF) in steps of 1/256 dB or 0.00390625 dB (0x0001). The settings for the RES attribute can only have positive values and range from 1/256 dB (0x0001) to +127.9961 dB (0x7FFF).

The Control Selector field shall be set to DR_TRESHOLD_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, "Layout 2 Parameter Block.")

5.2.1.10.4.5 ATTACK TIME CONTROL

The Attack Time Control is used to determine the response of the compressor to a step increase in the input signal level. The Attack Time Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX, and RES attributes can range from 0 ms (0x0000) to 255.9961 ms (0xFFFF) in steps of 1/256 ms or 0.00390625 ms (0x0001).

The Control Selector field shall be set to DR_ATTACK_TIME_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, "Layout 2 Parameter Block.")

5.2.1.10.4.6 RELEASE TIME CONTROL

The Release Time Control is used to determine the recovery response of the compressor to a step decrease in the input signal level. The Release Time Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX, and RES attributes can range from 0 ms (0x0000) to 255.9961 ms (0xFFFF) in steps of 1/256 ms or 0.00390625 ms (0x0001).

The Control Selector field shall be set to DR_RELEASE_TIME_CONTROL and the Channel Number field indicates the desired Channel.

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, "Layout 2 Parameter Block.")

5.2.1.10.4.7 UNDERFLOW CONTROL

See Section 5.2.1.4.2, "Underflow Control" for a detailed description. XX shall be replaced by DR.

5.2.1.10.4.8 OVERFLOW CONTROL

See Section 5.2.1.4.3, "Overflow Control" for a detailed description. XX shall be replaced by DR.

5.2.1.11 PROCESSING UNIT CONTROL REQUEST

This request is used to manipulate the Controls inside a Processing Unit of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, "Control Request Parameter Block Layout."

The following paragraphs present a detailed description of all possible Controls a Processing Unit can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.10, "Processing Unit Control Selectors."

5.2.1.11.1 UP/DOWN-MIX PROCESSING UNIT

5.2.1.11.1.1 MODE SELECT CONTROL

The Mode Select Control is used to change the behavior of the Up/Down-mix Processing Unit. A Mode Select Control shall have only the CUR attribute. The valid range for the CUR attribute is from one to the number of modes, supported by the Entity (reported through the **bNrModes** field of the Up/Down-mix Processing Unit descriptor).

The Control Selector field shall be set to UD_MODE_SELECT_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.11.1.2 UNDERFLOW CONTROL

See Section 5.2.1.4.2, "Underflow Control" for a detailed description. XX shall be replaced by UD.

5.2.1.11.1.3 OVERFLOW CONTROL

See Section 5.2.1.4.3, "Overflow Control" for a detailed description. XX shall be replaced by UD.

5.2.1.11.2 STEREO EXTENDER PROCESSING UNIT

5.2.1.11.2.1 WIDTH CONTROL

The Width Control is used to change the spatial appearance of the stereo image, produced by the Stereo Extender Processing Unit. The Width Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The valid range for the CUR, MIN, MAX, and RES attributes is from 0 to 255.

The Control Selector field shall be set to ST_EXT_WIDTH_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, "Layout 1 Parameter Block.")

5.2.1.11.2.2 UNDERFLOW CONTROL

See Section 5.2.1.4.2, "Underflow Control" for a detailed description. XX shall be replaced by ST.

5.2.1.11.2.3 OVERFLOW CONTROL

See Section 5.2.1.4.3, "Overflow Control" for a detailed description. XX shall be replaced by ST.

5.2.1.12 EXTENSION UNIT CONTROL REQUESTS

Because this specification has no knowledge about the inner workings of an Extension Unit, it is impossible to define requests that are able to manipulate specific Extension Unit Controls. However, this specification defines a number of Controls an Extension Unit shall or can support.

This request is used to manipulate the Controls inside an Extension Unit of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, "Control Request Parameter Block Layout."

The following paragraphs present a detailed description of all possible Controls a Processing Unit can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control

Selector value and the layout type of the parameter blocks are listed. Issuing non-supported Control Selectors to an Extension Unit leads to a control pipe stall. The Control Selector codes are defined in Appendix A.23.11, “Extension Unit Control Selectors.”

5.2.1.12.1.1 UNDERFLOW CONTROL

See Section 5.2.1.4.2, “Underflow Control” for a detailed description. XX shall be replaced by XU.

5.2.1.12.1.2 OVERFLOW CONTROL

See Section 5.2.1.4.3, “Overflow Control” for a detailed description. XX shall be replaced by XU.

5.2.1.13 CLOCK SOURCE CONTROL REQUEST

This request is used to manipulate the Controls inside a Clock Source Entity of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, “Control Request Parameter Block Layout.”

The following paragraphs present a detailed description of all possible Controls a Clock Source Entity can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.2, “Clock Source Control Selectors.”

5.2.1.13.1 SAMPLING FREQUENCY CONTROL

The Sampling Frequency Control is used to manipulate the actual sampling frequency of the clock signal that is generated by the Clock Source Entity. At a minimum, this Control shall be supported as Read-Only by every Clock Source Entity.

The Sampling Frequency Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX and RES attributes can range from 0 Hz (0x00000000) to 4,294,967,295 Hz (0xFFFFFFFF) in steps of 1 Hz (0x00000001).

Note: A discrete list of supported sampling frequencies can be expressed using the method as explained in Section 5.2.1.1, “Control Attributes.”

In many cases, the Clock Source Entity represents a crystal oscillator based generator with a single fixed frequency. In that case, the Set request is not supported. Additionally, the Set request may not be supported if the Clock Source entity represents an external clock which cannot be controlled by the Audio Function hardware.

The Control Selector field shall be set to CS_SAM_FREQ_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 3 (See Section 5.2.1.3.3, “Layout 3 Parameter Block.”)

5.2.1.13.2 CLOCK VALIDITY CONTROL

The Clock Validity Control is used to indicate if the clock signal that is generated by the Clock Source Entity is valid (stable and reliable). Only the Get request is supported for this Control. The Clock Validity Control shall have only the CUR attribute. The value of a Clock Validity Control CUR attribute shall be either TRUE (clock valid) or FALSE (clock invalid).

The Control Selector field shall be set to CS_CLOCK_VALID_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.14 CLOCK SELECTOR CONTROL REQUEST

This request is used to manipulate the Controls inside a Clock Selector Entity of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, “Control Request Parameter Block Layout.”

