Digital Media Programming Guide

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About This Guide

The *Digital Media Programming Guide* describes the Silicon Graphics® digital media development environment (DMdev) software. The DMdev is a family of libraries that provides application program interfaces (APIs) for digital media I/O, file operations, playback, and conversions. This guide describes the libraries and gives technical information on their design and proper use. A companion guide, *Digital Media Programmer's Examples*, contains code samples based on the DMdev to assist your development efforts. It can be viewed online using the IRIS InSight™ viewer.

Silicon Graphics also supplies end user desktop media tools, which use the DMdev. Tools for capturing, editing, recording, playing, compressing, and converting audio data and images, and control panels, such as the Video Panel and the Audio Control Panel, are described in the *Digital Media Tools User's Guide*, which you can view both documents from the InSight viewer.

What This Guide Contains

In the *Digital Media Programming Guide*:

Chapter 1, "Introduction to the Digital Media Libraries" gives an overview of the digital media development environment.

Chapter 2, "Digital Media Essentials" provides a foundation for understanding digital media data characteristics. It reviews how data is represented digitally and then explains how to express data attributes in the Digital Media Libraries.

Chapter 3, "Digital Media Parameters" explains how to use the digital media data structures that facilitate data specification and setting, getting, and passing parameters.

Chapter 4, "Digital Media I/O" describes using the digital media library routines that facilitate real-time input and output between live media devices.

Chapter 5, “Digital Media Buffers” explains the Digital Media buffers (DMbuffers) real-time visual data transport facility. The facility establishes a unified approach to providing data flow between live video devices.

Chapter 6, “Digital Media Data Conversion” tells how to use the digital media conversion libraries to implement data format conversion in your application.

Chapter 7, “Digital Media Audio File Operations” explains how to use the Audio File (AF) Library for reading and writing audio disk files, and for controlling the format of audio data buffers.

Appendix A, “Digital Media Conversion Libraries” contains the APIs of the individual image and audio conversion libraries. These libraries are not discussed in detail, but reference pages to the member functions are cited.

How to Use This Guide

This guide is written for C language programmers that have some knowledge of digital media concepts. Readers unfamiliar with the basic concepts can refer to Chapter 2, “Digital Media Essentials,” or to the “Suggestions for Further Reading” listed below.

Where to Start

If you’re not sure which library to use for a certain application, read Chapter 1, “Introduction to the Digital Media Libraries,” to get a brief overview of the uses and features of each library.

If you want to find some code that does what you want your application to do, browse through the *Digital Media Programmer’s Examples* online book to locate a sample program that performs a particular task.

Style Conventions

These style conventions are used in this guide:

Bold	functions, routines
Italics	arguments, variables, commands, program and file names, book titles, and emphasis
`Courier`	function prototypes, sample code
`Courier Bold`	user input entered from the keyboard

How to Use the Sample Programs

Code fragments and complete sample programs are used throughout this guide to demonstrate programming concepts. Source code for the sample programs is provided in the */usr/share/src/dmedia* directory, which is further organized in directories according to topic.

README files in each directory provide descriptions of the sample programs and instructions for compiling and running them. You must have the IRIS Development Option, *dev*, and the C language software, *c*, loaded before you can compile the sample programs. Use the *versions* command to find out which software is loaded on your system. See the release notes for each library for additional system software requirements for those libraries.

You should copy any program that you intend to modify to your home directory before making any changes.

Suggestions for Further Reading

This section lists references containing information on programming topics beyond the scope of this guide, which you may find helpful for developing your digital media application. Additional reference materials are listed in the introductory chapters for each library.

References for Using Digital Media with Other Libraries

If you are planning to integrate your digital media application with calls from the OpenGL™, IRIS Graphics Library™ (GL) or X Window System™ application, you may want to consult the following manuals:

- *OpenGL Programming Guide*, by Jackie Neider, Tom Davis, and Mason Woo, Addison-Wesley, 1994
- *OpenGL Reference Manual*, by Jackie Neider, Tom Davis, and Mason Woo, Addison-Wesley, 1994
- *Graphics Library Programming Guide*, by Patricia McLendon Creek, Silicon Graphics, 1992
- *Graphics Library Programming Tools and Techniques*, by Patricia McLendon Creek and Ken Jones, Silicon Graphics, 1993
- *IRIS IM Programming Notes*, by Patricia McLendon Creek and Ken Jones, Silicon Graphics, 1993
- *The X Window System, Volume 1: Xlib Programming Manual*, O'Reilly and Associates, 1990
- *The X Window System, Volume 4: X Toolkit Intrinsics, Motif Edition*, O'Reilly and Associates, 1990
- *X Window System: The Complete Reference to Xlib, X Protocol, ICCCM, XLFD*, Third Edition, by Robert W. Scheifler and James Gettys, Digital Press, 1992
- *X Window System Toolkit: The Complete Programmer's Guide and Specification*, Paul J. Asente and Ralph R. Swick, Digital Press, 1992

References for Adding a User Interface to Your Program

The Digital MediaLibraries don't impose any particular user interface (UI), so you can use any graphical UI toolkit, such as IRIS IM™ to build your interface. IRIS IM is Silicon Graphics' port of the industry-standard OSF/Motif™ software. Consult these OSF/Motif manuals for more information:

- *OSF/Motif Programmer's Guide*, Revision 1.2, Prentice-Hall, 1993
- *OSF/Motif Programmer's Reference,* Revision 1.2, Prentice-Hall, 1992
- *OSF/Motif Style Guide,* Revision 1.2, Prentice-Hall, 1992

Technical References and Standards

The references listed below are some of the more important standards mentioned throughout this book. For more complete listings, you can check the Web sites of organizations such as the Society of Motion Picture & Television Engineers at *http://www.smpte.org/*, and the International Telecommunication Union at *http://www.itu.ch/index.html*.

SMPTE Standard for Television—Composite Analog Video Signal—NTSC for Studio Applications, SMPTE 170M-1994, The Society of Motion Picture and Television Engineers

SMPTE Standard for Television—10-Bit 4:2:2 Component and 4fsc NTSC Composite Digital Signals—Serial Digital Interface, SMPTE 259M-1993, The Society of Motion Picture and Television Engineers

SMPTE Standard for Television—Component Video Signal 4:2:2 - Bit-Parallel Digital Interface, SMPTE 125M-1995, The Society of Motion Picture and Television Engineers

4:2:2 Digital Video: Background and Implementation Revised Edition, The Society of Motion Picture and Television Engineers, l995

Recommendation ITU-R BT.601-5—Studio Encoding Parameters of Digital Television for Standard 4:3 and Wide-Screen 16:9 Aspect Ratios, The International Telecommunication Union, 1995

JPEG Still Image Data Compression Standard, by William B. Pennebaker and Joan L. Mitchell, Van Nostrand Reinhold, 1993

Chapter 1

Introduction to the Digital Media Libraries

The digital media development environment (DMdev) is a family of libraries that provides application program interfaces (APIs) for digital media I/O, file operations, playback, and conversions.

Figure 1-1 shows how the libraries in the digital media development environment are related to each other and to other development libraries. Lines in Figure 1-1 indicate where some libraries make internal calls to other libraries.

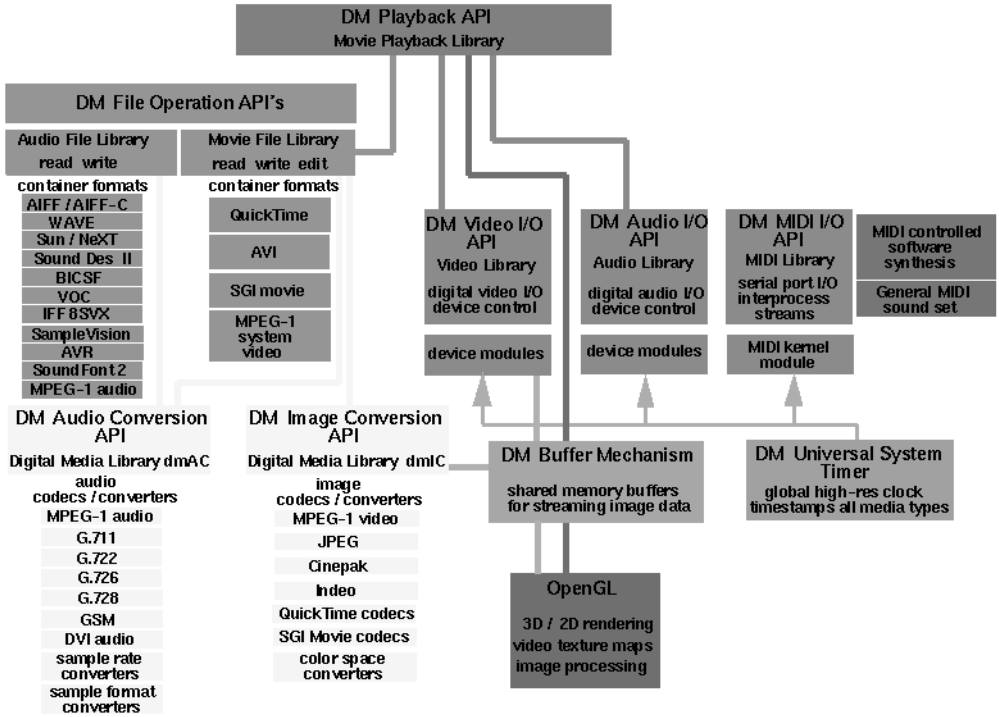


Figure 1-1 Silicon Graphics Digital Media Programming Environment

Together, the family of Digital Media Libraries encompass facilities for data format description, live audio and video I/O with built-in conversion capabilities, file operations such as reading, writing, and editing multimedia files, data conversion and compression, and playback. The following sections describe these facilities.

Digital Media Data Specification Facilities

What distinguishes the Digital Media Libraries from other multimedia developer software is the ability to accept any data regardless of format. You aren't limited to working with a particular format or the constraints it imposes. One of the things that makes this possible are the extensive data specification facilities offered by the DMdev.

The DM Library, *libdmedia.so*, provides global data type and parameter definitions for specifying digital media data attributes. You can use DM parameters to describe data held in memory, passed among the Digital Media Libraries, and imported and exported externally. File formats for on-disk data are described in "Digital Media File Operation and Conversion Facilities."

The DM Library features

- type definitions for digital media
- routines for creating and configuring digital media parameters
- routines for creating and configuring digital media buffers
- a debugging version of the library that lets you check for proper usage

See Chapter 3, "Digital Media Parameters," for a complete explanation of using the DM data types and parameters.

Digital Media I/O Facilities

The Audio Library (AL), the Video Library (VL), and the MIDI Library enable real-time I/O by providing the interface between your application program, the workstation CPU, and external devices. An overview of the digital media I/O capabilities follows, and Chapter 4, "Digital Media I/O," provides a complete explanation of using the digital media I/O APIs.

Audio I/O

The AL provides a device-independent C language API for programming audio I/O on all Silicon Graphics workstations. It provides routines for configuring the audio hardware, managing audio I/O between the application program and the audio hardware, specifying attributes of digital audio data, and facilitating real-time programming.

Use the AL for

- capturing audio from your workstation's microphone, line-level inputs, or a digital audio input source
- playing audio to your workstation's internal speaker, line-level outputs, headphones, or a digital output
- managing audio I/O between multiple audio devices
- adding audio to any application program

Video I/O

The VL provides an API for transporting live video on Silicon Graphics workstations equipped with on-board video and video options. The VL enables live video flow into a program.

Use the VL for

- displaying live video input in an onscreen window
- capturing video from your workstation's camera input, S-video input, composite analog inputs, or a digital input port into program memory
- playing video from program memory to your workstation's analog or digital video output
- combining video with computer graphics

Note: The range of video and VL capabilities you can use depends on the capabilities of your workstation and the video options installed in it.

MIDI I/O

The MIDI Library provides an API for sending, receiving, processing, and synthesizing musical instrument digital interface (MIDI) messages through the serial interface of Silicon Graphics workstations.

The MIDI Library enables

- content creation through an API to sound synthesizer and sequencer tools
- live interaction through a virtual 3D keyboard

Digital Media Live Data Transport Facilities

The digital media buffers live data transport system provides data types and operations for sharing and exchanging time-sensitive visual data in real time between video I/O devices, compression devices, graphics rendering and texturing operations, and the host processor(s). It includes

- Digital media buffers (DMbuffers) for carrying digital representations of images
- Digital media buffer pools (DMbufferpools) for reserving, apportioning, and allocating dedicated system and physical memory under direct application control for live visual data processing/transport devices

See Chapter 4, “Digital Media I/O,” for a complete explanation of using DMbuffers.

Digital Media File Operation and Conversion Facilities

Both the Movie Library and the Audio File Library provide file operations (identifying, opening, reading, writing, and editing files). These routines are capable of supporting a variety of file formats. Note the distinction between a file or container format, which is associated with media stored on disk (or tape), and the data format, which is a collection of attributes that describes the data. File format information is typically contained in a header that immediately precedes the data.

Audio Files

The Audio File Library, *libaudiofile*, provides a uniform C language API for indentifying, opening, reading, writing, and converting digital audio files of a variety of storage formats.

Use the Audio File Library for

- identifying audio files (multimedia file recognition)
- opening and creating audio files
- seeking, reading, and writing audio files
- setting and retrieving information in audio file headers
- setting and retrieving characteristics of the audio file or the data it contains
- converting audio file formats

Movie Files

The Movie File Library, *libmoviefile*, provides a file format-independent C language API for reading, writing, editing, and playing movies on Silicon Graphics workstations.

Use the Movie Library for:

- reading, writing, and editing movie files
- converting movie files from one container format to another
- compressing and decompressing movie files
- supporting movies embedded in applications programs

The Movie File Library provides a uniform interface to movies of various formats and lets you convert movies from one format to another. Currently, the Movie Library supports the following file formats:

- Apple® Computer QuickTime™ movie format, including uncompressed data, and JPEG, Indeo®, Cinepak®, Apple Animation, and Apple Video, compression
- Microsoft® audio-video interleaved (AVI) format, including uncompressed data, and JPEG, Indeo, and Cinepak compression

- MPEG-1 systems and video bitstreams
- Silicon Graphics movie format

Note: The Digital Media Libraries do not provide a QuickTime API, rather, they provide a file format-independent API that supports QuickTime and several other formats.

Note: Some QuickTime track types are not supported by the Movie File Library.

Digital Media Data Conversion Facilities

The Digital Media Library, *libdmedia*, provides data conversion support for real-time I/O and file operations. Many of the file operations and real-time I/O routines perform automatic data format conversions, by calling these lower level converter APIs, which are also accessible directly from your application:

- dmAC, an audio conversion API that performs audio compression and decompression and audio sample rate conversion
- dmIC, an image conversion API that performs image compression and decompression and color space conversion.

Digital Media Playback Facilities

The Digital Media Libraries provide a playback API, capable of playing audio, video, movies, and MIDI.

Movies

The Movie Playback Library, *libmovieplay*, is implemented using calls from the OpenGL and Digital Media Libraries. It provides a scheduler and modules to communicate with output devices. You can take advantage of its built-in playback support in your application.

The movie library playback engine provides:

- asynchronous playback support
- flexible playback control (start, stop, speed, looping)

- the ability to properly combine and blend multiple image and audio tracks
- software scaling of audio

The main advantage of the built-in playback support is that it performs audio and video synchronization for you. Otherwise you would have to calculate the rate for each track and determine the proper display timing. You can still take advantage of this synchronization capability even if you want to use your own display method, by turning off the movie library display and using your own event loop to respond to events. You may also choose to create your own playback using routines from the other libraries.

Chapter 2

Digital Media Essentials

Before writing a digital media application, it's essential to understand the basic attributes of digital image, video, and audio data. This chapter provides a foundation for understanding digital media data characteristics, and how to realize those qualities when creating your application. It begins by reviewing how data is represented digitally and then explains how to express data attributes in the Digital Media Libraries.

Digital Media Concepts

Data input from analog devices must be digitized in order to store to, retrieve from, and manipulate within a computer. Two of the most important concepts in digitizing are sampling and quantization.

Sampling and Quantization

Sampling involves partitioning a continuous flow of information, with respect to time or space (or both), into discrete pieces. Quantization involves representing the contents of such a sample as an integer value. Both operations are performed to obtain a digital representation.

The topic of exactly how many integers to use for quantizing and how many samples to take (and when or where to take them) in representing a given continuum is important because these choices affect the accuracy of the digital representation. Mathematical formulas exist for determining the correct amount of sampling and quantization needed to accurately re-create a continuous flow of data from its constituent pieces. A treatise on sampling theory is beyond the scope of this book, but you should be familiar with concepts such as the Nyquist theorem, pulse code modulation (PCM), and so on. For more information about digitization and related topics, see:

- Poynton, Charles A. *A Technical Introduction to Digital Video*. New York: John Wiley & Sons, 1995 (ISBN 0-471-12253-X).
- Watkinson, John. *An Introduction to Digital Video*. New York: Focal Press, 1994.

Quantities such as the sampling rate and number of quantization bits are called *attributes*; they describe a defining characteristic of the data that has a certain physical meaning. An important point about data attributes is that while they thoroughly describe data characteristics, they do not impose or imply a particular file format. In fact, one file format might encompass several types of data with several changeable attributes.

File format, also called container format, applies to data stored on disk or removable media. Data stored in a particular file format usually has a file header that contains information identifying the file format and auxiliary information about the data that follows it. Applications must be able to parse the header in order to recognize the file at a minimum, and to optionally open, read, or write the file. Similarly, data exported using a particular file format usually has a header prepended to it when output.

In contrast, *data format*, which is described by a collection of attributes, is meaningful for data I/O and exchange, and for data resident in memory. Because the Digital Media Libraries provide extensive data type and attribute specification facilities, they offer a lot of flexibility for recognizing, processing, storing, and retrieving a variety of data formats.

Parameters for Specifying Data Attributes

Parameters in the DM Library include files (*dmedia/dm_*.h)* provide a common language for specifying data attributes for the Digital Media Libraries. Not all of the libraries require or use all of the DM parameters.

Most of the Digital Media Libraries provide their own library-specific parameters for describing data attributes that are often meaningful only for the routines contained within each particular library. These library-specific parameters are prefaced with the initials of their parent library, rather than the initials DM. For example, the Video Library defines its own image parameters in *vl.h*, which are prefaced with the initials VL.

Many of the parameters defined in the DM Library have clones in the other libraries (except for the prefix initials). This makes it easy to write applications that use only one library. Some libraries provide convenience routines for converting a list of DM parameters to a list of library-specific parameters.

It's essential to understand the physical meaning of the attributes defined by each parameter. Knowing the meanings of the attributes helps you get the intended results from your application and helps you recognize and be able to use DM parameters and their clones throughout the family of Digital Media Libraries.

The sections that follow describe the essential attributes of digital image and audio data, their physical meanings, and the DM parameters that define them.

Digital Image Essentials

This section presents essential image concepts about color and video.

Color Concepts

Important color concepts discussed in this section are:

- Colorspace
- Intensity
- Gamma
- Luma and Luminance
- Chroma and Chrominance
- Chromaticity

Colorspace

Two dimensional digital images are composed of a number of individual picture elements (pixels) obtained by sampling an image in 2D space. Each pixel contains the intensity and, for color images, the color information for the region of space it occupies.

Color data is usually stored on a per component basis, with a number of bits representing each component. A color expressed in RGB values is said to exist in the RGB colorspace. To be more precise, a pixel is actually a vector in colorspace. There are other ways to encode color data, so RGB is just one type of colorspace.

There are four colorspaces to know about for the Digital Media Libraries:

- full-range RGB with the following properties:
  - component R, G, B
  - each channel ranges from [0.0 - 1.0]; mapped onto [0..2^n-1]
  - alpha is [0.0 - 1.0]; mapped onto [0..2^n-1]

- compressed-range RGB with the following properties:
  - component R, G, B
  - each channel ranges from [0.0 - 1.0]; mapped onto [64..940] (10-bit mode) and [16..235] (8-bit mode)
  - alpha ranges from[0.0 - 1.0]; mapped onto [64..940] (10-bit mode) and [16..235] (8-bit mode)
- full-range YUV with the following properties:
  - component Y, Cr, Cb
  - Y (luma) channel ranges from [0.0 - 1.0] mapped onto [0..2^n-1]
  - Cb and Cr (chroma) channels range from [-0.5 - +0.5] mapped onto [0..2^n-1]
  - alpha ranges from [0.0 - 1.0] mapped onto [0..2^n-1]
  - colorspace is as defined in Rec 601 specification
- compressed-range YUV with the following properties:
  - component Y, Cr, Cb
  - Y (luma) channel ranges [0.0 - 1.0] mapped onto [64..940] (10-bit mode) and [16..235] (8-bit mode)
  - Cb and Cr (chroma) channels range [-0.5 - +0.5] mapped onto [64..960] (10 bit mode) and [16..240] (8-bit mode)
  - alpha is [0.0 - 1.0] mapped onto [64..940] (1- bit mode) and [16..235] (8-bit mode)
  - colorspace is as defined in Rec. 601 specification

On the display screen, each pixel is actually a group of three phosphors (red, green, and blue) located in close enough proximity that they are perceived as a single color. There are some issues with anomalies related to the physical properties of screens and phosphors that are of interest for programmers. See “Freezing Video” in Chapter 4 for more information about these issues.

Intensity

Humans perceive brightness in such a way that certain colors appear to be brighter or more intense than others. For example, red appears to be brighter than blue, and green appears to be brighter than either blue or red. Secondary colors (cyan, magenta, and yellow), each formed by combining two primary colors, appear to be even brighter still. This phenomenon is simulated in Figure 2-1.

Figure 2-1 is a plot that simulates the human eye's response to the light intensity of different wavelengths (colors) of light. Pure colors (red, green, blue, yellow, cyan, and magenta) are plotted in YCrCb colorspace with brightness plotted along the horizontal axis. The rightmost colors are perceived as brighter than the leftmost colors, even though all the colors are of equal brightness.

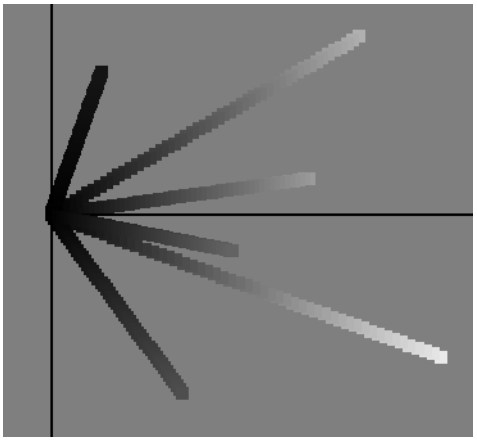


Figure 2-1 Plot Simulating Human Visual Perception of Brightness vs. Color

The brightness of a color is measured by a quantity called *saturation*. Figure 2-2 connects the maximum saturation points of each color. The lines tracing the path of maximum saturation from the values at each corner are called hue lines.

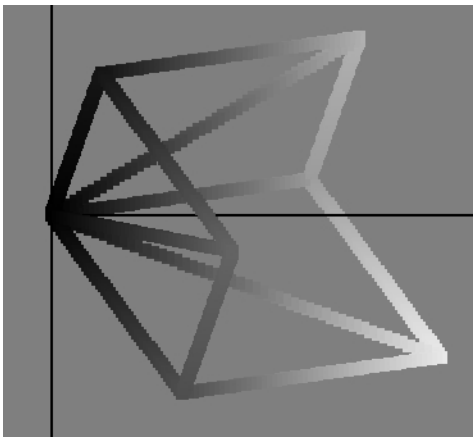


Figure 2-2 Hue and Saturation

The program that generated Figure 2-1 and Figure 2-2, *colorvu*, uses the Colorspace API described in "The Color Space Library" in Appendix A.

Both the human visual perception of light intensity and the physical equipment used to convey the sensation of brightness in a computer display are inherently nonlinear, but in different ways. As it happens, the differences actually complement each other, so that the nonlinearity inherent in the way a computer display translates voltage to brightness almost exactly compensates for the way the human eye perceives brightness, but some correction is still necessary, as explained in the next section, "Gamma."

Gamma

Cameras, computer displays, and video equipment must account for both the human visual response and the physical realities of equipment in order to create a believable image. Typically a *gamma* factor is applied to color values to correctly represent the visual perception of color, but the way in which the gamma correction is applied, the reason for doing so, and the actual gamma value used depend on the situation.

It is helpful to understand when in the image processing path gamma is applied. Computers apply a gamma correction to output data to correctly reproduce intensity when displaying visual data. Cameras compensate for the nonlinearity of human vision by applying a gamma correction to the source image data as it is captured.

The key points to know when working with image data are whether the data has been (or should be) gamma-corrected, and what is the value of the gamma factor. Silicon Graphics monitors apply a default gamma correction of 1.7 when displaying RGB images.

You can customize the gamma function by specifying gamma coefficients for image converters as described in "The Digital Media Color Space Library" in Chapter 6.

Luma and Luminance

Video uses a *nonlinear* quantity, referred to as *luma*, to convey brightness. Luma is computed as a weighted sum of gamma-corrected RGB components. Color science theory represents the sensation of brightness as a *linear* quantity called *luminance*, which is computed by adding the red, green, and blue components, each weighted by a linear gamma factor that mimics human visual response.

In video and software documentation, the terms luma and luminance are often used interchangeably, and the letter Y can represent either quantity. Some authors (Poynton and others) use a prime symbol to denote a nonlinear quantity, and so represent luma as Y', and luminance as Y. This type of notation emphasizes the difference between a nonlinear and linear quantity, but it is not common practice, so it is important to realize that there are differences. Unless otherwise noted, in this document and in the Digital Media Libraries, Y refers to the nonlinear luma.

Chroma and Chrominance

To supply the color information for video signal encoding, the luma value (Y) is subtracted from the red and blue color components, giving two color difference signals: B–Y (B minus Y) and R–Y (R minus Y), which together are called chroma. As in the case of luma and luminance, the term chrominance is sometimes mistakenly used to refer to chroma, but the two terms signify different quantities. In the strictest sense, chrominance refers to a representation of a color value expressed independently of luminance, usually in terms of chromaticity.

Chromaticity

Color science uses *chromaticity* values to express absolute color in the absence of brightness. Chromaticity is a mathematical abstraction that is not represented in the physical world, but is useful for computation. Chrominance is often expressed in terms of chromaticity. A CIE chromaticity diagram is an (x, y) plot of colors in the wavelengths

of visible light (400 nm to 700 nm). Color matching and similar applications require an understanding of chromaticity, but you probably don't need to use it for most applications written using the Digital Media Libraries.

Video Concepts

Important video concepts to be familiar with are the distinction between digital and analog video, video formats, black level, fields, and interlacing. This section contains these topics, which highlight key video concepts:

- YCrCb and Component Digital Video
- YUV and Composite Analog Video
- Black Level
- Video Fields
- Field Dominance

YCrCb and Component Digital Video

Because the human perception of brightness varies depending on color, some image encoding formats can take advantage of that difference by separating image data into separate components for brightness and color. One such method is the component digital video standard established by ITU-R BT.601 (also formerly known as CCIR Recommendation 601, or often simply Rec. 601).

For the digital video formats, Rec. 601 defines some basic properties common to digital component video, such as pixel sampling rate and colorspace, regardless of how it is transmitted. Then, the more specific documents (SMPTE 125M, SMPT259M, and ITU-R BT.656) define how the data format defined by Rec. 601 is to be transmitted over various kinds of links (serial, parallel) with various numbers of lines (525 or 625).

Component digital video uses scaled chroma values, called Cr and Cb, which are combined with luma into a signal format called YCrCb. This YCrCb refers to a signal format that is transmitted over a wire. It is related to, but separate from, the YCrCb colorspace used to store samples in computer memory.

Recommendation 601 defines methods for subsampling chroma values. The most common subsampling method is 4:2:2, where there is one pair of Cr, Cb samples for every other Y sample. In 4:4:4 subsampling, there is a luma sample for every chroma sample.

YUV and Composite Analog Video

There are also composite analog video encoding signals, YUV and YIQ, which are based on color-difference signals. In analog composite video, the two color-difference signals (U,V or I,Q) are combined into a chroma signal, which is then combined with the luma for transmission. NTSC and PAL are the two main standards for encoding and transmitting analog composite video. See SMPTE 170M for more information about analog video broadcast standards.

The important point to realize about YUV is that, like YCrCb, it is calculated by scaling color difference values, but different scale factors are used to obtain YUV than are used for YCrCb. The YUV and YCrCb colorspaces are similar to each other; they differ primarily in the ranges of acceptable values for the three components when represented as digital integers. The values of Y, U and V are in the range 0..255 (the SMPTE YUV ranges), while the range for Rec. 601 YCrCb is 16..235/240.

It is important to keep these differences in mind when selecting the colorspace for storing data in memory. While the terms YUV and YCrCb are used interchangeably and describe both video signals and a colorspace for encoding data in computer memory, they are separate concepts. Knowing and being able to specify precisely the colorspace of input data and the memory format you want are the keys to obtaining satisfactory results.

Analog video input to your workstation through an analog video connector is digitized and often converted to YCrCb in memory. YCrCb is also the colorspace used in many compression schemes (for example, MPEG and JPEG). On some Silicon Graphics video devices and connectors, component analog such as BetaSP and MII formats are digitized into a full-range YUV representation.

When working with analog video, the main points to be aware of are:

- bandwidth limitations (composite analog video uses a method devised to cope with bandwidth restrictions of early transmission methods for broadcast color television)
- chroma/luma crosstalk
- chroma aliasing

Refer to the video references listed in the introduction of this guide for more information.

Black Level

A common problem when importing video data for computer graphics display (or outputting synthesized computer graphics to video) is that pictures can look darker than expected or can look somewhat hazy because video and computer graphics use a different color scale. In Rec. 601 video, the *black level* (blackest black) is 16, but in computer graphics, 0 is blackest black. If a picture whose blackest black is 16 is displayed by a system that uses 0 as the blackest black, the image colors are all grayed out as a result of shifting the colors to this new scale. The best results are obtained by choosing the correct colorspace. The black level is related to bias, which sets the reference level for a color scale.

Video Fields

Video is sampled both spatially and temporally. Video is sampled and displayed such that only half the lines needed to create a picture are scanned at a particular instant in time. This is a result of the historical bandwidth limitations of broadcast video, but it is an important video concept.

A video field is a set of image samples, practically coincident in time, that is composed of every other line of an image. Each field in a video sequence is sampled at a different time, determined by the video signal's field rate. Upon display, two consecutive fields are *interlaced*, a technique whereby a video display scans every other line of a video image at a rate fast enough to be undetectable to the human eye. The persistence of the phosphors on the display screen holds the impression of the first set of scan lines just long enough for them to be perceived as being shown simultaneously to the second set of scan lines. (Actually, this is only strictly true of tube-based display devices whose electron beams take a whole field time to scan each line across the screen (from left-to-right then top-to-bottom). Array-based display devices change the state of all the pixels on the screen (or all the pixels on a given line) simultaneously)

The human eye cannot detect and resolve the two fields in a moving image displayed in this manner, but it can detect them in a still image, such as that shown when you pause a videotape. When you attempt to photograph or videotape a computer monitor using a camera, this effect is visible.

Most video signals in use today, including the major video signal formats you are likely to encounter and work with on a Silicon Graphics computer (NTSC, PAL, and 525- and 625-line Rec. 601 digital video), are field-based rather than frame-based. Correctly handling fields in software involves understanding the effects of temporal and spatial sampling.

Suppose you have an automatic film advance camera that can take 60 pictures per second, with which you take a series of pictures of a moving ball. Figure 2-3 shows 10 pictures from that sequence (different colors are used to emphasize the different positions of the ball in time). The time delay between each picture is a 60th of a second, so this sequence lasts 1/6th of a second.

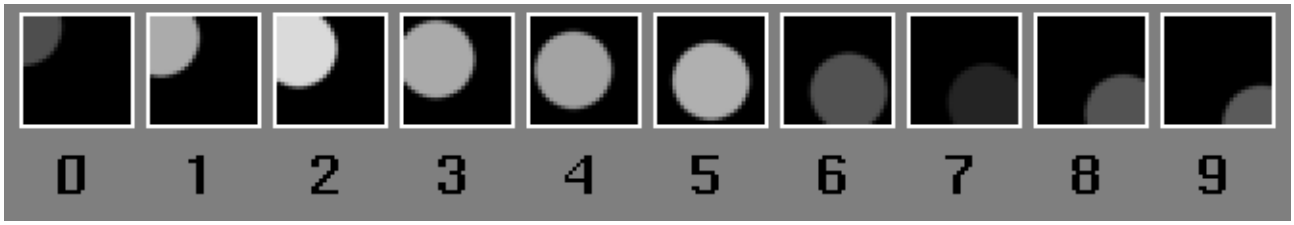


Figure 2-3 10 Pictures From a Film Camera Taken at 60 Pictures Per Second

Now suppose you take a modern NTSC video camera and shoot the same sequence. NTSC video has 60 fields per second, so you might think that the video camera would record the same series of pictures as Figure 2-3, but it does not. The video camera does record 60 images per second, but each image consists of only half of the scanlines of the complete picture at a given time, as shown in Figure 2-4, rather than a filmstrip of 10 complete images.

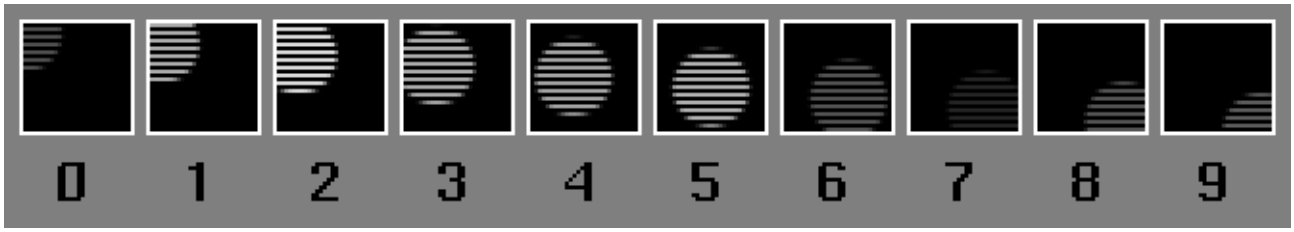


Figure 2-4 10 Fields From a 60-Field-Per-Second Video

Notice how the odd-numbered images contain one set of lines, and the even-numbered images contain the other set of lines (if you can't see this, click on the figure to bring up an expanded view).

Video data does not contain one complete image stored in every other frame, as shown in Figure 2-5.

Figure 2-5 One Common Misinterpretation of Video Fields

Nor does video data contain two consecutive fields, each containing every other line of an identical image, as shown in Figure 2-6.

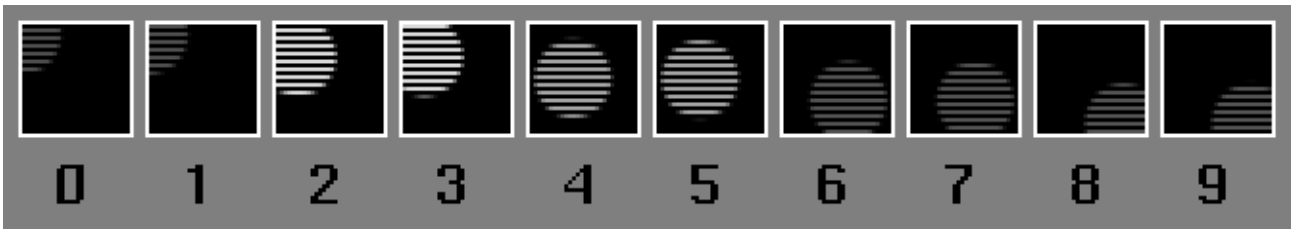


Figure 2-6 Video is not Pairs of Fields of Identical Images With Alternate Scanlines

Data in video fields are temporally and spatially distinct. In any video sequence, half of the spatial information is omitted for every temporal instant. This is why you cannot treat video data as a sequence of intact image frames. See "Freezing Video" in Chapter 4 for methods of displaying still frames of motion video.

Other video formats, many of which are used for computer monitors, have only one field per frame (often the term field is not used at all in these cases), which is called *noninterlaced* or *progressive scan*. Sometimes, video signals have fields, but the fields are not spatially distinct. Instead, each field contains the information for one color basis vector (R, G, and B for example); such signals are called *field sequential*.

It is important to use precise terminology when writing software or communicating with others regarding fields. Some terminology for describing fields is presented next.

Interlaced video signals have a natural two-field periodicity. F1 and F2 are the names given to each field in the sequence. When viewing the waveform of a video field on an

oscilloscope, you can tell whether it is an F1 field or an F2 field by the shape of its sync pulses.

ANSI/SMPTE 170M-1994 defines Field 1, Field 2, Field 3, and Field 4 for NTSC.

ANSI/SMPTE 125M-1992 defines the 525-line version of the bit-parallel digital Rec.-601 signal, using an NTSC waveform for reference. ANSI/SMPTE 259M-1993 defines the 525-line version of the bit-serial digital Rec.-601 signal in terms of the bit-parallel signal. 125M defines Field 1 and Field 2 for the digital signal.

Rec. 624-1-1978 defines Field 1 and Field 2 for 625-line PAL.

Rec. 656 Describes a 625-line version of the bit-serial and bit-parallel Rec.-601 digital video signal. It defines Field 1 and Field 2 for that signal.

The Digital Media Libraries define F1 as an instance of Field 1 or Field 3 and F2 as an instance of Field 2 or Field 4.

Field Dominance

Field dominance is relevant when transferring data in such a way that frame boundaries must be known and preserved, such as:

- GPI/VLAN/LTC-triggered capture or playback of video data
- edits on a VTR
- interpretation of fields in a VLBuffer for the purposes of interlacing or deinterlacing

Field dominance defines the order of fields in a frame and can be either F1 dominant or F2 dominant.

F1 dominant specifies a frame as an F1 field followed by an F2 field. This is the protocol recommended by all of the specifications listed at the end of the “Video Fields” section.

F2 dominant specifies a frame as an F2 field followed by an F1 field. This is the protocol followed by several New York production houses for the 525-line formats only.

Most older VTRs cannot make edits on any granularity finer than the frame. The latest generation of VTRs are able to make edits on arbitrary field boundaries, but can (and most often are) configured to make edits only on frame boundaries. Video capture or playback on a computer, when triggered, must begin on a frame boundary. Software

must interlace two fields from the same frame to produce a picture. When software deinterlaces a picture, the two resulting fields are in the same frame.

Regardless of the field dominance, if there are two contiguous fields in a VLBuffer, the first field is always temporally earlier than the second one: Under no circumstances should the temporal ordering of fields in memory be violated.

The terms even and odd can refer to whether a field's active lines end up as the even scanlines of a picture or the odd scanlines of a picture. In this case, you need to additionally specify how the scanlines of the picture are numbered (beginning with 0 or beginning with 1), and you may need to also specify 525 vs. 625 depending on the context.

Even and odd could also refer to the number 1 or 2 in F1 and F2, which is of course a different concept that only sometimes maps to the notion of whether a field's active lines end up as the even scanlines of a picture or the odd scanlines of a picture. This definition seems somewhat more popular.

For example:

- VL_CAPTURE_ODD_FIELDS captures F1 fields
- VL_CAPTURE_EVEN_FIELDS captures F2 fields

The way in which two consecutive fields of video should be interlaced to produce a picture depends on

- which field is an F1 field and which field is an F2 field
- whether the fields are from a 525- or 625-line signal.

The interlacing does not depend on

- the relative order of the fields, that is, which one is first
- anything relating to field dominance

Line numbering in memory does not necessarily correspond to the line numbers in a video specification. Software line numbering can begin with either a 0 or 1. Picture line numbering scheme in software is shown both 0-based (like the Movie Library) and 1-based.

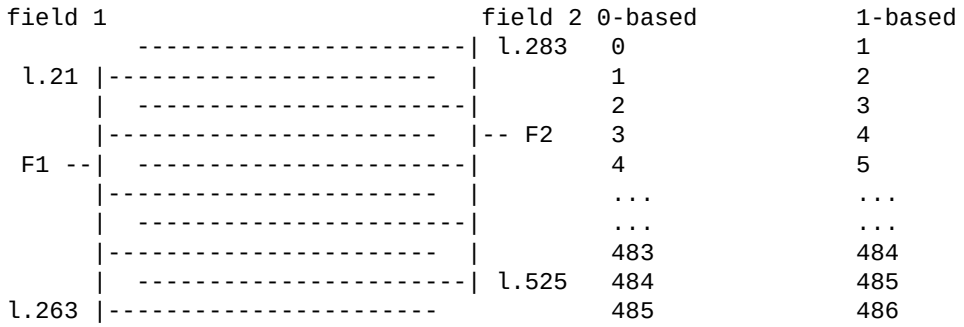
For 525-line analog signals, the picture should be produced in this manner (F1 has 243 active lines, F2 has 243 active lines, totaling 486 active lines):

```
field 1                          field 2  0-based          1-based
         (second half only)-----| l.283   0               1
 l.21 |-----------------------  |         1               2
      |  -----------------------|         2               3
      |-----------------------  |-- F2    3               4
 F1 --|  -----------------------|         4               5
      |-----------------------  |         ...             ...
      |  -----------------------|         ...             ...
      |-----------------------  |         483             484
      |  -----------------------| l.525   484             485
l.263 |------(first half only)            485             486
```

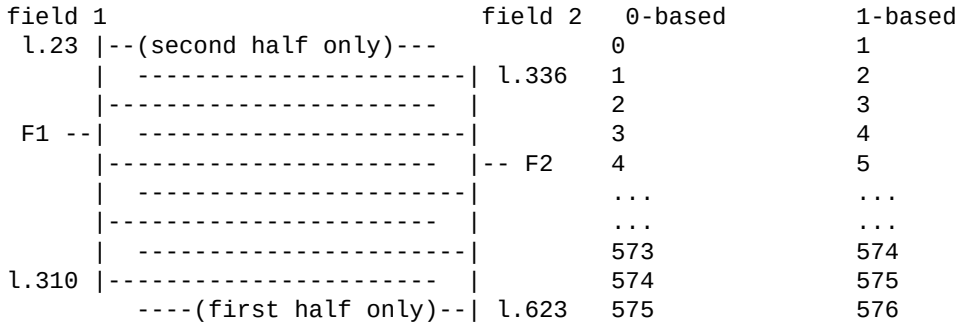
For official 525-line digital signals, the picture should be produced in this manner (F1 has 244 active lines, F2 has 243 active lines, totaling 487 active lines):

```
field 1                          field 2  0-based          1-based
 l.20 |-----------------------             0               1
      |  -----------------------| l.283   1               2
 l.21 |-----------------------  |         2               3
      |  -----------------------|         3               4
      |-----------------------  |-- F2    4               5
 F1 --|  -----------------------|         5               6
      |-----------------------  |         ...             ...
      |  -----------------------|         ...             ...
      |-----------------------  |         484             485
      |  -----------------------| l.525   485             486
l.263 |-----------------------             486             487
```

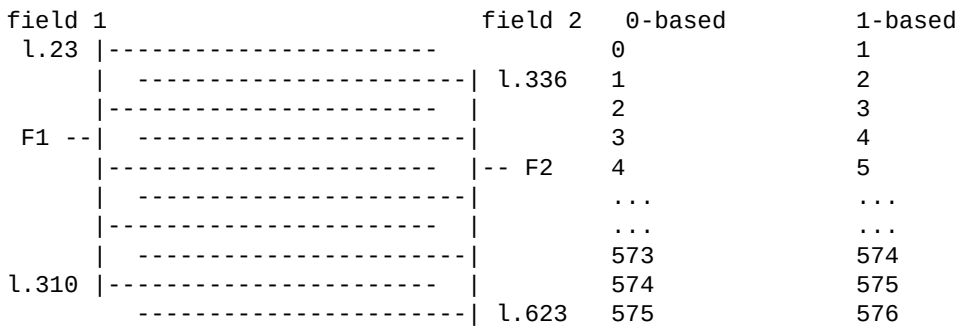
For practical 525-line digital signals, all current Silicon Graphics video hardware skips line 20 of the signal and pretends that the signal has 486 active lines. As a result, you can think of the digital signal as having exactly the same interlacing characteristics and line numbers as the analog signal (F1 has 243 active lines and F2 has 243 active lines, totaling 486 active lines):



For 625-line analog signals, the picture should be produced in this manner (F1 has 288 active lines, F2 has 288 active lines):



For 625-line digital signals, the picture should be produced in this manner (F1 has 288 active lines, F2 has 288 active lines):



All Field 1 and Field 2 line numbers match those in SMPTE 170M and Rec. 624. Both of the digital specs use identical line numbering to their analog counterparts. However, *Video Demystified* and many chip specifications use nonstandard line numbers in some

(not all) of their diagrams. A word of caution: 125M draws fictitious half-lines in its figure 3 in places that do not correspond to where the half-lines fall in the analog signal.

Digital Image Attributes

This section describes digital image data attributes and how to use them. Image attributes can apply to the image as a whole, to each pixel, or to a pixel component.

Parameters in *dmedia/dm_image.h* provide a common language for describing image attributes for the Digital Media Libraries. Not all of the libraries require or use all of the DM image parameters. Clones of some DM image parameters can be found in *vl.h*.

Digital image attributes described in this section are:

- Image Dimensions
- Pixel Aspect Ratio
- Image Rate
- Image Compression
- Image Quality
- Bitrate
- Keyframe/Reference Frame Distance
- Image Orientation
- Image Interlacing
- Image Layout
- Image Pixel Attributes, including
  - Pixel Packing
  - Pixel Component Data Type
  - Pixel Component Order and Interleaving
- Image Sample Rate

These attributes and the parameters that represent them are discussed in detail in the sections that follow.

Image Dimensions

Image size is measured in pixels: DM_IMAGE_WIDTH is the number of pixels in the x (horizontal) dimension, and DM_IMAGE_HEIGHT is the number of pixels in the y (vertical) dimension.

Video streams and movie files contain a number of individual images of uniform size. The image size of a video stream or a movie file refers to the height and width of the individual images contained within it, and is often referred to as *frame size*.

Some image formats require that the image dimensions be integral multiples of a factor, necessitating either cropping or padding of images that don't conform to those requirements.

Note: To determine the size of an image in bytes, use **dmImageFrameSize(3dm)**.

Pixel Aspect Ratio

Pixels aren't always perfectly square, in fact they often aren't. The shape of the pixel is defined by the *pixel aspect ratio*. The pixel aspect ratio is obtained by dividing the pixel height by the pixel width and is represented by DM_IMAGE_PIXEL_ASPECT.

Square pixels have a pixel aspect ratio of 1.0. Some video formats use nonsquare pixels, but computer display monitors typically have square pixels, so a square/nonsquare pixel conversion is needed for the image to look correct when displaying digital video images on the graphics monitor.

In general graphics rendering and display devices typically generate/accept only square pixels, but video I/O devices can typically generate/accept either square or nonsquare formats. It is probably preferable to use/retain a nonsquare format for an application whose purpose is to produce video, while it is probably preferable for an application whose ultimate intent is producing computer graphics to use/retain a square format. Whether a conversion is necessary or optimal depends on the original image source, the final destination, and, to a certain extent, the hardware path transporting the signal.

For example, the digital sampling of analog video in accordance to Rec. 601 yields a nonsquare pixel. On the other hand, graphics displays render each pixel as square. This means that a Rec. 601 nonsquare or video input stream sent directly (without filtering) to the workstation's video output displays correctly on an external video monitor, but does

not display correctly when sent directly (without filtering) to an onscreen graphics window.

Conversely, computer-originated digital video (640x480 and 768x576) displays incorrectly when sent to video out in nonsquare mode, but displays correctly when sent to an onscreen graphics window or to video out in square mode.

Some Silicon Graphics video devices sample natively using only one format, either square or nonsquare, and some Silicon Graphics video devices filter signals on certain connectors. See the video device reference pages for details.

Some video options for Silicon Graphics workstations perform square/nonsquare filtering in hardware; refer to your owner's manual to determine whether your video option supports this feature. Software filtering is also possible.

Image Rate

DM_IMAGE_RATE is the native display rate of a movie file in frames per second.

Image Compression

Compression is a method of encoding data more efficiently without changing its content significantly.

A codec (compressor/decompressor) defines a compressed data format. In some cases such as MPEG, the codec also defines a standard file format in which to contain data of that format. Otherwise, there is a set of file formats that can hold data of that format.

A "stateful" algorithm works by encoding the differences between multiple frames, as opposed to encoding each frame independently of the others. Stateful codecs are hard to use in an editing environment but generally produce better compression results because they get access to more redundancy in the data.

A "tile-based" algorithm (such as MPEG) divides the image up into (what is usually) a grid of fixed sections, usually called blocks, macroblocks, or macrocells. The algorithm then compresses each region independently. Tile-based algorithms are notorious for producing output with visible blocking artifacts at the tile boundaries. Some algorithms specify that the output is to be blurred to help hide the artifacts.

A "transform-based" algorithm (such as JPEG) takes the pixels of the image (which constitute the spatial domain) and transforms them into another domain—one in which data is more easily compressed using traditional techniques (such as RLE, Lempel-Ziv, or Huffman) than the spatial domain. Such algorithms generally do a very good job at compressing images. The computational cost of the transformation is generally high, so:

- Transform-based algorithms are typically more expensive than spatial domain algorithms.
- Transform-based algorithms are typically also tile-based algorithms (since the computation is easier on small tiles), and thus suffer the artifacts of tile-based algorithms.

For most compression algorithms, the compressed data stream is designed so that the video can be played forward or backward, but some compression schemes, such as MPEG, are predictive and so are more efficient for forward playback.

Note: In general, JPEG, MPEG, Cinepak, Apple Video and other video compression algorithms are better for compressing camera-generated images; RLE, Apple Animation and other color-cell techniques are better for compressing synthetic (computer-generated) images.

JPEG Still Video Compression

Although any algorithm can be used for still video images, the JPEG *(Joint Photographic Experts Group*)-baseline algorithm, which is referred to simply as JPEG for the remainder of this guide, is the best for most applications. JPEG is denoted by the DM parameter DM_IMAGE_JPEG.

JPEG is a compression standard for compressing full-color or grayscale digital images. It is a lossy algorithm, meaning that the compressed image is not a perfect representation of the original image, but you may not be able to detect the differences with the naked eye. Because each image is coded separately (intra-coded), JPEG is the preferred standard for compressed digital nonlinear editing.

JPEG is based on psychovisual studies of human perception: Image information that is generally not noticeable is dropped out, reducing the storage requirement anywhere from 2 to 100 times. JPEG is most useful for still images; it is usable, but slow when performed in software, for video. (Silicon Graphics hardware JPEG accelerators are available for compressing video to and decompressing video from memory, or for compressing to and decompressing from a special video connection to a video board. These JPEG hardware accelerators implement a subset of the JPEG standard (baseline

JPEG, interleaved YCrCb 8-bit components) especially for video-originated images on Silicon Graphics workstations.

The typical use of JPEG is to compress each still frame during the writing or editing process, with the intention of applying another type of compression to the final version of the movie or to leave it uncompressed. JPEG works better on high-resolution, continuous-tone images such as photographs, than on crisp-edged, high-contrast images such as line drawings.

The amount of compression and the quality of the resulting image are independent of the image data. The quality depends on the compression ratio. You can select the compression ratio that best suits your application needs.

For more information, see jpeg(4). See also Pennebaker, William B. and Joan L. Mitchell, *JPEG Still Image Data Compression Standard*, New York: Van Nostrand Reinhold, 1993 (ISBN 0-442-01272-1).

MPEG-1

MPEG-1 (ISO/IEC 11172) is the *Moving Pictures Expert Group* standard for compressing audio, video, and systems bitstreams. The MPEG-1 systems specification defines multiplexing for compressed audio and video bitstreams without performing additional compression. An MPEG-1 encoded systems bitstream contains compressed audio and video data that has been packetized and interleaved along with timestamp and decoder buffering requirements. MPEG-1 allows for multiplexing of up to 32 compressed audio and 16 compressed video bitstreams. Each bitstream type has its own syntax, as defined by the standard.

MPEG-1 video (ISO/IEC 11172-2) is a motion video compression standard that minimizes temporal and spatial data redundancies to achieve good image quality at higher compression ratios than either JPEG or MVC1.

MPEG-1 video uses a technique called *motion estimation* or *motion search*, which compresses a video stream by comparing image data in nearby image frames. For example, if a video shows the same subject moving against a background, it's likely that the same foreground image appears in adjacent frames, offset by a few pixels. Compression is achieved by storing one complete image frame, which is called a *keyframe* or *I* frame, then comparing an $n \times n$ block of pixels to nearby pixels in proximal frames, searching for the same (or similar) block of pixels, and then storing only the offset for the frames where a match is located. Images from the intervening frames can then be reconstructed by combining the offset data with the keyframe data.

There are two types of intervening frames:

- P (predictive) frames, which require information from previous P or I frames in order to be decoded. P frames are also sometimes considered forward reference frames because they contain information needed to decode other P frames later in the video bitstream.
- B (between) frames, which require information from both the previous and next P or I frame.

Figure 2-7 shows the relationships between I, P, and B frames.

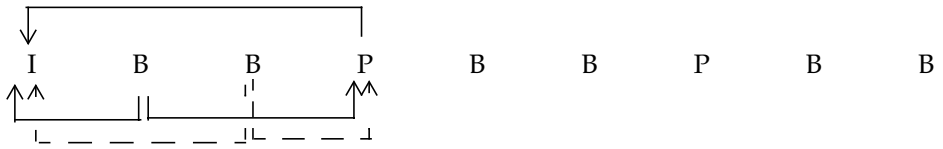


Figure 2-7 MPEG I, P, and B Frames

For example, suppose an MPEG-1 video bitstream contains the sequence I_0 P_3 B_1 B_2 P_6 B_4 B_5 P_9 B_7 B_8, where the subscripts indicate the order in which the frames are to be displayed. You must first display I_0 and retain its information in order to decode P_3, but you cannot yet display P_3 because you must first decode and display the two between frames (B_1 and B_2), which also require information from P_3, as well as from each other, to be decoded. Once B_1 and B_2 have been decoded and displayed, you can display P_3, but you must retain its information in order to decode P_6, and so on.

MPEG is an asymmetric coding technique—compression requires considerably more processing power than decompression because MPEG examines the sequence of frames and compresses them in an optimized way, including compressing the difference between frames using motion estimation. This makes MPEG well suited for video publishing, where a video is compressed once and decompressed many times for playback. Because MPEG is a predictive scheme, it is tuned for random access (editing) due to its inter-coding, or for forward playback rather than backward. MPEG is used on Video CD, DVD, Direct TV, and is the proposed future standard for digital broadcast TV.

For more information, see mpeg(4).

Run-Length Encoding

Run-length encoding (RLE) compresses images by replacing pixel values that are repeated for several pixels in a row with a single pixel at the first occurrence of a particular value, followed by a run-length (a count of the number of subsequent pixels of the same value) every time the color changes. Although this algorithm is lossless, it doesn't save as much space as the other compression algorithms—typically less than 2:1 compression is achieved. It is a good technique for animations where there are large areas with identical colors. The Digital Media Libraries have two RLE methods:

DM_IMAGE_RLE
: Specifies lossless RLE for 8-bit RGB data. It is the only algorithm currently available to directly compress 8-bit RGB data.

DM_IMAGE_RLE24
: Specifies lossless RLE for 24-bit RGB data.

Silicon Graphics Motion Video Compressor

Motion Video Compressor (MVC) is a Silicon Graphics proprietary algorithm that is a good general-purpose compression scheme for movies. MVC is a color-cell compression technique that works well for video, but can cause fuzzy edges in high-contrast animation. There are 2 versions:

DM_IMAGE_MVC1
: A fairly lossy algorithm that does not produce compression ratios as high as JPEG, but it is well suited to movies.

DM_IMAGE_MVC2
: Provides results similar to MVC1 in terms of image quality. MVC2 compresses the data more than MVC1, but takes longer to perform the compression. Playback is faster for MVC2, because there is less data to read in, and decompression is faster than for MVC1.

QuickTime Compression

QuickTime is an Apple Macintosh® system software extension that can be installed in the Macintosh system to extend its capabilities so as to allow time-based (audio, video, and animation) data for multimedia applications.

Movies compressed with QuickTime store and play picture tracks and soundtracks independently of each other, analogous to the way the Movie Library stores separate image and audio tracks. You can't work with pictures and sound as separate entities using the QuickTime Starter Kit utilities on the Macintosh, but you can use the Silicon Graphics Movie Library to work with the individual image and audio tracks in a QuickTime movie.

QuickTime movie soundtracks are playable on Macintosh and Silicon Graphics computers, but each kind of system has a unique audio data format, so audio playback is most efficient when using the native data format and rate for the computer on which the movie is playing.

The Macintosh QuickTime system software extension includes five codecs:

- Apple None (uncompressed)
- Apple Photo (JPEG standard)
- Apple Animation
- Apple Video
- Apple Compact Video (Cinepak)

Apple None

Apple None creates an uncompressed movie and can be used to change the number of colors in the images and/or the recording quality. Both the number of colors and the recording quality can affect the size of the movie.

To create an uncompressed QuickTime movie on the Macintosh, click on the "Apple None" choice in the QuickTime Compression Settings dialog box.

Note: Because the Macintosh software compresses QuickTime movies by default, you must set the compression to Apple None and save the movie again to create an uncompressed movie.

Apple Photo

Apple Photo uses the JPEG standard. JPEG is best suited for compressing individual still frames, because decompressing a JPEG image can be a time-consuming task, especially if the decompression is performed in software. JPEG is typically used to compress each still frame during the writing or editing process, with the intention of applying another type of compression to the final version of the movie or leaving it uncompressed.

Apple Animation

Apple Animation uses a lossy run-length encoding (RLE) method, which compresses images by storing a color and its run-length (the number of pixels of that color) every time the color changes. Apple Animation is not a true lossless RLE method because it stores colors that are close to the same value as one color. This method is most appropriate for compressing images such as line drawings that have highly contrasting color transitions and few color variations.

Apple Video

Apple Video uses a method whose objective is to decompress and display movie frames as fast as possible. It compresses individual frames and works better on movies recorded from a video source than on animations.

Note: Both Apple Animation and Apple Video compression have a restriction that the image width and height be a multiple of 4. Before transferring a movie from a Macintosh to a Silicon Graphics computer, make sure that the image size is a multiple of 4.

Cinepak

Cinepak (developed by Radius, Inc.), otherwise known as “Compact Video,” is a compressed data format that can be stored inside QuickTime movies. It achieves better compression ratios than QuickTime but takes much more CPU time to compress.

The Cinepak format is designed to control its own bitrate, and thus it is extremely common on the World Wide Web and is also used in CD authoring.

Cinepak is not a transform-based algorithm. It uses techniques derived from “vector quantization” (which technically is also what color-cell compression techniques such as MVC1 and MVC2 use) to represent small tiles of pixels using a small set of scalars. Cinepak builds and constantly maintains a “codebook,” which it uses to map the

compressed scalars back into pixel tiles. The codebook evolves over time as the image changes, thus this algorithm is stateful.

Indeo

Indeo (developed by Intel Corporation) is a compressed data format that can be used in QuickTime and AVI movies.

Image Quality

Compressed data isn't always a perfect representation of the original data. Information can be lost in the compression process. A *lossless* compression method retains all of the information present in the original data. Algorithms can be either numerically lossless or mathematically lossless. Numerically lossless means that the data is left intact. Mathematically lossless means that the compressed data is acceptably close to the original data.

A *lossy* compression method does not preserve 100% of the information in the original method.

Image quality is a measure of how true the compression is to the original image. Image quality is one of the conversion controls that you can specify for an image converter. Image quality is specified in both the spatial and temporal domains.

In a spatial approximation, pixels from a single image are compared to each other, and identical (or similar) pixels are noted as repeat occurrences of a stored representative pixel. Spatial quality, denoted by DM_IMAGE_QUALITY_SPATIAL, conveys the exactness of a spatial approximation.

In a temporal approximation, pixels from an image stream are compared across time, and identical (or similar) pixels are noted as repeat occurrences of a stored representative pixel, but offset in time. Temporal quality, denoted by DM_IMAGE_QUALITY_TEMPORAL, conveys the exactness of a temporal approximation.

Some lossless algorithms may require a quality factor, so specify DM_IMAGE_QUALITY_LOSSLESS.

Quality values range from 0 to 1.0, where 0 represents complete loss of the image fidelity and 1.0 represents lossless image fidelity. You can set both quality factors numerically, or you can use the following rule-of-thumb factors to set quality informally:

DM_IMAGE_QUALITY_MIN	approximately equal to 0 quality factor
DM_IMAGE_QUALITY_LOW	approximately equal to 0.25 quality factor
DM_IMAGE_QUALITY_NORMAL	approximately equal to 0.5 quality factor
DM_IMAGE_QUALITY_HIGH	approximately equal to 0.75 quality factor
DM_IMAGE_QUALITY_MAX	approximately equal to 0.99 quality factor

Using these "fuzzy" quality factors can be useful if your application uses a thumbwheel or slider to let the user indicate quality. These quality factors can be assigned to intermediate steps in the slider or thumbwheel to give the impression of infinitely adjustable quality.

Bitrate

The compression ratio is a tradeoff between the quality and the bitrate. Adjusting either one of these parameters affects the other, and, if both are set, bitrate usually takes precedence in the Silicon Graphics Digital Media Libraries.

For applications that require a constant bitrate, such as applications that send data over fixed data rate carriers or playback image streams at a minimum threshold rate, set DM_IMAGE_BITRATE. The picture quality is then adjusted to achieve the stated rate. Some Silicon Graphics algorithms guarantee the bitrate, some try to achieve the stated rate, and some do not support a bitrate parameter.

Keyframe/Reference Frame Distance

Certain compression algorithms such as MPEG use a technique called *motion estimation*, which compresses an image stream by storing a complete keyframe and then encoding related image data in nearby image frames, as described in "MPEG-1." Images from the encoded frames are decoded based on the keyframes or other encoded frames that precede or follow the frame being decoded.

The Digital Media Libraries have their own terminology to define three types of frames possible in a motion estimation compression method:

Intra	depends only on itself; contains all data needed to construct a complete image. Also called I frame or keyframe.
Inter	depends on a previous inter- or intra-frame. Also called reference frame, P (predictive) frame, or delta frame.
Between	depends on previous *and* next inter- or intra-frame; cannot be reconstructed using another between frame. Also called B frame.

There are two parameters for setting the distance between keyframes and reference frames:

DM_IMAGE_KEYFRAME_DISTANCE	specifies the distance between keyframes
DM_IMAGE_REFFRAME_DISTANCE	specifies the distance between reference frames

Image Orientation

Image orientation refers to the relative ordering of the horizontal scan lines within an image. The scanning order depends on the image source and can be either top-to-bottom or bottom-to-top, but it is important to know which. The default DM_IMAGE_ORIENTATION for images created on a Silicon Graphics workstation is bottom-to-top, denoted by DM_IMAGE_BOTTOM_TO_TOP. Video and compressed video is typically oriented top-to-bottom.

Image Interlacing

Interlacing is a video display technique that minimizes the amount of video data necessary to display an image by exploiting human visual acuity limitations. Interlacing weaves alternate lines of two separate fields of video at half the scan rate. For an explanation of interlacing, see “Video Fields.”

Generally, interlacing refers to a technique for signal encoding or display, and interleaving refers to a method of laying out the lines of video data in memory.

Interleaving can also refer to how the samples of an image's different color basis vectors are arranged in memory, or how audio and video are arranged together in memory. Interleaving image pixel data is described in "Pixel Component Order and Interleaving."

A movie file encodes pairs of fields into what it calls frames, and all data transfers are on frame boundaries. A two-field image in a movie file does not always represent a complete video frame because it could be clipped or not derived from video. This is further complicated by that fact that both top-to-bottom and bottom-to-top ordering of video lines in images are supported.

DM_IMAGE_INTERLACING describes the original interlacing characteristics of the signal that produced this image (or lack of interlacing characteristics).

In a zero-based picture line numbering scheme for noninterlaced images:

- In a DM_IMAGE_INTERLACED_ODD image, the scanlines of the first field occupy the odd-numbered lines (1, 3, 5, 7, and so on).
- In a DM_IMAGE_INTERLACED_EVEN image, the scanlines of the first field occupy the even-numbered lines (0, 2, 4, 8, and so on).

In this sense, first field means the image that is first temporally and in memory.

Note: If the DM_IMAGE _ORIENTATION is DM_BOTTOM_TO_TOP instead of DM_TOP_TO_BOTTOM, then all temporal ordering and memory ordering rules are reversed.

For an example of how DM_IMAGE_INTERLACING relates to video, consider a top-to-bottom buffer containing unclipped video data (a buffer containing all the video lines described for analog 525, practical digital 525, analog 625, and digital 625-line signals). The buffer's DM_IMAGE_INTERLACING depends on many factors.

For a signal with F1 dominance, a frame consists of an F1 field followed by an F2 field (temporally and in memory). The DM_IMAGE_INTERLACING parameter determines which picture lines contain the first field's data:

- for an analog or practical digital 525-line image, DM_IMAGE_INTERLACED_ODD
- for an analog or digital 625-line image, DM_IMAGE_INTERLACED_EVEN

However, if the signal has F2 dominance, where a frame consists of F2 followed by F1, the first field is now an F2 field so:

- for an analog or practical digital 525-line image, DM_IMAGE_INTERLACED_EVEN
- for an analog or digital 625-line image, DM_IMAGE_INTERLACED_ODD

Image Layout

DM_IMAGE_LAYOUT describes how pixels are arranged in an image buffer. In the DM_IMAGE_LAYOUT_LINEAR layout, lines of pixels are arranged sequentially. This is the typical image layout for most image data.

DM_IMAGE_LAYOUT_GRAPHICS and DM_IMAGE_LAYOUT_MIPMAP are two special layouts optimized for presentation to Silicon Graphics hardware. Both are passthrough formats; they are intended for use with image data that is passed untouched from a Silicon Graphics graphics or video input source directly to hardware. Use DM_IMAGE_LAYOUT_GRAPHICS to format image data sent to graphics display hardware. Use DM_IMAGE_LAYOUT_MIPMAP to format image data that represents a texture mipmap that is sent to texture memory, such as a video texture.

Image Pixel Attributes

This section describes image attributes that are specified on a per-pixel or per-pixel-component basis. Understanding these attributes requires some familiarity with the color concepts described in “Digital Image Essentials.”

Pixel Packing

Pixel packing formats define the bit ordering used for packing image pixels in memory. Native packings are supported directly in hardware. In other words, native packings don’t require a software conversion.

DM_IMAGE_PACKING parameters describe pixel packings recognized by the dmIC and Movie Library APIs. In addition to the DM_IMAGE_PACKING formats, there is also a set of VL_PACKING parameters in *vl.h* that describe image packings. There are some VL_PACKINGS that have no corresponding DM_IMAGE_PACKINGS.

For some packings, the DM_IMAGE_DATATYPE parameter controls how data is packed within the pixel. For example, 10-bit-per-pixel data can be left or right-justified in a 16-bit word.

The most common ways of packing data into memory are YCrCb and 32-bit RGBA.

YCrCb (4:2:2) Video Pixel Packing

Rec. 601 component digital video (4:2:2 subsampled) is composed of one 8-bit Y (luma) component per pixel, and two chroma samples coincident with alternate luma samples, supplying one 8-bit Cr component per two pixels, and one 8-bit Cb sample per two pixels. This results in 2 bytes per pixel. This is the Silicon Graphics native format for storing video image data in memory, which is represented by the DM_IMAGE_PACKING parameter DM_IMAGE_PACKING_CbYCrY, and the VL_PACKING parameter VL_PACKING_YVYU_422_8.

Note: The SMPTE 259M (specification for transmitting Rec. 601 over a link) digital video stream contains 10 bits in each component. An 8-bit packing format such as VL_PACKING_YVYU_422_8 uses only 8 of the 10 bits. This often generates acceptable results for strictly video data, but in order to parse some forms of ancillary data (such as embedded audio data) from a video stream, it is necessary to input all 10 bits. Because 10 bits is an atypical quantity for computers, the most common technique is to left-shift each 10-bit quantity to a 16-bit value, resulting in a 4-byte per component format called VL_PACKING_YVYU_422_10, where the extra bits are zero-padded on input and ignored on output. Storing data in this format takes more memory space, but may be preferable to the cost of manipulating 10-bit packed data on the CPU.

The pixel packing is independent of the colorspace. The use of a packing named “YUV” or “YVYU” does not imply that the data packed is YUV data, as opposed to YCrCb data. When YCrCb data is being packed with a YUV packing, the Cr component is packed as U, and the Cb component is packed as V. The VL_PACKING_YVYU_422_8 packing is the only packing that is natively supported in hardware (requiring no software conversion) on all VL video devices.

The 422 designation in the packing name means that the pixels are packed so that each horizontally adjacent pair of pixels share one common set of chroma (for example, UV, or alternatively, CrCb) data. Each pixel has its own value of luma (Y) data. So, data is packed in pairs of two pixels, two Y values, and one U and one V (or alternatively, one Cr and one Cb) value pair, in each pixel pair. This pixel packing always has an even number of pixels in each row.

The YUV and YCrCb colorspaces are similar, but they differ primarily in the ranges of acceptable values for the three components when represented as digital integers. The values of Y, U and V are in the range 0..255 (the SMPTE YUV ranges), while the range for Rec. 601 YCrCb is 16..235/240.

The set of VL packings presently defined does not enable the application to choose between the YUV and Rec. 601 YCrCb colorspaces. When an application specifies VL_PACKING_YVYU_422_8, the resultant colorspace is either YUV or YCrCb, depending on the device and the source node from which the data is coming. Most external digital sources produce YCrCb data. IndyCam produces Rec. 601-compliant YCrCb. There is no way to tell, from the VL_PACKING control, which of those two spaces (YUV or YCrCb) is used.

Each type of VL video device has a different set of colorspaces and packings implemented in hardware. Any other colorspaces and/or packings are implemented by means of a software conversion. Table 2-1 shows which color space and packing combinations are implemented in hardware or software, or not at all, for each device.

The chipset used in VINO and EV1 to convert analog input to digital pixels produces YUV output, not YCrCb output. That is, the values of Y, U and V are in the range 0..255 (the SMPTE YUV ranges), not the smaller 16..235/240 range specified for Rec. 601 YCrCb. For some devices that can't convert colorspace in hardware, such as EV1, the VL converts from YUV to RGBX/RGBA in software.

The VL routines used for this purpose assume the input is Rec. 601 YCrCb, not YUV, regardless of what the hardware actually produces. Therefore, if the hardware doesn't support the desired colorspace, and you require an accurate colorspace conversion, then specify pixels in a colorspace supported by the hardware, and do the colorspace conversion using dmIC or similar software converter, rather than relying on an automatic software colorspace conversion.

With Sirius Video, colorspace and packing are independent. Colorspace is chosen by the settings of the VL_FORMAT on the memory drain node, according to table below, and any packing can be applied to any colorspace, whether it makes sense or not. Colorspace conversion occurs when the VL_FORMAT of the video source node and the VL_FORMAT of the memory drain node imply different colorspaces.

32-Bit RGBA Graphics Pixel Packing

In 32-bit RGBA, the A may be a “don’t care” or it may be an alpha channel, synthesized on the computer. This results in 4 bytes per pixel. In the VL, this is called VL_PACKING_RGBA_8, VL_PACKING_RGB_8, and VL_PACKING_ABGR_8.

Table 2-1 shows the results in memory of reading pixels (or the source for writing pixels) in various formats. Pixel 0 is the leftmost pixel read or written. An ‘x’ means don’t care (this bit is not used).

Memory layout is presented in 32-bit words, with the MSB on the left and the LSB on the right (read the bit numbers vertically).

Table 2-1 Pixel Packing Formats

MSB			LSB	Packing Format
33222222	**22221111**	**111111**		
10987654	**32109876**	**54321098**	**76543210**	<----Bit numbers
bbgggrrr	**bbgggrrr**	**bbgggrrr**	**bbgggrrr**	DM_IMAGE_PACKING_8BGR VL_PACKING_RGB_332_P
aaaaaaaa	**bbbbbbbb**	**gggggggg**	**rrrrrrrr**	DM_IMAGE_PACKING_ABGR VL_PACKING_RGBA_8
xxxxxxxx	**bbbbbbbb**	**gggggggg**	**rrrrrrrr**	DM_IMAGE_PACKING_XBGR VL_PACKING_RGB_8
uuuuuuuu	**yyyyyyyy**	**vvvvvvvv**	**yyyyyyyy**	DM_IMAGE_PACKING_CbYCrY
uuuuuuuu	**yyyyyyyy**	**vvvvvvvv**	**yyyyyyyy**	VL_PACKING_YVYU_422_8
				DM_IMAGE_PACKING_RBG323 VL_PACKING_RBG_323
xxxxxxxx	**xxxxxxxx**	**xxxxxxxx**	**bbgggrrr**	DM_IMAGE_PACKING_BGR233 VL_PACKING_RGB_332
xxxxxxxx	**xxxxxxxx**	**xxxxxxxx**	**rrrgggbb**	VL_PACKING_BGR_332
bbgggrrr	**bbgggrrr**	**bbgggrrr**	**bbgggrrr**	VL_PACKING_RGB_332_IP
bbgggrrr	**bbgggrrr**	**bbgggrrr**	**bbgggrrr**	
rrrgggbb	**rrrgggbb**	**rrrgggbb**	**rrrgggbb**	VL_PACKING_BGR332_P
rrrgggbb	**rrrgggbb**	**rrrgggbb**	**rrrgggbb**	

Table 2-1 (continued) Pixel Packing Formats

MSB			LSB	Packing Format
xxxxxxx	xxxxxxx	bbbbbggg	gggrrrrr	VL_PACKING_RGB-565
bbbbbggg	gggrrrrr	bbbbbggg	gggrrrrr	VL_PACKING_RGB_565_P
rrrrrrrr gggggggg	bbbbbbbb rrrrrrrr	gggggggg bbbbbbbb	rrrrrrrr gggggggg	VL_PACKING_RGB_565_IP
xxbbbbbb	bbbbgggg	gggggrr	rrrrrrrr	VL_PACKING_RGB_10
yyyyyyyy	yyyyyyyy	yyyyyyyy	yyyyyyyy	DM_IMAGE_PACKING_LUMINANCE VL_PACKING_Y_8_IP
xxxxxxxx	uuuuuuuu	yyyyyyyy	vvvvvvvv	DM_IMAGE_PACKING_CbYCr VL_PACKING_YUV_444_8
aaaaaaaa	uuuuuuuu	yyyyyyyy	vvvvvvvv	VL_PACKING_YUV_4444_8
xxuuuuuu	uuuuyyyy	yyyyyyvv	vvvvvvvv	VL_PACKING_YUV_444_10
rrrrrrrr	gggggggg	bbbbbbbb	aaaaaaaa	DM_IMAGE_PACKING_RGBA VL_PACKING_ABGR_8
vvvvvvvv	yyyyyyyy	uuuuuuuu	aaaaaaaa	DM_IMAGE_PACKING_CbYCrA VL_PACKING_AUYV_8
rrrrrrrr	rrgggggg	ggggbbbb	bbbbbbaa	VL_PACKING_A_2_BGR_10
vvvvvvvv	vvyyyyyy	yyyyuuuu	uuuuuuaa	VL_PACKING_A_2_UYV_10
uuuuuuuu	uuyyyyyy	yyyyaaaa	aaaaaaxx	VL_PACKING_AYU_AYV_10

Table 2-2 lists DM_IMAGE_PACKING formats.

Table 2-2 DM Pixel Packing Formats

Pixel Packing Format
DM_IMAGE_PACKING_RGB
DM_IMAGE_PACKING_BGR
DM_IMAGE_PACKING_RGBX
DM_IMAGE_PACKING_RGBA
DM_IMAGE_PACKING_XRGB
DM_IMAGE_PACKING_ARGB
DM_IMAGE_PACKING_XBGR
DM_IMAGE_PACKING_ABGR
DM_IMAGE_PACKING_RBG323
DM_IMAGE_PACKING_BGR233
DM_IMAGE_PACKING_XRGB1555
DM_IMAGE_PACKING_CbYCr
DM_IMAGE_PACKING_CbYCrA
DM_IMAGE_PACKING_CbYCrY
DM_IMAGE_PACKING_CbYCrYYY
DM_IMAGE_PACKING_LUMINANCE
DM_IMAGE_PACKING_LUMINANCE_ALPHA

Pixel Component Data Type

DM_IMAGE_DATATYPE describes the number of bits per component and the alignment of the bits within the pixel. Table 2-3 lists the data type parameters and the attributes they describe.

Table 2-3 Image Data Types

Image Data Type Parameter	Attributes
DM_IMAGE_DATATYPE_BIT	Nonuniform number of bits per component
DM_IMAGE_DATATYPE_CHAR	8 bits per component
DM_IMAGE_DATATYPE_SHORT10L	10 bits per component, left-aligned
DM_IMAGE_DATATYPE_SHORT10R	10 bits per component, right-aligned
DM_IMAGE_DATATYPE_SHORT12L	12 bits per component, left-aligned
DM_IMAGE_DATATYPE_SHORT12R	12 bits per component, right-aligned

Pixel Component Order and Interleaving

DM_IMAGE_ORDER describes the order of pixel components or blocks of components within an image and has one of the following formats:

- DM_IMAGE_ORDER_INTERLEAVED orders pixels component by component.
- DM_IMAGE_ORDER_SEQUENTIAL groups like components together line by line.
- DM_IMAGE_ORDER_SEPARATE groups like components together per image.