The following paragraphs present a detailed description of all possible Controls a Clock Selector Entity can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.3,”Clock Selector Control Selectors.”

5.2.1.14.1 CLOCK SELECTOR CONTROL

A Clock Selector Entity represents a multi-input source selector, capable of selecting among a number of clock signals. The Clock Selector Control shall be present and at least be implemented as Read-Only. (Means external to USB may change this Control.) The valid range for the CUR attribute is from one up to the number of Clock Input Pins of the Clock Selector. This value can be found in the **bNrInPins** field of the Clock Selector descriptor.

The Control Selector field shall be set to CX_CLOCK_SELECTOR_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.1.15 CLOCK MULTIPLIER CONTROL REQUEST

This request is used to manipulate the Controls inside a Clock Multiplier Entity of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, “Control Request Parameter Block Layout.”

The following paragraphs present a detailed description of all possible Controls a Clock Multiplier Entity can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.4,”Clock Multiplier Control Selectors.”

5.2.1.15.1 NUMERATOR CONTROL

At a minimum, the Numerator Control shall be supported as Read-Only. It is used to manipulate the factor P by which the incoming sampling clock signal is multiplied. This Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX and RES attributes can range from 1 (0x0001) to 2^{16}-1 (0xFFFF).

The Control Selector field shall be set to CM_ NUMERATOR_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, “Layout 2 Parameter Block.”)

5.2.1.15.2 DENOMINATOR CONTROL

At a minimum, the Denominator Control shall be supported as Read-Only. It is used to manipulate the factor Q by which the incoming sampling clock signal is divided. This Control shall support the CUR and RANGE(MIN, MAX, RES) attributes. The settings for the CUR, MIN, MAX and RES attributes can range from 1 (0x0001) to 2^{16}-1 (0xFFFF).

The Control Selector field shall be set to CM_ DENOMINATOR_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 2 (See Section 5.2.1.3.2, “Layout 2 Parameter Block.”)

5.2.2 AUDIOSTREAMING REQUESTS

The following sections describe the requests an Audio Function can support for its AudioStreaming interfaces. The same layout of the parameter blocks is used for both the Set and Get requests.

AudioStreaming requests can be directed either to the AudioStreaming interface or to the associated isochronous data endpoint, depending on the location of the Control to be manipulated.

5.2.2.1 INTERFACE CONTROL REQUESTS

These requests are used to manipulate the Controls inside an AudioStreaming interface of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, “Control Request Parameter Block Layout.”

The following paragraphs present a detailed description of all possible Controls an AudioStreaming Interface can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in Appendix A.23.12, “AudioStreaming Interface Control Selectors.”

5.2.2.1.1 ACTIVE ALTERNATE SETTING CONTROL

This Control is used to inform Host software which Alternate Setting of an AudioStreaming interface is currently active. The main purpose of this Control is to notify the Host (through an interrupt) that the last selected Alternate Setting is no longer valid. There can be a variety of reasons why this could happen. For instance, increasing the sampling frequency of a Clock Source Entity might render the current Alternate Setting of an interface connected to that Clock Source invalid because more bandwidth is now needed than is available in the current Alternate Setting.

This specification does not allow an interface to change from one active Alternate Setting to another without Host intervention. Whenever an Alternate Setting becomes invalid, the interface is required to switch to (idle) Alternate Setting zero. If this situation may occur in the Audio Function, this Control (and the Valid Alternate Settings Control) shall be present and implemented as Read-Only. It always provides the currently active Alternate Setting for the interface. The Host software needs to then take appropriate action to reactivate the interface by switching to a valid Alternate Setting. An Active Alternate Setting Control shall have only the CUR attribute. The value of an Active Alternate Setting Control CUR attribute shall only be either the last set Alternate Setting or zero.

The Control Selector field shall be set to AS_ACT_ALT_SETTING_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.2.1.2 VALID ALTERNATE SETTINGS CONTROL

This Control is used to inform Host software what the currently possible valid Alternate Settings are for an AudioStreaming interface. If the Active Alternate Setting Control is present, then this Control shall also be present and be implemented as Read-Only. It always provides a list of currently valid Alternate Settings for the interface. A Valid Alternate Settings Control shall have only the CUR attribute. The value of a Valid Alternate Setting Control CUR attribute is a bitmap, returned in the **bmValidAltSettings** field, that contains a bit for each possible active Alternate Setting. The **bSize** field indicates the length in bytes of the **bmValidAltSettings** field.

A bit set means that this Alternate Setting is currently valid. A bit cleared means that this Alternate Setting is currently not valid. Bit D0 corresponds to Alternate Setting 0 and shall always be set since it is always a possible valid setting. Bit D1 corresponds to Alternate Setting 1. Bit Dm corresponds to Alternate Setting m. All bits that do not correspond to an existing Alternate Setting shall be set to 0. An attempt to set the interface to an invalid Alternate Setting (through the standard Set Interface request) will result in a control pipe stall.

The Control Selector field shall be set to AS_VAL_ALT_SETTINGS_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for the CUR attribute of the Valid Alternate Settings Control is as follows:

Table 5-13: Valid Alternate Settings Control CUR Parameter Block

wLength		1+n		
Offset	Field	Size	Value	Description
0	bSize	1	Number	Indicates the number of bytes in the bmValidAltSettings bitmap: n
1	bmValidAltSettings	n	Bitmap	Each set bit represents a currently valid Alternate Setting for the interface.

5.2.2.1.3 AUDIO DATA FORMAT CONTROL

The Audio Data Format Control is used to indicate which Audio Data Format is currently being used by the AudioStreaming interface. Only the Get request is supported for this Control. The Audio Data Format Control shall have only the CUR attribute. The returned value for the Audio Data Format Control CUR attribute follows the definition of the **bmFormats** field in the class-specific AS interface descriptor. Only one bit can be set in the bitmap, indicating exactly which Audio Data Format is being used by the interface.

The Control Selector field shall be set to AS_AUDIO_DATA_FORMAT_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request is as follows:

Table 5-14: Audio Data Format Control CUR Parameter Block

wLength		8		
Offset	Field	Size	Value	Description
0	bmFormats	8	Bitmap	The Audio Data Format(s) that can be used to communicate with this interface. See the *USB Audio Data Formats* document for further details.

5.2.2.2 ENDPOINT CONTROL REQUEST

This request is used to manipulate the Controls inside an AudioStreaming endpoint of the Audio Function. The exact layout of the request is defined in Section 5.2.1.3, "Control Request Parameter Block Layout."

The following paragraphs present a detailed description of all possible Controls an Endpoint can incorporate. For each Control, the supported attributes and their value ranges are specified. Also, the appropriate Control Selector value and the layout type of the parameter blocks are listed. The Control Selector codes are defined in AppendixA.23.13, "Endpoint Control Selectors."

5.2.2.2.1 PITCH CONTROL

The Pitch Control enables or disables the ability of an adaptive endpoint to dynamically track its sampling frequency. The Control is necessary because the clock recovery circuitry must be informed whether it should allow for relatively large swings in the sampling frequency. A Pitch Control shall have only the CUR attribute. The value of a Pitch Control CUR attribute shall be either TRUE or FALSE.

The Control Selector field shall be set to EP_PITCH_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.2.2.2 DATA OVERRUN CONTROL

The Data Overrun Control is used to indicate the occurrence of a data overrun (buffer overflow) condition within an Endpoint *since the last Get Data Overrun request*. If implemented, this Control shall be Read-Only. A Data Overrun Control shall have only the CUR attribute. The value of a Data Overrun Control CUR attribute shall be either TRUE (overrun condition occurred) or FALSE (normal).

The Control Selector field shall be set to EP_DATA_OVERRUN_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.2.2.3 DATA UNDERRUN CONTROL

The Data Underrun Control is used to indicate the occurrence of a data underrun (buffer underflow) condition within an Endpoint *since the last Get Data Underrun request*. If implemented, this Control shall be Read-Only. A Data Underrun Control shall have only the CUR attribute. The value of a Data Underrun Control CUR attribute shall be either TRUE (underrun condition occurred) or FALSE (normal).

The Control Selector field shall be set to EP_DATA_UNDERRUN_CONTROL and the Channel Number field shall be set to zero (master Control).

The parameter block for this Control request uses Layout 1 (See Section 5.2.1.3.1, “Layout 1 Parameter Block.”)

5.2.3 ADDITIONAL REQUESTS

5.2.3.1 MEMORY REQUESTS

The Host can interact with an addressable Entity (Clock Entity, Terminal, Unit, Power domain, Interface or endpoint) within the Audio Function in a very generic way. The Entity presents a memory space to the Host whose layout depends on the implementation. The Memory request provides full access to this memory space.

This request is used to upload or download a parameter block into a particular Entity of the Audio Function.

Table 5-15: Memory Request Values

bmRequest Type	bRequest	wValue	wIndex	wLength	Data
00100001B 10100001B	MEM INTEN	Offset	Entity ID and Interface	Length of parameter block	Parameter block
00100010B 10100010B			Zero and Endpoint		

The **bRequest** field indicates that the MEM or INTEN attribute of the Entity is addressed.

The **wValue** field specifies a zero-based offset value that can be used to access only parts of the Entity's memory space.

The **wIndex** field specifies the interface or endpoint to be addressed in the low byte and the Entity ID (Clock Entity ID, Unit ID, Terminal ID, or Power Domain ID) or zero in the high byte. In case an interface is addressed, the virtual Entity 'interface' can be addressed by specifying zero in the high byte. The values in the **wIndex** field shall be appropriate to the recipient. Only existing Entities in the Audio Function can be addressed and only appropriate interface or endpoint numbers may be used. If the request specifies an unknown or non-Entity ID or an unknown interface or endpoint number, the control pipe shall indicate a stall.

The layout of the parameter block is implementation dependent. A device is required to reevaluate its memory space at the end of each Set Memory request.

The Interrupt Enable attribute is used to manipulate the memory space's ability to generate an interrupt when any of its memory locations change other than through Host manipulation. The Interrupt Enable attribute shall be supported for all Entity memory spaces that are able to generate an interrupt. When supported, the Interrupt Enable attribute is always Read-Write and therefore, both the Get and Set request shall be supported. The Interrupt Enable attribute can take only two values: 0 (False, Disabled) or 1 (True, Enabled). All Interrupt Enable attributes shall have a default value of 1 (Enabled).

5.2.3.2 CLASS-SPECIFIC STRING REQUEST

To overcome the limitations of the standard Get String descriptor request (limited to 255 device-wide strings), this specification provides a class-specific extension to retrieve a larger set of potentially larger String descriptors from the Audio Function.

The Get Class-specific String request is used to retrieve class-specific String descriptors and shall be supported if the Audio Function contains at least one class-specific String descriptor. These String descriptors are uniquely identified within the Audio Function through their **wStrDescrID** and **iLangID** values.

Table 5-16: String Request

bmRequest Type	bRequest	wValue	wIndex	wLength	Data
10100001B	STRING	wStrDescrID	iLangID and Interface	Length of parameter block	Parameter block

The **bmRequestType** field shall be set to 0b10100001 to indicate that this is a class-specific Get Request, directed to the AudioControl interface.

The **bRequest** field indicates that the class-specific set of String descriptors is targeted.

The **wValue** field specifies the String descriptor's **wStrDescrID**. For maximum interoperability between this class-specific method to retrieve String descriptors from the Audio Function and the standard Get String descriptor request, all Audio Function related standard String descriptors, including the LANGID code array at index 0, that can be retrieved using the standard Get String descriptor request shall also be retrievable using this class-specific String request by specifying zero in the high byte and the standard String descriptor index in the low byte of the **wStrDescrID** value. Whether other than Audio Function related standard String descriptors present in the device can be retrieved using this new method is implementation-dependent. Any newly defined class-specific String descriptor that uses the **wStrDescrID** value as an index, shall use **wStrDescrID** values starting from 256 onwards.

The **wIndex** field specifies the **iLangID** in the high byte and the AudioControl interface number in the low byte. The **iLangID** field contains a zero-based index into the LANGID code array as returned by the device. A device can at most support 126 different languages since the LANGID code array is restricted to 254 bytes and each LANGID code takes up 2 bytes. The range of the **iLangID** is therefore from 0 to 125 maximum.