Table 2-4 shows the resultant pixel component order for each interleaving method for some example image formats.

Table 2-4 Pixel Interleaving Examples

Packing Format	Interleaved	Sequential	Separate
ABGR	ABGRABGR ABGRABGR ABGRABGR	AAABBBGGGRRR AAABBBGGGRRR	AAAAAAAA BBBBBBBB GGGGGGGG RRRRRRRR
444 YCrCb, with CbYCr packing	CbYCrCbYCr CbYCr	CbCbCbYYYCrCrCr CbCbCbYYYCrCrCr	CbCbCbCbCbCb YYYYYY CrCrCrCrCrCr
420 YCrCb, with CbYCrY packing	CbYCrYYYCb YCrYYYCbY CrYYY	CbCbCbYYYYYYYYYYYYCrCrCr CbCbCbYYYYYYYYYYYYCrCrCr	CbCbCbCbCbCb YYYYYYYYYYYY YYYYYYYYYYYY CrCrCrCrCrCr

Image Sample Rate

DM_IMAGE_RATE is the native display rate in frames per second of a movie file.

Digital Audio Essentials

The digital representation of an audio signal is generated by periodically sampling the amplitude (voltage) of the audio signal. The samples represent periodic "snapshots" of the signal amplitude. The Nyquist Theorem provides a way of determining the minimum sampling frequency required to accurately represent the information (in a given bandwidth) contained in an analog signal. Typically, digital audio information is sampled at a frequency that is at least double the highest interesting analog audio frequency. See *The Art of Digital Audio* or a similar reference on digital audio for more information.

Parameters in *dmedia/dm_audio.h* provide a common language for describing digital audio attributes for the digital media libraries.

This section contains these topics, which describe digital audio attributes:

- Audio Channels
- Audio Sample Rate
- Audio Compression Scheme
- Audio Sample Format (for example, twos-complement binary, floating point)
- PCM Mapping
- Audio Sample Width (number of bits per sample)

Audio Channels

A *sample frame* is a set of audio samples that are coincident in time. A sample frame for mono data is a single sample. A sample frame for stereo data consists of a left-right sample pair.

Stereo samples are interleaved; left-channel samples alternate with right-channel samples. 4-channel samples are also interleaved, with each frame usually having two left-right sample pairs, but there can be other arrangements.

Figure 2-8 shows the relationship between the number of channels and the frame size of audio sample data.

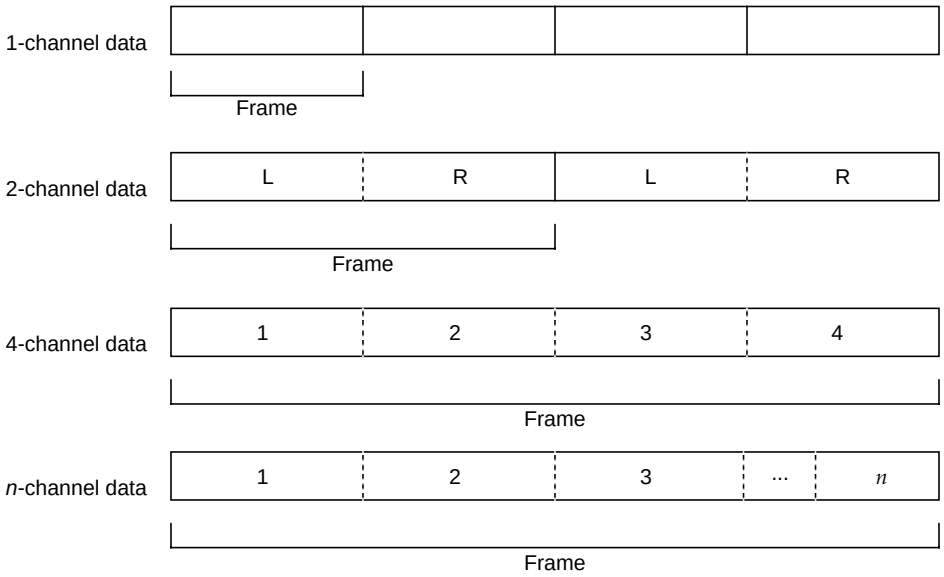


Figure 2-8 Audio Samples and Frames

Audio Sample Rate

The *sample rate* is the frequency at which samples are taken from the analog signal. Sample rates are measured in hertz (Hz). A sample rate of 1 Hz is equal to one sample per second. For example, when a mono analog audio signal is digitized at a 48 kilohertz (kHz) sample rate, 48,000 digital samples are generated for every second of the signal.

To understand how the sample rate relates to sound quality, consider the fact that a telephone transmits voice-quality audio in a frequency range of about 320 Hz to 3.2 kHz. This frequency range can be represented accurately with a sample rate of 6.4 kHz. The range of human hearing, however, extends up to approximately 18–20 kHz, requiring a sample rate of at least 40 kHz.

The sample rate used for music-quality audio, such as the digital data stored on audio CDs is 44.1 kHz. A 44.1 kHz digital signal can theoretically represent audio frequencies from 0 kHz to 22.05 kHz, which adequately represents sounds within the range of normal human hearing. The most common sample rates used for DATs are 44.1 kHz and 48 kHz.

Higher sample rates result in higher-quality digital signals; however, the higher the sample rate, the greater the signal storage requirement.

Audio Compression Scheme

All audio data on Silicon Graphics systems is considered to have a compression scheme. The scheme may be an industry standard such as MPEG-1 audio, or it may be no compression at all. For more information, see "The Digital Media Color Space Library" in Chapter 6.

Audio Sample Format

Uncompressed audio data is encoded in a digital data format called linear *pulse code modulation* (PCM) (see the audio references for a definition of this term) to represent digital audio samples.

The formats supported by the audio system are:

- 8-bit and 16-bit signed integer
- 24-bit signed, right-justified within a 32-bit integer
- 32-bit and 64-bit floating point

Note: The audio hardware supports 16-bit I/O for analog data and 24-bit I/O for AES/EBU digital data.

For floating point data, the application program specifies the desired range of values for the samples; for example, from –1.0 to 1.0. A method for relating data from one range of values to data with a different range of values is described next.

PCM Mapping

PCM mapping describes the relationship between data with differing sample ranges. If the input and output mappings are different, a conversion consisting of clipping and transformation of values is necessary.

PCM mapping defines a reference value, denoted by DM_AUDIO_PCM_MAP_INTERCEPT, that is the midway point between a signal

swing. It is convenient to assign a value of zero to this point. Adding a slope value, denoted by DM_PCM_MAP_SLOPE, to the intercept obtains the full-scale deflection.

The values DM_AUDIO_PCM_MAP_MINCLIP and DM_AUDIO_PCM_MAXCLIP define the minimum and maximum legal PCM values. Input and output values are clipped to these values. If *maxclip* <= *minclip*, then no clipping is done because all PCM values are legal, even if they are outside the true full-scale range.

Audio Sample Width

The native data format used by the audio hardware is 24-bit two's complement integers. The audio hardware sign-extends each 24-bit quantity into a 32-bit word before delivering the samples to the Audio Library.

Audio input samples delivered to the Audio Library from the Indigo, Indigo^{2}™, and Indy™ audio hardware have different levels of resolution, depending on the input source that is currently active; the AL provides samples to the application at the desired resolution. You can also write your own conversion routine if desired.

Microphone/line-level input samples come from analog-to-digital (A/D) converters, which have 16-bit resolution. These samples are treated as 24-bit samples with 0's in the low 8 bits.

AES/EBU digital input samples have either 20-bit or 24-bit resolution, depending on the device connected to the digital input; for the 20-bit case (the most common), samples are treated as 24-bit samples, with 0's in the least significant 4 bits. The AL passes these samples through to the application if 24-bit two's complement is specified. If two's complement with 8-bit or 16-bit resolution is specified, the AL right-shifts the samples so that they fit into a smaller word size. For floating point data, the AL converts from the 24-bit format to floating point, using a scale factor specified by the application to map the peak integer values to peak float values.

For audio output, the AL delivers samples to the audio hardware as 24-bit quantities sign-extended to fill 32-bit words. The actual resolution of the samples from a given output port depends on the application program connected to the port. For example, an application may open a 16-bit output port, in which case the 24-bit samples arriving at the audio processor contains 0's in their least significant 8 bits.

The Audio Library is responsible for converting between the output sample format specified by an application and the 24-bit native format of the audio hardware. For 8-bit or 16-bit integer samples, this conversion is accomplished by left-shifting each sample written to the output port by 16 bits and 8 bits, respectively. For 32-bit or 64-bit floating point samples, this conversion is accomplished by rescaling each sample from the range of floating point values specified by the application to the full 24-bit range and then rounding the sample to the nearest integer value.

Table 2-5 lists the audio parameters and the valid values for each (not all values are supported by all libraries).

Table 2-5 Audio Parameters

Parameter	Type	Values
DM_AUDIO_CHANNELS	Integer	1, 2, or 4
DM_AUDIO_COMPRESSION	String	DM_AUDIO_UNCOMPRESSED (default) DM_AUDIO_G711_U_LAW DM_AUDIO_G711_A_LAW DM_AUDIO_MPEG DM_AUDIO_MPEG1 DM_AUDIO_MULTIRATE DM_AUDIO_G722 DM_AUDIO_G726 DM_AUDIO_G728 DM_AUDIO_DVI DM_AUDIO_GSM
DM_AUDIO_FORMAT	DMaudioformat	DM_AUDIO_TWOS_COMPLEMENT (default) DM_AUDIO_UNSIGNED DM_AUDIO_FLOAT DM_AUDIO_DOUBLE
DM_AUDIO_RATE	Double	Native rates are 8000, 11025, 16000, 22050, 32000, 44100, and 48000 Hz
DM_AUDIO_WIDTH	Integer	8, 16, or 24

Digital Media Synchronization Essentials

Most digital media applications use more than one medium at a time, for example, audio and video. This section explains how the data can be related for the various digital media functions that perform capture and presentation of concurrent media streams.

Timecodes

Timecodes are important for synchronizing and editing audio and video data.

There are different types of encoding methods and standards. In general, a timecode refers to a number represented as *hours:minutes:seconds:frames*. This numerical representation is used in a variety of ways (in both protocols and user interfaces), but in all cases the numbering scheme is the same. The SMPTE 12M standard provides definitions and specifications for a variety of timecodes and timecode signal formats.

The numerical ranges for each field in a timecode are as follows:

hours	00 to 23
minutes	00 to 59
seconds	00 to 59
frames	depends on the signal type

Some signals use a drop-frame timecode, where some “hours:minutes:seconds:frame” combinations are not used; they are simply skipped in a normal progression of timecodes.

A timecode can refer to a

- timer
- timestamp
- signal on wire
- signal on tape

One example application where timecode is used merely as a way to express a time is *mediaplayer*, which displays the offset from the beginning of the movie either in seconds or as a timecode.

Another common computer application of timecode is as a timestamp on particular frames in a movie file. The Silicon Graphics movie file format and the QuickTime format offer the ability to associate each image in the file with a timecode. Sometimes these timecodes are synthesized by the computer, and sometimes they are captured with the source material. These timecodes are often used as markers so that edited or processed material can be later correlated with material edited or processed on another system. A/V professionals use an edit decision list (EDL) to indicate the timecodes frames to be recorded.

Longitudinal Time Code

Longitudinal time code (LTC), sometimes ambiguously referred to as SMPTE time code, is a self-clocking signal defined separately for 525- and 625-line video signals, where the corresponding video signal itself is carried on another wire. The signal occupies its own channel and resembles an audio signal in voltage and bandwidth.

LTC is the most common way of slaving one system's transport to that of another system (by ensuring that both systems are on the same frame, not by genlocking signals on both machines). In some audio and MIDI applications, LTC is useful even though there is no video signal.

In an LTC signal, there is one codeword for each video frame. Each LTC codeword contains a timecode and other useful information.

See dmLTC(3dm) for routines for decoding LTC.

Vertical Interval Time Code

Vertical interval time code (VITC) is a standardized part of a 525- or 625-line video signal. The code itself occupies some lines in the vertical blanking interval of each field of the video signal (not normally visible on monitors). It's a good idea to provide data redundancy by recording the VITC on two nonconsecutive lines in case of video dropout.

Each VITC codeword contains a timecode and a group of flag bits that include

- Dropframe
- Colorframe
- Parity
- Field mark

The field mark bit is an F1/F2 field indicator; it is asserted for a specific field.

VITC also provides 32 user bits, where users can store information such as reel and shot number. This information can be used to help index footage after it is shot, and under the right circumstances (not always trivial), the original VITC recorded along with footage can even tag along with that footage as it is edited, allowing you to produce an edit list or track assets, given a final prototype edit.

See dmVITC(3dm) for routines for decoding VITC.

MIDI Time Code

MIDI time code is part of the standard MIDI protocol, which is carried over a serial protocol that is also called MIDI. Production studios often need to synchronize the transports of computers with the transports of multitrack audio tape recorders and dedicated MIDI sequencers. Sometimes LTC is used for this, and sometimes the MIDI time code is the clock signal of choice.

Time Code in AES Digital Audio Streams

The AES standard allows embedded timecodes in a digital audio signal.

Unadjusted System Time and Media Stream Count

The Digital Media Libraries provide their own temporal reference, called unadjusted system time (UST). The UST is an unsigned 64-bit number that measures the number of nanoseconds since the system was booted. UST values are guaranteed to be monotonically increasing and are readily available for all the Digital Media Libraries.

Typically, the UST is used as a timestamp, that is, it is paired with a specific item or location in a digital media stream. Because each type of media, and similarly each of the libraries, possesses unique attributes, the UST information is presented in a different

manner in each library. Table 2-6 describes how UST information is provided by each of the libraries.

Table 2-6 Methods for Obtaining Unadjusted System Time

Library	UST Method
Digital Media Library	**dmGetUST()** and **dmGetUSTCurrentTimePair()**
Audio Library	**ALgetframenumber()** and **ALgetframetime()**
MIDI Library	**mdTell()** and **mdSetTimestampMode()**
Video Library	*ustime* field in the DMediaInfo structure

Synchronization and UST/MSC

The media stream count (MSC), with the UST, is used to synchronize buffered media data streams. UST/MSC pairs are used with libraries, such as the Audio Library and the Video Library, that provide timing information about the sampled data. The MSC is a monotonically increasing, unsigned 64-bit number that is applied to each sample in a media stream. This means the MSC of the most recent data sample has the largest value. By using the UST/MSC facility, an application can schedule accurately the use of data samples, and also can detect data underflow and overflow. To see how these things are done, here is some background.

A media stream can be seen as travelling a *path*. An input path comprises electrical signals on an input *jack* (an electrical connection) being converted by a *device* to digital data that is placed in an input buffer for use by an application. An output path goes from the application to an output jack via an output buffer. As implied by this description, the data placed in a buffer by the device (input path) or application (output path) has the highest MSC, and the data taken out by the application or device, respectively, has the lowest.

There are two kinds of MSCs, device and frontier. The *device* MSC is the basis for the other. An input device assigns a device MSC to a sample about to be placed in the input buffer. An output device assigns one to a sample about to be removed from the output buffer. The MSC of the sample at the application's end of a buffer is the *frontier* MSC. It is calculated based on the device MSC. In an input path, the frontier MSC is equal to the device MSC minus the number of samples waiting in the input buffer. In an output path, the frontier MSC equals the device MSC plus the number of waiting output buffer samples.

What does using MSCs enable your application to do? Assuming the data stream going to the buffer is not underflowing or overflowing, your application can precisely control the sample flow by using MSCs to determine corresponding USTs. Your application can synchronize data streams, such an audio stream and a video stream, by matching the USTs of their samples. Also, it can compensate for IRIX™ scheduling interruptions by using the USTs of the samples for controlling the contents of the buffer.

As shown in this Video Library code sample, you can determine the time (UST) a media stream sample came in from or went out to a jack by using the functions **vlGetFrontierMSC()** and **vlGetUSTMSCPair()**.

```
double      ust_per_msc;
USTMSCpair  pair;
stamp_t     frontier_msc, desired_ust;
int         err;

ust_per_msc = vlGetUSTPerMSC(server, path);
err = vlGetUSTMSCPair(server, path, video_node, &pair);
frontier_msc = vlGetFrontierMSC(server, path, memNode);
desired_ust = pair.ust + ((frontier_msc - pair.msc) * ust_per_msc);
```

This sample works for both input and output paths. In either case, the sample indicated by *desired_ust* is the one with the frontier MSC. Thus for an input path, *desired_ust* is the UST of the next sample to be taken from the buffer by your application. For an output path, it is the UST of the next sample your application will place in the buffer. The USTs of other samples in the buffer can be found by adjusting the calculation on the last line.

These techniques assume that there is no data underflow or overflow to the buffer. If there is an underflow or overflow condition, calculations like these become unreliable. This is because the frontier MSC is based on the current device MSC. It is not a constant value attached to a specific data sample. For example, consider an overflow condition in an input path. The device MSC, and thus the frontier MSC, is increased by one every time the device is ready to place a sample in the input buffer. Because the buffer is full, the sample is discarded, but the MSCs retain their new values. Therefore, the UST/MSC pair associated with a given sample has changed and calculations like the one in the earlier code sample are no longer reliable.

This situation also demonstrates one of the advantages of the UST/MSC pairing. The design enables your application to determine buffer overflow or underflow immediately, based on the value of the frontier MSC. In the above example, your application can check for data overflow immediately after putting data samples into the buffer by checking if the difference between the current frontier MSC and the previous frontier MSC is greater

than the number of samples just queued. If it is greater, there is an overflow condition. The size of the discrepancy is the actual magnitude of the overflow because putting the samples into the buffer relieved the overflow. Your application can make analogous determinations for input underflow, and output overflow and underflow. Notice that the overflow condition can be found without waiting for the samples with the discontinuous data to get to the front of the buffer. This allows your application to take corrective action immediately.

Table 2-7 Methods for Using UST/MSC

Function	Description
vlGetFrontierMSC() **ALgetframenumber()**	Get the frontier MSC associated with a particular node. See also vlGetFrontierMSC(3dm) and ALgetframenumber(3dm).
vlGetUSTMSCPair() **ALgetframetime()**	Get the time at which a field or frame came in or will go out. See also vlGetUSTMSCPair(3dm) and ALgetframetime(3dm).
vlGetUSTPerMSC()	Get the time interval between fields or frames in a path. See also vlGetUSTPerMSC(3dm).

Counting Video Fields With MSCs

The VL presents field numbers to a VL application in two contexts:

- For video-to-memory or memory-to-video paths whose VL_CAP_TYPE is set to VL_CAPTURE_NONINTERLEAVED (fields separate, each in its own buffer), the functions **vlGetFrontierMSC()** and **vlGetUSTMSCPair()** return MSCs that count fields (in other VL_CAP_TYPEs, the returned MSCs do not count fields).
- For any video-to-memory path, the user can use **vlGetDMediaInfo()** to return the **DMediaInfo** structure contained in an entry in a **VLBuffer**. This structure contains a member called *sequence,* which always counts fields (regardless of VL_CAP_TYPE).

In both of these cases, there should be the following correlation:

- These values should be 0%2 if they represent an F1 field
- These values should be 1%2 if they represent an F2 field

This is a relatively new convention and is not yet implemented on all devices.

Digital Media File Format Essentials

Currently, the Digital Media Libraries support the following file formats:

Image Containers

- RGB
- FIT
- GIF
- JFIF
- PNG
- PPM
- TIFF
- Photo CD

Audio Containers

- Raw audio data
- AIFF/AIFC
- Microsoft Waveform Audio Format WAVE (RIFF)
- NeXT® .snd
- Sun® .au
- Berkeley IRCAM/CARL (BICSF)
- Digidesign Sound Designer II™
- Raw MPEG1 audio bitstream
- Audio Visual Research AVR™
- Amiga® IFF 8SVX™
- Creative Labs VOC
- Sample Vision
- E-mu Systems® SoundFont2®

In addition, the Digital Media Libraries recognize but do not support

- Sound Designer I
- NIST Sphere

Movie Containers

A movie is a collection of digital media data contained in tracks, temporally organized in a storage medium, which is captured from and played to audio and video devices. Saying that movies are composed of time-based data means that each piece of data is associated (and usually timestamped) with a particular instant in time, and has a certain duration in time. Movies can contain multiple tracks of different media.

Movies are generally stored in a file format that contains both a descriptive header and the movie data. When a movie is opened, only the header information exists in memory. A movie also has properties or attributes independent of the file format and may not necessarily be stored in a file.

Parameters in *dmedia/dm_image.h* and *dmedia/dm_audio.h* provide a common language for specifying movie data attributes. The Movie Library also provides its own parameters in *libmovie/movifile.h*.

The Movie Library currently supports these file formats:

- QuickTime
- MPEG-1 systems and video bitstreams
- Silicon Graphics movie format

Chapter 3

Digital Media Parameters

This chapter explains how to use digital media data structures that facilitate data specification and setting, getting, and passing parameters.

Digital Media Data Type Definitions

The DM Library provides type definitions for digital media that are useful when programming with the family of Digital Media Libraries. Data types and constant names have an uppercase DM prefix; routines have a lowercase dm prefix.

The *dmedia/dmedia.h* header file provides these type definitions:

DMboolean	integer for conditionals; DM_FALSE is 0 and DM_TRUE is 1
DMfraction	integer numerator divided by integer denominator
DMstatus	enumerated type consisting of DM_SUCCESS and DM_FAILURE

It is good programming practice to check the return values of functions. DMstatus provides a way to check return values. When a function succeeds, DM_SUCCESS is returned. When a function fails, DM_FAILURE is returned and a system error code is set that can be interpreted using the functions described in the next section.

Digital Media Error Handling

Errors encountered while using the Digital Media Libraries can be diagnosed with the help of two routines. The function **dmGetError()** retrieves the number, summary, and detailed description of an error generated by the execution of the current process. The current process in this case is the same as determined by **getpid()**. The companion

function, **dmGetErrorForPID()**, gets the same type of error information for a process your application specifies.

```
const char *dmGetError ( int *errornum,
                         char error_detail[DM_MAX_ERROR_DETAIL] )
const char *dmGetErrorForPID ( pid_t pid, int *errornum,
                               char error_detail[DM_MAX_ERROR_DETAIL] )
```

The functions **dmGetError()** and **dmGetErrorForPID()** enable your application to handle in a consistent manner errors generated while using the digital media libraries. Both functions return a pointer to a null-terminated character string that summarizes the error. The setting of the errors by the libraries and the retrieval of them by your application is guaranteed to be thread-safe. Only the most recent error for a given thread is returned. If there are no errors, the functions return NULL.

The parameter *pid* in **dmGetErrorForPID()** is the id of the process in which to check for an error. The last parameter in both functions, *error_detail*, is the address of a null-terminated character array of size DM_MAX_ERROR_DETAIL. If one exists, a detailed description of the error is loaded into the array. If you set *error_detail* to NULL, no description is loaded. The remaining parameter, *errornum*, is a pointer to an integer into which the number of the current error is loaded. If your application sets *errornum* to NULL, no number is loaded. The error numbers returned in *errornum* fall into ranges according to the digital media libraries that generated them. The currently defined error ranges and their libraries are as follows:

0-999	UNIX® System (The error numbers are identical to those returned by oserror(3C).)
1000-1999	Color Space Library in *libdmedia*
2000-2999	Movie Library in *libmoviefile* or *libmovieplay*
3000-3999	Audio File Library in *libaudiofile*
4000-4999	DMbuffer in *libdmedia*
5000-5999	Audio Converter in *libdmedia*
6000-6999	Image Converter in *libdmedia*
10000-10999	Global Digital Media Library in *libdmedia*
11000-11999	FX Plug-in Utility Library in *libfxplugutils*
12000-12999	FX Plug-in Manager Library in *libfxplugmgr*

Digital Media Parameter Types

The DM Library provides definitions for the digital media parameter data types. Table 3-1 lists the digital media parameter type definitions that are defined in *dmedia/dm_params.h*.

Table 3-1 Digital Media Parameter Data Types

Parameter Type	Meaning
DM_TYPE_BINARY	Binary data
DM_TYPE_ENUM	Enumerated type
DM_TYPE_ENUM_ARRAY	Array of enumerated types
DM_TYPE_FLOAT	Floating point value (double)
DM_TYPE_FLOAT_ARRAY	Array of floats
DM_TYPE_FLOAT_RANGE	Range of floats
DM_TYPE_FRACTION	Ratio
DM_TYPE_FRACTION_ARRAY	Array of fractions
DM_TYPE_FRACTION_RANGE	Range of fractions
DM_TYPE_INT	Integer value
DM_TYPE_INT_ARRAY	Array of integers
DM_TYPE_INT_RANGE	Range of integers
DM_TYPE_LONG_LONG	Long long (64-bits)
DM_TYPE_PARAMS	DMparams list
DM_TYPE_STRING	String
DM_TYPE_STRING_ARRAY	Array of strings
DM_TYPE_TOC_ENTRY	Table-of-contents entry for ring buffers

Digital Media Parameter Lists

Parameter-value lists, which are contained in a DMparams structure supply configuration information for digital media objects such as audio ports, movie tracks, and video devices. A DMparams list is a list of pairs, where each pair contains the name of a parameter and the corresponding value for that parameter.

You can use a DMparams list to

- configure a digital media structure upon initialization by passing a complete list containing all the parameters and values needed to configure that object to a creation routine
- change the settings of an existing digital media structure by providing a list of parameters and corresponding values to replace

Most Digital Media Libraries provide convenience routines for setting, adjusting, and getting relevant parameter values.

Every DMparams list that describes a format includes the parameter DM_MEDIUM to indicate what kind of data it describes. DM_MEDIUM is an enumerated type consisting of:

DM_IMAGE	which represents image data
DM_AUDIO	which represents audio data
DM_TIMECODE	which represents a timecode
DM_TEXT	which represents text

Another common parameter, DM_CODEC, is an enumerated type that describes whether a codec is synchronous (DM_SYNC_CODEC) or asynchronous (DM_ASYNC_CODEC). The compressor and decompressor of a *synchronous codec* are linked such that there must be both uncompressed input available to the compressor and compressed input available to the decompressor before either can generate output. An *asynchronous codec* has no such linkage.

This section explains how to use the DM Library routines for

- creating and destroying DMparams lists
- creating default audio and image configurations
- setting and getting values in DMparams lists

- manipulating DMparams lists

The routines described in this section follow the general rule that ownership of data is not passed during procedure calls, except in the routines that create and destroy DMparams lists. Functions that take strings copy the strings if they want to keep them. Functions that return strings or other structures retain ownership and the caller must not free them.

In the initialization section of your application, you create and use DMparams lists to configure data structures for your application as described in the following steps:

1. Create an empty DMparams list by calling **dmParamsCreate()**.
2. Set the parameter values by one of the methods listed below:
    - Use a function that sets up a standard configuration for a particular type of data: **dmSetImageDefaults()** for images, **dmSetAudioDefaults()** for audio.
    - Use a generic function such as **dmParamsSetInt()** to set the values of individual parameters within an empty DMparams list or one that has already been initialized with the standard audio or image configuration. See “Setting and Getting Individual Parameter Values” on page 65 for a description of this method.
    - Use a library function such as **mvSetMovieDefaults()** to set a group of parameters specific to that library.
3. Free the DMparams list and its contents by calling **dmParamsDestroy()**.

These steps are described in detail in the sections that follow.

Creating and Destroying DMparams Lists

Some libraries require you to allocate memory for DMparams lists, but with the DM library, you need not allocate memory for DMparams lists, because memory management is provided for you by the **dmParamsCreate()** and **dmParamsDestroy()** routines. These routines work together as a self-contained block within which you create the DMparams list, set the parameter value(s) and use them, and then destroy the structure, freeing its associated memory.

Only the **dmParamsCreate()** function can create a DMparams list, and only the **dmParamsDestroy()** function can free one. This means that DMparams lists are managed correctly when every call to create one is balanced by a call to destroy one. The creation

function can fail because of lack of memory, so it returns an error code. The destructor can never fail.

To create an empty DMparams list, call **dmParamsCreate()**. Its function prototype is:

```
DMstatus dmParamsCreate ( DMparams** returnNewList )
```

where:

returnNewList is a pointer to a handle that is returned by the DM Library

If there is sufficient memory to allocate the structure, a pointer to the newly created structure is put into **returnNewList* and DM_SUCCESS is returned; otherwise, DM_FAILURE is returned.

When you have finished using the DMparams list, you must destroy it to free the associated memory. To free both the DMparams list structure and its contents, call **dmParamsDestroy()**. Its function prototype is:

```
void dmParamsDestroy ( DMparams* params )
```

where:

params is a pointer to the DMparams list you want to destroy

Example 3-1 is a code fragment that creates a DMparams list called *params*, then calls a Movie Library routine, **mvSetMovieDefaults()**, to initialize the default movie parameters, and finally destroys the list, freeing both the structure and its contents.

Example 3-1 Creating and Destroying a DMparams List

```
DMparams* params;
if ( dmParamsCreate( &params ) != DM_SUCCESS ) {
    printf( "s\n", dmGetError(NULL, NULL) );
    exit( 1 );
}
if ( mvSetMovieDefaults(params, MV_FORMAT_SGI_3) != DM_SUCCESS ) {
    printf( “s\n”, mvGetErrorStr(mvGetErrno()));
    exit( 1 );
}
dmParamsDestroy ( params );
```

Setting and Getting Individual Parameter Values

After creating an empty DMparams list or a default audio or image configuration, you can use the routines described in this section to set and get values for individual elements of a DMparams list.

There is a routine for setting and getting the parameter values for each parameter data type defined in the DM Library, as listed in Table 3-1.

All of these functions store and retrieve entries in a DMparams list. They assume that the named parameter is present and is of the specified type; the debugging version of the library asserts that this is the case. All functions that can possibly fail return an error code indicating success or failure. Insufficient memory is the only reason these routines can fail. Type mismatch causes a failed assertion in the debug library and undefined results in the non-debug library.

Table 3-2 lists the DM Library routines for setting parameter values. All the routines except **dmParamsSetBinary()** require three arguments:

params a pointer to a DMparams list

paramName the name of the parameter whose value you want to set

value a value of the appropriate type for the given parameter

Table 3-2 DM Library Routines for Setting Parameter Values

Routine	Purpose
dmParamsSetBinary()	Sets the contents of a data buffer. See dmParamsSetInt(3dm).
dmParamsSetEnum()	Sets the value of an enum parameter whose type is int.
dmParamsSetEnumArray()	Sets the value of a parameter whose type is DMenumarray.
dmParamsSetFloat()	Sets the value of a parameter whose type is double.
dmParamsSetFloatArray()	Sets the value of a parameter whose type is DMfloatarray.
dmParamsSetFloatRange()	Sets the value of a parameter whose type is DMfloatrange.
dmParamsSetFract()	Sets the value of a parameter whose type is DMfraction.
dmParamsSetFractArray()	Sets the value of a parameter whose type is DMfractionarray.
dmParamsSetFractRange()	Sets the value of a parameter whose type is DMfractionrange.

Table 3-2 (continued) DM Library Routines for Setting Parameter Values

Routine	Purpose
dmParamsSetInt()	Sets the value of a parameter whose type is int.
dmParamsSetIntArray()	Sets the value of a parameter whose type is DMintarray.
dmParamsSetIntRange()	Sets the value of a parameter whose type is DMintrange.
dmParamsSetLongLong()	Sets the value of a parameter whose type is long long.
dmParamsSetParams()	Sets the value of a parameter whose type is DMparam.
dmParamsSetString()	Sets the value of a parameter whose type is a character string.
dmParamsSetStringArray()	Sets the value of a parameter whose type is DMstringarray.

These routines return either DM_SUCCESS or DM_FAILURE.

Table 3-3 lists the DM Library routines for getting parameter values. All the routines except **dmParamsGetBinary()** require two arguments:

params a pointer to a DMparams list

paramName the name of the parameter whose value you want to get

Routines that get values return either a pointer to a value or the value itself. For strings, DMparams lists, and table-of-contents entries, the pointer that is returned points into the internal data structure of the DMparams list. This pointer should never be freed and is only guaranteed to remain valid until the next time the list is changed. In general, if you need to keep a string value around after getting it from a DMparams list, it should be copied.

Table 3-3 DM Library Routines for Getting Parameter Values

Routine	Purpose
dmParamsGetBinary()	Returns a pointer to binary data. See dmParamsSetInt(3dm).
dmParamsGetEnum()	Returns an integer value for the given enum parameter.
dmParamsGetEnumArray()	Returns a pointer to a value of type DMenumarray.
dmParamsGetFloat()	Returns a value of type double for the given parameter.
dmParamsGetFloatArray()	Returns a pointer to a value of type DMfloatarray.

Table 3-3 (continued) DM Library Routines for Getting Parameter Values

Routine	Purpose
dmParamsGetFloatRange()	Returns a pointer to a value of type DMfloatrange.
dmParamsGetFract()	Returns a value of type DMfraction for the given parameter.
dmParamsGetFractArray()	Returns a pointer to a value of type DMfractionarray.
dmParamsGetFractRange()	Returns a pointer to a value of type DMfractionrange.
dmParamsGetInt()	Returns an integer value for the given parameter.
dmParamsGetIntArray()	Returns a pointer to a value of type DMintarray for the parameter.
dmParamsGetIntRange()	Returns a pointer to a value of type DMintrange for the parameter.
dmParamsGetLongLong()	Returns a 64-bit long for the given parameter.
dmParamsGetParams()	Returns a pointer to a value of type DMparams for the parameter.
dmParamsGetString()	Returns a pointer to a value of type const char for the parameter.
dmParamsGetStringArray()	Returns a pointer to a value of type DMstringarray.

Setting Parameter Defaults

Setting Image Defaults

To initialize a DMparams list with the default image configuration, call **dmSetImageDefaults()**, passing in the width and height of the image frame, and the image packing format. Its function prototype is:

```
DMstatus dmSetImageDefaults ( DMparams* params, int width,
                              int height, DMpacking packing )
```

where:

params is a pointer to a DMparams list that was returned by **dmParamsCreate()**

width is the width of the image in pixels

height is the height of the image in pixels

packing is the image packing format

Table 3-4 lists the parameters and values set by **dmSetImageDefaults()**.

Table 3-4 Image Defaults

Parameter	Default
DM_MEDIUM	DM_IMAGE
DM_IMAGE_WIDTH	*width*
DM_IMAGE_HEIGHT	*height*
DM_IMAGE_RATE	15.0 frames per second (Hz)
DM_IMAGE_INTERLACING	DM_IMAGE_NONINTERLACED
DM_IMAGE_PACKING	*packing*
DM_IMAGE_ORIENTATION	DM_BOTTOM_TO_TOP
DM_IMAGE_COMPRESSION	DM_IMAGE_UNCOMPRESSED

Determining the Buffer Size Needed to Store an Image Frame

To determine the image frame size for a given DMparams list, call **dmImageFrameSize()**. **dmImageFrameSize()** returns the number of bytes needed to store one uncompressed image frame in the given format. Its function prototype is:

```
size_t dmImageFrameSize ( const DMparams* params )
```

Example 3-2 is a code fragment that creates a DMparams list, fills in the image defaults, and then frees the structure and its contents.

Example 3-2 Setting Image Defaults

```
DMparams* imageParams;

if ( dmParamsCreate( &imageParams ) != DM_SUCCESS ) {
    printf( "s\n", dmGetError(NULL, NULL) );
    exit( 1 );
}
if ( dmSetImageDefaults( imageParams,
       320,  /* width */
       240,  /* height */
       DM_PACKING_RGBX ) != DM_SUCCESS ) {
    printf( "s\n", dmGetError(NULL, NULL) );
    exit( 1 );
}
printf( "%d bytes per image frame.\n",
        dmImageFrameSize( imageParams ) );
dmParamsDestroy( imageParams );
```

Setting Audio Defaults

To initialize a DMparams list with the default audio configuration, call **dmSetAudioDefaults()**, passing in the desired sample width, sample rate, and number of channels. Its function prototype is:

```
DMstatus dmSetAudioDefaults ( DMparams* params, int width,
                              double rate, int channels )
```

where:

params is a pointer to a DMparams list that was returned from **dmParamsCreate()**

width is the number of bits per audio sample: 8, 16, or 24

rate is the audio sample rate; the native audio sample rates are 8000, 11025, 16000, 22050, 32000, 44100, and 48000 Hz

channels is the number of audio channels

dmSetAudioDefaults() returns DM_SUCCESS if there was enough memory available to set up the parameters; otherwise, it returns DM_FAILURE.

Table 3-5 lists the parameters and values set by **dmSetAudioDefaults()**.

Table 3-5 Audio Defaults

Parameter	Default
DM_MEDIUM	DM_AUDIO
DM_AUDIO_WIDTH	*width*
DM_AUDIO_FORMAT	DM_AUDIO_TWOS_COMPLEMENT
DM_AUDIO_RATE	*rate*
DM_AUDIO_CHANNELS	*channels*
DM_AUDIO_COMPRESSION	DM_AUDIO_UNCOMPRESSED

Determining the Buffer Size Needed to Store an Audio Frame

To determine the audio frame size for a given DMparams list, call **dmAudioFrameSize()**. **dmAudioFrameSize()** returns the number of bytes needed to store one audio frame (one sample from each channel). Its function prototype is:

```
size_t dmAudioFrameSize ( DMparams* params )
```

Example 3-3 is a code fragment that creates a DMparams list, fills in the audio defaults, and then frees the structure and its contents.

Example 3-3 Setting Audio Defaults

```
DMparams* audioParams;
if ( dmParamsCreate( &audioParams ) != DM_SUCCESS ) {
    printf( "s\n", dmGetError(NULL, NULL) );
    exit( 1 );
}
if ( dmSetAudioDefaults ( audioParams,
                          16,    /* width (in bits/sample) */
                          22050, /* sampling rate */
                          2      /* # channels (stereo) */
                          ) != DM_SUCCESS ) {
    printf( "s/n", dmGetError(NULL, NULL) );
    exit( 1 );
}
printf( "%d bytes per audio frame.\n",
         dmAudioFrameSize( audioParams ) );
dmParamsDestroy( audioParams );
```

Example 3-4 shows two equivalent ways of setting up a complete image format description; the first sets the parameter values individually, the second creates a default image configuration with the appropriate values.