The values specified in the **wValue** and **wIndex** fields shall be appropriate to the recipient. Only existing **wStrDescrID** values in the Audio Function can be indexed and only appropriate **iLangID** and AudioControl interface numbers may be used. If the request specifies an unknown **wStrDescrID** or **iLangID** value or an unknown AudioControl interface number, the control pipe shall indicate a stall.

The length of the descriptor to return is indicated in the **wLength** field of the request. If the descriptor is longer than what is indicated in the **wLength** field, only the initial bytes of the descriptor are returned. If the descriptor is shorter than what is indicated in the **wLength** field, the device indicates the end of the control transfer by sending a short packet when further data is requested.

The layout of the parameter block follows the class-specific String descriptor definition as outlined in Section 4.9, "Class-specific String descriptors."

5.2.3.3 HIGH CAPABILITY DESCRIPTOR REQUEST

To overcome the limitations of the standard Get Descriptor request (limited to maximum 256 bytes long), this specification provides a class-specific method to retrieve larger descriptors from the Audio Function. Also, since High Capability descriptors have the ability to report changes dynamically, they can be used whenever there is a need for the descriptor to indicate that some of its values have changed (even when its length is less than 256 bytes).

The Get High Capability Descriptor request is used to retrieve High Capability descriptors and shall be supported whenever the Audio Function uses at least one High Capability descriptor. A High Capability descriptor is uniquely identified within an interface or endpoint by the **wDescriptorID** value.

Table 5-17: High Capability Descriptor Request

bmRequest Type	bRequest	wValue	wIndex	wLength	Data
10100001B	HIGH_CAPABILITY _DESCRIPTOR	wDescriptorID	Zero and Interface	Length of parameter block	Parameter block
10100010B			Zero and Endpoint		

Bit D7 of the **bmRequestType** field specifies that this is a Get request (D7 = 0b1). It is a class-specific request (D6..5 = 0b01), directed to either an interface (AudioControl or AudioStreaming) of the Audio Function (D4..0 = 0b00001) or the isochronous endpoint of an AudioStreaming interface (D4..0 = 0b00010).

The **bRequest** field indicates that a class-specific descriptor is targeted.

The **wValue** field specifies the descriptor's **wDescriptorID** value.

When the request addresses an interface (**bmRequestType** = 10100001), the **wIndex** field specifies the interface in the low byte and zero in the high byte.

When the request addresses an endpoint (**bmRequestType** = 10100010), the **wIndex** field specifies the endpoint to be addressed in the low byte and zero in the high byte.

The values in the **wValue** and **wIndex** fields shall be appropriate to the recipient. Only existing **wDescriptorID** values can be used and only appropriate interface or endpoint numbers may be used. If the request specifies an unknown **wDescriptorID** value or an unknown interface or endpoint number, the control pipe shall indicate a stall.

The length of the descriptor to return is indicated in the **wLength** field of the request. If the descriptor is longer than what is indicated in the **wLength** field, only the initial bytes of the descriptor are returned. If the descriptor is shorter than what is indicated in the **wLength** field, the device indicates the end of the control transfer by sending a short packet when further data is requested.

The layout of the parameter block follows the High Capability descriptor definitions as outlined in Section 4.2, "Class-Specific Descriptors."

6 INTERRUPTS

Interrupts are used as a means to inform the Host that a change has occurred in the current state of the Audio Function. This specification currently defines two different types of interrupts:

- Memory Change: Some internal Entity's memory location has been updated. Host software can be notified so that the appropriate action can be taken.
- Control Change: Some addressable Control inside the Audio Function changed one or more of its attribute values.

The Audio Controls inside a Clock Entity, Unit, Terminal or Power Domain can be the source of an interrupt. Likewise, any addressable Control inside the AudioControl interface or any of the AudioStreaming interfaces can generate an interrupt. Finally, all addressable Controls related to an audio endpoint can be the cause of an interrupt.

A change of state in the Audio Function is most often caused by a certain event that takes place. An event can either be user-initiated or device-initiated. User-initiated connector insertion or removal is a typical example of a user-initiated event. The Host could then switch selectors or mixers so as to play audio out of the just inserted device (e.g. a headphone) and stop playing audio out of the current device (e.g. a speaker set).
An example of a device-initiated event could be the following: An external device (e.g. an A/V receiver) could switch from PCM to AC-3 encoded data on its optical digital output, depending on the material that is currently being played. If this device is connected to the optical digital input of an Audio Function that has auto-detect capabilities, the interface on that Audio Function might need to be reconfigured (e.g. to start the AC-3 decoding process), maybe causing all other interfaces to change some aspect of their format, or even become unusable. The device could issue an interrupt, letting the Host know that the Audio Function needs reconfiguration.

6.1 INTERRUPT DATA MESSAGE

The actual type of interrupt (Memory Change, Control Change, Descriptor Change) and its originator is conveyed to the Host through the interrupt data message that is sent over the interrupt endpoint. It is then the responsibility of the Host to query the Audio Function for more detailed information about the cause of the interrupt through a Get Memory request or one of the Get Control or Get Descriptor requests as defined in Section 5.2, "Class-Specific Requests".

Note that if the Host directly changes the CUR attribute on any Control, that Control shall not generate an interrupt. Likewise, if the Host directly modifies a memory location within an Entity, that change shall not generate an interrupt.

Interrupts are considered to be of the 'edge-triggered' type, meaning that an interrupt is generated whenever an event occurs, but there is no specific action required from the Host to clear the interrupt condition. When the Host issues a Get request in response to the interrupt, the most current value of the addressed Control's attribute will be returned.

The interrupt data message is always 6 bytes in length. The first **bInfo** field is required for all interrupt data messages. It contains information in D0 indicating whether this is a vendor-specific interrupt (D0 = 0b1) or a class-specific interrupt (D0 = 0b0). Bit D1 indicates whether the interrupt originated from an interface (D1 = 0b0) or an endpoint (D1 = 0b1). Bits D7..2 of the **bInfo** field are reserved. For vendor-specific interrupts, the layout of the remainder of the interrupt message is undefined. For class-specific interrupts, the layout is defined as follows.

When the interrupt originates from an Entity in an interface (D1 = 0b0 in the **bInfo** field), the **wIndex** field specifies the interface in the low byte and the Entity ID (Clock Entity ID, Unit ID, Terminal ID, or Power Domain ID). For indicating the interface itself, an Entity ID of zero shall be specified in the high byte.