Example 3-4 Setting Individual Parameter Values

```
DMparams* format;
dmParamsCreate( &format );
dmParamsSetInt ( format, DM_IMAGE_WIDTH, 320 );
dmParamsSetInt ( format, DM_IMAGE_HEIGHT, 240 );
dmParamsSetFloat ( format, DM_IMAGE_RATE, 15.0 );
dmParamsSetString( format, DM_IMAGE_COMPRESSION, DM_IMAGE_UNCOMPRESSED );
dmParamsSetEnum( format, DM_IMAGE_INTERLACING, DM_IMAGE_NONINTERLEAVED );
dmParamsSetEnum ( format, DM_IMAGE_PACKING, DM_PACKING_RGBX );
dmParamsSetEnum ( format, DM_IMAGE_ORIENTATION, DM_BOTTOM_TO_TOP );dmParamsDestroy
   ( format );
```

The following is equivalent:

```
DMparams* format;
dmParamsCreate ( &format );
dmSetImageDefaults ( format, 320, 240, DM_PACKING_RGBX );
dmParamsDestroy ( format );
```

Manipulating DMparams Lists

This section explains how to manipulate DMparams lists. Some of the tasks you can two with DMparams lists include:

- testing two parameter values for equality
- copying either individual parameter-value pairs or entire DMparams lists
- determine how many parameter-value pairs are in a particular DMparams list
- getting information about parameter names, data types

Table 3-6 lists the routines that perform operations on DMparams lists and the entries within them.

Table 3-6 Routines for Manipulating DMparams Lists and Entries

Routine	Purpose
dmParamsAreEqual()	Determine if the values of two parameters are equal
dmParamsCopyAllElems()	Copy the entire contents of one list to another
dmParamsCopyElem()	Copy one parameter-value pair from one DMparams list to another
dmParamsGetElem()	Get the name of a given parameter
dmParamsGetElemType()	Get the data type of a given parameter
dmParamsGetNumElems()	Get the number of parameters in a list
dmParamsGetType()	Get the data type of the named parameter
dmParamsIsPresent()	Determine if a given parameter is in the list
dmParamsRemoveElem()	Remove a given parameter from the list
dmParamsScan()	Scan all the entries of a digital media parameter list

The sections that follow explain how to use each routine.

Determining DMparams Equivalence

The function **dmParamsAreEqual()** compares two DMparams structures and tests for equality. Its function prototype is:

```
DMboolean dmParamsAreEqual ( const DMparams *params1,
                             const DMparams *params2 )
```

If *params1* and *params2* have the same number of parameter-value pairs, and if the parameters of the same name have the same type and value in both lists, then the function returns DM_TRUE.

Determining the Number of Elements in a DMparams List

To perform any task that requires your application to loop through the contents of a DMparams list (for example, to print out a list of parameters and their values) you need to know how many parameters are in the list in order to set up a loop to step through the entries one-by-one.

To get the total number of elements present in a DMparams list, call **dmParamsGetNumElems()**. Its function prototype is:

```
int dmParamsGetNumElems ( const DMparams* params )
```

The number of elements and their position in a list is guaranteed to remain stable unless the list is changed by using one of the “set” functions, by copying an element into it, or by removing an element from it.

There is also a convenience function, **dmParamsScan()**, for looping through the contents of a DMparams list and performing the same operation on each element of the list. See for more information on

Copying the Contents of One DMparams List into Another

To copy the entire contents of the *fromParams* list into the *toParams* list, call **dmParamsCopyAllElems()**. Its function prototype is:

```
DMstatus dmParamsCopyAllElems ( const DMparams* fromParams,
                                DMparams* toParams )
```

If there are any parameters of the same name in both lists, the corresponding value(s) in the destination list are overwritten. DM_SUCCESS is returned if there is enough memory to hold the copied data; otherwise, DM_FAILURE is returned. Type mismatch causes a failed assertion in the debug version of the library.

Copying an Individual Parameter Value from One List into Another

If a parameter appears in more than one DMparams list, it is sometimes more convenient to copy the individual parameter or group of parameters from one list to another, rather than individually setting the parameter value(s) for each list.

To copy the parameter-value pair for the parameter named *paramName* from the *fromParams* list into the *toParams* list, call **dmParamsCopyElem()**. Its function prototype is:

```
DMstatus dmParamsCopyElem ( const DMparams* fromParams,
                            const char* paramName,
                            DMparams* toParams )
```

If there is a preexisting parameter with the same name in the destination list, that value is overwritten. DM_SUCCESS is returned if there is enough memory to hold the copied element; otherwise, DM_FAILURE is returned.

Determining the Name of a Given Parameter

To get the name of the entry occupying the position given by *index* in the *params* list, call **dmParamsGetElem()**. Its function prototype is:

```
const char* dmParamsGetElem ( const DMparams* params, const int index )
```

The *index* must be from 0 to one less than the number of elements in the list.

Determining the Data Type of a Given Parameter

To get the data type of the value occupying the position given by *index* in the *params* list, call **dmParamsGetElemType()**. Its function prototype is:

```
DMparamtype dmParamsGetElemType ( const DMparams* params,
                                  const int index )
```

Similarly, to get the data type of the parameter given by *name* in the *params* list, call **dmParamsGetType()**. Its function prototype is:

```
DMparamtype dmParamsGetType ( const DMparams* params,
                              const char* paramName )
```

See Table 3-1 for a list of valid return values.

Determining if a Given Parameter Exists

To determine whether the element named *paramName* exists in the *params* list, call **dmParamsIsPresent()**. Its function prototype is:

```
DMboolean dmParamsIsPresent  ( const DMparams* params,
                                const char* paramName )
```

DM_TRUE is returned if *paramName* is in *params*; otherwise, DM_FALSE is returned.

Scanning a DMparams List

Instead of creating your own loop to cycle through the contents of a DMparams list, you can use the convenience routine **dmParamsScan()**, which performs a specified operation on each element of the list. Its function prototype is:

```
DMstatus dmParamsScan ( const DMparams* params,
  DMstatus (*scanFunc) ( const DMparams* params,
                         const char* paramName,
                         void* scanArg,
                         DMboolean* stopScan ),
  void* scanArg )
```

The function **dmParamsScan()** passes the name of each entry in a DMparams list and *scanArg* as parameters to *scanFunc*. If *scanFunc* sets the value of *stopScan* to DM_TRUE, **dmParamsScan()** stops the DMparams list scan and returns the value returned by *scanFunc*. Otherwise, **dmParamsScan()** processes all elements in the list and returns DM_SUCCESS.

Removing an Element from a DMparams List

To remove the *paramName* entry from the *params* list, call **dmParamsRemoveElem()**. Its function prototype is:

```
const char* dmParamsRemoveElem ( DMparams* params,
                                 const char* paramName)
```

The element named *paramName* must be present.

Example 3-5 prints the contents of a DMparams list.

Example 3-5 Printing the Contents of a Digital Media DMparams List

```
void PrintParams( DMparams* params ) {
    int i;
    int numElems = dmParamsGetNumElems( params );

   for ( i = 0;  i < numElems;  i++ ) {
        const char* name = dmParamsGetElem( params, i );
        DMparamtype type = dmParamsGetElemType( params, i );
        printf( "    %20s: ", name );
        switch( type ) {
            case DM_TYPE_ENUM:
                printf( "%d", dmParamsGetEnum( params, name ) );
                break;
            case DM_TYPE_INT:
                printf( "%d", dmParamsGetInt( params, name ) );
                break;
            case DM_TYPE_STRING:
                printf( "%s", dmParamsGetString( params, name ) );
                break;
            case DM_TYPE_FLOAT:
                printf( "%f", dmParamsGetFloat( params, name ) );
                break;
            case DM_TYPE_FRACTION:
                DMfraction f = dmParamsGetFract( params, name );
                printf( "%d/%d", f.numerator, f.denominator );
                break;
            case DM_TYPE_PARAMS:
                printf( "... param list ... " );
                break;
            case DM_TYPE_TOC_ENTRY:
                printf( "... toc entry ..." );
                break;
```

```
            default:
                assert( DM_FALSE );
            }
        printf( "\n" );
    }
}
```

Compiling and Linking a Digital Media Library Application

Applications that call DM Library routines must include the *libdmedia* header files to obtain definitions for the library; however, these files are usually included in the header file of the library you are using.

This code fragment includes all the *libdmedia* header files:

```
#include <dmedia/dmedia.h>
#include <dmedia/dm_audio.h>
#include <dmedia/dm_image.h>
#include <dmedia/dm_params.h>
#include <dmedia/dm_buffer.h>
#include <dmedia/dm_imageconvert.h>
#include <dmedia/dm_audioconvert.h>
```

Link with the DM Library when compiling an application that makes DM Library calls by including **-ldmedia** on the link line. It's likely that you'll be linking with other libraries as well, and because the linking order is usually specific, follow the linking instructions for the library you are using.

Debugging a Digital Media Library Application

The debugging version of the DM Library checks for library usage violations by setting assertions that state the requirements for a parameter or value.

To debug your DM application, link with the debugging version of the DM Library, *libdmedia.so.1*, by setting your LD_LIBRARY_PATH environment variable to the directory containing the debug library before linking with **-ldmedia**, and then run your program. For example, use **`setenv LD_LIBRARY_PATH /usr/lib/debug`** to set the path.

Your application will abort with an error message if it fails an assertion. The message explains the situation that caused the error and, by implication or by explicit description, suggests a corrective action.

When you have finished debugging your application, you should relink with the nondebugging library, *libdmedia.a*, because the runtime checks imposed by the debugging library cause an undesirable size and performance overhead for a packaged application.

Chapter 4

Digital Media I/O

This chapter explains how to use the Digital Media Library routines that facilitate real-time input and output between live media devices. In particular, it describes using the Video Library (VL) and the Audio Library (AL) to create the interface between your application program, the workstation CPU, and external devices.

Video I/O Concepts

The VL enables live video flow into a program. This section explains basic video I/O concepts of the VL.

Video I/O programming with the VL involves

- *devices*, for processing video (each including sets of nodes)
- *nodes*, for defining endpoints or internal processing points of a video transport path
- *paths*, for routing video data by connecting nodes
- *ports*, for producing or consuming video data
- *controls*, for modifying the behavior of video nodes and transport paths
- *events*, for monitoring video I/O status
- *buffers*, for sending video data to and receiving video data from host memory; these can be either VLbuffers, as described in this chapter or DMbuffers, as described in Chapter 5, “Digital Media Buffers.”

Each of these topics is discussed in a separate section.

The manner in which video data transfer is accomplished differs slightly depending on the buffering method, but the essential concepts of using paths, nodes, controls, and events apply to both methods.

Devices

There are two types of video devices: external devices that are connected to a video jack on the workstation, and VL video devices, which are internal video boards and options for processing video data. The application should perform a query to determine which external video devices are connected and powered on, by calling **vlGetDeviceList()**

```
int vlGetDeviceList ( VLServer svr, VLDevList *devlist )
```

which fills the supplied VLDevList structure with a list of available devices, including the number of devices available and an array of VLDevice structures describing the available devices. A VLDevice structure contains the index of the device, the device name, the number of nodes available, and a list of VLNodeInfo structures describing the nodes available on that device.

To select the desired node, find the entry in the node list for the device name you want in the return argument of **vlGetDeviceList()**, then pass in the corresponding node number to **vlGetNode()**.

Nodes

A node is an endpoint or internal processing element of the video transport path, such as a video *source* such as a camera, a video *drain* (such as to the workstation screen), a *device* (video), or the *blender* in which video sources are combined for output to a drain.

Nodes have three attributes:

- *type*, which specifies the node's function in a path
- *class*, which identifies the type of system resource associated with the node
- *number*, which differentiates among multiple node instances and typically corresponds to the numbering of the video connectors on the video board

Node types are

VL_SRC the origination point (source) of a video stream

VL_DRN the destination point (drain) to which video is sent

VL_INTERNAL
a mid-stream filter such as a blender

VL_DEVICE — a special node for device-global controls shared by all paths

Note: For VL_DEVICE, set the node class to 0.

Putting a VL_DEVICE node on a path gives that path access to global device controls that can affect all paths on the device.

Node classes are

VL_VIDEO — a hardware video port that connects to a piece of video equipment such as a video tape deck or camera. All video devices have at least one port. The VL_SRC node type signifies an input port; VL_DRN signifies an output port.

VL_MEM — a memory buffer used to send or receive video data

VL_GFX — a direct connection between a video device and a graphics framebuffer

VL_SCREEN — a direct connection between a video device and a graphics display device, but different from VL_GFX because the video data does not interact directly with the graphics framebuffer and cannot be manipulated with graphics routines

VL_TEXTURE — an interface to graphics hardware for transferring video data directly to or from texture memory

VL_BLENDER — a filter that operates on data flowing from source to drain

VL_CSC — an interface to an optional real-time color space converter on systems that support it (and that have the option board installed)

VL_FB — an internal framebuffer node for freezing video on certain systems

Additional node classes may be available on certain video options; refer to the documentation that came with your video option for details.

To create a video node, call **vlGetNode().** Its function prototype is

```
VLNode vlGetNode ( VLServer vlSvr, int type, int class, int number )
```

Upon successful completion, **vlGetNode()** returns a VL Node (a handle to a node), which identifies the node for functions that perform an action on a node.

To use the default node for a device, specify its number as VL_ANY:

```
nodehandle = vlGetNode( svr, VL_SRC, VL_VIDEO, VL_ANY );
```

Paths

A path is a route between video nodes for directing the flow of video data.

Using a path involves

- creating the path
- getting the device ID
- adding nodes (if needed)
- specifying the data transfer characteristics of the path
- setting up the data path

These steps are explained in the sections that follow.

Creating a Video Data Transfer Path

Use **vlCreatePath()** to create the video data transfer path. Its function prototype is

```
VLPath vlCreatePath ( VLServer svr, VLDev dev, VLNode source, VLNode drain )
```

You can create a path using any available node by specifying the generic value VL_ANY for the device. This code fragment creates a path if the device is unknown:

```
if ((path = vlCreatePath(vlSvr, VL_ANY, src, drn)) < 0) {
    vlPerror(_progName);
    exit(1);
}
```

This code fragment creates a path that uses a device specified by parsing a *devlist*:

```
if ((path = vlCreatePath(vlSvr, devlist.devices[devicenum].dev, src,
    drn)) < 0) {
    vlPerror(_progName);
    exit(1);
}
```

Note: If the path contains one or more invalid nodes, **vlCreatePath()** returns VLBadNode.

Getting the Device ID

If you specify VL_ANY as the device when you create the path, use **vlGetDevice()** to discover the device ID selected. Its function prototype is

```
VLDev vlGetDevice ( VLServer vlSvr, VLPath path )
```

For example:

```
devicenum = vlGetDevice(vlSvr, path);
deviceName = devlist.devices[devicenum].name;
printf("Device is: %s/n", deviceName);
```

Adding Nodes to an Existing Video Path

You can add nodes to an existing path to provide additional processing or I/O capabilities. For this optional step, use **vlAddNode()**. Its function prototype is

```
int vlAddNode ( VLServer vlSvr, VLPath vlPath, VLNodeId node )
```

where

vlSvr	names the server to which the path is connected
vlPath	is the path as defined with **vlCreatePath()**
node	is the node ID

Specifying Video Data Transfer Path Characteristics

Path attributes specify usage rules for video controls and data transfers. Even though the names are the same, the intent and function of the usage attributes depend on whether they specify control or stream (data) usage.

Control usage attributes are

- VL_SHARE, meaning other paths can set controls on this node; this control is the desired setting for other paths, including *vcp*, to work

  Note: When using VL_SHARE, pay attention to events. If another user has changed a control, a VLControlChanged event occurs.

- VL_READ_ONLY, meaning controls cannot be set, only read; for example, this control can be used to monitor controls

- VL_LOCK, which prevents other paths from setting controls on this path; controls cannot be used by another path
- VL_DONE_USING, meaning the resources are no longer required; the application releases this set of paths for other applications to acquire

Stream (data) usage attributes are

- VL_SHARE, meaning transfers can be preempted by other users; paths contend for ownership

  Note: When using VL_SHARE, pay attention to events. If another user has taken over the node, a VLStreamPreempted event occurs.
- VL_READ_ONLY, meaning the path cannot perform transfers, but other resources are not locked; set this value to use the path for controls
- VL_LOCK, which prevents other paths that share data transfer resources with this path from transferring (except that two paths can share a video source when locked); existing paths that share resources with this path are preempted
- VL_DONE_USING, meaning the resources are no longer required; the application releases this set of paths for other applications to acquire

Setting Up a Video Transfer Data Path

Once the path has been created and usage attributes assigned, its settings do not go into effect until the path is set up with **vlSetupPaths()**. Its function prototype is

```
int vlSetupPaths ( VLServer vlSvr, VLPathList paths,
                   u_int count, VLUsageType ctrlusage,
                   VLUsageType streamusage )
```

where

vlSvr	names the server to which the path is connected
paths	specifies a list of paths you are setting up
count	specifies the number of paths in the path list
ctrlusage	specifies usage for path controls
streamusage	specifies usage for the data

This example fragment sets up a path with shared controls and a locked stream:

```
if (vlSetupPaths(vlSvr, (VLPathList)&path, 1, VL_SHARE,
    VL_LOCK) < 0)
{
    vlPerror(_progName);
    exit(1);
}
```

Note: The Video Library infers the connections on a path if **vlBeginTransfer()** is called and no drain nodes have been connected using **vlSetConnection()** (implicit routing). To specify a path that does not use the default connections, use **vlSetConnection()** (explicit routing).

- For each internal node on the path, all unconnected input ports are connected to the first source node added to the path. Pixel ports are connected to pixel ports and alpha ports are connected to alpha ports.
- For each drain node on the path, all unconnected input ports are connected to the first internal node placed on the path, if there is an internal node, or to the first source node placed on the path. Pixel ports are connected to pixel ports and alpha ports are connected to alpha ports.

Note: Do not combine implicit and explicit routing.

Controls

Controls determine the behavior of a node or path and provide information about them. Controls are specific to the path and node, and can also be device-dependent, depending on the control type. In general, controls on a video node are independent of controls on a memory or screen node. Even though controls on different types of nodes have the same names, they have different meanings, different units, and different behavior, depending on what node class they control.

The type definition of a VL control is:

```
typedef int VLControlType;
```

To get the value of a control, call **vlGetControl()**:

```
int vlGetControl ( VLserver svr, VLPath path, VLnode node,
                   VLControlType type, VLControlValue *value )
```

The control is located according to the *svr, path, node,* and *type* and its value is returned in a pointer to a VLControlValue structure:

```
typedef union {
 VLFraction       fractVal;
 VLBoolean        boolVal;
 int              intVal;
 VLXY             xyVal;
 char             stringVal[96];
 float            matrixVal[3][3];
 uint             pad[24];
 VLExtendedValue  extVal;
} VLControlValue;

typedef struct {
 int  x, y;
} VLXY;

typedef struct {
 int  numerator;
 int  denominator;
} VLFraction;
```

To obtain information about the valid values for a given control, call **vlGetControlInfo()**:

```
VLControlInfo *vlGetControlInfo ( VLserver svr, VLPath path, VLnode node,
                                  VLControlType type )
```

The control is located according to the *svr, path, node,* and *type,* and its value is returned in a pointer to a VLControlInfo structure:

```
typedef struct __vlControlInfo {
 char                name[VL_NAME_SIZE]; /* name of control */
 VLControlType       type;            /* e.g. WINDOW, HUE */
 VLControlClass      ctlClass;       /* SLIDER, DETENT, KNOB, BUTTON */
 VLControlGroup      group;          /* BLEND, VISUAL QUALITY, SYNC */
 VLNode              node;           /* associated node */
 VLControlValueType  valueType;      /* what kind of data */
 int                 valueCount;     /* how many data items */
 int                 numFractRanges; /* number of ranges */
 VLFractionRange     *ranges;        /* range of values of control */
 int                 numItems;       /* number of enumerated items */
 VLControlItem       *itemList;      /* the actual enumerations */
} VLControlInfo;
```

These controls are highly interdependent, so the order in which they are set is important. In most cases, the value being set takes precedence over other values that were previously set.

There are two types of controls: “path” controls and “device” controls:

- Path controls are those such as VL_SIZE, VL_OFFSET, and VL_ZOOM, which are capable of actively controlling a transfer. These controls are private to a path and any changes (with some exceptions) cause events to be sent *only* to the process owning the path. These controls are active while the path is transferring, and retain their values when the transfer is suspended for any reason. In practice, this means that the user program can set up the desired transfer controls, and then restart a preempted transfer without restoring controls to their previous values.
- Device controls are those such as VL_BRIGHTNESS and VL_CONTRAST, which are outside the realm of a “path” and can possibly affect the data that another path is processing. Because most of these controls directly affect some hardware change, they retain their values after the paths are removed.

Establishing the Default Input Source

VL_DEFAULT_SOURCE specifies which of the input nodes is to be considered the “default” input. This is automatically set up when the video driver is loaded according to Table 4-1, which indicates which input signal(s) is active.

Table 4-1 Default Video Source

S-video	Composite	Camera	Default_Source
yes	x	x	svideo
no	yes	x	composite
no	no	yes	camera
no	no	no	composite

For example, if a VCR is connected to the S-Video input and it is powered on, then it is the default input.

When the VL_DEFAULT_SOURCE is changed, a VLDefaultSource event is sent to all processes that have this event enabled in their vlEventMask.

Getting Video Source Controls

Most source controls are read-only values that are set either by the user (from the Video Control Panel) or automatically, according to the characteristics of the video input signal. However, reading the values of these controls is useful for obtaining information about the input video stream that is necessary for setting controls on the drain node.

Getting Video Input Format Using the VL_FORMAT Control

The VL_FORMAT control on the video source node is usually set using the Video Control Panel. It is often of no concern to a vid-to-mem application, except with Sirius Video™, where it is used to determine color space conversion.

VL_FORMAT selects the input video format (use VL_MUXSWITCH if there is more than one to select):

- VL_FORMAT_COMPOSITE selects analog composite video.
- VL_FORMAT_SVIDEO selects analog composite video.
- VL_FORMAT_DIGITAL_COMPONENT and VL_FORMAT_DIGITAL_COMPONENT_SERIAL select digital video.
- VL_FORMAT_DIGITAL_INDYCAM and VL_FORMAT_DIGITAL_CAMERA select the connected camera.

Getting Video Input Timing Using the VL_TIMING Control

The VL_TIMING control on the video source node is usually set from the Video Control Panel. The input source timing also affects the value returned by the VL_SIZE control on the video source node.

Use VL_TIMING to determine whether the input source timing is PAL or NTSC, and whether the input pixels are square or not. Knowing whether the input signal is PAL or NTSC timing is useful for setting the VL_RATE control on the memory drain node. (For Sirius Video, it is also used to determine the value for the VL_TIMING control on the memory drain node.) An easy way to set the VL_TIMING value for the memory node is to read the value of the VL_TIMING control from the video source node, and then set that value into the VL_TIMING control for the memory node.

The VL_TIMING control is an integer value that adjusts the video filter for different video standards.

The 525 (NTSC) or 625 (PAL) timing standards are specified, and the pixels are considered to be in the accepted video aspect ratio for those standards (also known as "non-square") for VL_TIMING_525_CCIR601 and VL_TIMING_625_CCIR601.

The 525 (NTSC) or 625 (PAL) timing standards are specified and, depending on the VL video device and the connector type, a non-square-to-square pixel filter can be engaged so that in memory, the pixels are in a 1:1 aspect ratio (which is compatible with OpenGL) for VL_TIMING_525_SQ_PIX and VL_TIMING_625_SQ_PIX.

When these timings are applied to a path that has a standard digital camera attached, then the 525 (NTSC) or 625 (PAL) timing standards are interpreted to mean that the external pixels are in a 1:1 aspect ratio, and there is no non-square format available for the internal pixels.The pixel conversion applies a ratio of 11/10 for NTSC, and a ratio of 11/12 for PAL.

Note: The application program should always check the default VL_SIZE after a timing change to determine the size of the resultant images.

Getting Video Input SIze Using the VL_SIZE Control

The VL_SIZE control on the video source node is a read-only control. The *x* and *y* values returned by this control are affected by the setting of the VL_TIMING control on the video source node. The *x* and *y* values of this control are not, in general, affected by the settings of any controls in the memory drain node, including VL_ZOOM, VL_SIZE, and VL_CAP_TYPE.

The *x* component value of this control reveals the width, in pixels, of the unzoomed, unclipped video input images (in fields or frames, depending on the VL_CAP_TYPE). The meaning of the *y* component value of the video source node's VL_SIZE control depends on the video device. On Sirius, the *y* value is the number of pixel rows in each field, and includes the count of rows of pixel samples taken from the field's Vertical Retrace Interval. On EV1 and VINO, the *y* value is the number of pixel rows in each frame (pair of fields), and does not include any pixel rows from the Vertical Retrace Interval.

Setting Memory Drain Node Controls

This section describes setting controls on the memory drain node.

Setting the Memory Packing Controls Using the VL_PACKING Control

A vid-to-mem application chooses the color space (that is, the set of components that make up each pixel–for example–RGB, RGBA, YUV, YCrCb, Y, YIQ) and the particular packing of those pixel components into memory using the VL_PACKING control (on all video devices) and also with the VL_FORMAT control on Sirius video.

On all VL video devices except Sirius, VL_FORMAT is not applicable to memory drain nodes, and VL_PACKING is used to select the color space as well as the packing. Packings that imply RGB or RGBA color spaces select those spaces. Packings that imply Y, YUV, or YCrCb color spaces select one of those spaces.

Setting the Memory Capture Mode Using the VL_CAP_TYPE Control

On all VL video devices except Sirius, the capture mode can be set by the application. Its setting determines whether the images in the buffers returned by the VL are individual fields, interleaved frames, or pairs of non-interleaved fields.

VL_CAP_TYPE specifies the capture mode:

- VL_CAPTURE_INTERLEAVED captures or sends buffers that contain both the F1 and F2 fields interlaced in memory. A side effect of changing from non-interleaved to interleaved is that the VL_RATE is halved.
- VL_CAPTURE_NONINTERLEAVED captures or sends buffers that contain only one field each but are transferred in pairs keeping the F1 and even field of a picture together. A side effect of this characteristic is, if a transfer error occurs in the second field, then the first is not transferred.
- VL_CAPTURE_FIELDS captures or sends buffers that contain only one field each and are transferred individually. Since these are separate fields, VL_RATE is effective on individual fields, and a single field may be dropped. Also, changing from interleaved to fields doubles the VL_RATE.
- VL_CAPTURE_EVEN_FIELDS captures only the F1 fields. For output, the field is transferred during both field times.
- VL_CAPTURE_ODD_FIELDS captures only the F1 fields. For output, the field is transferred during both field times

There is no single VL_CAP_TYPE that is available, and implemented in the same way, on all VL video devices. VL_CAPTURE_NONINTERLEAVED is available on all devices, but has different meanings on different platforms. VL_CAPTURE_INTERLEAVED, VL_CAPTURE_EVEN_FIELDS, and VL_CAPTURE_ODD_FIELDS are available and common to all VL video devices except Sirius.

On Sirius Video, VL_CAP_TYPE is read-only, and is permanently set to VL_CAPTURE_NONINTERLEAVED. Each captured buffer contains exactly one field, unclipped, unzoomed, with *n* leading pixel rows of samples from the Vertical Retrace Interval.

EV1 implements VL_CAPTURE_NONINTERLEAVED differently from all other VL video devices. On all VL video devices except EV1, when VL_CAP_TYPE is set to VL_CAPTURE_NONINTERLEAVED, each image buffer that the VL gives to the application contains one field, either F1 or F2, and VL_RATE (the rate at which these buffers are returned) is in fields per second, not frames per second. But on EV1 video devices, when VL_CAP_TYPE is set to VL_CAPTURE_NONINTERLEAVED, each image buffer contains two non-interleaved fields, and VL_RATE is in frames per second.

Setting the Memory Capture Target Rate Using the VL_RATE Control

On all VL video devices except Sirius, VL_RATE sets the target rate (upper bound) of image buffers per second to be captured and returned to the application. The VL does not deliver more buffers per second than the rate you specify, but it can deliver less.

The contents of each image buffer is either a frame or a field, as determined by the VL_CAP_TYPE control. Accordingly, VL_RATE is in units of fields per second or frames per second, as determined by the VL_CAP_TYPE control.

VL_RATE is effective on a pair of fields, though it is still interpreted as a field rate. What this means is that if a field is to be dropped because of the effects of VL_RATE, then both fields are dropped (for output, if the VL_RATE causes some fields to be dropped, then the preceding fields are repeated). Also, changing from interleaved to non-interleaved mode doubles the VL_RATE.

VL_RATE is expressed as a fractional value (an integer numerator divided by an integer denominator) and ranges from the maximum rate (60/1 for NTSC, 50/1 for PAL, and half of each value for VL_CAPTURE_INTERLEAVED) down to 1/0xffff in any increment. Both the numerator and denominator must be specified. The usual value for the denominator is 1. Some devices convert the fraction to an integer number of images per second by truncating rather than rounding, so using values that are equivalent to integer

values is the safest thing to do. Because VL_RATE is a fraction, **vlGetControlInfo()** cannot be used to obtain the minimum or maximum values for VL_RATE.

Acceptable values are determined from the following list of devices:

- VL_CAPTURE_NONINTERLEAVED for all devices except EV1 and Sirius
  - NTSC: all multiples of 10 and 12 between 10 and 60
  - PAL: all multiples of 10 between 10 and 50
- VL_CAPTURE_NONINTERLEAVED (EV1)
  - NTSC: all multiples of 5 and 6 between 5 and 30
  - PAL: all multiples of 5 between 5 and 25
- VL_CAPTURE_INTERLEAVED, VL_CAPTURE_EVEN_FIELDS, and VL_CAPTURE_ODD_FIELDS
  - NTSC: all multiples of 5 and 6 between 5 and 30
  - PAL: all multiples of 5 between 5 and 25
- VL_CAPTURE_NONINTERLEAVED for Sirius Video. This control is read-only. Its value is determined by the setting of the VL_TIMING control on the memory node.
  - NTSC: 60 fields per second
  - PAL: 50 fields per second

VINO's VL_RATE cannot be set to a value less than 5/1.

Setting Video Capture Region Controls

Figure 4-1 shows a diagram of an NTSC F1 field.

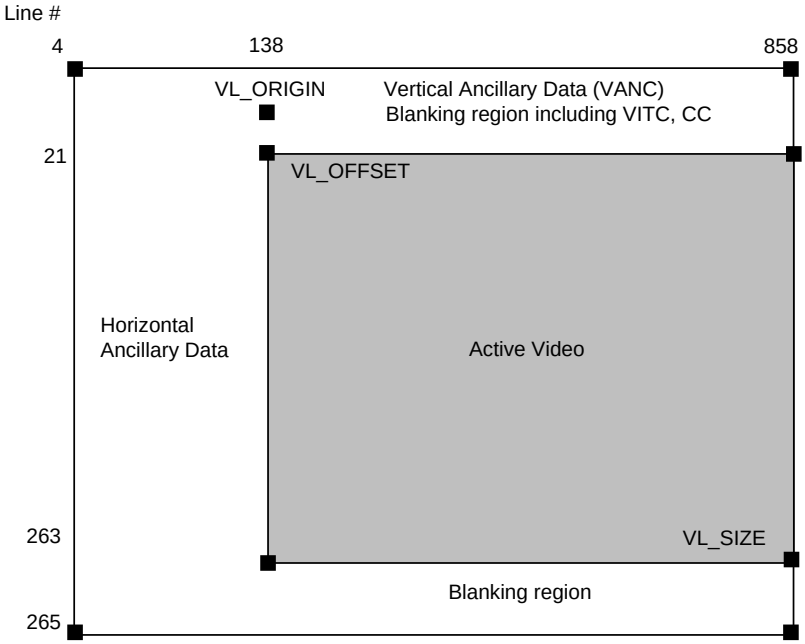


Figure 4-1 Video Image Parameter Controls

The data contained within the area labeled "Active Video" is the default data transferred to and from memory, but the hardware and video driver allow the transfer to include most all the portion of the "hidden" video, or the Horizontal and/or Vertical Ancillary Data (HANC/VANC).

The following controls specify the capture region (all these controls are path controls):

- VL_ORIGIN is used on the screen capture device to specify the origin of the capture area. For Video input, the VL_ORIGIN can be used to specify a "black fill" region.
- VL_OFFSET is used on a source or drain memory node to specify an (*x*, *y)* value that signifies the upper left corner of the active video region. For input, the area to the left and above the VL_OFFSET is omitted. For output, the same region is filled with "black."

The VL_OFFSET values are in "ZOOMED" coordinates (see "Using VL_ZOOM on the Memory Drain Node" below). VL_OFFSET has a default of 0,0. Negative values of VL_OFFSET specify non-picture data such as horizontal and vertical ancillary data, which must be decoded separately from the picture data.

Certain restrictions apply to the value of VL_OFFSET. The resultant offset must be on a 2-pixel boundary, and the minimum offset is restricted to the values listed in the reference pages for the VL video devices. See also "Using VL_SIZE and VL_OFFSET on the Memory Drain Node," for detailed information about these values for memory drain nodes.

Note: The actual minimum offset is affected by VL_ZOOM and VL_ASPECT.

- VL_SIZE is used on a source or drain memory node to specify an (*x*, *y)* value that defines the extent of the active video region. Adding the VL_SIZE coordinates to the VL_OFFSET coordinates gives the coordinates of the lower right corner of the active video region (VL_OFFSET + VL_SIZE = lower right corner). For input, the area to the right and below this corner is omitted. For output, the same region is filled with "black."

  The VL_SIZE values are in "ZOOMED" coordinates. See "Using VL_SIZE and VL_OFFSET on the Memory Drain Node," for details about VL_SIZE values.

  Certain restrictions apply to the value of VL_SIZE: The resultant size must be on a 2-pixel boundary and the number of bytes to be transferred must be a multiple of 8.

  The maximum VL_SIZE is defined by the total number of lines in the video standard. Increasing the VL_SIZE beyond the maximum horizontal dimension causes VL_OFFSET to assume negative values. Out-of-range values return vlErrno VLValueOutOfRange.

The use of all these controls is explained in the sections that follow.

Using VL_SIZE and VL_OFFSET on the Memory Drain Node

This section discusses the VL_SIZE control and the VL_OFFSET control on the memory drain node.

The VL_SIZE control on the memory drain node determines the number of rows of pixels, and the number of pixels in each row, in each image buffer (field or frame) that the VL returns to the application. If zooming (decimation) is being done, the VL_SIZE control on the memory drain node specifies the size of the image after it has been decimated.

The VL_SIZE control on the memory drain node can be used to "clip" a region out of an image by setting the *x* and/or *y* components to values that are smaller than the size of the captured (and decimated, if applicable) image.

When the (possibly decimated) image is being clipped, the clipped region does not have to come from the upper left hand corner of the (possibly decimated) source image. The VL_OFFSET control on the memory drain node determines the number of top pixel rows to skip and the number of leading pixels to skip in each row to find the first pixel in the (possibly decimated) image to place in the image buffer, the first pixel of the clipping region.

When zooming (decimation) is being used, VL_OFFSET is always in coordinates of the zoomed image. It is as if the entire source image is decimated down, and then the clipping function is applied to the decimated image. In practice, the hardware usually clips before decimating, but the VL API always specifies the VL_OFFSET in the coordinates of the decimated (virtual) image.

On all VL devices except Sirius, the vertical (*y*) component of VL_OFFSET may be specified with a negative value. This causes the clipping region to include row of samples taken before the top of the image, for example, rows from the Vertical Retrace Interval. This feature is usually used with VL_ZOOM of 1/1, since the information in the Vertical Retrace Interval isn't an image and doesn't make sense to decimate or average, at least not in the vertical direction.

The VL imposes these requirements on the values of VL_OFFSET and VL_SIZE:

- The sum of the vertical components of VL_OFFSET and VL_SIZE must not exceed the height of the virtual (zoomed) image, and
- The sum of the horizontal components of VL_OFFSET and VL_SIZE must not exceed the width of the virtual (zoomed) image.

  When an attempt to set either one of these controls violates either of these rules, the call to **vlSetControl()** fails with the vlErrno VLValueOutOfRange, and the offending component (horizontal or vertical) is set to the largest non-negative value that does not violate the rule, or to zero if no such non-negative value exists.

Both VL_OFFSET and VL_SIZE cannot be set in one atomic operation. A change in either component of either control could violate one of the rules, especially after VL_ZOOM is set to a smaller fraction. It may be necessary to alternately and repeatedly set VL_OFFSET and VL_SIZE until no VLValueOutOfRange errors are reported.

Every VL video device places additional limitations on the range of acceptable values of VL_SIZE and VL_OFFSET. Each device has different limitations.

- Sirius doesn't clip at all. VL_SIZE and VL_OFFSET are read-only in Sirius.
- EV1 supports clipping only in the vertical (Y) direction. The entire width of the (possibly decimated) image is always placed in the image buffer. Application-specified horizontal clipping values are ignored.
- VINO imposes an additional list of requirements on VL_SIZE and VL_OFFSET, along with the following clipping requirements:
  - The right side edge of the clipped image must always coincide with the right side edge of the virtual (possibly decimated) image. That is, the clipped image must always come from the right side of the (possibly decimated) source image. Consequently, when **vlSetControl()** is called to set the VL_OFFSET or VL_SIZE control on a memory node, if the sum of the horizontal components of the (new) settings of VL_OFFSET and VL_SIZE is less than the width of the virtual (zoomed) image, the **vlSetControl()** call succeeds, and the horizontal component of the other control is adjusted so that the sum of the two components exactly equals the width of the virtual (zoomed) image. This is done only in the horizontal direction.
  - Each pixel row in the image buffer must be a multiple of 8 bytes in length. This means that the horizontal component of VL_SIZE must be a multiple of 2, 4, or 8 pixels, depending on the pixel packing (size of the individual pixels in memory).

Using VL_ZOOM on the Memory Drain Node

VL_ZOOM controls the expansion or decimation of the video image. Values greater than 1 expand the video; values less than 1 perform decimation. The only value of VL_ZOOM that works on all VL devices is 1/1. Acceptable values for vid-to-mem applications follow.

VINO may exhibit the following effects at these decimation factors: 1/4, 1/5, 1/6, 1/7, and 1/8:

- Y values that are not adjacent horizontally are averaged together
- The decimated images appear extremely green.

As a workaround, the VINO driver implements decimation by 1/4 and 1/6 by decimating in hardware by 1/2 or 1/3, and then decimating by an additional factor of

1/2 in software. This produces acceptable looking images, but at significant cost in CPU time. The three other VL_ZOOM factors, 1/5, 1/7, and 1/8, also exhibit the green image effect.

For example, the listed zoom factors on VINO may behave as follows:

1/1, 1/2, 1/3	Implemented in hardware. Looks OK.
1/4, 1/6	Implemented partially in hardware, partially in software. Looks OK, but is slower and uses 10% of an R4600 CPU.
1/5, 1/7, 1/8	Implemented in hardware. Exhibits green shift.

For example, the listed zoom factors on EV1 may behave as follows:

1/1, 1/2, 1/4, 1/8	Works for vid-to-mem
1/3, 1/5, 1/7	Works only for vid-to-screen, not vid-to-mem, and only with VL_CAPTURE_INTERLEAVED
2/1, 4/1	Works only for vid-to-screen, not vid-to-mem

Note: Sirius and Galileo 1.5 accept only a 1/1 zoom factor (Sirius and Galileo 1.5 don't zoom).

VL_ZOOM specifies the decimation of the input video to a fraction of its original size. Scaling from 1/1 down to 1/256 is available; the actual increments are: 256 to 1/256. The actual zoom value is affected by VL_ASPECT.

Note: VL_ZOOM is available only on the VL_DRN/VL_MEM (input) node.

VL_SYNC selects the type of sync used for video output. The choices are:

- VL_SYNC_INTERNAL means that the timing for the output is generated using an internal oscillator appropriate for the timing required (NTSC or PAL).
- VL_SYNC_GENLOCK means that the timing for the output is "genlocked" to the VL_SYNC_ SOURCE.
- VL_SYNC_SOURCE selects which sync source is used when VL_SYNC is set to VL_SYNC_GENLOCK.

VL_LAYOUT specifies the pixel layout (same as DM_IMAGE_LAYOUT):

- VL_LAYOUT_LINEAR means that video pixels are arranged in memory linearly.
- VL_LAYOUT_GRAPHICS means that video pixels are arranged in memory in a Pbuffer fashion that is compatible with the O2 OpenGL.
- VL_LAYOUT_MIPMAP means that video pixels are arranged in memory in a texture or mipmapped fashion that is compatible with the O2 OpenGL.

Signal Quality Controls

The following signal quality controls are available (as supported by the video device):

- VL_BRIGHTNESS
- VL_CONTRAST
- VL_H_PHASE
- VL_HUE
- VL_SATURATION
- VL_RED_SETUP
- VL_GREEN_SETUP
- VL_GRN_SETUP
- VL_BLUE_SETUP
- VL_BLU_SETUP
- VL_ALPHA_SETUP
- VL_V_PHASE

Each of these controls is defined if they are provided in the analog encoder or decoder. They are not available in the digital domains.

VL_SIGNAL can be either VL_SIGNAL_NOTHING, VL_SIGNAL_BLACK, or VL_SIGNAL_REAL_IMAGE.

VL_FLICKER_FILTER enables or disables the “flicker” filter.

VL_DITHER_FILTER enables or disables the “dither” filter.

VL_NOTCH_FILTER enables or disables the “notch” filter.

To determine default values, use **vlGetControl()** to query the values on the video source or drain node before setting controls. For all these controls, it pays to track return codes. If the value returned is VLValueOutOfRange, the value set is not what you requested.

Table 4-2 summarizes the VL controls. For each control, the ASCII name of the control, the type of value it takes, and the node types and classes to which it can be applied is listed.

Table 4-2 Summary of VL Controls

Control	ASCII Name	Value	Node Type/Class
VL_DEFAULT_SOURCE	default_input	intVal	VL_SRC/VL_VIDEO
VL_TIMING	timing	intVal	VL_SRC/VL_VIDEO
VL_ORIGIN	origin	xyVal	VL_ANY/VL_MEM
VL_SIZE	size	xyVal	VL_ANY/VL_MEM
VL_RATE	fieldrate	fractVal	VL_ANY/VL_MEM
VL_ZOOM	zoom	fractVal	VL_ANY/VL_MEM
VL_ASPECT	aspect	fractVal	VL_ANY/VL_MEM
VL_CAP_TYPE	fieldmode	intVal	VL_ANY/VL_MEM
VL_PACKING	packing	intVal	VL_ANY/VL_MEM
VL_FORMAT	format	intVal	VL_SRC/VL_VIDEO, VL_ANY/VL_MEM
VL_SYNC	sync	intVal	VL_DRN/VL_VIDEO
VL_SYNC_SOURCE	sync_source	intVal	VL_DRN/VL_VIDEO
VL_LAYOUT	layout	intVal	VL_ANY/VL_MEM
VL_SIGNAL	signal	intVal	VL_DRN/VL_VIDEO
VL_FLICKER_FILTER	flicker_filter	boolVal	VL_SRC/VL_SCREEN
VL_DITHER_FILTER	dither_filter	boolVal	VL_SRC/VL_VIDEO
VL_NOTCH_FILTER	notch_filter	boolVal	VL_DRN/VL_VIDEO

The ASCII name is used to assign values to controls in the VL Resources file and can also be found in the control table returned by **vlGetControlList()**.

The following list is a key to which nodes the control can be applied:

- VL_SRC/VL_VIDEO—source video node
- VL_DRN/VL_VIDEO—drain video node
- VL_ANY/VL_VIDEO—source or drain video node
- VL_SRC/VL_SCREEN—source screen node
- VL_SRC/VL_MEM—source memory node
- VL_DRN/VL_MEM—drain memory node
- VL_ANY/VL_MEM—source or drain memory node

Video Events

Video events provide a way to monitor the status of a video I/O stream. Typically, a number of events are combined into an event mask that describes the events of interest. Use **vlSelectEvents()** to specify the events you want to receive. Its function prototype is

```
int vlSelectEvents( VLServer vlSvr, VLPath path, VLEventMask eventmask )
```

where

vlSvr	names the server to which the path is connected
path	specifies the data path
eventmask	specifies the event mask; Table 4-3 lists the possibilities

Table 4-3 lists and describes the VL event masks.

Table 4-3 VL Event Masks

Symbol	Meaning
VLStreamBusyMask	Stream is locked
VLStreamPreemptedMask	Stream was grabbed by another path
VLStreamChangedMask	Video routing on this path has been changed by another path
VLAdvanceMissedMask	Time was already reached
VLSyncLostMask	Irregular or interrupted signal
VLSequenceLostMask	Field or frame dropped
VLControlChangedMask	A control has changed
VLControlRangeChangedMask	A control range has changed
VLControlPreemptedMask	Control of a node has been preempted, typically by another user setting VL_LOCK on a path that was previously set with VL_SHARE
VLControlAvailableMask	Access is now available
VLTransferCompleteMask	Transfer of field or frame complete
VLTransferFailedMask	Error; transfer terminated; perform cleanup at this point, including **vlEndTransfer()**
VLEvenVerticalRetraceMask	Vertical retrace event, even field
VLOddVerticalRetraceMask	Vertical retrace event, odd field
VLFrameVerticalRetraceMask	Frame vertical retrace event
VLDeviceEventMask	Device-specific event, such as a trigger
VLDefaultSourceMask	Default source changed

When transferring video, the main event is a VLTransferComplete.

Video I/O Model

In the traditional video I/O model, you use the buffering, data transfer, and event handling routines supplied by the VL. One of the consequences of this approach is that it might require you to copy data passed outside the VL. (See Chapter 5 for the DMbuffers I/O method for O2™ workstations.)

A basic VL application has the following components:

Preliminary path setup:

- **vlOpenVideo()**—open the video server
- **vlGetDeviceList()**—discover which devices and nodes are connected to this system
- **vlGetNode()**—get the source and drain nodes
- **vlCreatePath()**—create a video path with the source and drain nodes specified
- **vlSetupPath()**—set the path up to be usable given the access requested
- **vlDestroyPath()**—remove a video path

Specific control settings:

- **vlSetControl()**—set various parameters associated with the video transfer
- **vlGetControl()**—get various parameters associated with the video transfer

Preparing to capture or output video to/from memory:

- **vlCreateBuffer()**—create a VLbuffer
- **vlRegisterBuffer()**—register this buffer with the path

Starting and controlling the video transfer:

- **vlBeginTransfer()**—initiate the transfer
- **vlEndTransfer()**—terminate the transfer
- **vlNextEvent()**—handle events from the video device
- **vlGetNextValid()**—get incoming buffers with captured video
- **vlPutValid()**—send outgoing buffers with inserted video

Freezing Video

Showing a still frame from a recorded video sequence (either uncompressed or compressed using JPEG) presents an enigma. Displaying a still frame requires a complete set of spatial information at a single instant of time—the data is simply not available to display a still frame correctly.

One way to display a still frame is to combine the lines from two adjacent fields, as shown in Figure 4-2. No matter which pair of fields you choose, the resulting still frame exhibits artifacts.

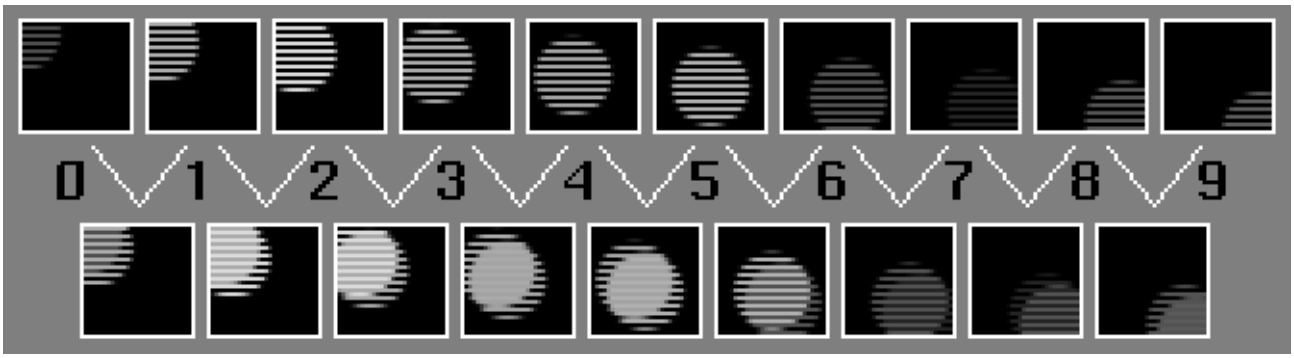


Figure 4-2 Tearing

Figure 4-2 shows a display artifact known as *tearing* or *fingering*, which is an inevitable consequence of putting together an image from bits of images snapped at different times. You don't notice the artifact if the fields are flashed in rapid succession at the field rate, but when you try to freeze motion and show a frame, the effect is visible. You wouldn't notice the artifact if the objects being captured were not moving between fields.

These types of artifacts cause trouble for most compressors. If you are capturing still frames to pass frame-sized images on to a compressor, you definitely should avoid tearing. A compressor wastes a lot of bits trying to encode the high-frequency information in the tearing artifacts and fewer bits encoding your actual picture. Depending on the size and quality of compressed image you want, you might consider sending every other field (perhaps decimated horizontally) to the compressor, rather than trying to create frames that compress well.

Another possible technique for producing still frames is to double the lines in a single field, as shown in Figure 4-3.

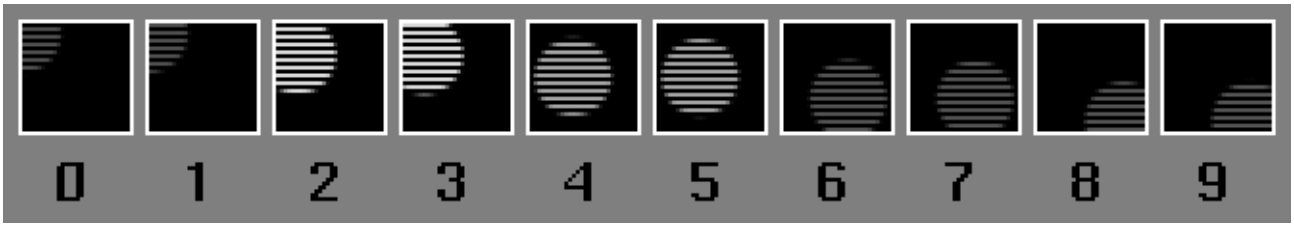


Figure 4-3 Line Doubling on a Single Field

This looks a little better, but there is an obvious loss of spatial resolution (as evidenced by the visible “jaggies” and vertical blockiness).

To some extent, this can be reduced by interpolating adjacent lines in one field to get the lines of the other field, as shown in Figure 4-4.

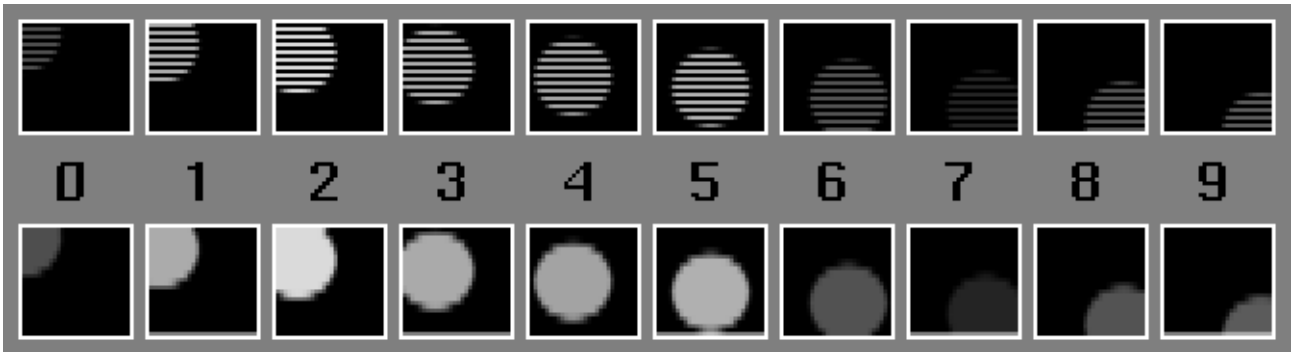


Figure 4-4 Interpolating Alternate Scan Lines from Adjacent Fields

There are an endless variety of more elaborate tricks you can use to come up with good still frames, all of which come under the heading of “de-interlacing methods.” Some of these tricks attempt to use data from both fields in areas of the image that are not moving (so you get high spatial resolution), and double or interpolate lines of one field in areas of the image that are moving (so you get high temporal resolution). Many of the tricks take more than two fields as input. Since the data is simply not available to produce a spatially complete picture for one instant, there is no perfect solution. But depending on why you want the still frame, the extra effort may well be worth it.

When a CRT-based television monitor displays interlaced video, it doesn't flash one frame at a time on the screen. During each field time (each 50th or 60th of a second), the CRT lights up the phosphors of the lines of that field only. Then, in the next field interval, the CRT lights up the phosphors belonging to the lines of the other field. So, for example, at the instant when a pixel on a given picture line is refreshed, the pixels just above and below that pixel have not been refreshed for a 50th or 60th of a second, and will not be refreshed for another 50th or 60th of a second.

So if that's true, then why don't video images flicker or jump up and down as alternate fields are refreshed?

This is partially explained by the persistence of the phosphors on the screen. Once refreshed, the lines of a given field start to fade out slowly, and so the monitor is still emitting some light from those lines when the lines of the other field are being refreshed. The lack of flicker is also partially explained by a similar persistence in your visual system.

Unfortunately though, these are not the only factors. Much of the reason you do not perceive flicker on a video screen is that good-looking video signals themselves have built-in characteristics that reduce the visibility of flicker. It is important to understand these characteristics, because when you synthesize images on a computer or process digitized images, you must produce an image that also has these characteristics. An image that looks good on a non-interlaced computer monitor can look inferior on an interlaced video monitor.

A complete understanding of when flicker is likely to be perceived and how to get rid of it requires an in-depth analysis of the properties of the phosphors of a particular monitor (not only their persistence but also their size, overlap, and average viewing distance), it requires more knowledge of the human visual system, and it may also require an in-depth analysis of the source of the video (the persistence, size, and overlap of the CCD elements used in the camera, the shape of the camera's aperture, and so on). This description is intended to give only a general sense of the issues.

Standard analog video (NTSC and PAL) has characteristics (such as bandwidth limitations) that can introduce many artifacts similar to the ones described here into the final result of video output from a computer. Describing these artifacts is beyond the scope of this document, but they are important to consider when creating data to be converted to an analog video signal. An example of this would be antialiasing (blurring) data in a computer to avoid chroma aliasing when the data is converted to analog video.

Here are some of the major areas to be concerned about when creating data for video output:

- Abrupt vertical transitions: 1-pixel-high lines

  First of all, typical video images do not have abrupt vertical changes. For example, say you output an image that is entirely black except for one, 1-pixel-high line in the middle.

  Since the non-black data is contained on only one line, it appears in only one field. A video monitor updates the image of the line only 30 times a second, and the line flickers on and off quite visibly. To see this on a video-capable machine, run *videoout*, turn off the antiflicker filter, and point *videoout*'s screen window at the line.

  You do not have to have a long line for this effect to be visible: Narrow, non-antialiased text exhibits the same objectionable flicker.

  Typical video images are more vertically blurry; even where there is a sharp vertical transition (the bottom of an object in sharp focus, for example), the method typical cameras use to capture the image will cause the transition to blur over more than one line. It is often necessary to simulate this blurring when creating synthetic images for video.

- Abrupt vertical transitions: 2-pixel-high lines

  These lines include data in both fields, so part of the line is updated each 50th or 60th of a second. Unfortunately, when you actually look at the image of this line on a video monitor, the line appears to be solid in time, but it appears to jump up and down, as the top and bottom line alternate between being brighter and darker. You can also see this with the *videoout* program.

  The severity of these effects depends greatly on the monitor and its properties, but they are generally objectionable. One partial solution is to vertically blur the data you are outputting. Turning on the "flicker filter" option to *videoout* causes some boards (such as EV1) to vertically prefilter the screen image by a simple 3-tap (1/4,1/2,1/4) filter. This noticeably improves (but does not remove) the flickering effect.

There is no particular magic method that produces flicker-free video. The more you understand about the display devices you care about, and about when the human vision system perceives flicker and when it does not, the better you can produce a good image.

Synthetic Imagery and Fields

When you modify digitized video data or synthesize new video data, the result must consist of fields with all the same properties, but temporally offset and spatially disjointed. This may not be trivial to implement in a typical renderer without wasting a lot of rendering resources (rendering 50/60 images a second, throwing out unneeded lines in each field) unless the developer has fields in mind from the start.

You might think that you can generate synthetic video by taking the output of a frame-based renderer at 25/30 frames per second and pulling two fields out of each frame image. This does not work well: the motion in the resulting sequence on an interlaced video monitor noticeably stutters, due to the fact that the two fields are scanned out at different times, yet represent an image from a single time. Your renderer must know that it is rendering 50/60 temporally distinct images per second.

Slow-motion Playback and Synthesizing Dropped Fields

Two relatively easy tasks to do with frame-based data, such as movies, are playing it slowly (by outputting some frames more than once) or dealing with frames that are missing in the input stream by duplicating previous frames. There are more elaborate ways to generate better-looking results in these cases, and they, too, are not difficult to implement on frame-based data.

Suppose you are playing a video sequence, and run up against a missing field, as shown in Figure 4-5 (the issues discussed here also come up when you want to play back video slowly).

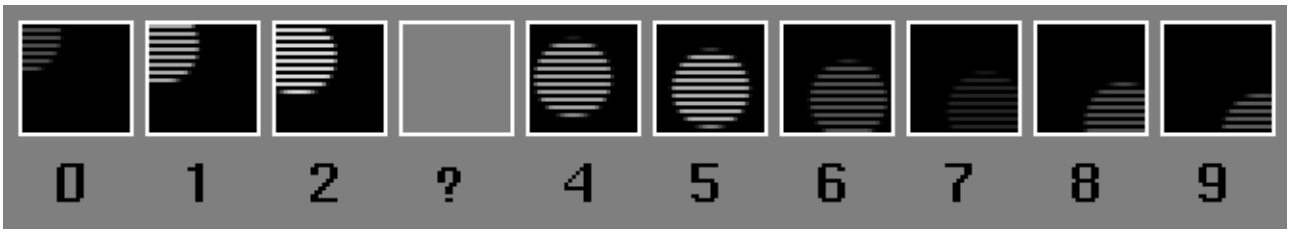


Figure 4-5 Dropped Frame

To keep the playback rate of the video sequence constant, put some video data in that slot. Which field do you choose?

Suppose you choose to duplicate the previously displayed field (field 2), as shown in Figure 4-6.

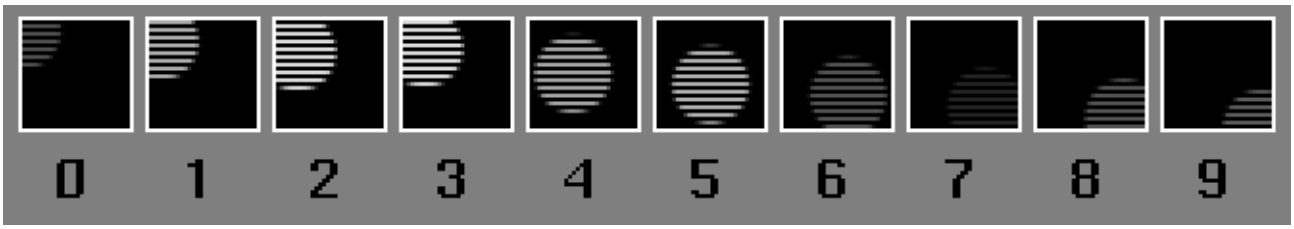


Figure 4-6 Field Duplication

You can also try duplicating field 4 or interpolating between 2 and 4, but with all of these methods there is a crucial problem: The surrounding fields contain data from a different spatial location than the missing field. If you view the resulting video, you immediately notice that the image visually jumps up and down at this point. This is a large-scale version of the same problem that made the 2-pixel-high line jump up and down: Your eye is very good at picking up on the vertical "motion" caused by an image being drawn to the lines of one field, then being drawn again one picture line higher, into the lines of the other field. You would see this even if the ball was not in motion.

Suppose instead you choose to fill in the missing field with the last non-missing field that occupies the same spatial locations, as shown in Figure 4-7.

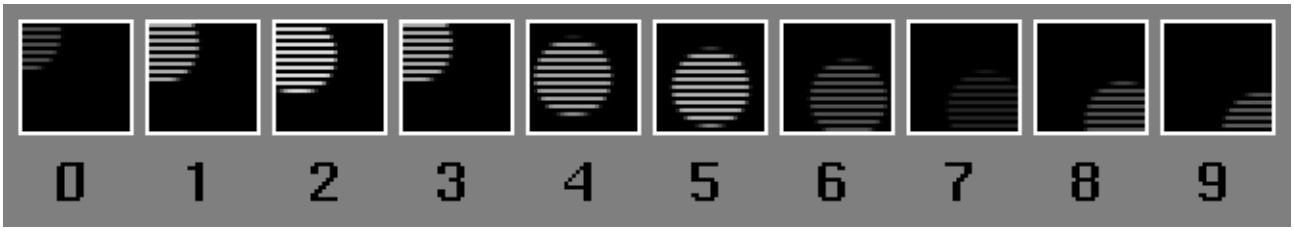


Figure 4-7 Field Replacement

Now you have a more obvious problem: You are displaying the images temporally out of order. The ball appears to fly down, fly up again for a bit, and then fly down. Clearly, this method is not good for video that contains motion. But for video containing little or no motion, it works pretty well, and does not suffer the up-and-down jittering of the previous approach.

Which of these two methods is best thus depends on the video being used. For general-purpose video where motion is common, you'd be better off using the first technique, the "temporally correct" technique. For certain situations such as computer screen capture or video footage of still scenes, however, you can often get guarantees that the underlying image is not changing, and the second technique, the "spatially correct" technique, is best.

As with de-interlacing methods, there are many more elaborate methods for interpolating fields that use more of the input data. For example, you can interpolate fields 2 and 4 and then interpolate the result of that vertically to guess at the content of the other field's lines. Depending on the situation, these techniques may or may not be worth the effort.

Still Frames on Video Output

The problem of getting a good still frame from a video input has a counterpart in video output. Suppose you have a digitized video sequence and you want to pause playback of the sequence. Either you, the video driver, or the video hardware must continue to output video fields even though the data stream has stopped, so which fields do you output?

If you choose the "temporally correct" method and repeatedly output one field (effectively giving you the "line-doubled" look), then you get an image with reduced vertical resolution. But you also get another problem: As soon as you pause, the image appears to jump up or down, because your eye picks up on an image being drawn into the lines of one field, and then being drawn one picture line higher or lower, into the lines of another field. Depending on the monitor and other factors, the paused image may appear to jump up and down constantly or it may appear to jump only when you enter and exit pause.

If you choose the "spatially correct" method and repeatedly output a pair of fields, then if there is any motion at the instant where you pause, you see that motion happening back and forth, 60 times a second. This can be very distracting.

There are, of course, more elaborate heuristics that can be used to produce good looking pauses. For example, vertically interpolating an F1 to make an F2 or vice versa works well for slow-motion, pause, and vari-speed play. In addition, vertical interpolation can be combined with inter-field interpolation for "super slow-motion" effects.

Audio I/O Concepts

The Audio Library (AL) provides a device-independent C language API for programming audio I/O on all Silicon Graphics workstations. It provides routines for configuring the audio hardware, managing audio I/O between the application program and the audio hardware, specifying attributes of digital audio data, and facilitating real-time programming. This section describes how to set up and use the AL facilities that provide audio I/O capability.

Audio Library Programming Model

Programming audio I/O involves three basic concepts:

Audio device(s) The audio hardware used by the AL, which is shared among audio applications. Audio devices contain settings pertaining to the configuration of both the internal audio system and the external electrical connections.

ALport A one-way (input or output) audio data connection between an application program and the host audio system. An ALport contains:

- an audio sample queue, which stores audio sample frames awaiting input or output
- settings pertaining to the attributes of the digital audio data it transports

Some of the settings of an ALport are static; they cannot be changed once the ALport has been opened. Other settings are dynamic; they can be changed while an ALport is open.

ALconfig An opaque data structure for configuring the settings of an ALport:

- audio device (static setting)
- size of the audio sample queue (static setting)
- number of channels (static setting)
- format of the sample data (dynamic setting)
- width of the sample data (dynamic setting)
- range of floating point sample data (dynamic setting)

Audio Ports

An ALport provides a one-way (input or output) interface between an application program and the host audio system. More than one ALport can be opened by the same application; the number of ALports that can be active at the same time depends on the hardware and software configurations you are using. Open ALports use CPU resources, so be sure to close an ALport when I/O is completed and free the ALconfig when it is no longer needed.

An ALport consists of a queue, which stores audio data awaiting input or output, and static and dynamic state information.

Audio I/O is accomplished by opening an audio port and reading audio data from or writing audio data to the port. For audio input, the hardware places audio sample frames in an input port's queue at a constant rate, and your application program reads the sample frames from the queue. Similarly, for audio output, your application writes audio sample frames to an output port's queue, and the audio hardware removes the sample frames from the queue. A minimum of two ALports are needed for input and output capability for an audio application.

Using ALconfig Structures to Configure ALports

You can open an ALport with the default configuration, or you can customize an ALconfig for configuring an ALport suited to your application needs.

The default ALconfig has:

- a buffer size of 100,000 sample frames
- stereo data
- a two's complement sample format
- a 16-bit sample width

These settings provide an ALport that is compatible with CD- and DAT-quality data, but if your application requires different settings, you must create an ALconfig with the proper settings before opening a port. The device, channel, and queue-size settings for an ALport are static—they cannot be changed after the port has been opened.

The steps for configuring and opening an ALport are listed below.

1. If the default ALconfig settings are satisfactory, you can simply open a default ALport by using 0 for the configuration in the **alOpenPort()** routine; otherwise, create a new ALconfig by calling **alNewConfig()**.
2. If nondefault values are needed for any of the ALconfig settings, set the desired values as follows:
   - Call **alSetChannels()** to change the number of channels.
   - Call **alSetQueueSize()** to change the sample queue size.
   - Call **alSetSampFmt()** to change the sample data format.
   - Call **alSetWidth()** to change the sample data width.
   - Call **alSetFloatMax()** to set the maximum amplitude of floating point data (not necessary for integer data formats).
3. Open an ALport by passing the ALconfig to the **alOpenPort()** routine.
4. Create additional ALports with the same settings by using the same ALconfig to open as many ports as are needed.

To create a new ALconfig structure that is initialized to the default settings, call **alNewConfig()**. Its function prototype is

```
ALconfig alNewConfig ( void )
```

The ALconfig that is returned can be used to open a default ALport, or you can modify its settings to create the configuration you need. In Example 4-1, the channel, queue size, sample format, and floating point data range settings of an ALconfig named *audioconfig* are changed.

alNewConfig() returns an ALconfig structure upon successful completion; otherwise, it returns 0 and sets an error code that you can retrieve by calling oserror(3C). A possible error is:

AL_BAD_OUT_OF_MEM	insufficient memory available to allocate the ALconfig structure

Audio ports are opened and closed by using **alOpenPort()** and **alClosePort()**, respectively. Unless you plan to use the default port configuration, set up an ALconfig structure by using **alNewConfig()** and then use the routines for setting ALconfig fields, such as **alSetChannels()**, **alSetQueueSize()**, and **alSetWidth()** before calling **alOpenPort()**.

Example 4-1 demonstrates how to configure and open an output ALport that accepts floating point mono sample frames.

Example 4-1 Configuring and Opening an ALport

```
ALconfig audioconfig;
ALport audioport;
int err;

void audioinit /* Configure an audio port */
{
audioconfig = alNewConfig();

alSetSampFmt(audioconfig, AL_SAMPFMT_FLOAT);
alSetFloatMax(audioconfig, 10.0);
alSetQueueSize(audioconfig, 44100);
alSetChannels(audioconfig,AL_MONO);

audioport = alOpenPort("surreal","w",audioconfig);
if (audioport == (ALport) 0) {
    err = oserror();
    if (err == AL_BAD_NO_PORTS) {
        fprintf(stderr, " System is out of audio ports\n");
    } else if (err == AL_BAD_DEVICE_ACCESS) {
        fprintf(stderr, " Couldn't access audio device\n");
    } else if (err == AL_BAD_OUT_OF_MEM) {
        fprintf(stderr, " Out of memory\n");
    }
    exit(1);
}
```

Audio Sample Queues

Audio sample frames are placed in the sample queue of an ALport to await input or output. The audio system uses one end of the sample queue; the audio application uses the other end.

During audio input, the audio hardware continuously writes audio sample frames to the tail of the input queue at the selected input rate, for example, 44,100 sample frames per second for 44.1 kHz stereo data. If the application can't read the sample frames from the head of the input queue at least as fast as the hardware writes them, the queue fills up and some incoming sample data is irretrievably lost.

During audio output, the application writes audio sample frames to the tail of the queue. The audio hardware continuously reads sample frames from the head of the output queue at the selected output rate, for example, 44,100 sample pairs per second for 44.1 kHz stereo data, and sends them to the outputs. If the application can't put sample frames in the queue as fast as the hardware removes them, the queue empties, causing the hardware to send 0-valued sample frames to the outputs (until more data is available), which are perceived as pops or breaks in the sound.

For example, if an application opens a stereo output port with a queue size of 100,000, and the output sample rate is set to 48 kHz, the application needs to supply ($2 \times 48{,}000 = 96{,}000$) sample frames to the output port at the rate of at least 1 set of sample frames per second, because the queue contains enough space for about one second of audio at that rate. If the application fails to supply data at this rate, an audible break occurs in the audio output.

On the other hand, if an application tries to put 40,000 sample frames into a queue that already contains 70,000 sample frames, there isn't enough space in the queue to store all the new sample frames, and the program *blocks* (waits) until enough of the existing sample frames have been removed to allow for all 40,000 new sample frames to be put in the queue. The AL routines for reading and writing block; they do not return until the input or output is complete.

To allocate and initialize an ALport structure, call **alOpenPort()**. Its function prototype is

```
ALport alOpenPort ( char *name, char *direction, ALconfig config )
```

where

name is an ASCII string used to identify the port for humans (much like a window title in a graphics program). The name is limited to 20 characters and should be both descriptive and unique, such as an acronym for your company name or the application name, followed by the purpose of the port.

direction specifies whether the port is for input or output:

- "*r*" configures the port for reading (input)
- "*w*" configures the port for writing (output)

config is an ALconfig that you have previously defined or is null (0) for the default configuration.

Upon successful completion, **alOpenPort()** returns an ALport structure for the named port; otherwise, it returns a null-valued ALport, and sets an error code that you can retrieve by calling oserror(3C). Possible errors include:

AL_BAD_CONFIG	*config* is either invalid or null
AL_BAD_DIRECTION	*direction* is invalid
AL_BAD_OUT_OF_MEM	insufficient memory available to allocate the ALport structure
AL_BAD_DEVICE_ACCESS	audio hardware is inaccessible
AL_BAD_NO_PORTS	no audio ports currently available

alClosePort() closes and deallocates an audio port—any sample frames remaining in the port are not output.

Example 4-2 opens an input port and an output port and then closes them.

Example 4-2 Opening Input and Output ALports

```
input_port = alOpenPort("waycoolinput", "r", 0);
if (input_port == (ALport) 0 {
    err = oserror();
    if (err == AL_BAD_NO_PORTS) {
        fprintf(stderr, " System is out of audio ports\n");
    } else if (err == AL_BAD_DEVICE_ACCESS) {
        fprintf(stderr, " Couldn't access audio device\n");
    } else if (err == AL_BAD_OUT_OF_MEM) {
        fprintf(stderr, " Out of memory: port open failed\n");
    }
    exit(1);
}
...
output_port = alOpenPort("killeroutput", "w", 0);
if (input_port == (ALport) 0 {
    err = oserror();
    if (err == AL_BAD_NO_PORTS) {
        fprintf(stderr, " System is out of audio ports\n");
    } else if (err == AL_BAD_DEVICE_ACCESS) {
        fprintf(stderr, " Couldn't access audio device\n");
    } else if (err == AL_BAD_OUT_OF_MEM) {
        fprintf(stderr, " Out of memory: port open failed\n");
    }
    exit(1);
```

```
...
alClosePort(input_port);
alClosePort(output_port);
```

Reading and Writing Audio Data

This section explains how an audio application reads and writes audio sample frames to and from ALports.

Audio input is accomplished by reading audio data sample frames from an input ALport's sample queue. Similarly, audio output is accomplished by writing audio data sample frames to an output ALport's sample queue.

alReadFrames() and **alWriteFrames()** provide mechanisms for transferring audio sample frames to and from sample queues. They are *blocking* routines, which means that a program halts execution within the **alReadFrames()** or **alWriteFrames()** call until the request to read or write sample frames can be completed.

Reading Sample Frames From an Input ALport

alReadFrames() reads a specified number of sample frames from an input port to a sample data buffer, blocking until the requested number of sample frames have been read from the port. Its function prototype is

```
int alReadFrames ( const ALport port, void *samples, const int framecount )
```

where

port is an audio port configured for input

samples is a pointer to a buffer into which you want to transfer the sample frames read from input. *samples* is treated as one of the following types, depending on the configuration of the ALport:

char *	for integer sample frames of width AL_SAMPLE_8
short *	for integer sample frames of width AL_SAMPLE_16
long *	for integer sample frames of width AL_SAMPLE_24
float *	for floating point sample frames
double *	for double-precision floating point sample frames

framecount is the number of sample frames to read

To prevent blocking, *framecount* must be less than the return value of **alGetFilled()**.

When 4-channel data is input on systems that do not support 4 line-level electrical connections, that is, when setting AL_CHANNEL_MODE to AL_4CHANNEL is not possible, **alReadFrames()** provides 4 sample frames per frame, but the second pair of sample frames is set to 0.

Table 4-4 shows the input conversions that are applied when reading mono, stereo, and 4-channel input in stereo mode (default) and in 4-channel mode hardware configurations. Each entry in the table represents a sample frame.

Table 4-4 Input Conversions for alReadFrames()

	Hardware Configuration	
Input	**Indigo, and Indigo^2 or Indy in Stereo Mode**	**Indigo^2 or Indy in 4-channel Mode**
Frame at physical inputs	(L_1, R_1)	(L_1, R_1, L_2, R_2)
Frame as read by a mono port	(L_1 + R_1) /2	(Clip (L_1 + L_2), Clip (R_1 + R_2)) /2
Frame as read by a stereo port	(L_1, R_1)	(Clip (L_1 + L_2), Clip (R_1 + R_2))
Frame as read by a 4-channel port	(L_1, R_1, 0, 0)	(L_1, R_1, L_2, R_2)

Note: If the summed signal is greater than the maximum allowed by the audio system, it is clipped (limited) to that maximum, as indicated by the Clip function.

Writing Sample Frames to an Output ALport

Sample frames placed in an output queue are played by the audio hardware after a specific amount of time, which is equal to the number of sample frames that were present in the queue before the new sample frames were written, divided by the (sample rate × number of channels) settings of the ALport.

alWriteFrames() writes a specified number of sample frames to an output port from a sample data buffer, blocking until the requested number of sample frames have been written to the port. Its function prototype is

```
int alWriteFrames ( ALport port, void *samples, long framecount )
```

where

port	is an audio port configured for input
samples	is a pointer to a buffer from which you want to transfer the sample frames to the audio port
framecount	is the number of sample frames you want to read

Table 4-5 shows the output conversions that are applied when writing mono, stereo, and 4-channel data to stereo mode (default) and 4-channel mode hardware configurations.

Table 4-5 Output Conversions for alWriteFrames()

		Hardware Configuration	
Output	**Frame as Written into Port**	**Indigo, and Indigo^2 or Indy in Stereo Mode**	**Indigo^2 or Indy in 4-channel Mode**
Mono Port	(L_1)	(L_1, L_1)	(L_1, L_1, 0, 0)
Stereo Port	(L_1, R_1)	(L_1, R_1)	(L_1, R_1, 0, 0)
4-channel Port	(L_1, R_1, L_2, R_2)	(Clip (L_1 + L_2), Clip (R_1 + R_2))	(L_1, R_1, L_2, R_2)

Audio I/O Control

This section describes facilities for audio I/O control.

The AL programming model encompasses both the tangible elements of the audio system, such as the system itself, the audio boards and the devices on them and the conceptual elements of the software, such as the ALport from the application program to the audio hardware, methods of specifying the flow of data and control information, and synchronization. In the AL, both the tangible and the conceptual elements are represented as resources.

Resources are organized in a hierarchy. Data and control information flow through the hierarchy from the application to the hardware and vice versa.

At the top of the hierarchy is a port. The port is the application's handle to the audio hardware. Ports transport data and control information to and from the application. The port can be thought of as straddling the application level and the driver level. Information is routed from the application through the port to the driver level and hardware level resources along a software-configurable route.

Software resources called connections provide point-to-point routing from a port to a device or from device to device. Multiple connections to a single device are possible.

Within the driver level is the system and subsystem(s). Subsystems contain devices, which are the audio hardware's handle to the system. Devices transport data and control information to and from the machine.

At the hardware level, device resources are the central processing point for both data and control information. A software resource called an interface provides a mapping to the external jacks.

A master clock is a resource that generates timing information. A master clock provides a baseline rate, such as the output of a crystal, a video signal, or the timing information encoded in a digital audio stream.

The reference rate of a master clock can be transformed by a software resource called a clock generator. Timing information flows from a master clock to a device as transformed by a clock generator.