When the interrupt originates from an endpoint (D1 = 0b1 in the **bInfo** field), the **wIndex** field specifies the endpoint to be addressed in the low byte and zero in the high byte.

The **wValue** field interpretation is qualified by the value in the **wIndex** field. The layout of the **wValue** field changes depending on the addressed Entity. The **wValue** field follows exactly the same rules as outlined in Section 5, "Requests" for each of the supported Get Control requests. The **wValue** field returns the Control Selector (CS) in the high byte and the Channel Number (CN) in the low byte. The Control Selector and the Channel Number (CN) together indicate exactly which Control generated the interrupt. If a Control is channel-independent, then the Control is considered to be a master Control and the virtual channel zero is returned to indicate it (CN = 0).

There are two exceptions to the above. The first is when a Mixer Unit Control request returns CS = MU_MIXER_CONTROL in the high byte. Then the Mixer Control Number (MCN) is returned in the low byte. The second is the Memory request where the **wValue** field specifies a zero-based offset value that indicates the address of the location in the Entity's memory space that generated the interrupt. If the offset value is zero, this indicates that multiple memory locations might have changed and the entire memory space needs to be examined.

The **bSourceType** field contains a constant, identifying the specific source type (attribute of the addressed Control or Entity, String, or High Capability descriptor) is causing the interrupt. Possible source types are:

• Current setting attribute	(CUR)
• Range attribute	(RANGE)
• Memory space attribute	(MEM)
• String	(STRING)
• High Capability Descriptor	(HIGH_CAPABILITY_DESCRIPTOR)

When there are no interrupts pending, the interrupt endpoint shall NAK when polled.

The following table specifies the format of the interrupt message when D0 = 0b0:

Table 6-1: Interrupt Data Message Format

Offset	Field	Size	Value	Description
0	bInfo	1	Bitmap	D0: Vendor-specific. D1: Interface or Endpoint D7..2: Reserved.
1	bSourceType	1	Constant	The source type that caused the interrupt
2	wValue	2	Number	CS in the high byte and CN or MCN in the low byte for CUR and RANGE source type. Zero-based offset into memory space for MEM source type. **wStrDescrID** value for String source type. **wDescriptorID** value for High Capability Descriptor source type.
4	wIndex	2	Number	Entity ID or zero in the high byte and Interface or Endpoint in the low byte.

6.2 INTERRUPT SOURCES

Any Control, High Capability descriptor, class-specific String descriptor, or memory location within an addressable Entity of the Audio Function can be the source of an interrupt. The interrupt message contains enough information to determine exactly which Control or memory location caused the interrupt. The Host can then issue the normal Control or Memory requests to further qualify the cause of the interrupt.

APPENDIX A. AUDIO DEVICE CLASS CODES

A.1 AUDIO FUNCTION CLASS CODE

Table A-1: Audio Function Class Code

Audio Function Class Code	Value
AUDIO_FUNCTION	AUDIO

A.2 AUDIO FUNCTION SUBCLASS CODES

Table A-2: Audio Function Subclass Codes

Audio Function Subclass Code	Value
FUNCTION_SUBCLASS_UNDEFINED	0x00
FULL_ADC_3_0	0x01
GENERIC_I/O	0x20
HEADPHONE	0x21
SPEAKER	0x22
MICROPHONE	0x23
HEADSET	0x24
HEADSET_ADAPTER	0x25
SPEAKERPHONE	0x26

A.3 AUDIO FUNCTION PROTOCOL CODES

Table A-3: Audio Function Protocol Codes

Audio Function Protocol Code	Value
FUNCTION_PROTOCOL_UNDEFINED	0x00
AF_VERSION_01_00	IP_VERSION_01_00
AF_VERSION_02_00	IP_VERSION_02_00
AF_VERSION_03_00	IP_VERSION_03_00

A.4 AUDIO INTERFACE CLASS CODE

Table A-4: Audio Interface Class Code

Audio Interface Class Code	Value
AUDIO	0x01

A.5 AUDIO INTERFACE SUBCLASS CODES

Table A-5: Audio Interface Subclass Codes

Audio Interface Subclass Code	Value
INTERFACE_SUBCLASS_UNDEFINED	0x00
AUDIOCONTROL	0x01
AUDIOSTREAMING	0x02
MIDISTREAMING	0x03

A.6 AUDIO INTERFACE PROTOCOL CODES

Table A-6: Audio Interface Protocol Codes

Audio Interface Protocol Code	Value
IP_VERSION_01_00	0x00
IP_VERSION_02_00	0x20
IP_VERSION_03_00	0x30

A.7 AUDIO FUNCTION CATEGORY CODES

Table A-7: Audio Function Category Codes

Audio Function Subclass Code	Value
FUNCTION_SUBCLASS_UNDEFINED	0x00
DESKTOP_SPEAKER	0x01
HOME_THEATER	0x02
MICROPHONE	0x03
HEADSET	0x04
TELEPHONE	0x05
CONVERTER	0x06
VOICE/SOUND_RECORDER	0x07
I/O_BOX	0x08
MUSICAL_INSTRUMENT	0x09
PRO-AUDIO	0x0A
AUDIO/VIDEO	0x0B
CONTROL_PANEL	0x0C
HEADPHONE	0x0D
GENERIC_SPEAKER	0x0E
HEADSET_ADAPTER	0x0F
SPEAKERPHONE	0x10
Reserved	0x11..0xFE

Audio Function Subclass Code	Value
OTHER	0xFF

A.8 AUDIO CLASS-SPECIFIC DESCRIPTOR TYPES

Table A-8: Audio Class-specific Descriptor Types

Descriptor Type	Value
CS_UNDEFINED	0x20
CS_DEVICE	0x21
CS_CONFIGURATION	0x22
CS_STRING	0x23
CS_INTERFACE	0x24
CS_ENDPOINT	0x25
CS_CLUSTER	0x26