Resources have attributes that are represented by parameters. Because parameters are resource-specific, only certain resources recognize certain parameters. The resource hierarchy permits parameter inheritance, whereby a parameter is passed down the hierarchy until it reaches a resource that recognizes it. In this way, control information flows through the resource hierarchy until it reaches a resource that is capable of implementing that particular control. Control requests can be sent to a particular resource or to a class of resources.

Audio Parameters

The following parameters and the resources they are associated with are supported by the AL.

All resources support this set of universal parameters

- AL_TYPE
- AL_NAME
- AL_PARAMS
- AL_NO_OP

System (AL_SYSTEM) Parameters are

- AL_DEFAULT_INPUT
- AL_DEFAULT_OUTPUT
- AL_MAX_PORTS
- AL_UNUSED_PORTS
- AL_MAX_SETSIZE

Device Parameters are

- AL_INTERFACE
- AL_CLOCK_GEN
- AL_CHANNELS
- AL_PORT_COUNT
- AL_MAX_SETSIZE

Clock Generator Parameters are

- AL_RATE
- AL_RATE_FRACTION
- AL_MASTER_CLOCK
- AL_VIDEO_LOCK

Master Clock Parameters are

- AL_RATE
- AL_CLOCK_TYPE

Interface Parameters are

- AL_GAIN
- AL_GAIN_REF
- AL_WORDSIZE
- AL_CHANNELS

Table 4-6 lists universal parameters which apply to all resources. The parameter token, its type, supported operations, and a description are listed for each parameter.

Table 4-6 Universal Parameters

Parameter	Type	Operations	Description
AL_TYPE	32-bit integer	Get	All resources have types. Some types are subtypes of another; in this case, parameters which apply to the supertype also apply to the subtype. See the alIsSubtype(3dm) manual page for more information.
AL_NAME	Character string	Get	Each resource on each system has a unique name. This name is typically used internally in applications, for example, for saving parameters to files. AL_NAME can be passed into alGetResourceByName(3dm) to retrieve the original resource.
AL_LABEL	Character string	Get, Set	This is a human-readable description of the resource. Use this, for example, for resource selection in menus, or for otherwise displaying resources to users. It is unique on a given system. I t is also user-configurable. Attempts to set its value to equal another existing label will be rejected.
AL_NO_OP	None	Get, Set	The ALpv structure associated with this parameter is ignored. This effectively allows a program to "comment out" individual parameter/value pairs in a larger list without restructuring the list.
AL_PARAMS	Set of enumerated types	Query	Returns the set of parameters supported by this resource.

Techniques for Working With Audio Parameters

Control parameters change the hardware behavior of the audio system in real time, as opposed to changing the expected format of on-disk data the way an ALconfig does.

The AL provides its own parameter-value structure, called an ALpv, for working with AL parameters. The ALpv type definition is:

```
typedef struct {
    int     param;          /* parameter */
    ALvalue value;          /* value */
    short   sizeIn;         /* size in -- 1st dimension */
    short   size2In;        /* size out -- 2nd dimension */
    short   sizeOut;        /* size out */
    short   size2Out;       /* size out -- 2nd dimension */
} ALpv;
```

The ALvalue structure is a union of three types:

```
typedef union {
    int         i;          /* 32-bit integer values */
    long long   ll;         /* 64-bit integer and fixed-point values */
    void*       ptr;        /* pointer values */
} ALvalue;
```

The AL supports various data types, each of which uses a specific field of ALvalue:

32-bit integer	Uses the *value.i* field
64-bit integer	Uses the *value.ll* field
resource	Uses the *value.i* field
enumeration	Uses the *value.i* field
fixed-point	Uses the *value.ll* field **Note:** The convenience functions alFixedToDouble(3dm) and alDoubleToFixed(3dm) can be used to convert between double-precision floating-point and 64-bit fixed-point.
vectors, sets, matrices, strings	The value.ptr field must point to the actual value.

There are also required size fields for these data types.

The AL routines for working with parameters are:

alQueryValues()	determines possible hardware parameters
alGetParams()	gets current settings of hardware parameters
alSetParams()	sets hardware parameters
alGetParamInfo()	gets defaults and bounds of hardware parameters

Some methods for using these routines are:

- If you need a complete list of all available parameters on a particular resource, call **alQueryValues()**.
- If you are interested only in certain values, create an array that is twice the size of the number of parameters you are querying, and fill the even locations with the parameters of interest, then:
  - call **alGetParams()** to determine the current settings of the state variables.
  - fill in the even entries with the values that you want to change, and then call **alSetParams()** to change the values.
- Some parameters might exist but might not allow the needed settings, so call **alGetParamInfo()** to get the parameter bounds and check to be sure that the values you want to use exist.

Chapter 5

Digital Media Buffers

This chapter describes the Digital Media buffers (DMbuffers) real-time visual data transport facility, which is currently supported on O2 workstations. Time-sensitive visual data is moved through system memory using DMbuffers. This method provides a unified approach to facilitating data flow between live video devices. DMbuffers provide a pipelined I/O model—the application can direct the flow of multiple images on multiple paths simultaneously. This chapter builds on video I/O concepts presented in Chapter 4, "Digital Media I/O."

About Digital Media Buffers

DMbuffers feature

- an operating system-generic live image data storage and transport facility
- a library-transparent interface that allows real-time data interchange with compression/decompression (dmIC) and OpenGL
- an application-centered nonblocking I/O method that provides event processing by means of a single **select()** loop
- a software-configurable memory allocation method that provides the dedicated memory and throughput resources to suit visual data transport requirements using general-purpose system memory

Because the bandwidth of digital video signals far exceeds the bandwidth of typical data storage devices and communications links, video is usually compressed when storing to disk or sending over a network. This software interface supports specialized multimedia hardware on O2 workstations which further boosts visual data processing performance.

DMbuffers and DMbufferpools provide the method for allowing your application to allocate and use general-purpose system memory for transporting visual data.

A DMbufferpool is a custom storage facility created by the application. Video I/O devices, compression devices and algorithms, and graphics devices have direct access to

this storage on a compartmental basis called a DMbuffer. The application can define what this compartment represents, but in general, a DMbuffer represents an image. An image can be in the form of raw uncompressed pixel data for a video field or frame, or in the form of a picture's worth of compressed data (JPEG, MPEG, or so on).

When creating the storage facility, the application describes its data requirements in a DMparams structure. The application also queries every device that plans on using the facility about their needs, and obtains from them a DMparams list describing their requirements for allocating and apportioning memory. Using this information, the storage facility(DMbufferpool) is configured with the proper number, size, and type compartments (DMbuffers) sufficient for containing the data and its associated bookkeeping information.

Because a DMbuffer is created according to all necessary data and device requirements, and with universal content descriptors, every device that needs access to it can tell what's stored inside and can share and use the contents without making modifications.

To transfer the contents of a compartment, it is much easier and faster to transport only a pointer to the storage location rather than moving the actual contents. DMbuffers are really placeholders which contain only the pointers to the actual data; the data itself is stored elsewhere. When data is transferred, only pointers describing the data are passed, not the actual data. This means that the process doesn't have to handle or copy the data in order to transport it.

DMbufferpools contain a fixed number of DMbuffers, all the same size. It's up to you to decide how many DMbuffers to use, but, in general, three DMbuffers are sufficient for most applications: one to receive data, one to send data, and one available in case of contention between the sending and receiving processes.

DMbuffers give the application access to memory which is

- reserved at start-up and guaranteed for the life of the application
- not visited by the page daemon and therefore can't be swapped

When an image needs to be transported, a DMbuffer is allocated from the pool. Once allocated, a DMbuffer can go to a video I/O device, an image converter, or graphics I/O in any order, regardless of the order of allocation. Multiple DMbuffers with independent agendas can coexist. When the transfer is complete, a DMbuffer can be returned to the pool for reuse or retained for future use. It is not necessary to keep the same DMbuffer waiting to complete processing before accepting more input because another DMbuffer can be allocated from the pool.

Compared with the traditional VLbuffers, DMbuffers

- use the same event mechanism to deliver input and output buffer events to the application, rather than using different file descriptors for input and output
- deliver data buffers ordered with VL events on the video input path
- allow buffers to be sent to video multiple times, held by the application, or reordered

Currently, the DM buffers method is supported by:

- Image Converter Library (dmIC)
- Video Library
- Movie Library
- OpenGL

DMbuffers have the following attributes:

- Type
- Size
- MSC/UST

DMbuffer Live Data Transport Paths

This section presents the framework for eight types of live data transport paths that can be realized using DMbuffers:

- Memory to Video
- Video to Memory
- Memory to Image Converter
- Image Converter to Memory
- Memory to Movie File
- Movie File to Memory
- Memory to OpenGL
- OpenGL to Memory

Figure 5-1 shows the eight live data transport paths that can use DMbuffers.

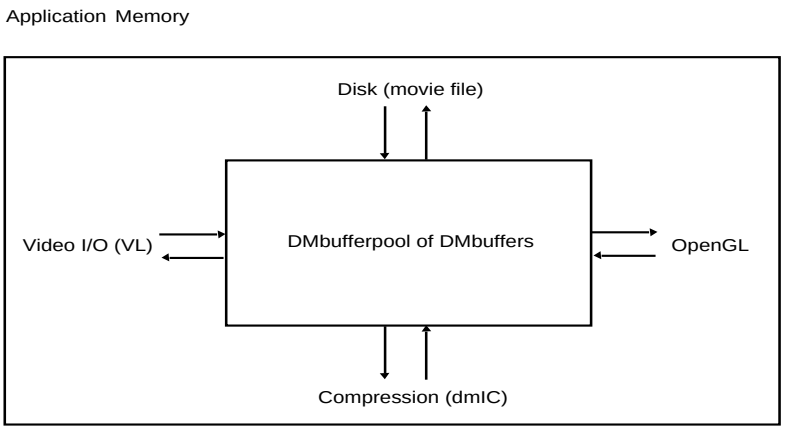


Figure 5-1 DMbuffer Live Data Transport Paths

Each path can be used as a stage in a more complex path. For example:

video-to-memory—>memory-to-dmIC—>dmIC-to-memory—>memory-to-movie file

are the 4 paths involved in recording compressed live video to disk.

In general, the calls outlined in this section simply move data around; they do not directly encode, decode, or otherwise process the images. This means that data is typically passed untouched along these pathways.

Note: The outlines presented in this section are intended to provide a sketch of the calls necessary for each path; the sequence of the calls may vary within each step and the actual coding may involve more routines. See the reference pages for details about using these routines.

Memory to Video

This section presents a basic sketch for memory to video output.

1. Open video output by calling:

   vlOpenVideo(3dm), to open the video server

   vlGetNode(3dm), to get the video drain node (VL_DRN, VL_VIDEO)

   vlGetNode(3dm), to get the memory source node (VL_SRC, VL_MEM)

2. Create a DMbufferpool for output of video fields or frames by calling:

   dmBufferSetPoolDefaults(3dm), to initialize the DMparams list

   vlDMPoolGetParams(3dm), to determine the video I/O device requirements

   dmBufferCreatePool(3dm), to create a pool using list modified by previous query

3. Associate the DMbufferpool with this path by calling:

   vlDMPoolRegister(3dm)

4. Prepare the pool to use **select()** to be notified of a free DMbuffer by calling:

   vlGetTransferSize(3dm), to get the size of one field or frame in this path

   dmBufferSetPoolSelectSize(3dm), to set the memory available threshold for waking from select

   dmBufferGetPoolFD(3dm), to get the input file descriptor for the buffer pool

5. Get a video output path file descriptor (for notification of errors/drops) by calling:

   vlGetFD(3dm)

6. Start the video flow by calling:

   vlBeginTransfer(3dm)

7. Get notification of a free DMbuffer in the output pool by calling:

   select(2)

8. Fill the DMbuffer with pixel data from the desired DMbuffer path

9. Enqueue the DMbuffer for video output by calling:

   vlDMBufferSend(3dm)

Video to Memory

This section presents a basic sketch for video input to memory using DMbuffers:

1. Open video input by calling:

   vlOpenVideo(3dm), to open the video server

   vlGetNode(3dm), to get the video source node (VL_SRC, VL_VIDEO)

   vlGetNode(3dm), to get the memory drain node (VL_DRN, VL_MEM)

2. Create a DMbufferpool for input of video fields or frames by calling:

   dmBufferSetPoolDefaults(3dm), to initialize the DMparams list

   vlDMPoolGetParams(3dm), to determine the video I/O device requirements

   dmBufferCreatePool(3dm), to create a pool with a list modified by previous query

3. Associate the DMbufferpool with this path by calling:

   vlDMPoolRegister(3dm)

4. Get a video input file descriptor (for use with **select(2)**) by calling:

   vlPathGetFD(3dm)

5. Start the video flow by calling:

   vlSelectEvents(3dm), to select events, namely *VLTransferCompleteMask*, and *VLSequenceLostMask* at a minimum, and others that the process might require

   vlBeginTransfer(3dm), to initiate video transfer

6. Get notification of each new field or frame by calling:

   select(2), to wait for an event

7. Get a new field or frame in the form of a DMbuffer by calling:

   vlEventRecv(3dm), to dequeue the next VLEvent

   vlEventToDMBuffer(3dm), to convert the VLEvent into a DMbuffer

Memory to Image Converter

1. Find the appropriate image converter by calling:

   dmICGetNum(3dm), to get the number of image converters installed in the system

   dmICGetDescription(3dm), to get the description of a given converter, then search them all (using the total returned) to find the index of the one that performs the desired conversion

2. Create the image converter context/instance by calling:

   dmICCreate(3dm)

3. Configure the image converter by calling:

   dmSetImageDefaults(3dm), to initialize the DMparams list with image defaults

   dmICSetSrcParams(3dm), to configure the source (input) image parameters

   dmICSetDstParams(3dm), to configure the destination (output) image parameters

   dmICSetConvParams(3dm), to configure the conversion algorithm (quality, etc.)

4. Create a DMbufferpool for the image converter input by calling:

   dmBufferSetPoolDefaults(3dm), to initialize the DMparams list

   dmICGetSrcPoolParams(3dm), to modify the list to reflect the image converter's buffering requirements (other DMbuffer paths using this pool must also be queried for their requirements before creating the buffer pool)

   dmBufferCreatePool(3dm), to create a buffer pool with the required attributes

5. Prepare the pool to use **select()** to be notified of a free DMbuffer by calling:

   dmBufferSetPoolSelectSize(3dm), to set the memory available threshold for waking from select

   dmBufferGetPoolFD(3dm), to get the buffer pool file descriptor

6. Get notification of a free DMbuffer in the input pool by calling:

   select(2), to wait for a free DMbuffer

   dmBufferAllocate(3dm), to allocate a DMbuffer

7. Fill the DMbuffer with pixel data (if encoding) or bits (if decoding) by calling:

   dmBufferMapData(3dm), to get a pointer to the data area of the DMbuffer

dmBufferSetSize(3dm), to set the image size, which is possibly obtained from one of: **dmImageFrameSize(3dm)**, **vlGetTransferSize(3dm)**, **mvGetTrackDataInfo(3dm)**, or **mvGetTrackDataFieldInfo(3dm)**

dmBufferSetUSTMSCpair(3dm), to set the sequence number of this image

8. Enqueue the DMbuffer for input to the converter by calling:

   dmICSend(3dm)

Image Converter to Memory

1. Find the appropriate image converter by calling:

   dmICGetNum(3dm), to get the number of image converters installed in the system

   dmICGetDescription(3dm), to get the description of a given converter, then search them all (using the total returned) to find the index of the one that performs the desired conversion

2. Create the image converter context/instance by calling:

   dmICCreate(3dm)

3. Configure the image converter by calling:

   dmSetImageDefaults(3dm), to initialize the DMparams list with image defaults

   dmICSetSrcParams(3dm), to configure the source (input) image parameters

   dmICSetDstParams(3dm), to configure the destination (output) image parameters

   dmICSetConvParams(3dm), to configure the conversion algorithm (quality, etc.)

4. Create a DMbufferpool for the image converter output by calling:

   dmBufferSetPoolDefaults(3dm), to initialize the DMparams list

   dmICGetDstPoolParams(3dm), to modify the list to reflect the destination parameters (other DMbuffer paths using this pool must also be queried for their requirements before creating the buffer pool)

   dmBufferCreatePool(3dm), to create a pool with the required attributes

   dmICSetDstPool(3dm), to attach this pool to the image converter

5. Get a converter file descriptor for notification of new output from the image converter by calling:

   dmICGetDstQueueFD(3dm)

6. Get notification of a new DMbuffer of converted data by calling:

   select(2)

7. Get the new DMbuffer by calling:

   dmICReceive(3dm), to dequeue the converted image from the converter

   dmBufferGetUSTMSCpair(3dm), to get the sequence number of this image

   dmBufferGetSize(3dm), to get the size of the image data in bytes

   dmBufferMapData(3dm), to get a pointer to the image data

Memory to Movie File

1. Open or create a movie file with an image track using one of the following methods:
   - Create a new movie file by calling:

     mvCreateFile(3dm), to get a handle to the new movie file

     mvAddTrack(3dm), to insert an empty image track into the movie file
   - Open an existing movie file by calling:

     mvOpenFile(3dm), to get a handle to the existing movie

     mvFindTrackByMedium(3dm), to find the movie's image (DM_IMAGE) track

     mvGetTrackLength(3dm), to find the playing time of the current track
2. Determine the image size by calling:

   dmBufferMapData(3dm), to get a pointer to the image

   dmBufferGetSize(3dm), to get the size of the image in bytes

   Note: Perform these operations twice for field-based data (once per field)
3. Save the image to the movie file using one of the following methods (pass the pointer and the size returned in the previous step to the function):
   - Write the data from a single DMbuffer as a frame by calling:

     mvInsertTrackData(3dm)
   - Write the data from 2 DMbuffers (one per field) by calling:

     mvInsertTrackDataFields(3dm)

Movie File to Memory

1. Open a movie file for reading by calling:

   mvOpenFile(3dm)

   mvFindTrackByMedium(3dm), to find the DM_IMAGE track

   mvGetImageWidth(3dm), to get the width of the image

   mvGetImageHeight(3dm), to get the height of the image

   mvGetTrackMaxFieldSize(3dm), to get the size (in bytes) of the largest compressed field, which can be used for configuring DMbuffer size when creating a buffer pool

2. Get a free DMbuffer by calling:

   dmBufferAllocate(3dm), to allocate a DMbuffer

   dmBufferMapData(3dm), to get a pointer to the data area of the DMbuffer

3. Get the movie file *index* for the desired image by calling:

   mvGetTrackDataIndexAtTime(3dm)

4. Import the compressed image data into memory using one of the following methods:

   - If image data is stored as a frame, call:

     mvGetTrackDataInfo(3dm), to get the size of the compressed data

     mvReadTrackData(3dm), to copy the compressed data from the movie file directly into the DMbuffer

   - If image data is stored as fields, call:

     mvGetTrackDataFieldInfo(3dm), to get the sizes of the compressed fields

     mvReadTrackDataFields(3dm), to copy the 2 compressed fields directly into 2 DMbuffers

5. Set the DMbuffer size(s) by calling:

   dmBufferSetSize(3dm)

Memory to OpenGL

This section presents a sketch of transporting video images to a window on the graphics display.

1. Get a pointer to the image data in a DMbuffer by calling:

   dmBufferMapData(3dm)

2. Display image in a graphics context (window or offscreen buffer) by calling:

   glPixelZoom(*zoomx*, *-zoomy*), to set a negative y zoom indicating top-to-bottom orientation (needed for video)

   glDrawPixels(3G), using

   GL_YCRCB_422_SGIX, (to convert from YCrCb to RGB)

   GL_INTERLACE_SGIX (as supported—an OpenGL interlacing extension for sending two noninterlaced fields of video to an interlaced graphics display by automatically interlacing alternate lines from two video fields)

OpenGL to Memory

The OpenGL can render to offscreen memory (pbuffer) which can be accessed directly as a DMbuffer.

1. Create an OpenGL/X pbuffer by calling:

   glXCreateGLXPbufferSGIX()

2. Relate the two buffers by calling:

   glXMakeCurrent()

   glXAssociateDMPbufferSGIX() with *pbuffer*, *dmbuffer*

A Detailed Look at Recording Compressed Live Video to Disk

This section explains how to get video into memory using DMbuffers. Refer to the sample program *dmplay.dmic* in Video Capture with Compression in Chapter 4 of "Digital Media Programmer's Examples" for a demonstration of this method.

Figure 5-2 shows the video compression path.

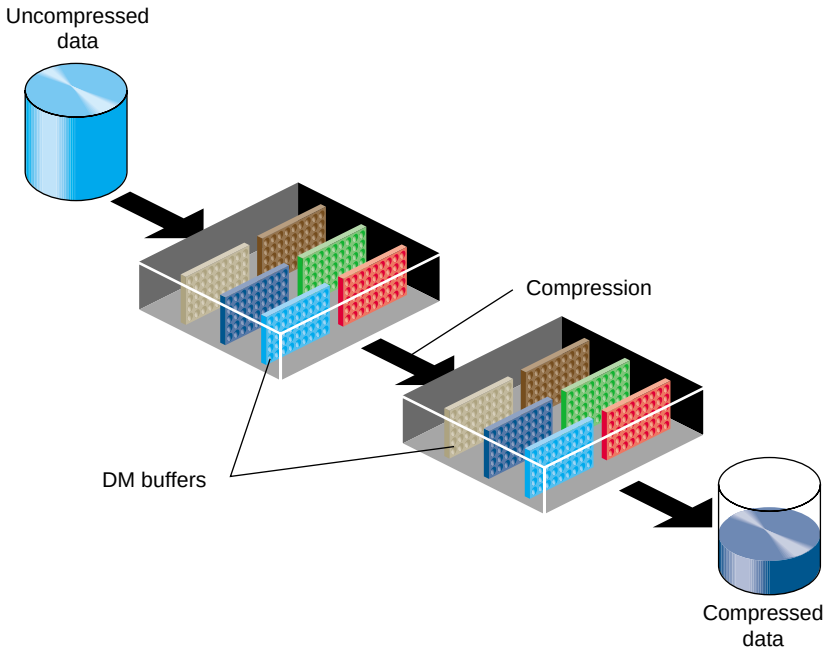


Figure 5-2 Compression Path Using DMbuffers

The first step in video I/O is getting the video data into a DMbuffer. Once you have a buffer of video data, you can send it to dmIC to compress it, send it to graphics for display, applying effects along the way, or write it to a movie file. You can create separate threads for the video input and image conversion to ensure that the processing remains event-driven. In addition, the DMbuffers method provides the capability of performing multiple image conversion operations simultaneously within the same application with no performance penalty.

Getting video into memory requires

- a video source (input) node
- a memory drain (output) node
- a path that connects the two nodes
- an video input DMbufferpool associated with the memory drain node

This method of compressing video into memory also uses

- an input DMbufferpool on the input to the DM image converter (this is the DM buffer pool associated with the memory drain node)
- an output DMbufferpool to contain the output from the converter

The first three items are the same regardless of whether you use a DMbuffer or a VLbuffer to transport visual data, and are explained in Chapter 4.

The basic model for video input to memory using DMbuffers is:

1. Initialize the DMparams list by calling **dmBufferSetPoolDefaults(3dm).**
2. Query each device about its requirements for memory allocation:
   - Determine video I/O device requirements using **vlDMPoolGetParams(3dm)**
   - Determine image converter requirements using **dmICGetSrcPoolParams(3dm)** and **dmICGetDstPoolParams(3dm)** as described in "3. Creating Data Buffers Using the Image Conversion API" in Chapter 6.
3. Create buffer pools using the DMParams lists returned by the previous queries, by calling **dmBufferCreatePool(3dm)**.
4. Register the input buffer pool as the video input destination (drain) by calling **vlDMPoolRegister(3dm)**.

This application can be divided into the following major areas (each of which is further broken down into detailed steps):

- Video initialization, which involves
  - opening a connection to the video server (**vlOpenVideo()**)
  - if necessary, determining which device the application will use (**vlGetDevice()**, **vlGetDeviceList()**)
  - getting input information, such as the size of the video images

Note: Image size (in pixels) can be determined from the video timing and packing. Use **dmImageGetSize()** to determine the size of an image in bytes. The *dmplay.dmic* program first sets the capture mode to noninterlaced to get the size of each field, and then resets the capture mode to interlaced before initiating video transfer.

- Creating and setting up a video data transfer path, which involves
  - specifying nodes on the data path (**vlGetNode()**)
  - creating a path connecting the specified nodes (**vlCreatePath()**)
  - setting up the hardware for the path (**vlSetupPaths()**)
- Getting and setting controls for video data transfer, which involves
  - getting input parameters (controls) for the nodes on the path (**vlGetControl()**)
  - setting output parameters (controls) for the nodes on the path (**vlSetControl()**)
- Creating and registering DMbufferpools to handle video data, which involves
  - initializing the DMparams list by calling **dmBufferSetPoolDefaults()**
  - querying each device about its requirements for memory allocation:

    Determine video I/O device requirements using **vlDMPoolGetParams().**

    Determine image converter requirements using **dmICGetSrcPoolParams()** and **dmICGetDstPoolParams()** as described in "3. Creating Data Buffers Using the Image Conversion API" in Chapter 6.
  - creating buffer pools using the DMParams lists modified by the previous queries, by calling **dmBufferCreatePool()**
  - registering the input buffer pool as the video input destination (drain) by calling **vlDMPoolRegister()**
- Transfer initiation, which involves:
  - setting the video transfer mode using VL_CAP_TYPE
  - specifying path-related events to be captured by calling **vlSelectEvents()**
  - obtaining a VL file descriptor to wait upon by calling **vlGetFD()**
  - initiating the video transfer by calling **vlBeginTransfer()**

- Main loop, which involves
  - waiting for video events using **select()**
  - dequeueing the next video event using **vlEventReceive()**
  - obtaining the video field or frame as a DMbuffer for further transport to the appropriate destination (dmIC, OpenGL, etc.) using **vlEventtoDMbuffer()**
- Cleanup, which involves
  - freeing the DMBuffers and destroying the DMbufferpools

Chapter 6

Digital Media Data Conversion

The file input/output routines of Digital Media Libraries, such as the Audio File Library and the Movie Library, use digital media converters to provide automatic data format conversion. This chapter describes how to use the digital media conversion libraries to create and use these converters in your application.

About Digital Media Data Conversion

Digital media *data conversion* includes compression and decompression based on industry standards, such as JPEG and MPEG-1. It also encompasses transforming image color spaces, changing audio sample rates, and other basic data modifications. There are two generalized *conversion libraries* to effect digital media data conversions: the Image Conversion Library for video data, and the Audio Conversion Library for audio data. Applications can access these libraries to convert streams of digital media data. (Because they incur some overhead, the conversion libraries usually are not used for small amounts of data such as single still images.)

The conversion library APIs provide an interface to *codecs* (*co*mpressor-*dec*ompressors) such as JPEG or MPEG-1, to the Color Space Library and to other transformation libraries. The codecs and transformation libraries do the actual data conversions. Codecs may be either software modules or hardware devices with software interfaces. The hardware codecs are faster, but the software codecs may offer more options. If your application chooses a software codec in lieu of a hardware one, you may want to have your application notify the user of this fact. For more information about specific codecs see Chapter 2, “Digital Media Essentials.”

The digital media conversion libraries offer a number of benefits:

- They permit memory-to-memory data format conversions. These are more flexible than conversions between memory and I/O devices because the converted data is retained in memory for reuse.
- They integrate the use of software codecs, hardware codecs, and transformation libraries.

- They enable applications to perform conversions by merely specifying three sets of parameters:
  - one that describes the source data format
  - one that describes the destination data format
  - one that gives codec-specific settings
- They enable an application to use a number of codecs simultaneously.
- They have a modular design that allows codecs to be loaded dynamically as the application needs them.
- They allow applications to obtain a codec's parameters and default values, and to change these values appropriately.

Using Digital Media Converters

A digital media *converter* is the *encoder* (compressor) or *decoder* (decompressor) of a codec. Converters are grouped by the conversion library that uses them. As you might expect, the Image Conversion Library uses the converters for image data, while the Audio Conversion Library uses the converters for audio data. Despite these groupings, converters are used in similar ways.

An application creates a *converter instance* by using the API of either of the conversion libraries. A converter instance includes a converter, any necessary transformation libraries, an input and an output buffer, and state information in the form of parameters that describe the input data, output data, and conversion process. A number of converter instances can use the same converter simultaneously. A converter instance that includes an encoder takes uncompressed data, applies the transformation libraries needed to make the data usable to the encoder, and then uses the encoder to produce the converted data. A converter instance with a decoder reverses this process.

A converter instance can be viewed as a pipeline. On one end of the pipeline is uncompressed data in a specified format. On the other end is data in the format native to the converter. The uncompressed data is the pipeline's input if the converter is a encoder, otherwise it is the output. The pipeline processing is done by the transformation libraries and the converter. When multiple transformation libraries are needed, the conversion libraries make sure that they are used in an order that best maintains the quality of the data.

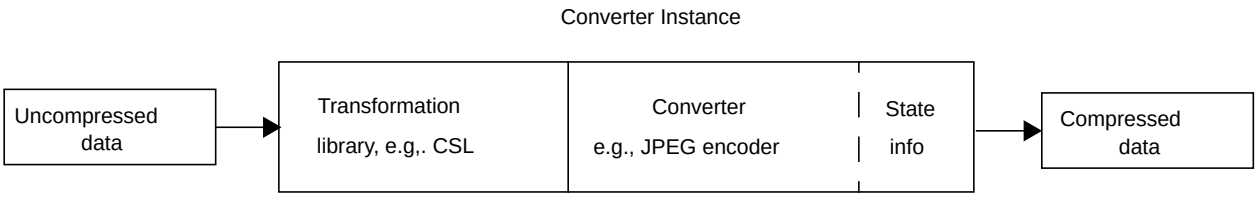


Figure 6-1 Conversion Pipeline

To perform a data conversion your application follows these five steps:

1. Create a converter instance.
2. Configure the converter instance.
3. Create data buffers for the converter instance (image conversion only).
4. Convert the data using the converter instance.
5. Destroy the converter instance.

To use the digital media conversion libraries to create a converter instance, you must link your application with *libdmedia.so*.

Image Data Conversion

This section describes the Digital Media Image Conversion Library and its converters. What follows is a discussion of how to use the Image Conversion API to execute the five steps of an image data conversion.

Digital Media Image Conversion Library

The API of the Digital Media Image Conversion Library provides an interface for memory-to-memory image conversion that is independent of the algorithm. The Image Conversion Library, see also dmic(4), lets you create an image converter instance based on a codec's encoder or decoder. Table 6-1 lists commonly installed codecs that can be accessed through the Image Conversion API. As shown in the table, these codecs usually have both types of converters, and may be performed by hardware on properly equipped systems. The columns DM_IC_ENGINE Value and DM_IC_ID Value contain

identification values that are used with **dmICGetDescription()** as described in the section "1. Creating a Converter Instance Using the Image Conversion API."

Table 6-1 Digital Media Image Converters

Codec Name	DM_IC_ENGINE Values	DM_IC_ID Value
Apple QuickTime Animation	"The Apple Animation compressor" "The Apple Animation decompressor"	'rle '
Cinepak	"The Cinepak compressor" "The Cinepak decompressor"	'cvid'
Intel Indeo	"The Indeo Video compressor" "The Indeo Video decompressor"	'IV32'
H.261	"The H261 software encoder" "The H261 software decoder"	'h261'
JPEG	"Software JPEG Encoder" "Software JPEG Decoder" "Vice"	'jpeg'
MPEG-1 Video	"Vice"	'mpeg'
Motion Video Compressor 1	"The MVC1 compressor" "The MVC1 decompressor"	'mvc1'
Motion Video Compressor 2	"The MVC2 compressor" "The MVC2 decompressor"	'mvc2'
8-bit Run Length Encode	"The RLE compressor" "The RLE decompressor"	'rle1'
24-bit Run Length Encode	"The RLE24 compressor" "The RLE24 decompressor"	'rle2'
Apple QuickTime Video	"The Apple Video compressor" "The Apple Video decompressor"	'rpza'

In addition to the operations done by the codecs, an image converter instance may also perform transformations involving

- color space
- size
- clipping

- orientation
- interlacing
- image rate

For some of these, the image converter instance calls the Color Space Library during the conversion process. This library is discussed in "The Digital Media Color Space Library." Also shown is how the Image Conversion Library provides access to it independently of a converter instance.

As the next five sections demonstrate, The Image Conversion Library enables you to effectively use codecs and the Color Space Library by following the five steps listed in "Using Digital Media Converters." To use the Image Conversion API, you must include these header files:

```
#include <dmedia/dmedia.h>
#include <dmedia/dm_imageconvert.h>
```

1. Creating a Converter Instance Using the Image Conversion API

The Image Conversion Library allows you to choose among the image converters in the system. To find a converter and create an instance of it, the Image Conversion Library enables you to do the following:

- Get the number of image converters present in the system with **dmICGetNum()**
- Get the description of a specific converter with **dmICGetDescription()**
- Choose a converter based on your application's conversion needs with **dmICChooseConverter()**
- Create a converter instance with **dmICCreate()**

Note: Many of the Image Conversion Library functions return a DMstatus value. The value is DM_SUCCESS if they succeed, DM_FAILURE if not. After a receiving a DM_FAILURE, your application can call the function **dmGetError()** or **dmGetErrorForPID()** to return an error message and error number. See "Digital Media Error Handling" in Chapter 3 for more information.

Use the function **dmICGetNum()** to find the number of image converters available:

```
int dmICGetNum ( void )
```

The function returns an integer that is the number of available image converters.

Use the function **dmICGetDescription()** to obtain the description of a converter:

```
DMstatus dmICGetDescription ( int i, DMparams *converterParams )
```

The parameter *i,* which is the index of the converter, is the key to the description returned. The converter's index is an integer between 0 and *n*-1 where *n* is the number of image converters returned by **dmICGetNum()**. The parameter *converterParams* points to a previously created DMparams data structure. If **dmICGetDescription()** is successful, the data structure contains six parameters whose values characterize the converter.

Tip: The application controls the creation and destruction of DMparams structures. See the upcoming section "Using the Image Conversion API" for examples.

To extract the converter's description from the structure pointed to by *converterParams*, you submit parameter keywords to the appropriate DMparams-querying functions. The left side of the following table lists the six parameter keywords and the relevant DMparams-querying functions. On the right side are possible values returned by the querying functions.

DM_IC_ID and **dmParamsGetInt()**	The codec's DM_IC_ID value, as shown in Table 6-1
DM_IC_ENGINE and **dmParamsGetString()**	The codec's DM_IC_ENGINE value, as shown in Table 6-1
DM_IC_VERSION and **dmParamsGetInt()**	The codec's version number, for example 1.
DM_IC_REVISION and **dmParamsGetInt()**	The codec's revision number, for example 2.
DM_IC_CODE_DIRECTION and **dmParamsGetEnum()**	DM_IC_CODE_DIRECTION_UNDEFINED DM_IC_CODE_DIRECTION_ENCODE DM_IC_CODE_DIRECTION_DECODE
DM_IC_SPEED and **dmParamsGetEnum()**	DM_IC_SPEED_UNDEFINED DM_IC_SPEED_REALTIME DM_IC_SPEED_NONREALTIME

Note: Although they are not shown here, DMparams-querying functions have corresponding setting functions. For example, **dmParamsSetEnum()** corresponds with **dmParamsGetEnum()**. However, some parameter values, such as those above, cannot be set by your application. See Chapter 3, "Digital Media Parameters," for a discussion of using DMparams.

Your application should check the value of DM_IC_CODE_DIRECTION to determine if the codec is an encoder or decoder. The contents of DM_IC_ID and DM_IC_ENGINE may not have enough information to make this decision. If the value of DM_IC_SPEED is DM_IC_SPEED_REALTIME, the codec is suitable for real-time encoding or decoding. Usually, this indicates a hardware codec.

Use the function **dmICChooseConverter()** to choose a converter based on parameter values:

```
int dmICChooseConverter ( DMparams *srcParams, DMparams *dstParams,
                          DMparams *convParams )
```

The function returns the index of the converter that matches the requirements specified by the DMparams structures pointed to by *srcParams*, *dstParams*, and *convParams*. For **dmICChooseConverter()** to succeed, either the DM_IC_CODE_DIRECTION and DM_IC_ID parameters must be set in *convParams*, or the DM_IMAGE_COMPRESSION parameter must be set in *srcParams* and *dstParams*.

This function opens the converter that matches the specified parameters and values. If **dmICChooseConverter()** is successful, *srcParams*, *dstParams*, and *convParams* can be used as inputs to **dmICSetSrcParams()**, **dmICSetDstParams()**, and **dmICSetConvParams()** respectively. (See "2. Configuring a Converter Instance Using the Image Conversion API.") The converter chosen is guaranteed to be optimal only for the given parameters. If the user changes any of the parameter values, your application may need to call **dmICChooseConverter()** again.

Use the function **dmICCreate()** to create an image converter instance:

```
DMstatus dmICCreate ( int i, DMimageconverter *converter )
```

This function opens the converter corresponding to *i*, an index returned by **dmICChooseConverter()**, and initializes *converter* (the address of a previously declared DMimageconverter variable) with the *handle* (the address of a pointer) of a converter instance. The handle is declared as follows:

```
typedef struct _DMimageconverter *DMimageconverter;
```

An application normally may open multiple converter instances based on one or more converters. However, some converters may limit the number of instances based on them. A converter is always used as part of a converter instance.

Tip: Because converters can be loaded dynamically, a converter's index can vary from system to system, or even from time to time on the same system. Before using an index, your application should use either **dmICGetDescription()** to check its validity, or **dmICChooseConverter()** to establish its value. Once determined, the value of the index will not change while an application is running.

The code that follows the comment "// 1. Creating a converter instance" in the section "Using the Image Conversion API" makes use of some of the above functions.

2. Configuring a Converter Instance Using the Image Conversion API

Once a converter instance has been created, it must be configured. The application does this by using DMparams data structures to specify a set of source parameters, a set of destination parameters, and, optionally, a set of conversion parameters. The *source parameters* describe the image data that is the converter instance's input. The input is uncompressed data if the converter is a compressor, otherwise it is in the format native to the converter. The *destination parameters* describe the converter instance's output. The output is in the format native to the converter if the converter is a compressor, otherwise it is uncompressed data. The *conversion parameters* fine tune settings that are specific to the converter.

The Image Conversion Library enables you to do the following:

- Get the source, destination, and conversion parameters, and their default settings, with **dmICGetDefaultSrcParams()**, **dmICGetDefaultDstParams()**, and **dmICGetDefaultConvParams()**
- Get the actual settings of the source, destination, and conversion parameters with **dmICGetSrcParams()**, **dmICGetDstParams()**, and **dmICGetConvParams()**
- Change the settings of the source, destination, and conversion parameters with **dmICSetSrcParams()**, **dmICSetDstParams()**, and **dmICSetConvParams()**

In each of the nine functions described in this section, the first parameter, *converter,* is a DMimageconverter previously initialized by **dmICCreate()**. The second parameter is a pointer to a DMparams structure.