A.9 CLUSTER DESCRIPTOR SUBTYPES

Table A-9: Audio Class-Specific Cluster Descriptor Subtypes

Descriptor Subtype	Value
SUBTYPE_UNDEFINED	0x00

A.10 CLUSTER DESCRIPTOR SEGMENT TYPES

Table A-10: Cluster Descriptor Segment Types

Segment Type	Value
SEGMENT_UNDEFINED	0x00
CLUSTER_DESCRIPTION	0x01
CLUSTER_VENDOR_DEFINED	0x1F
CHANNEL_INFORMATION	0x20
CHANNEL_AMBISONIC	0x21
CHANNEL_DESCRIPTION	0x22
CHANNEL_VENDOR_DEFINED	0xFE
END_SEGMENT	0xFF

A.11 CHANNEL PURPOSE DEFINITIONS

Table A-11: Channel Purpose Definitions

Channel Purpose	Value
PURPOSE_UNDEFINED	0x00
GENERIC_AUDIO	0x01

Channel Purpose	Value
VOICE	0x02
SPEECH	0x03
AMBIENT	0x04
REFERENCE	0x05
ULTRASONIC	0x06
VIBROKINETIC	0x07
NON_AUDIO	0xFF

A.12 CHANNEL RELATIONSHIP DEFINITIONS

Table A-12: Channel Relationship Definitions

USB Audio Channel Relationship		CEA-861.2 Channel Allocation	
Description	Acronym	Description	Acronym
RELATIONSHIP_UNDEFINED	UND	---	---
MONO	M	---	---
LEFT	L	---	---
RIGHT	R	---	---
ARRAY	AR	---	---
PATTERN_X	PX	---	---
PATTERN_Y	PY	---	---
PATTERN_A	PA	---	---
PATTERN_B	PB	---	---
PATTERN_M	PM	---	---
PATTERN_S	PS	---	---
Front Left	FL	Front Left	FL
Front Right	FR	Front Right	FR
Front Center	FC	Front Center	FC
Front Left of Center	FLC	Front Left of Center	FLc
Front Right of Center	FRC	Front Right of Center	FRc
Front Wide Left	FWL	Front left Wide	FLw
Front Wide Right	FWR	Front Right Wide	FRw
Side Left	SL	Side Left	SiL
Side Right	SR	Side Right	SiR
Surround Array Left	SAL	Left Surround	LS
Surround Array Right	SAR	Right Surround	RS

Back Left	BL	Back Left	BL
Back Right	BR	Back Right	BR
Back Center	BC	Back Center	BC
Back Left of Center	BLC	---	---
Back Right of Center	BRC	---	---
Back Wide Left	BWL	---	---
Back Wide Right	BWR	---	---
Top Center	TC	Top Center	TpC
Top Front Left	TFL	Top Front Left	TpFL
Top Front Right	TFR	Top Front Right	TpFR
Top Front Center	TFC	Top Front Center	TpFC
Top Front Left of Center	TFLC	---	---
Top Front Right of Center	TFRC	---	---
Top Front Wide Left	TFWL	---	---
Top Front Wide Right	TFWR	---	---
Top Side Left	TSL	Top Side Left	TpSiL
Top Side Right	TSR	Top Side Right	TpSiR
Top Surround Array Left	TSAL	Top Left Surround	TpLS
Top Surround Array Right	TSAR	Top Right Surround	TpRS
Top Back Left	TBL	Top Back Left	TpBL
Top Back Right	TBR	Top Back Right	TpBR
Top Back Center	TBC	Top Back Center	TpBC
Top Back Left Of Center	TBLC	---	---
Top Back Right Of Center	TBRC	---	---
Top Back Wide Left	TBWL	---	---
Top Back Wide Right	TBWR	---	---
Bottom Center	BC	---	---
Bottom Front Left	BFL	Bottom Front Left	BtFL
Bottom Front Right	BFR	Bottom Front Right	BtFR
Bottom Front Center	BFC	Bottom Front Center	BtFC
Bottom Front Left Of Center	BFLC	---	---
Bottom Front Right Of Center	BFRC	---	---
Bottom Front Wide Left	BFWL	---	---
Bottom Front Wide Right	BFWR	---	---
Bottom Side Left	BSL	---	---
Bottom Side Right	BSR	---	---
Bottom Surround Array Left	BSAL	---	---

Bottom Surround Array Right	BSAR	---	---
Bottom Back Left	BBL	---	---
Bottom Back Right	BBR	---	---
Bottom Back Center	BBC	---	---
Bottom Back Left Of Center	BBLC	---	---
Bottom Back Right Of Center	BBRC	---	---
Bottom Back Wide Left	BBWL	---	---
Bottom Back Wide Right	BBWR	---	---
Low Frequency Effects	LFE	Low Frequency Effects 1	LFE1
Low Frequency Effects Left	LFEL	---	---
Low Frequency Effects Right	LFER	Low Frequency Effects 2	LFE2
Headphone Left	HPL	---	---
Headphone Right	HPR	---	---

A.13 AMBISONIC COMPONENT ORDERING CONVENTION TYPES

Table A-13: Ambisonic Component Ordering Convention Types

Ambisonic Component Ordering Convention Type	Value
ORD_TYPE_UNDEFINED	0x00
AMBISONIC_CHANNEL_NUMBER (ACN)	0x01
FURSE_MALHAM	0x02
SINGLE_INDEX DESIGNATION (SID)	0x03

A.14 AMBISONIC NORMALIZATION TYPES

Table A-14: Ambisonic Normalization Types

Ambisonic Normalization Type	Value
NORM_TYPE_UNDEFINED	0x00
maxN	0x01
SN3D	0x02
N3D	0x03
SN2D	0x04
N2D	0x05

A.15 AUDIO CLASS-SPECIFIC AC INTERFACE DESCRIPTOR SUBTYPES

Table A-15: Audio Class-Specific AC Interface Descriptor Subtypes

Descriptor Subtype	Value
AC_DESCRIPTOR_UNDEFINED	0x00
HEADER	0x01
INPUT_TERMINAL	0x02
OUTPUT_TERMINAL	0x03
EXTENDED_TERMINAL	0x04
MIXER_UNIT	0x05
SELECTOR_UNIT	0x06
FEATURE_UNIT	0x07
EFFECT_UNIT	0x08
PROCESSING_UNIT	0x09
EXTENSION_UNIT	0x0A
CLOCK_SOURCE	0x0B
CLOCK_SELECTOR	0x0C
CLOCK_MULTIPLIER	0x0D
SAMPLE_RATE_CONVERTER	0x0E
CONNECTORS	0x0F
POWER_DOMAIN	0x10

A.16 AUDIO CLASS-SPECIFIC AS INTERFACE DESCRIPTOR SUBTYPES

Table A-16: Audio Class-Specific AS Interface Descriptor Subtypes

Descriptor Subtype	Value
AS_DESCRIPTOR_UNDEFINED	0x00
AS_GENERAL	0x01
AS_VALID_FREQ_RANGE	0x02

A.17 AUDIO CLASS-SPECIFIC STRING DESCRIPTOR SUBTYPES

Table A-17: Audio Class-Specific String descriptor Subtypes

Descriptor Subtype	Value
SUBTYPE_UNDEFINED	0x00

A.18 EXTENDED TERMINAL SEGMENT TYPES

Table A-18: Extended Terminal Segment Types

Segment Type	Value
SEGMENT_UNDEFINED	0x00
TERMINAL_VENDOR_DEFINED	0x1F
CHANNEL_BANDWIDTH	0x20
CHANNEL_MAGNITUDE_RESPONSE	0x21
CHANNEL_MAGNITUDE/PHASE_RESPONSE	0x22
CHANNEL_POSITION_XYZ	0x23
CHANNEL_POSITION_ RΘΦ	0x24
CHANNEL_VENDOR_DEFINED	0xFE
END_SEGMENT	0xFF

A.19 EFFECT UNIT EFFECT TYPES

Table A-19: Effect Unit Effect Types

wEffectType	Value
EFFECT_UNDEFINED	0x0000
PARAM_EQ_SECTION_EFFECT	0x0001
REVERBERATION_EFFECT	0x0002
MOD_DELAY_EFFECT	0x0003
DYN_RANGE_COMP_EFFECT	0x0004

A.20 PROCESSING UNIT PROCESS TYPES

Table A-20: Processing Unit Process Types

wProcessType	Value
PROCESS_UNDEFINED	0x0000
UP/DOWNMIX_PROCESS	0x0001
STEREO_EXTENDER_PROCESS	0x0002
MULTI_FUNCTION_PROCESS	0x0003

A.21 AUDIO CLASS-SPECIFIC ENDPOINT DESCRIPTOR SUBTYPES

Table A-21: Audio Class-Specific Endpoint Descriptor Subtypes

Descriptor Subtype	Value
DESCRIPTOR_UNDEFINED	0x00
EP_GENERAL	0x01

A.22 AUDIO CLASS-SPECIFIC REQUEST CODES

Table A-22: Audio Class-Specific Request Codes

Class-Specific Request Code	Value
REQUEST_CODE_UNDEFINED	0x00
CUR	0x01
RANGE	0x02
MEM	0x03
INTEN	0x04
STRING	0x05
HIGH_CAPABILITY_DESCRIPTOR	0x06

A.23 CONTROL SELECTOR CODES

A.23.1 AUDIOCONTROL INTERFACE CONTROL SELECTORS

Table A-23: AudioControl Interface Control Selectors

Control Selector	Value
AC_CONTROL_UNDEFINED	0x00
AC_ACTIVE_INTERFACE_CONTROL	0x01
AC_POWER_DOMAIN_CONTROL	0x02

A.23.2 CLOCK SOURCE CONTROL SELECTORS

Table A-24: Clock Source Control Selectors

Control Selector	Value
CS_CONTROL_UNDEFINED	0x00
CS_SAM_FREQ_CONTROL	0x01
CS_CLOCK_VALID_CONTROL	0x02

A.23.3 CLOCK SELECTOR CONTROL SELECTORS

Table A-25: Clock Selector Control Selectors

Control Selector	Value
CX_CONTROL_UNDEFINED	0x00
CX_CLOCK_SELECTOR_CONTROL	0x01

A.23.4 CLOCK MULTIPLIER CONTROL SELECTORS

Table A-26: Clock Multiplier Control Selectors

Control Selector	Value
CM_CONTROL_UNDEFINED	0x00

Control Selector	Value
CM_NUMERATOR_CONTROL	0x01
CM_DENOMINATOR_CONTROL	0x02

A.23.5 TERMINAL CONTROL SELECTORS

Table A-27: Terminal Control Selectors

Control Selector	Value
TE_CONTROL_UNDEFINED	0x00
TE_INSERTION_CONTROL	0x01
TE_OVERLOAD_CONTROL	0x02
TE_UNDERFLOW_CONTROL	0x03
TE_OVERFLOW_CONTROL	0x04
TE_LATENCY_CONTROL	0x05

A.23.6 MIXER CONTROL SELECTORS

Table A-28: Mixer Control Selectors

Control Selector	Value
MU_CONTROL_UNDEFINED	0x00
MU_MIXER_CONTROL	0x01
MU_UNDERFLOW_CONTROL	0x02
MU_OVERFLOW_CONTROL	0x03
MU_LATENCY_CONTROL	0x04

A.23.7 SELECTOR CONTROL SELECTORS

Table A-29: Selector Control Selectors

Control Selector	Value
SU_CONTROL_UNDEFINED	0x00
SU_SELECTOR_CONTROL	0x01
SU_LATENCY_CONTROL	0x02

A.23.8 FEATURE UNIT CONTROL SELECTORS

Table A-30: Feature Unit Control Selectors

Control Selector	Value
FU_CONTROL_UNDEFINED	0x00
FU_MUTE_CONTROL	0x01
FU_VOLUME_CONTROL	0x02
FU_BASS_CONTROL	0x03

Control Selector	Value
FU_MID_CONTROL	0x04
FU_TREBLE_CONTROL	0x05
FU_GRAPHIC_EQUALIZER_CONTROL	0x06
FU_AUTOMATIC_GAIN_CONTROL	0x07
FU_DELAY_CONTROL	0x08
FU_BASS_BOOST_CONTROL	0x09
FU_LOUDNESS_CONTROL	0x0A
FU_INPUT_GAIN_CONTROL	0x0B
FU_INPUT_GAIN_PAD_CONTROL	0x0C
FU_PHASE_INVERTER_CONTROL	0x0D
FU_UNDERFLOW_CONTROL	0x0E
FU_OVERFLOW_CONTROL	0x0F
FU_LATENCY_CONTROL	0x10