Use the functions **dmICGetDefaultSrcParams()**, **dmICGetDefaultDstParams()**, and **dmICGetDefaultConvParams()** to get the default parameter settings:

```
DMstatus dmICGetDefaultSrcParams ( DMimageconverter converter,
                                   DMparams *srcParams )
```

```
DMstatus dmICGetDefaultDstParams ( DMimageconverter converter,
                                   DMparams *dstParams )

DMstatus dmICGetDefaultConvParams ( DMimageconverter converter,
                                    DMparams *convParams )
```

After each of these functions returns, the DMparams structure contains the parameters, along with their default values, that your application can set for the converter instance indicated by *converter* using the setting functions described in this section. Although the conversion parameters, pointed to by *convParams*, vary markedly from converter to converter, those shown below are supported by many. The left side of the table lists each conversion parameter and its DMparams-querying function. Possible values for the parameter are listed on the right. For a more complete discussion of retrieving parameter values, see Chapter 3, "Digital Media Parameters."

DM_IMAGE_BITRATE and **dmParamsGetInt()**	An integer value that indicates the rate, in bits per second, of a compressed video stream.
DM_IMAGE_QUALITY_SPATIAL and **dmParamsGetFloat()**	DM_IMAGE_QUALITY_NORMAL DM_IMAGE_QUALITY_LOSSLESS several others.
DM_IMAGE_QUALITY_TEMPORAL and **dmParamsGetFloat()**	DM_IMAGE_QUALITY_NORMAL DM_IMAGE_QUALITY_LOSSLESS several others.

Use the functions **dmICGetSrcParams()**, **dmICGetDstParams()**, and **dmICGetConvParams()** to get the actual parameter settings:

```
DMstatus dmICGetSrcParams ( DMimageconverter converter,
                            DMparams *srcParams )

DMstatus dmICGetDstParams ( DMimageconverter converter,
                            DMparams *dstParams )

DMstatus dmICGetConvParams ( DMimageconverter converter,
                             DMparams *convParams )
```

After these functions return successfully, the DMparams structures contain the parameters and values that describe the input image, the output image, and the conversion settings for the data in the buffer obtained by the last successful call to **dmICReceive()**. The parameters are not defined prior to a successful call to **dmICReceive()**. See the section "4. Converting Data Using the Image Conversion API" for more information on **dmICReceive()**.

Use the functions **dmICSetSrcParams()**, **dmICSetDstParams()**, and **dmICSetConvParams()** to change the parameter values:

```
DMstatus dmICSetSrcParams ( DMimageconverter converter,
                            DMparams *srcParams )

DMstatus dmICSetDstParams ( DMimageconverter converter,
                            DMparams *dstParams )

DMstatus dmICSetConvParams ( DMimageconverter converter,
                             DMparams *convParams )
```

The parameters and values to be used are specified in the DMparams structures pointed to by *srcParams*, *dstParams*, and *convParams*. The settings describe the input image, the output image, and the conversion settings for the data in the buffer that will be sent by the next **dmICSend()** operation. See the section "4. Converting Data Using the Image Conversion API" for more information on **dmICSend()**.

The left side of the following table lists the image conversion parameters that *must* be specified in the source and destination DMparams structures. Also listed are the appropriate DMparams-setting functions. Possible values of the parameters are listed on the right. See *dm_image.h* for all the image parameters, and Chapter 3, "Digital Media Parameters" for a complete discussion of them.

DM_IMAGE_WIDTH and **dmParamsSetInt()**	Image pixel width, e.g. 640.
DM_IMAGE_HEIGHT and **dmParamsSetInt()**	Image pixel height, e.g. 480.
DM_IMAGE_PACKING and **dmParamsSetEnum()**	DM_IMAGE_PACKING_RGBX many others.
DM_IMAGE_ORIENTATION and **dmParamsSetEnum()**	DM_IMAGE_TOP_TO_BOTTOM DM_IMAGE_BOTTOM_TO_TOP

The code that follows the comment "// 2. Configuring a converter instance" in the section "Using the Image Conversion API" makes use of some of the above functions.

3. Creating Data Buffers Using the Image Conversion API

The Image Conversion Library uses a buffering system to transmit data between your application and the converter instance. Typically, a data buffer contains a single video image, either compressed or uncompressed. You use the Image Conversion Library to

create an source (input) *buffer pool* and an destination (output) buffer pool. Your application allocates individual buffers from the source buffer pool, and uses them to send data to the converter instance. The converter instance returns processed data to your application with buffers it creates from the destination buffer pool. The buffer pools, which are fixed in size, are created during an application's initialization. Before creating the buffer pools with **dmBufferCreatePool()**, your application must make sure they satisfy the requirements of all the libraries that are going to use them. This last point is explained more fully below with the functions **dmICGetSrcPoolParams()** and **dmICGetDstPoolParams()**.

The Image Conversion Library allows you to determine the buffering requirements of a converter instance prior to creating a buffer pool. To find a converter instance's buffering needs, the Image Conversion Library enables you to do the following:

- Get the source and destination pool requirements with **dmICGetSrcPoolParams()** and **dmICGetDstPoolParams()**
- Select the buffer pool from which to receive data with **dmICSetDstPool()**

Use the functions **dmICGetSrcPoolParams()** and **dmICGetDstPoolParams()** to determine a converter instance's buffering needs:

```
DMstatus dmICGetSrcPoolParams ( DMimageconverter converter,
                                DMparams *poolParams )

DMstatus dmICGetDstPoolParams ( DMimageconverter converter,
                                DMparams *poolParams )
```

The converter instance indicated by *converter* modifies the DMparams structure pointed to by *poolParams* to describe its buffering needs. Your application should also pass this *poolParams* to other libraries that may share the same buffer pool. For example, if the buffer pool must be shared with the Video Library or the Graphics Library, *poolParams* should be passed to **vlDMPoolGetParams()** or **dmBufferGetGLPoolParams()**. This function returns an error if it detects that requirements set in *poolParams* by another library conflict with those of the converter instance.

Use **dmICSetDstPool()** to choose the buffer pool for a converter instance's output:

```
DMstatus dmICSetDstPool ( DMimageconverter converter,
                          DMbufferpool *pool )
```

The variable *pool* points to the destination buffer pool from which the converter instance, *converter*, allocates output buffers. Your application is responsible for creating and disposing of *pool*. The function returns an error if the converter instance does not meet

the requirements of *pool*. This function must be called prior to **dmICSend()** or **dmICReceive()**, which are described in the next section.

The code that follows the comment "// 3. Creating data buffers for the converter instance" in the section "Using the Image Conversion API" makes use of some of the above functions.

4. Converting Data Using the Image Conversion API

During the actual conversion process, your application sends data buffers to a converter instance, and receives buffers of processed data from it. A converter instance has a source and destination queue to handle buffer traffic. Your application allocates, fills, and sends data buffers to the source queue. The instance attaches the buffers in the queue, that is prevents them from being deleted, until it can process the data. The destination queue, containing data buffers created by the instance from the destination buffer pool selected with **dmICSetDstPool()**, holds the processed data until your application is ready to receive it. The source and destination queues allow the converter instance to interact asynchronously with your application.

The Image Conversion Library enables your application to do the following:

- Send data to and receive it from a converter instance with **dmICSend()** and **dmICReceive()**
- Use a file descriptor to be notified of data arriving from a converter instance with **dmICGetDstQueueFD()**
- Determine the number of buffers that are ready to be received from the converter instance with **dmICGetDstQueueFilled()** and the number that are ready to be processed by the converter instance with **dmICGetSrcQueueFilled()**

Use the function **dmICSend()** to send data to a converter instance for conversion:

```
DMstatus dmICSend ( DMimageconverter converter, DMbuffer srcBuffer,
                    int numRefBuffers, DMbuffer *refBuffers )
```

This function adds the data buffer *srcBuffer* to the input queue of the converter instance *converter*. It is an asynchronous operation, and the conversion may not have taken place when **dmICSend()** returns. However, if it is no longer needed by your application, the buffer can be freed with **dmBufferFree()** immediately after **dmICSend()** returns. The buffer is attached by the converter instance until it can be used, and will not be deleted until the converter instance is finished with it. The integer *numRefBuffers* is the number

of reference buffers in an array pointed to by *refBuffers*. The converter instance uses the *reference buffers* to interpret subsequent buffers which are coded in terms of them.

The **dmICSend()** function returns errors if the queue is full, or if any of the parameters are invalid. If the requirements of the source, destination, and conversion parameters set previously cannot be met during the instance's processing, an error is returned on a subsequent call to **dmICReceive()**.

Use the function **dmICReceive()** to receive data from a converter instance:

```
DMstatus dmICReceive ( DMimageconverter converter,
                       DMbuffer *dstBuffer )
```

This function removes a completed data buffer, pointed to by *dstBuffer,* from the output queue of *converter*. The buffer is automatically attached to the caller. When your application no longer needs it, the buffer must be freed with **dmBufferFree()**. The function returns an error of DM_IC_Q_EMPTY if there are no buffers ready.

Use the function **dmICGetDstQueueFD()** to obtain a file descriptor associated with the data from a converter instance:

```
int dmICGetDstQueueFD ( DMimageconverter converter )
```

The function returns the file descriptor associated with the converter instance *converter*. Your application can use this file descriptor with **select()** or **poll()** to be notified when data is available to be retrieved with **dmICReceive()**. The code sample in "Using the Image Conversion API" shows an example of this technique.

Use **dmICGetDstQueueFilled()** to determine how many data buffers have been processed by a converter instance and are now available to your application. Use the function **dmICGetSrcQueueFilled()** to find how many data buffers have been sent to the converter instance, but not processed:

```
int dmICGetDstQueueFilled ( DMimageconverter converter )
int dmICGetSrcQueueFilled ( DMimageconverter converter )
```

The function **dmICGetDstQueueFilled()** returns an integer that is the number of buffers from *converter* that can be removed by **dmICReceive()**. The **dmICGetSrcQueueFilled()** function returns an integer that is the number of buffers that have been sent to *converter* with **dmICSend()** and have yet to be processed. Buffers that are being processed by the converter instance are not counted by either function.

The code following the comment "// 4. Converting the data using a converter instance" in the section "Using the Image Conversion API" uses some of the above functions.

5. Destroying a Converter Instance Using the Image Conversion API

The Image Conversion Library enables you to free a converter instance's resources with **dmICDestroy()**:

```
void dmICDestroy ( DMimageconverter converter )
```

This function closes the converter instance indicated by *converter*. The converter instance's internal storage and other resources are freed and *converter* is no longer valid.

The code that follows the comment "// 5. Destroying a converter instance" in the next section uses this function.

Using the Image Conversion API

What follows is a code sample that demonstrates the use of the Image Conversion API.

```
#include <stdio.h>
#include <unistd.h>
#include <dmedia/dmedia.h>
#include <dmedia/dm_image.h>
#include <dmedia/dm_buffer.h>
#include <dmedia/dm_imageconvert.h>

//
// main()
//
int main( int argc, char *argv[] )
{
    int               nNumConverters = 0;
    DMimageconverter  ImgCvt;
    DMparams          *pParams;
    DMbufferpool      InPool, OutPool;
    DMbuffer          InBuffer, OutBuffer;
    int               i;

// 1. Creating a converter instance

    // Get the number of converters
    nNumConverters = dmICGetNum();
    if (nNumConverters <= 0)  {
```

```
        fprintf( stderr, “ICGetNum() = (%d)\n”, nNumConverters );
        return( -1 );
    }

    // Open the first converter
    if (dmICCreate( 0, &ImgCvt ) != DM_SUCCESS)  {
        fprintf( stderr, “ICCreate() failed.\n” );
        return( -1 );
    }

    // Get description of first converter and print it
    dmParamsCreate( &pParams );
    if (dmICGetDescription( 0, pParams ) != DM_SUCCESS)  {
        fprintf( stderr, “GetDescription() Failed\n” );
        return( -1 );
    }
    PrintDescription( pParams );
    dmParamsDestroy( pParams );

// 2. Configuring a converter instance

    // Set SrcParams and print
    dmParamsCreate( &pParams );
    dmSetImageDefaults( pParams, 320, 240, DM_IMAGE_PACKING_RGBX );
    if (dmICSetSrcParams( ImgCvt, pParams ) != DM_SUCCESS)  {
        fprintf( stderr, “SetSrcParams() Failed\n” );
        return( -1 );
    }
    if (dmICGetSrcParams( ImgCvt, pParams ) != DM_SUCCESS)  {
        fprintf( stderr, “GetSrcParams() Failed\n” );
    }
    PrintImageParams( pParams, “Src” );
    dmParamsDestroy( pParams );

    // Set DstParams and print
    dmParamsCreate( &pParams );
    dmSetImageDefaults( pParams, 160, 120, DM_IMAGE_PACKING_RGBX );
    if (dmICSetDstParams( ImgCvt, pParams ) != DM_SUCCESS)  {
        fprintf( stderr, “SetDstParams() Failed\n” );
        return( -1 );
    }
    if (dmICGetDstParams( ImgCvt, pParams ) != DM_SUCCESS)  {
        fprintf( stderr, “GetDstParams() Failed\n” );
    }
    PrintImageParams( pParams, “Dst” );
```

```
    dmParamsDestroy( pParams );

// 3. Creating data buffers for the converter instance

    // Create Pool
    dmParamsCreate( &pParams );
    dmBufferSetPoolDefaults( pParams, 5, 1024, DM_TRUE, DM_FALSE );
    if (dmBufferCreatePool( pParams, &InPool ) != DM_SUCCESS)  {
        fprintf( stderr, "Create Input Pool Failed\n" );
        return( -1 );
    }
    if (dmBufferCreatePool( pParams, &OutPool ) != DM_SUCCESS)  {
        fprintf( stderr, "Create Output Pool Failed\n" );
        return( -1 );
    }
    dmParamsDestroy( pParams );

    if (dmICSetDstPool( ImgCvt, OutPool ) != DM_SUCCESS)  {
        fprintf( stderr, "SetDstPool() Failed\n" );
        return( -1 );
    }

// 4. Converting data using the converter instance

    // We loop twice to test whether the codec does a coredump
    // when no src/dst/conv params are sent.
    // A real application would not have this arbitrary cut off.
    for ( i = 0 ; i < 2; ++i )  {
        dmBufferAllocate( InPool, &InBuffer );

        if (dmICSend( ImgCvt, InBuffer, 0, NULL ) != DM_SUCCESS)  {
            fprintf( stderr, "ICSend() Failed\n" );
            return( -1 );
        }

        if (dmBufferFree( InBuffer ) != DM_SUCCESS)  {
            fprintf( stderr, "dmBufferFree() Failed\n" );
            return( -1 );
        }

        {
            int     FD;
            fd_set  fdset;

            FD = dmICGetDstQueueFD( ImgCvt );
```

```
            FD_ZERO( &fdset );
            FD_SET( FD, &fdset );
            fprintf( stderr, "Waiting on FD %d\n", FD );
            select( FD+1, &fdset, NULL, NULL, NULL );
        }
        if (dmICReceive( ImgCvt, &OutBuffer ) != DM_SUCCESS)  {
           fprintf( stderr, "ICReceive() Failed\n" );
           return( -1 );
        }
        // OutBuffer can now be sent anywhere a DMbuffer is accepted.
        // We just free it here.
        if (dmBufferFree( OutBuffer ) != DM_SUCCESS)  {
           fprintf( stderr, "dmBufferFree() Failed\n" );
           return( -1 );
        }
    }

// 5. Destroying a converter instance

    // Close converter
    dmICDestroy( ImgCvt );
    return( 0 );
}

//
// PrintDescription()
//   Prints the description obtained by dmICGetDescription().
//
void PrintDescription( DMparams *pParams )
{
    fprintf( stderr, "Name: %s\n",
             dmParamsGetString( pParams, DM_IC_ENGINE ) );
    switch( dmParamsGetEnum( pParams, DM_IC_SPEED ) )  {
        case DM_IC_SPEED_UNDEFINED:
            fprintf( stderr, "Speed Undefined\n" );
            break;
        case DM_IC_SPEED_REALTIME:
            fprintf( stderr, "Speed Real Time\n" );
            break;
        case DM_IC_SPEED_NONREALTIME:
            fprintf( stderr, "Speed Not Real Time\n" );
            break;
        default:
            fprintf( stderr, "Speed Unknown\n" );
    }
```

```
    switch( dmParamsGetEnum( pParams, DM_IC_CODE_DIRECTION ) )  {
        case DM_IC_CODE_DIRECTION_UNDEFINED:
            fprintf( stderr, "Code Direction Undefined\n" );
            break;
        case DM_IC_CODE_DIRECTION_ENCODE:
            fprintf( stderr, "Code Direction Encode\n" );
            break;
        case DM_IC_CODE_DIRECTION_DECODE:
            fprintf( stderr, "Code Direction Decode\n" );
            break;
        default:
            fprintf( stderr, "Code Direction Unknown\n" );
    }
    fprintf( stderr, "Version: %d\n",
             dmParamsGetInt( pParams, DM_IC_VERSION ) );
    fprintf( stderr, "Revision: %d\n",
             dmParamsGetInt( pParams, DM_IC_REVISION ) );
}

//
// PrintImageParams()
//   Prints the descriptions obtained by dmICGetxxxParams().
//
void PrintImageParams( DMparams *pParams, const char *pOrigin )
{
    fprintf( stderr, "Width: %d   Height: %d\n",
             dmParamsGetInt( pParams, DM_IMAGE_WIDTH ),
             dmParamsGetInt( pParams, DM_IMAGE_HEIGHT ) );
    switch( dmParamsGetEnum( pParams, DM_IMAGE_PACKING ) )  {
        case DM_IMAGE_PACKING_RGBX:
            fprintf( stderr, "RGBX Packing\n" );
            break;
        case DM_IMAGE_PACKING_XBGR:
            fprintf( stderr, "XBGR Packing\n" );
            break;
        case DM_IMAGE_PACKING_RGBA:
            fprintf( stderr, "RGBA Packing\n" );
            break;
        case DM_IMAGE_PACKING_ABGR:
            fprintf( stderr, "ABGR Packing\n" );
            break;
        default:
            fprintf( stderr, "Unknown Packing\n" );
    }
}
```

The Digital Media Color Space Library

The Digital Media Color Space Library is a lower level library that is called by the Image Conversion Library. It provides the support for many of the parameter keyword operations discussed in "2. Configuring a Converter Instance Using the Image Conversion API,"such as DM_IMAGE_WIDTH, DM_IMAGE_HEIGHT, DM_IMAGE_PACKING, DM_IMAGE_ORDER, DM_IMAGE_ORIENTATION, and DM_IMAGE_MIRROR. More explicitly, the Color Space Library enables your application to do the following

- Convert between these color spaces: RGB, YCrCb, and Y
- Convert between many packings, such as DM_IMAGE_PACKING_RGB, DM_IMAGE_PACKING_CbYCr, and DM_IMAGE_PACKING_LUMINANCE
- Convert between many data types, such as DM_IMAGE_DATATYPE_BIT, DM_IMAGE_DATATYPE_CHAR, and DM_IMAGE_DATATYPE_SHORT10L
- Provide gamma correction for the RGB components of source and destination data
- Adjust the hue, saturation, brightness, contrast, bias, and scale of individual components
- Enable colorimetry adjustments of specific monitors

Normally, your application does not need to call the Color Space Library directly. The Image Conversion Library calls it as determined by the values in the source, destination, and conversion parameters of your converter instance. However, to enable easy access to the Color Space Library for conversions involving only uncompressed data, the Image Conversion Library provides the convenience function **dmICAnyToAny()**.

Use **dmICAnyToAny()** to make color space conversions on uncompressed data:

```
DMstatus dmICAnyToAny ( void *pBufferSrc, void *pBufferDst,
                        DMparams *pParamsSrc, DMparams *pParamsDst,
                        DMparams *pParamsConv )
```

The two variables *pBufferSrc* and *pBufferDst*, are pointers to the source and destination data which is uncompressed. Under some circumstances, *pBufferSrc* and *pBufferDst*, can be the same, allowing in-place data conversions. See dmColor(3dm) for more information. The variables *pParamsSrc*, *pParamsDst*, and *pParamsConv*, are pointers to DMparams structures which are set as shown in "Using the Image Conversion API." For example, the next sample changes the packing of a 640-by-480 pixel image from CbYCrY to XBGR:

```
/* Points to the source and destination buffers. */
void        *pBufferSrc, *pBufferDst;

/* Points to the source and destination parameters. */
DMparams    *pParamsSrc, pParamsDst;

/* Allocate buffers and fill source buffer. */
...
/* Do other preliminary processing */
...

/* Set the source parameters. */
dmParamsCreate( &pParamsSrc );
dmParamsSetInt( pParamsSrc,  DM_IMAGE_WIDTH, 640 );
dmParamsSetInt( pParamsSrc,  DM_IMAGE_HEIGHT,480 );
dmParamsSetEnum( pParamsSrc, DM_IMAGE_PACKING,
                             DM_IMAGE_PACKING_CbYCrY);

/* Set the destination parameters. */
dmParamsCreate( &pParamsDst );
dmParamsSetInt( pParamsDst,  DM_IMAGE_WIDTH, 640 );
dmParamsSetInt( pParamsDst,  DM_IMAGE_HEIGHT,480 );
dmParamsSetEnum( pParamsDst, DM_IMAGE_PACKING,
                             DM_IMAGE_PACKING_XBGR);

/* Do the conversion and clean up. */
dmICAnyToAny( pBufferSrc, pBufferDst,
              pParamsSrc, pParamsDst, NULL );
dmParamsDestroy( pParamsSrc );
dmParamsDestroy( pParamsDst );
```

For more information about the Color Space Library, see "The Color Space Library" in Appendix A.

Summary of the Digital Media Image Conversion Library

These are the functions of the Digital Media Image Conversion Library API. More details about specific functions, such as the errors they return, can be found by looking at the reference pages mention in the "Description" column.

Table 6-2 The Digital Media Image Conversion Library API

Function	Description
int **dmICGetNum** (void)	Return the number of image converters available. See also dmICGetNum(3dm).
int **dmICChooseConverter** (DMparams **srcParams*, DMparams **dstParams*, DMparams **convParams*)	Return the index of an image converter that matches the specified image parameters. See also dmICChooseConverter(3dm).
DMstatus **dmICGetDescription** (int *i*, DMparams **converterParams*)	Get the description of the converter indicated by index *i*. See also dmICGetDescription(3dm).
DMstatus **dmICCreate** (int *i*, DMimageconverter **converter*)	Create an instance of image converter *i*. See also dmICCreate(3dm).
void **dmICDestroy** (DMimageconverter *converter*)	Destroy the image converter instance. See also dmICDestroy(3dm).
DMstatus **dmICGetSrcParams** (DMimageconverter *converter*, DMparams **srcParams*)	Get the actual settings of source parameters of an image converter instance. See also dmICGetSrcParams(3dm).
DMstatus **dmICSetSrcParams** (DMimageconverter *converter*, DMparams **srcParams*)	Change the settings of source parameters of an image converter instance. See also dmICSetSrcParams(3dm).
DMstatus **dmICGetDefaultSrcParams** (DMimageconverter *converter*, DMparams **srcParams*)	Get the source parameters and default values of an image converter instance. See also dmICGetDefaultSrcParams(3dm).

Table 6-2 (continued) The Digital Media Image Conversion Library API

Function	Description
DMstatus **dmICGetDstParams** (DMimageconverter *converter*, DMparams **dstParams*)	Get the actual settings of destination parameters of an image converter instance. See also dmICGetDstParams(3dm).
DMstatus **dmICSetDstParams** (DMimageconverter *converter*, DMparams **dstParams*)	Change the settings of destination parameters of an image converter instance. See also dmICSetDstParams(3dm).
DMstatus **dmICGetDefaultDstParams** (DMimageconverter *converter*, DMparams **dstParams*)	Get the destination parameters and default values of an image converter instance. See also dmICGetDefaultDstParams(3dm).
DMstatus **dmICGetConvParams** (DMimageconverter *converter*, DMparams **convParams*)	Get the actual settings of conversion parameters of an image converter instance. See also dmICGetConvParams(3dm).
DMstatus **dmICSetConvParams** (DMimageconverter *converter*, DMparams **convParams*)	Change the settings of conversion parameters of an image converter instance. See also dmICSetConvParams(3dm).
DMstatus **dmICGetDefaultConvParams** (DMimageconverter *converter*, DMparams **convParams*)	Get the conversion parameters and default values of an image converter instance. See also dmICGetDefaultConvParams(3dm).
DMstatus **dmICGetSrcPoolParams** (DMimageconverter *converter*, DMparams **poolParams*)	Get the input buffering needs of the image converter instance. See also dmICGetSrcPoolParams(3dm).
DMstatus **dmICGetDstPoolParams** (DMimageconverter *converter*, DMparams **poolParams*)	Get the output buffering needs of the image converter instance. See also dmICGetDstPoolParams(3dm).

Table 6-2 (continued) The Digital Media Image Conversion Library API

Function	Description
DMstatus **dmICSetDstPool** (DMimageconverter *converter*, DMbufferpool *pool*)	Set the pool from which *converter* allocates each output DMbuffer. See also dmICSetDstPool(3dm).
int **dmICGetDstQueueFD** (DMimageconverter *converter*)	Get the queue file descriptor of the image converter instance. See also dmICGetDstQueueFD(3dm).
int **dmICGetDstQueueFilled** (DMimageconverter *converter*)	Get the number of buffers ready to be received from the converter instance. See also dmICGetDstQueueFilled(3dm).
int **dmICGetSrcQueueFilled** (DMimageconverter *converter*)	Get the number of buffers sent to the converter instance, but not yet processed. See also dmICGetSrcQueueFilled(3dm).
int **dmICSend** (DMimageconverter *converter*, DMbuffer *srcBuffer*, int *numRefBuffers*, DMbuffer **refBuffers*)	Transfer source buffer and reference buffers to the image converter instance. See also dmICSend(3dm).
DMstatus **dmICReceive** (DMimageconverter *converter*, DMbuffer **dstBuffer*)	Transfer data from the image converter instance. See also dmICReceive(3dm).
DMstatus **dmICAnyToAny** (void **src*, void **dst*, DMparams **srcParams*, DMparams **dstParams*, DMparams **convParams*)	A convenience function.

Audio Data Conversion

This section describes the Digital Media Audio Conversion Library, its converters, and how to use the Audio Conversion API to execute the four steps of an audio data conversion.

The Digital Media Audio Conversion Library

The Digital Media Audio Conversion Library provides data format conversion for applications that do real-time audio capture, playback and file conversion. It makes it possible to efficiently move data between any audio producer and any audio consumer, regardless of their native formats. The library provides a single API for performing memory-to-memory sound compression and conversion. Table 6-1 lists commonly installed codec options that can be accessed through the Audio Conversion API. The "DM_AUDIO_COMPRESSION Value" column contains the identification values that are used with **dmACSetParams()** as described in "Configuring a Converter Instance Using the Audio Conversion API."

Table 6-3 Digital Media Audio Codecs

DM_AUDIO_COMPRESSION Value	Description
DM_AUDIO_DVI	Intel's Digital Video Interactive. See also "The DVI Audio Compression Library" in Appendix A.
DM_AUDIO_G711_ALAW DM_AUDIO_G711_ULAW	International Telecommunication Union Standard G.711. See "The G.711 Audio Compression Library" in Appendix A.
DM_AUDIO_G722	International Telecommunication Union Standard G.722. See "The G.722 Audio Compression Library" in Appendix A.
DM_AUDIO_G726	International Telecommunication Union Standard G.726. See "The G.726 Audio Compression Library" in Appendix A.
DM_AUDIO_G728	International Telecommunication Union Standard G.728. See "The G.728 Audio Compression Library" in Appendix A.
DM_AUDIO_GSM	Global System for Mobile Telecommunications. See "The GSM Audio Compression Library" in Appendix A.

Table 6-3 (continued) Digital Media Audio Codecs

DM_AUDIO_COMPRESSION Value	Description
DM_AUDIO_MPEG1	MPEG-1 Audio. See "The MPEG-1 Audio Compression Library" in Appendix A.
DM_AUDIO_MULTIRATE	Aware MultiRate near-lossless compression.

In addition to the compression and decompression done by the codecs, an audio converter instance may also perform such transformations as

- converting between different numerical representations, such as unsigned integer and two's complement signed integer
- converting between big-endian and little-endian byte orders
- audio sampling rate conversion (see "The Audio Rate Conversion Library" in Appendix A)
- converting between different numbers of interleaved channels, such as mono and stereo
- Pulse Code Modulation (PCM) mapping

As the next sections demonstrate, the Audio Conversion Library enables you to effectively use the codecs and transformation libraries by following the steps listed in "Using Digital Media Converters." To use the Audio Conversion API, you must use these header files:

```
#include <dmedia/dm_audioconvert.h>
#include <dmedia/dm_audioutil.h>
```

Creating a Converter Instance Using the Audio Conversion API

Use **dmACCreate()** to create an audio converter instance:

```
DMstatus dmACCreate ( DMaudioconverter* converter )
```

This function creates and initializes *converter*, a handle to a DMaudioconverter instance. All the Audio Conversion Library functions use this handle which is declared as follows:

```
typedef struct _DMaudioconverter *DMaudioconverter;
```

Note: All of the Audio Conversion Library functions return a DMstatus value of DM_SUCCESS if they succeed, DM_FAILURE if not. After a receiving a DM_FAILURE, your application can call the function the function **dmGetErrorForPID()** or **dmGetError()** to return an error message and error number. See "Digital Media Error Handling" in Chapter 3 for more information.

Configuring a Converter Instance Using the Audio Conversion API

Once a converter instance has been created, it must be configured. As with the Image Conversion Library, your application does this by using DMparams data structures to specify a set of source parameters, a set of destination parameters, and, optionally, a set of conversion parameters.

The Audio Conversion Library enables you to do the following:

- Configure an audio converter instance by setting the source, destination, and conversion parameters with **dmACSetParams()**
- Get the source, destination, and conversion parameter settings of a configured audio converter instance with **dmACGetParams()**
- Reset an audio converter instance to its original configuration with **dmACReset()**

Use **dmACSetParams()** to set the DMaudioconverter parameter values:

```
DMstatus dmACSetParams ( DMaudioconverter converter,
                         DMparams *sourceparams, DMparams *destparams,
                         DMparams *conversionparams )
```

The handle *converter* indicates an audio converter instance created by a previous call to **dmACCreate()**. The DMparams structures pointed to by *sourceparams* and *destparams* describe the formats of the audio data prior to and after conversion. The variable *conversionparams* points to a DMparams structure that contains parameters specific to the conversion process. The variables *destparams* and *conversionparams* are optional and may be set to NULL.

Use **dmACGetParams()** to get the DMaudioconverter parameter values:

```
DMstatus dmACGetParams ( DMaudioconverter converter,
                         DMparams *sourceparams, DMparams *destparams,
                         DMparams *conversionparams )
```

The handle *converter* indicates an audio converter instance previously configured by a call to **dmACSetParams()**. The DMparams structures pointed to by *sourceparams* and

destparams describe the formats of the audio data prior to and after conversion. The variable *conversionparams* points to a DMparams structure that contains parameters specific to the conversion process. The variables *destparams* and *conversionparams* are optional and may be set to NULL.

Use **dmACReset()** to reset a DMaudioconverter handle to its original configuration:

```
DMstatus dmACReset ( DMaudioconverter converter )
```

The handle *converter* indicates an audio converter instance already configured by a call to **dmACSetParams()**.

The next three sections describe the DMparams structures for source, destination, and conversion parameters in more detail. They are followed by a section that discusses parameters relevant to specific converters.

Source Parameters

The source parameters, which describe the data to be converted, are contained in the DMparams structure indicated by *sourceparams*. The source parameters that must be specified are shown below. There are no default values. The parameters and their DMparams-setting functions follow the bullets. Legal values for the parameters follow the dashes.

- DM_AUDIO_FORMAT and **dmParamsSetEnum():**

  DM_AUDIO_TWOS_COMPLEMENT

  DM_AUDIO_UNSIGNED

  DM_AUDIO_FLOAT

  DM_AUDIO_DOUBLE

- DM_AUDIO_WIDTH and **dmParamsSetInt()**:

  The width of the data in bits. An integer value between 1 and 32.

- DM_AUDIO_BYTE_ORDER and **dmParamsSetEnum()**:

  DM_AUDIO_BIG_ENDIAN

  DM_AUDIO_LITTLE_ENDIAN

- DM_AUDIO_CHANNELS and **dmParamsSetInt()**:

  The number of audio channels. An integer value greater than 0.

- DM_AUDIO_RATE and **dmParamsSetFloat()**:

  The audio sampling rate in Hz. A DM_TYPE_FLOAT value greater than 0.0.

- DM_AUDIO_COMPRESSION and **dmParamsSetString()**:

  DM_AUDIO_UNCOMPRESSED or one of the values shown in Table 6-1.

Destination Parameters

The destination parameters, which describe the output audio data, are contained in the DMparams structure indicated by *destparams*. Any destination parameter not specified defaults to its source parameter value, except DM_AUDIO_COMPRESSION which defaults to DM_AUDIO_UNCOMPRESSED.

There is a set of four parameters for PCM mapping whose values, although they can be set for source data, are normally specified only for destination data. The parameters are based on a model where there is a PCM value that corresponds to zero voltage and a differential value that corresponds to full voltage. The set consists of the following parameters, which are shown with their DMparams setting functions and appropriate values.

- DM_AUDIO_PCM_MAP_SLOPE and **dmParamsSetFloat()**:

  The full voltage PCM value. (Default is 32767.0)

- DM_AUDIO_PCM_MAP_INTERCEPT and **dmParamsSetFloat()**:

  The zero voltage PCM value. (Default is 0.0)

- DM_AUDIO_PCM_MAP_MAXCLIP and **dmParamsSetFloat()**:

  Clip all PCM values to this maximum value. (Default is 32767.0)

- DM_AUDIO_PCM_MAP_MINCLIP and **dmParamsSetFloat()**:

  Clip all PCM values to this minimum value. (Default is -32768.0)

The function **dmACSetParams()** automatically sets their default input and output values from the input and output data format specifications. Your application needs to set them only if it has special mapping requirements, such as input data with a fixed offset like a DC bias. If your application sets any of these four parameters, it must set all of them. See afIntro(3dm) for more information on PCM mapping.

Conversion Parameters

The conversion parameters, which modify the codec settings and other aspects of the conversion process, are contained in the DMparams structure indicated by *conversionparams*. There are five categories of conversion parameters.

1. The Processing Mode Parameter

   This parameter, whose keyword is DM_AUDIO_PROCESS_MODE, is used to determine the converter's Processing mode. It can also be used to set the processing mode when both the input and output data are uncompressed. There are two processing modes: push and pull. In *pull* mode, your application requests a given number of output frames. Decompression must use pull mode and your application uses a buffer length parameter to specify how many frames of uncompressed data the converter instance should put in the output buffer of **dmACConvert()**. In *push* mode, your application gives the converter a specified number of input frames. Compression requires push mode and your application specifies how many frames of uncompressed data are in the input buffer. The two settings for DM_AUDIO_PROCESS_MODE are

   - DM_AUDIO_PROCESS_PULL and **dmParamsGetInt()**
   - DM_AUDIO_PROCESS_PUSH and **dmParamsGetInt()**

2. Buffer Length Parameters

   The three buffer length parameters are used to determine the number of frames in the input or output buffers. Your application specifies them only during compression, decompression, or rate conversion because the input and output buffer lengths are equal at all other times. Your application must set the parameter DM_AUDIO_MAX_REQUEST_LEN prior to calling **dmACConvert()** to specify the largest buffer the converter instance will have to process. Your application then calls **dmACGetParams()** to find the value of DM_AUDIO_MIN_INPUT_LEN or DM_AUDIO_MIN_OUTPUT_LEN. If the converter instance is in pull mode, DM_AUDIO_MIN_INPUT_LEN gives the minimum number of frames the converter instance requires in the input buffer. If the instance is in push mode, DM_AUDIO_MIN_OUTPUT_LEN gives the minimum number of frames the instance requires in the output buffer. Your application then allocates buffers appropriate to these sizes. The parameters, DMparams functions, and typical values are as follows:

   - DM_AUDIO_MAX_REQUEST_LEN and **dmParamsSetInt():**

     An integer value greater than 0. This value can only be set.

- DM_AUDIO_MIN_INPUT_LEN and **dmParamsGetInt():**

  An integer value that your application can only read.

- DM_AUDIO_MIN_OUTPUT_LEN and **dmParamsGetInt():**

  An integer value that your application can only read.

3. The Dithering Parameter

   The dithering parameter, whose keyword is DM_AUDIO_DITHER_ALGORITHM, is used only when data is converted from a larger to a smaller data type. An example of such a conversion would be going from a floating point to a 16-bit integer representation. The dithering algorithm is applied to reduce the quantization error distortion inherent in reducing resolution. The two possible values are as follows:

   - DM_AUDIO_DITHER_NONE (default)
   - DM_AUDIO_DITHER_LSB_TPDF (Least Significant Bit—Triangular Probability Density Function)

4. Rate Conversion Parameters

   There are three parameters that affect the rate conversion algorithm. They are used only when the input and output sampling rates are not equal. The three parameters, their DMparams setting functions, and their possible values are shown below. See dmAudioRateConverterSetParams(3dm) for more information about them.

   - DM_AUDIO_RC_ALGORITHM and **dmParamsSetString()**:

     DM_AUDIO_RC_JITTER_FREE (default)

     DM_AUDIO_RC_POLYNOMIAL_ORDER_1

     DM_AUDIO_RC_POLYNOMIAL_ORDER_3

   - DM_AUDIO_RC_JITTER_FREE_STOPBAND_ATTENUATION and **dmParamsSetFloat()**:

     DM_AUDIO_RC_JITTER_FREE_STOPBAND_ATTENUATION_78_DB (default)

     DM_AUDIO_RC_JITTER_FREE_STOPBAND_ATTENUATION_96_DB

     DM_AUDIO_RC_JITTER_FREE_STOPBAND_ATTENUATION_120_DB

- DM_AUDIO_RC_JITTER_FREE_TRANSITION_BANDWIDTH and **dmParamsSetFloat()**:

  DM_AUDIO_RC_JITTER_FREE_TRANSITION_BANDWIDTH_1_PERCENT

  DM_AUDIO_RC_JITTER_FREE_TRANSITION_BANDWIDTH_10_PERCENT

  DM_AUDIO_RC_JITTER_FREE_TRANSITION_BANDWIDTH_20_PERCENT

5. The Channel Conversion Parameter

   The channel conversion or channel matrix parameter is used to mix the channels associated with a track. The matrix is a one-dimensional array composed of a two-dimensional array in row-major order, where each row represents an output channel and each column represents an input channel. See the afSetChannelMatrix(3dm) reference page for a detailed explanation. The parameter keyword and its DMparams-setting function are as follows:

   - DM_AUDIO_CHANNEL_MATRIX and **dmParamsSetFloatArray()**:

     A DMfloatarray of double-precision floating point numbers.