A.23.9 EFFECT UNIT CONTROL SELECTORS

A.23.9.1 PARAMETRIC EQUALIZER SECTION EFFECT UNIT CONTROL SELECTORS

Table A-31: Reverberation Effect Unit Control Selectors

Control Selector	Value
PE_CONTROL_UNDEFINED	0x00
PE_ENABLE_CONTROL	0x01
PE_CENTERFREQ_CONTROL	0x02
PE_QFACTOR_CONTROL	0x03
PE_GAIN_CONTROL	0x04
PE_UNDERFLOW_CONTROL	0x05
PE_OVERFLOW_CONTROL	0x06
PE_LATENCY_CONTROL	0x07

A.23.9.2 REVERBERATION EFFECT UNIT CONTROL SELECTORS

Table A-32: Reverberation Effect Unit Control Selectors

Control Selector	Value
RV_CONTROL_UNDEFINED	0x00
RV_ENABLE_CONTROL	0x01
RV_TYPE_CONTROL	0x02
RV_LEVEL_CONTROL	0x03
RV_TIME_CONTROL	0x04

Control Selector	Value
RV_FEEDBACK_CONTROL	0x05
RV_PREDELAY_CONTROL	0x06
RV_DENSITY_CONTROL	0x07
RV_HIFREQ_ROLLOFF_CONTROL	0x08
RV_UNDERFLOW_CONTROL	0x09
RV_OVERFLOW_CONTROL	0x0A
RV_LATENCY_CONTROL	0x0B

A.23.9.3 MODULATION DELAY EFFECT UNIT CONTROL SELECTORS

Table A-33: Modulation Delay Effect Unit Control Selectors

Control Selector	Value
MD_CONTROL_UNDEFINED	0x00
MD_ENABLE_CONTROL	0x01
MD_BALANCE_CONTROL	0x02
MD_RATE_CONTROL	0x03
MD_DEPTH_CONTROL	0x04
MD_TIME_CONTROL	0x05
MD_FEEDBACK_CONTROL	0x06
MD_UNDERFLOW_CONTROL	0x07
MD_OVERFLOW_CONTROL	0x08
MD_LATENCY_CONTROL	0x09

A.23.9.4 DYNAMIC RANGE COMPRESSOR EFFECT UNIT CONTROL SELECTORS

Table A-34: Dynamic Range Compressor Effect Unit Control Selectors

Control Selector	Value
DR_CONTROL_UNDEFINED	0x00
DR_ENABLE_CONTROL	0x01
DR_COMPRESSION_RATE_CONTROL	0x02
DR_MAXAMPL_CONTROL	0x03
DR_THRESHOLD_CONTROL	0x04
DR_ATTACK_TIME_CONTROL	0x05
DR_RELEASE_TIME_CONTROL	0x06
DR_UNDERFLOW_CONTROL	0x07
DR_OVERFLOW_CONTROL	0x08
DR_LATENCY_CONTROL	0x09

A.23.10 PROCESSING UNIT CONTROL SELECTORS

A.23.10.1 UP/DOWN-MIX PROCESSING UNIT CONTROL SELECTORS

Table A-35: Up/Down-mix Processing Unit Control Selectors

Control Selector	Value
UD_CONTROL_UNDEFINED	0x00
UD_MODE_SELECT_CONTROL	0x01
UD_UNDERFLOW_CONTROL	0x02
UD_OVERFLOW_CONTROL	0x03
UD_LATENCY_CONTROL	0x04

A.23.10.2 STEREO EXTENDER PROCESSING UNIT CONTROL SELECTORS

Table A-36: Stereo Extender Processing Unit Control Selectors

Control Selector	Value
ST_EXT_CONTROL_UNDEFINED	0x00
ST_EXT_WIDTH_CONTROL	0x01
ST_EXT_UNDERFLOW_CONTROL	0x02
ST_EXT_OVERFLOW_CONTROL	0x03
ST_EXT_LATENCY_CONTROL	0x04

A.23.11 EXTENSION UNIT CONTROL SELECTORS

Table A-37: Extension Unit Control Selectors

Control Selector	Value
XU_CONTROL_UNDEFINED	0x00
XU_UNDERFLOW_CONTROL	0x01
XU_OVERFLOW_CONTROL	0x02
XU_LATENCY_CONTROL	0x03

A.23.12 AUDIOSTREAMING INTERFACE CONTROL SELECTORS

Table A-38: AudioStreaming Interface Control Selectors

Control Selector	Value
AS_CONTROL_UNDEFINED	0x00
AS_ACT_ALT_SETTING_CONTROL	0x01
AS_VAL_ALT_SETTINGS_CONTROL	0x02
AS_AUDIO_DATA_FORMAT_CONTROL	0x03

A.23.13 ENDPOINT CONTROL SELECTORS

Table A-39: Endpoint Control Selectors

Control Selector	Value
EP_CONTROL_UNDEFINED	0x00
EP_PITCH_CONTROL	0x01
EP_DATA_OVERRUN_CONTROL	0x02
EP_DATA_UNDERRUN_CONTROL	0x03

A.24 CONNECTOR TYPES

Table A-40: Connector Types

Connector Type	Value
UNDEFINED	0x00
2.5 MM PHONE CONNECTOR	0x01
3.5 MM PHONE CONNECTOR	0x02
6.35 MM PHONE CONNECTOR	0x03
XLR/6.35MM COMBO CONNECTOR	0x04
XLR	0x05
OPTICAL/3.5MM COMBO CONNECTOR	0x06
RCA	0x07
BNC	0x08
BANANA	0x09
BINDING POST	0x0A
SPEAKON	0x0B
SPRING CLIP	0x0C
SCREW TYPE	0x0D
DIN	0x0E
MINI DIN	0x0F
EUROBLOCK	0x10
USB TYPE-C	0x11
RJ-11	0x12
RJ-45	0x13
TOSLINK	0x14
HDMI	0x15
Mini-HDMI	0x16
Micro-HDMI	0x17
DP	0x18

Connector Type	Value
MINI-DP	0x19
D-SUB	0x1A
THUNDERBOLT	0x1B
LIGHTNING	0x1C
WIRELESS	0x1D
USB STANDARD A	0x1E
USB STANDARD B	0x1F
USB MINI-B	0x20
USB MICRO-B	0x21
USB MICRO-AB	0x22
USB 3.0 MICRO-B	0x23
OTHER (CONNECTOR TYPE NOT IN THIS LIST)	0xFF