Converting Data Using the Audio Conversion API

Use **dmACConvert()** to convert the audio data's format, sampling rate, and compression:

```
DMstatus dmACConvert ( DMaudioconverter converter, void *inbuffer,
                       void *outbuffer, int *in_amount,
                       int *out_amount )
```

This function performs the data format, sampling rate, and compression or decompression specified by **dmACSetParams()**. The variable *converter* is a handle to an audio converter instance previously created with **dmACCreate()** and configured with **dmACSetParams()**. The variables *inbuffer* and *outbuffer* point to the buffers that contain the audio data prior to and after conversion. As described in "Configuring a Converter Instance Using the Audio Conversion API," your application may need to determine the number of frames these buffers must hold using the DM_AUDIO_MIN_INPUT_LEN or DM_AUDIO_MIN_OUTPUT_LEN parameters. If *inbuffer* is NULL, the converter instance flushes any internal buffers to the output buffer.

The variable *in_amount* points to an integer containing the number of frames (bytes if the data is compressed) of input data available to the converter instance. This can be any value greater than 0. In pull mode, **dmACConvert()** resets this value to the number of frames (bytes) read from *inbuffer* by the converter instance. (To review the push and pull

modes, see the conversion parameter discussion in "Configuring a Converter Instance Using the Audio Conversion API.") The pointer *out_amount* indicates an integer containing the number of frames (bytes if the data is compressed) of converted data your application wants from the converter instance. The initial value is ignored in push mode. After processing the data, **dmACConvert()** resets the *out_amount* value to the number of frames (bytes) actually placed into *outbuffer*. If the conversion involved rate conversion, compression, or decompression, this value can vary significantly from the in_amount value. It can even be zero when the *in_amount* value was positive.

Compression Parameters

The compression parameters modify individual audio converters listed in Table 6-1. The parameters and their values are listed below. For more information about using specific parameters, please refer to the relevant reference pages.

- Digital Video Interactive (DVI) has one parameter.

  DM_DVI_AUDIO_BITS_PER_SAMPLE specifies the compression algorithm. Its possible values are

  – DM_DVI_AUDIO_3BITS_PER_SAMPLE

  – DM_DVI_AUDIO_4BITS_PER_SAMPLE (default)

- ITU-T G722 has three parameters.

  DM_AUDIO_CODEC_MAX_BYTES_PER_BLOCK

  DM_AUDIO_CODEC_FRAMES_PER_BLOCK

  DM_AUDIO_CODEC_FILTER_DELAY

- ITU-T G722 has two parameters.

  DM_AUDIO_BITRATE is an integer in units of bits per second with one of the following values: 16000, 24000, 32000, 40000.

  DM_G726_NATIVE_FORMAT specifies the input or output sample data format. Its possible values are

  – AUDIO_ENCODING_ULAW

  – AUDIO_ENCODING_ALAW

  – AUDIO_ENCODING_LINEAR

- ITU-T G728 has one parameter.

  DM_G728_POSTFILTERING_FLAG selects a decoder with or without post filtering. Its possible values are

  – DM_G728_POSTFILTERING_YES

  – DM_G728_POSTFILTERING_NO

- Global System for Mobile Telecommunications (GSM) has three parameters.

  DM_AUDIO_CODEC_MAX_BYTES_PER_BLOCK

  DM_AUDIO_CODEC_FRAMES_PER_BLOCK

  DM_AUDIO_CODEC_FILTER_DELAY

- MPEG-1 Audio has seven parameters for encoding and decoding.

  DM_AUDIO_RATE is the input or output sampling rate given in Hz. It is a double with possible values of 32000, 44100 (default), and 48000.

  DM_AUDIO_FORMAT is the format of each input or output sample. The only supported value is DM_AUDIO_TWOS_COMPLEMENT

  DM_AUDIO_WIDTH is the width of each input or output sample. The only supported value is DM_AUDIO_WIDTH_16

  DM_AUDIO_CHANNELS is the number of channels in the input or output data. It is an integer value of 1 or 2 (default).

  DM_MPEG1_AUDIO_LAYER is a flag specifying the basic algorithm to be used. There are two possible values.

  – DM_MPEG1_AUDIO_LAYER1

  – DM_MPEG1_AUDIO_LAYER2 (default)

  DM_AUDIO_CHANNEL_POLICY indicates how multiple channels should be treated. There are three possible values.

  – DM_AUDIO_STEREO—The channels are part of a single multichannel signal.

  – DM_AUDIO_JOINT_STEREO (default)—The algorithm may attempt to exploit redundancy between channels for greater coding gain.

  – DM_AUDIO_INDEPENDENT—The separate channels are unrelated and should be processed separately.

  DM_AUDIO_BIT_RATE specifies the desired bit rate, in bits per second, for the compressed data.

Supported values with DM_MPEG1_AUDIO_LAYER1 are 32000, 64000, 96000, 128000, 160000, 192000, 224000, 256000 (default), 288000, 320000, 352000, 384000, 416000, and 448000. Supported values with DM_MPEG1_AUDIO_LAYER2 are 32000, 48000, 56000, 64000, 80000, 96000, 112000, 128000, 160000, 192000, 224000, 256000 (default), 320000, and 38400

- MPEG-1 Audio has four parameters to use with encoding only.

  DM_MPEG1_AUDIO_PSYCHOMODEL selects which psychoacoustic model to use for calculating the safe masking thresholds for quantizing noise. There are two possible values.

  – DM_MPEG1_AUDIO_PSYCHOMODEL1

  – DM_MPEG1_AUDIO_PSYCHOMODEL2

  DM_MPEG1_AUDIO_PSYCHOMODEL1_ALPHA is a float value that specifies the alpha parameter for DM_MPEG1_AUDIO_PSYCHOMODEL1. It has a possible value within (0.0, 2.0]. Default is 2.0.

  DM_MPEG1_AUDIO_BITRATE_POLICY is a flag used for the interpretation of DM_AUDIO_BIT_RATE. There are two possible values.

  – DM_MPEG1_AUDIO_FIXRATE (default)

  – DM_MPEG1_AUDIO_CONSTANT_QUALITY

  DM_MPEG1_AUDIO_CONST_QUAL_NMR is the desired mask-to-noise ratio in decibels. It is a float value with a value within (-13.0, 13.0]. The default is 0.0.

- MPEG-1 Audio has three parameters to use with decoding only.

  DM_MPEG1_AUDIO_DECIMATION_SCALE specifies the decimation factor applied to reduce the complexity of decoding. It has three possible values.

  – DM_MPEG1_AUDIO_BANDWIDTH_FULL (default)

  – DM_MPEG1_AUDIO_BANDWIDTH_HALF

  – DM_MPEG1_AUDIO_BANDWIDTH_QUARTER

  DM_MPEG1_AUDIO_SCALE_FILTERSHAPE specifies the filter shape applied to reduce the complexity of decoding. It has three possible values.

  – DM_MPEG1_AUDIO_DEFAULT_FILTER (default)

  – DM_MPEG1_AUDIO_FILTER_SHAPE1

  – DM_MPEG1_AUDIO_FILTER_SHAPE2

DM_MPEG1_AUDIO_COMBINE_CHANS_FLAG enables single channel output. It is an integer with a default value of 0, that is JOINT_STEREO mode. A value of 1 forces the decoder to produce single channel output.

- MPEG-1 Audio has three parameters to use only in query mode.

  DM_AUDIO_CODEC_FRAMES_PER_BLOCK is an integer that specifies how many sample frames are put into each compressed data block.

  DM_AUDIO_CODEC_MAX_BYTES_PER_BLOCK is an integer that indicates the maximum number of bytes that will compose a compressed data block.

  DM_AUDIO_CODEC_FILTER_DELAY is an integer that indicates the delay, in sample frames, introduced by compression and decompression processing.

Destroying a Converter Instance Using the Audio Conversion API

The Audio Conversion library provides the function **dmACDestroy()** to destroy an audio converter instance:

```
DMstatus dmACDestroy ( DMaudioconverter converter )
```

The function frees the memory associated with the DMaudioconverter handle. The handle is not valid after this call returns.

Summary of the Digital Media Audio Conversion Library

These are the functions of the Digital Media Audio Conversion Library API. More details about specific functions, such as the errors they return, can be found by looking at the reference pages mention in the "Description" column.

Table 6-4 The Digital Media Audio Conversion API

Function	Description
DMstatus **dmACConvert** (DMaudioconverter *converter*, void **inbuffer*, void **outbuffer*, int **in_amount*, int **out_amount*)	Convert the audio data format, sampling rate, and compression. See also dmACConvert(3dm).
DMstatus **dmACCreate** (DMaudioconverter **converter*)	Create a DMaudioconverter handle to use for audio format conversion. See also dmACCreate(3dm).

Table 6-4 (continued) The Digital Media Audio Conversion API

Function	Description
DMstatus **dmACDestroy** (DMaudioconverter *converter*)	Destroy a DMaudioconverter handle used for audio format conversion. See also dmACDestroy(3dm).
DMstatus **dmACReset** (DMaudioconverter *converter*)	Reset a DMaudioconverter handle to its default state. See also dmACReset(3dm).
DMstatus **dmACSetParams** (DMaudioconverter *converter*, DMparams **sourceparams*, DMparams **destparams*, DMparams **conversionparams*)	Set the DMaudioconverter parameter values. See also dmACSetParams(3dm).
DMstatus **dmACGetParams** (DMaudioconverter *converter*, DMparams **sourceparams*, DMparams **destparams*, DMparams **conversionparams*)	Get the DMaudioconverter parameter values. See also dmACGetParams(3dm).

Chapter 7

Digital Media Audio File Operations

The Audio File (AF) Library provides a uniform programming interface for reading and writing audio disk files. It also allows applications to control the format of audio data buffers. The AF Library is implemented as a Dynamic Shared Object (DSO) which enables properly written applications to use the latest version of the library without recompiling and relinking. See "Querying the AF Library" for a further discussion.

The Audio File Library comprises routines that handle these fundamental tasks:

- Querying the AF Library
- Creating and Configuring Audio Files
- Opening, Closing, and Identifying Audio Files
- Reading and Writing Audio Track Information

This chapter covers these topics, explains basic audio file concepts, and provides programming tips for the AF Library.

About the Audio File Library

This section explains concepts fundamental to understanding and using the Audio File Library properly.

Audio File Library Programming Model

The AF Library has two basic data structures:

- AFfilesetup, an audio file setup that stores initialization parameters used when creating a new audio file handle
- AFfilehandle, an audio file handle that provides access to the audio file

The basic steps required for setting up an audio file for writing are:

1. Initialize an AFfilesetup, by calling **afNewFileSetup()**.
2. Configure the AFfilesetup for your data.
3. Open an audio file for reading or writing by calling either **afOpenFile()** or **afOpenFD()**. These routines return an AFfilehandle whose data configuration matches the settings in the AFfilesetup.

   Note: The AF Library currently supports read-only and write-only file access (but not both simultaneously). Therefore, to alter an existing file, you must create a new file and copy data from the original file.

About Audio Files

This section explains basic concepts for working with audio files. It describes data structures used by the Audio File Library and in particular, the structure of AIFF-C files and the higher-level abstraction that the AF Library API uses to read and write AIFF-C (and AIFF) files.

The AF Library breaks audio files into the following four functional components:

Audio file format	Allows applications to identify audio file formats and format versions.
Audio tracks	Contain audio sample data, parameters that characterize the data format (such as sample rate, channel configuration, and compression type), and *marker* structures that store sample frame locations in the track for looping and other purposes.
Instrument configurations	Contain instrument parameters for configuring digital samples when playing back audio track data, and loop markers for repeating tracks or portions of a track.
Miscellaneous data	Include text strings (author, copyright, name, annotation, and so on) and other non-audio information (such as MIDI data and application-specific data).

The two portions of an audio file you will make most use of are *audio tracks* and *instrument configurations*.

Audio File Formats

Audio file format is typically indicated by header information preceding the actual data that describes the nature of the data in that file. The file format of an audio file constrains the data format of each of its tracks to one of a set of track formats supported by that file format, but you do not necessarily know which one. You must therefore set and query the track format for each of an audio file's tracks independently of its file format. It is often possible and desirable to write your application so that it queries only the data format(s) of the track(s) (instead of querying the file format) of the audio files it opens.

Audio Tracks, Sample Frames, and Track Markers

Audio tracks contain the recorded samples that produce sound when sent to the audio hardware. These samples are stored linearly for mono recordings and as interleaved left-right pairs (left channel in even locations, right channel in odd locations) for stereo recordings. These pairs are called *sample frames* (this term is also used for mono tracks, but a sample frame is the same thing as a sample when mono data is used).

Audio tracks also contain *track markers,* which can be set to point to arbitrary locations in the audio track. These markers, which are identified by an integer ID number and (optionally) a name string, point to locations between sample frames.

Audio Track Format Parameters

Data format information, including sample rate, sample format, sample width, and sample compression type is stored as part of the audio track. Several kinds of compression are supported (you can also choose not to use compression). The AF Library automatically compresses samples being written to a file and decompresses samples read from a file. The ability of the AF Library to perform compression/decompression of audio data in real time is dependent on system overhead.

Instrument Configurations and Loops

Instrument configurations contain a set of parameters that define the aspects of a sampler, including detuning, key velocity, and gain. They also contain *loop markers*, which identify the beginning and ending points of loops that allow all or part of the audio track to be repeated. These loop markers point to previously created audio track markers, which in turn refer to locations in the audio track that comprise the beginning and ending of the loop. AIFF and AIFF-C files support two kinds of loops, *sustain* and

release, each with a beginning and ending marker, which can be used in audio tracks and track markers.

AIFF-C and the AF Library API

Silicon Graphics has adopted AIFF-C as its default digital audio file format. AIFF-C is based on Apple Computer's Audio Interchange File Format (AIFF), which conforms to the *EA IFF 85 Standard for Interchange Format Files* developed by Electronic Arts. Unlike the older AIFF standard, AIFF-C files can store compressed sample data as well as two's complement linear PCM sample data.

AIFF-C provides a standard file format for storing sampled sounds on magnetic media. The format can store any number of channels of sampled sound at a variety of sample rates and sample widths. The format is extensible, allowing support of new compression types and application-specific data, while maintaining backward compatibility.

An AIFF-C file is composed of a series of different kinds of data *chunks*. For the most part, the AF Library API handles low-level chunk manipulation. For complete information on the types of chunks supported by AIFF-C, see the *Audio Interchange File Format with Compression (AIFF-C) Specification*.

AIFF and AIFF-C files consist of similar component structures. The chunks in an AIFF-C file are grouped together inside a special *container chunk*. The *EA IFF 85* specification defines several types of container chunks, but the kind used by AIFF-C is of type 'FORM'.

Table 7-1 shows the mapping between the Audio File Library API functional components and the low-level AIFF-C/AIFF data chunks.

Table 7-1 Mapping of AF Library Components to AIFF-C/AIFF File Chunks

AF Library Functional Component	AIFF-C/AIFF Chunks
File format information	'FVER', 'FORM'
Audio tracks	'SSND', 'COMM', 'MARK', 'AESD', 'COMT'[a]
Instrument configurations	'INST'
Miscellaneous data	'AUTH', 'NAME', '(c)', 'ANNO',' MIDI ', 'APPL'

a. 'COMT' chunks are not currently supported by the AF Library.

Virtual Data Format

The Audio File Library allows the format of audio data in an application's buffer to be independent of that being read from or written to disk audio files. The format of the audio data in the buffer is called the data's *virtual format*. Once the virtual data format is set with the routines described in "Getting and Setting the Virtual Audio Format," conversion between the disk file format and the virtual format happens automatically. An application can thus specify the virtual format to be identical to the final format it wants the buffer data to be in, and ignore the original disk file format entirely.

The default virtual format of the data that is loaded by **afReadFrames()** into the application's data buffer is identical to the disk format with two important exceptions:

- the virtual byte order default is DM_AUDIO_BIG_ENDIAN
- the buffer data is always DM_AUDIO_UNCOMPRESSED

These exceptions were made to assure backwards compatibility with the first version of the AF Library. It is possible to use **afSetVirtualByteOrder()** to set the virtual byte order to DM_AUDIO_LITTLE_ENDIAN, but the virtual compression cannot be changed.

PCM Mapping

PCM mapping enables an application to specify the numerical mapping when it converts data from an integer format to a floating point format or vice versa. For example, an application may need to read files of 8-, 16-, 24-, or 32-bit integer data, which may be signed or compressed into a buffer as floating point data. Similarly, the application may want to write floating point data in buffers to a file and have the data converted to unsigned integers in real time.

In the latter case, the application probably expects the floating point (float) data to have values within a certain range, perhaps -1.0 to 1.0 or 0.0 to 1.0. If the float-to-integer conversion is specified by only a slope value, it is impossible to achieve some mappings because the intercept would then be a fixed value. Therefore the intercept must be specifiable. That way float data which ranges from [-1.0, 1.0], for example, can be mapped to the range [0, 65535]. However, unlike the floating point range, the unsigned integer range is not symmetric. Thus, minimum and maximum clipping values (*minclip* and *maxclip*) to which the AF Library clips all PCM values, are necessary to allow the application to be more specific about how the endpoints of the mapping line up.

This model assumes there exists one PCM value which corresponds to *zero volts*, and a differential PCM value which produces a *full-voltage* value when added to or subtracted from the zero volt value. The idea of *voltage* is a canonical form and does not correspond to any hardware. It does not matter what the full-voltage value is. To sum up:

slope	the full-voltage differential PCM value
intercept	the zero-volt PCM value
minclip	the minimum permitted PCM value
maxclip	the maximum permitted PCM value

Querying the AF Library

Your application can query the AF Library about features such as instrument parameters, file formats, and compression algorithms. The query functions return information about these features such as their names, descriptions, labels, default values, ID counts, and implementation status. See the reference page afQuery(3dm) for a complete list query parameters.

There are four query routines as shown in Table 7-2. Which routine to use depends on the data type of the parameter being asked about. For example, to retrieve a character string, use **afQueryPointer()** and cast the return value to (char *).

```
void* afQueryPointer ( int querytype, int arg1, int arg2,
                       int arg3, int arg4 )
```

The parameters to the query functions vary with the type of query. The first parameter, *querytype*, is a query type such as AF_QUERYTYPE_COMPRESSION or AF_QUERYTYPE_FILEFMT. *arg1* is the first sub-selector, such as AF_QUERY_TYPE or AF_QUERY_NAME. *arg2* can be either an additional sub-selector, such as AF_QUERY_DEFAULT, or the target variable being returned by the query, such as the file format or the compression type. The target of the query will always be the final non-zero parameter to **afQueryPointer()** and its relatives. All subsequent parameters must be set to 0.

Example 7-1 Creating a List of Supported Compression Type IDs

```
long          numCompressionTypes;
int           *compressionIDs = NULL;
AFfilehandle  handle     = afOpenFile ( "somefile.aifc", "r", NULL );
int           fileformat = afGetFileFormat ( handle, NULL );
```

```
/* Get the total number of compression types.*/
numCompressionTypes = afQueryLong ( AF_QUERYTYPE_FILEFMT,
                                    AF_QUERY_COMPRESSION_TYPES,
                                    AFQUERY_VALUE_COUNT,
                                    fileformat, 0 );

/* If the total is not zero, retrieve the array of IDs */
if( numCompressionTypes > 0 ) {
    compressionIDs = (int *) afQueryPointer ( AF_QUERYTYPE_FILEFMT,
                                              AF_QUERY_COMPRESSION_TYPES,
                                              AF_QUERY_VALUES,
                                              fileformat, 0 );
    /* Use the IDs here. */
}

/* When finished, free the array memory. */
if( compressionIDs != NULL )
    free ( compressionIDs );
```

Summary of the Query Functions

Table 7-2 is a summary of the Audio File Library query functions. Functions that are not covered in the preceding text have a notation in the Description column. Please refer to a function's reference page if a more detailed explanation is needed. For example, the reference page for **afQuery()** is afQuery(3dm).

Table 7-2 Audio File Library Query Functions

Function	Description
AUpvlist **afQuery** (int *querytype*, int *arg1*, int *arg2*, int *arg3*, int *arg4*)	Retrieve an AUpvlist static parameter associated with the Audio File Library formats. Not covered in text.
double **afQueryDouble** (int *querytype*, int *arg1*, int *arg2*, int *arg3*, int *arg4*)	Retrieve a double static parameter associated with the Audio File Library formats. Not covered in text.

Table 7-2 (continued) Audio File Library Query Functions

Function	Description
long **afQueryLong** (int *querytype*, int *arg1*, int *arg2*, int *arg3*, int *arg4*)	Retrieve a long static parameter associated with the Audio File Library formats. Not covered in text.
void* **afQueryPointer** (int *querytype*, int *arg1*, int *arg2*, int *arg3*, int *arg4*)	Retrieve a void* static parameter associated with the Audio File Library formats.

Creating and Configuring Audio Files

This section explains how to initialize an AF Library application, including how to create, configure, and free AF Library data structures for working with audio files.

Creating an Audio File Setup

The AFfilesetup structure stores initialization parameters used when creating a new audio file. When you open an audio file for reading or writing the AF Library returns another structure, an AFfilehandle, which provides access to the audio file and is used as an argument by all AF Library routines.

afNewFileSetup() creates and initializes an AFfilesetup structure that you configure for your data, and then use to open an audio file:

```
AFfilesetup afNewFileSetup ( void )
```

afNewFileSetup() returns a default AFfilesetup structure.

Table 7-3 lists the AFfilesetup configuration parameters and their defaults.

Table 7-3 AFfilesetup Parameters and Defaults

Parameter	Default
File format	AF_FILE_AIFFC
Audio track	AF_DEFAULT_TRACK
Audio track sample format, sample width	AF_SAMPFMT_TWOSCOMP, 16-bit

Table 7-3 AFfilesetup Parameters and Defaults

Parameter	Default
Audio track channels (interleaved)	2 (stereo)
Audio track compression	AF_COMPRESSION_NONE
Audio track markers	Four markers with IDs: 1,2,3,4
Instrument	AF_DEFAULT_INST
Instrument Parameters	(See Table 7-7)
Loops	Two loops with IDs: 1, 2; default mode is AF_LOOP_MODE_NOLOOP

Your application should free an AFfilesetup that is no longer needed. **afFreeFileSetup()** deallocates an AFfilesetup structure. Its function prototype is:

```
void afFreeFileSetup ( AFfilesetup setup )
```

where *setup* is an AFfilesetup previously created by a call to **afNewFileSetup()**. This does not affect any file previously opened using the same AFfilesetup structure.

Before using the new AFfilesetup to open an audio file, you might need to modify the default AFfilesetup in order to create the configuration you want. The sections that follow explain how to change the default AFfilesetup configuration.

Initializing the Audio File Format

You need to set the file format in an AFfilesetup structure before passing the structure to **afOpenFile()**.

afInitFileFormat() configures the file format parameter in an AFfilesetup structure. Its function prototype is:

```
void afInitFileFormat ( AFfilesetup setup, int filefmt )
```

where *setup* is the AFfilesetup structure, and *filefmt* is an integer constant which specifies an audio format supported by the AF Library. A new audio file that is opened by calling

afOpenFile() with this AFfilesetup as an argument will then be formatted accordingly. Valid format types are shown below.

AF_FILE_AIFFC	Extended Audio Interchange File Format (AIFF-C)
AF_FILE_AIFF	Audio Interchange File Format (AIFF)
AF_FILE_NEXTSND	NeXT .snd and Sun .au
AF_FILE_WAVE	Waveform Audio File Format (RIFF)
AF_FILE_BICSF	Berkeley/IRCAM/CARL Sound
AF_FILE_MPEG1BITSTREAM	MPEG-1 audio bitstream encoded data
AF_FILE_SOUNDESIGNER2	Digidesign Sound Designer II
AF_FILE_AVR	Audio Visual Research
AF_FILE_IFF_8SVX	Amiga IFF/8SVX
AF_FILE_VOC	Creative Labs VOC
AF_FILE_SAMPLEVISION	Sample Vision
AF_FILE_SOUNDFONT2	Creative Labs E-mu Systems SoundFont2
AF_FILE_RAWDATA	Headerless data (for raw read-access mode only)

Initializing Audio Track Data

This section explains how to change the default settings for audio track parameters in an AFfilesetup structure before passing the structure to **afOpenFile()**.

Note: Each of the functions in this section contains a *track* argument, which identifies an audio track in the AFfilesetup structure being initialized. In the current release of the AF Library, the value of *track* must always be AF_DEFAULT_TRACK because all supported file formats contain only one audio track.

Initializing the Audio Track Format

Use a DMparams structure and the function **afInitFormatParams()** to initialize a specified audio track's data format in an AFfilesetup structure.

```
DMstatus afInitFormatParams ( AFfilesetup setup, int track,
                              DMparams *params )
```

The parameter *setup* is an AFfilesetup structure previously created by a call to **afNewFileSetup()**. The *track* parameter is an integer identifying an audio track in *setup*. Finally, *params* is a DMparams list previously created by a call to **dmParamsCreate()**. (See "Creating and Destroying DMparams Lists" in Chapter 3 for more information on creating DmParams lists.) If successful, **afInitFormatParams()** returns DM_SUCCESS, otherwise it returns DM_FAILURE.

The **afInitFormatParams()** function initializes all the parameters associated with an audio track's data in an AFfilesetup structure. The parameters and associated values are shown below.

DM_AUDIO_FORMAT	Sample format: DM_AUDIO_DOUBLE, DM_AUDIO_UNSIGNED, DM_AUDIO_TWOSCOMP, or DM_AUDIO_FLOAT
DM_AUDIO_WIDTH	Width in bits for integer sample formats. Integer value between 1 and 32, inclusive.
DM_AUDIO_CHANNELS	Channel count. Integer value greater than or equal to 1.
DM_AUDIO_RATE	Sampling rate. A double-precision floating point positive value.
DM_AUDIO_COMPRESSION	Compression type: DM_AUDIO_GSM, DM_AUDIO_UNCOMPRESSED, DM_AUDIO_G711_ULAW, DM_AUDIO_G711_ALAW, DM_AUDIO_G722, DM_AUDIO_G726, DM_AUDIO_G728, others
DM_AUDIO_PCM_MAP_SLOPE	Slope value for PCM mapping. See afGetVirtualPCMMapping(3dm).
DM_AUDIO_PCM_MAP_INTERCEPT	Intercept value for PCM mapping.
DM_AUDIO_PCM_MAP_MAXCLIP	Maximum clipping for PCM mapping.
DM_AUDIO_PCM_MAP_MINCLIP	Minimum clipping for PCM mapping.

Note: **afInitFormatParams()** replaces **afInitRate()**, **afInitSampleFormat()**, **afInitChannels()**, **afInitCompression()**, and **afInitCompressionParams()**.

Initializing AES Data

Audio Engineering Society (AES) channel status bytes are embedded in AES audio samples to provide additional information about that data, such as whether an emphasis has been added to a sample. For example, on early CD recordings, high frequencies were sometimes emphasized to compensate for the nature of CD players. You might want to reverse compensate for that emphasis if you are loading AES stream data directly from a CD player through the AES serial input of your workstation for playback on a different source, such as DAT. See the *AES3-1985 (ANSI S4.40-1985)* document for more information about AES channel status bytes.

afInitAESChannelDataTo() sets a flag, which is off by default, in an AFfilesetup structure to indicate that space should be reserved for the 24 AES channel status bytes that are embedded in all AES data. Its function prototype is:

```
void afInitAESChannelDataTo ( AFfilesetup setup, int track,
                              int usedata )
```

where *setup* is the AFfilesetup structure, and *track* is an integer that identifies an audio track in *setup*. The *usedata* parameter is a flag indicating whether AES data will be stored in the file.

Note: afInitAESChannelDataTo() replaces **afInitAESChannelData()**.

afSetAESChannelData() sets the values of the AES channel status bytes. Its function prototype is:

```
void afSetAESChannelData ( AFfilehandle file, int track,
                           unsigned char buf[24])
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *track* is the ID for the audio track and should always be AF_DEFAULT_TRACK, and *buf* is a 24-element array that specifies the AES channel status bytes. If no header space has been reserved in the file (by calling **afInitAESChannelDataTo()** before creating the file), **afSetAESChannelData()** ignores the data and returns without error.

Initializing Audio Track Markers

Audio track marker structures store sample frame locations in the track for looping and other purposes. Markers are identified by an integer ID number and (optionally) a name string. Markers point to a location between two samples in the audio track: position 0 is

before the first sample, position 1 is between the first and second sample, and so on. You can assign positions to the markers by calling **afSetMarkPosition()**. By default, **afNewFileSetup()** allocates space for four markers, which is sufficient to store the beginning and end points for both a sustain loop and a release loop.

afInitMarkIDs() initializes a list of unique marker IDs corresponding to marker structures in a given audio track. Its function prototype is:

```
void afInitMarkIDs ( AFfilesetup setup, int trackID, int markids[],
                     int nmarks )
```

where *setup* is the AFfilesetup structure, *trackID* is an integer that identifies an audio track in *setup*, *markids* is an array of unique positive integers that will be used as handles for the marker structures in the file opened with *setup*, *nmarks* is an integer that specifies the number of marker IDs in the *markids* array, that is, the total number of marker structures that will be allocated for the audio track.

afInitMarkName() specifies a name string for a marker structure. Marker names default to empty strings. Its function prototype is:

```
void afInitMarkName ( AFfilesetup setup, int track, int markid,
                      const char *namestr )
```

where *setup* is the AFfilesetup structure, *track* is an integer that identifies an audio track in *setup*, *markid* is a positive integer that identifies a marker structure configured previously by **afInitMarkIDs()**, *namestr* is a constant string that will be written into the marker structure when an audio file is created by passing *setup* to **afOpenFile()**.

Initializing Instrument Data

This section explains how to initialize the instrument parameters in an AFfilesetup structure before passing the structure to **afOpenFile()**.

afInitInstIDs() initializes a list of unique instrument IDs that are used to reference the instrument configurations in an AFfilesetup. Its function prototype is:

```
void afInitInstIDs ( AFfilesetup setup, int instids[], int ninsts )
```

where *setup* is the AFfilesetup structure, *instids* is an array of positive integers that are used as handles for the instrument configurations in an audio file, and *ninsts* is the number of entries in *instids*.

afInitLoopIDs() initializes a list of unique instrument loop IDs that correspond to the loops supplied for a specified instrument in an audio file. Its function prototype is:

```
void afInitLoopIDs ( AFfilesetup setup, int inst, int loopids[],
                     int nloops )
```

where *setup* is the AFfilesetup structure, *inst* is an integer that identifies an instrument configuration in an audio track. In the current release of the AF Library, the value of *inst* should always be AF_DEFAULT_INST. *loopids* is an array of unique, positive integers that will identify individual loops within an audio file opened using *setup*. *nloops* is a integer that indicates the number of elements in *loopids*.

The values set in *loopids* can be used by other AF Library functions to set the start point, end point, and play mode for each loop (see "Reading and Writing Instrument Configurations").

Note: In the current release of the AF Library, both AIFF and AIFF-C files must contain exactly 2 loops: a *sustain* loop and a *release* loop. *nloops* is currently ignored, since its value is always 2.

Initializing Miscellaneous Data

Use these functions to initialize miscellaneous data chunks in an AFfilesetup structure, including file name, author, copyright, and annotation strings, MIDI data, and application-specific data.

afInitMiscIDs() initializes a list of unique miscellaneous chunk IDs that are then used to reference various file format-dependent data chunks in an audio file. Its function prototype is:

```
void afInitMiscIDs ( AFfilesetup setup, int miscids[], int nmisc )
```

where *setup* is the AFfilesetup structure, *miscids* is an array of unique, positive integers used to reference the miscellaneous data chunks in an audio file opened using *setup*, *nmisc* is the number of elements in *miscids*, that is, the total number of miscellaneous chunks in the file configuration. The default number of miscellaneous IDs in an AFfilesetup structure is 0.

afInitMiscType() initializes a miscellaneous data chunk with a given ID to one of a variety of supported chunk types. Its function prototype is:

```
void afInitMiscType ( AFfilesetup setup, int miscid, int type )
```

where *setup* is the AFfilesetup structure, *miscid* is a positive integer that identifies a miscellaneous chunk in *setup*, and *type* is an integer constant that defines the chunk type.

Table 7-4 lists valid parameters for each chunk type.

Table 7-4 Miscellaneous Chunk Types and Parameter Values

Parameter Value	Miscellaneous Chunk Type
AF_MISC_ANNO	Annotation string.
AF_MISC_APPL	Application-specific data.
AF_MISC_AUTH	Author string.
AF_MISC_COPY	Copyright string.
AF_MISC_MIDI	MIDI data.
AF_MISC_NAME	Name string.
AF_MISC_PCMMAP	PCM mapping information.
AF_MISC_NeXT	NeXT file info chunk.
AF_MISC_IRCAM_PEAKAMP	BICSF peak amplitude sfcode.
AF_MISC_COMMENT	Text comment string. The tags AF_MISC_IRCAM_COMMENT and AF_MISC_ICMT are also allowed.
AF_MISC_ICRD	Creation date string. This is usually of the form "YYYY-MM-DD".
AF_MISC_ISFT	Software name string. Usually set to the name of the software package which created the sound.
AF_MISC_UNRECOGNIZED	Unrecognized data chunk.

afInitMiscSize() initializes the amount of space reserved for miscellaneous chunks of data in an AFfilesetup structure. This space is then reserved (written as a zero-filled area) in the header structure of an audio file that is opened using the specified AFfilesetup structure. The application program is responsible for managing the contents of the header space reserved for each chunk. Its function prototype is:

```
void afInitMiscSize ( AFfilesetup setup, int miscid, int size )
```

where *setup* is the AFfilesetup structure, *miscid* is a positive integer that identifies a miscellaneous chunk in *setup*, and *size* is a non-negative integer that specifies the number of bytes to reserve for the chunk data identified by *miscid*. It is not necessary to add a trailing "zero pad byte" normally required by chunks in AIFF/AIFF-C files with odd numbers of data bytes (see the description for **afReadMisc()**); the AF Library handles this transparently.

Initializing PCM Mapping

The function **afInitPCMMapping()** configures the PCM mapping for a specified audio track in an AFfilesetup structure. Its prototype is:

```
void afInitPCMMapping ( AFfilesetup setup, int track,
                        double slope, double intercept,
                        double minclip, double maxclip )
```

The *setup* parameter is an AFfilesetup structure, previously created by a call to **afNewFileSetup()**, which is passed to **afOpenFile()** when a new audio file is created. The parameter *track* is an integer which identifies an audio track in *setup*, and should always have a value of AF_DEFAULT_TRACK. *slope* is a positive double-precision floating point value that specifies an amplitude scaling factor for the waveform associated with *track*. The parameter *intercept* is a double-precision floating point value which specifies the sample value which represents the vertical midpoint (zero crossing) of the waveform associated with *track*. Finally, *minclip* and *maxclip* are double-precision floating point values specifying the maximum desired negative and positive amplitudes for the waveform. Any values encountered outside the range [*minclip*, *maxclip*] are clipped to these values.

To configure *setup* for a floating point waveform with a total expected amplitude of 100.0 (that is, values between -100.0 and 100.0), a symmetrical clipping to match, and a 0.0 zero crossing value, the track is initialized with as follows:

```
afInitPCMMapping ( setup, AF_DEFAULT_TRACK, 100.0, 0.0, -100.0, 100.0 );
```

PCM mapping is useful to modify frames only as they are read into or written from buffers with **afReadFrames()** or **afWriteFrames()** respectively. None of the currently supported file formats can store the PCM mapping information, even though the mapping can be applied to the frames stored in those files. Therefore, it is important to specify mapping values carefully. In general, all two's complement and floating point sample formats are expected to be symmetrical about zero. The *intercept* value must be

0.0, and *minclip* and *maxclip* must be negative and positive *n*, where *n* is a non-zero positive value.

Summary of the Creation and Configuration Functions

The following table is a summary of the Audio File Library creation and configuration functions. Functions that are not covered in the preceding text have a notation in the Description column. Please refer to a function's reference page if a more detailed explanation is needed. For example, the reference page for **afInitTrackIDs()** is afInitTrackIDs(3dm).

Table 7-5 Audio File Creation and Configuration Functions

Function	Description
AFfilesetup **afNewFileSetup** (void)	Create and initialize an AFfilesetup structure.
void **afFreeFileSetup** (AFfilesetup *setup*)	Deallocate an AFfilesetup structure.
void **afInitFileFormat** (AFfilesetup *setup*, int *filefmt*)	Initialize the audio file format type in an AFfilesetup structure.
DMstatus **afInitFormatParams** (AFfilesetup *setup*, int *track*, DMparams **params*)	Use a DMparams structure to initialize the audio data format in an AFfilesetup structure for a specified audio track.
void **afInitRate** (AFfilesetup *setup*, int *track*, double *rate*)	Obsolete. See **afInitFormatParams()**.
void **afInitSampleFormat** (AFfilesetup *setup*, int *track*, int *sampfmt*, int *sampwidth*)	Obsolete. See **afInitFormatParams()**.
void **afInitChannels** (AFfilesetup *setup*, int *track*, int *channels*)	Obsolete. See **afInitFormatParams()**.

Table 7-5 (continued) Audio File Creation and Configuration Functions

Function	Description
void **afInitCompression** (AFfilesetup *setup*, int *track*, int *compression*)	Obsolete. See **afInitFormatParams()**.
void **afInitCompressionParams** (AFfilesetup *setup*, int *track*, int *compression*, AUpvlist *pvlist*, int *numitems*)	Obsolete. See **afInitFormatParams()**.
void **afInitByteOrder**(AFfilesetup *setup*, int *track*, int *byteorder*)	Initialize the byte order data format in an AFfilesetup structure for a specified audio track. Not covered in text.
void **afInitAESChannelDataTo** (AFfilesetup *setup*, int *track*, int *usedata*)	Set a flag in an AFfilesetup structure to reserve or remove storage space for AES channel status data in a file.
void **afInitAESChannelData** (AFfilesetup *setup*, int *track*)	Obsolete. See **afInitAESChannelDataTo()**.
void **afInitMarkIDs** (AFfilesetup *setup*, int *trackID*, int *markids*[], int *nmarks*)	Specify a list of marker IDs for a new audio file in an AFfilesetup structure.
void **afInitMarkName** (AFfilesetup *setup*, int *track*, int *markid*, const char **namestr*)	Initialize the name for a specified marker in an AFfilesetup configuration structure.
void **afInitMarkComment** (AFfilesetup *setup*, int *track*, int *markid*, const char **commstr*)	Initialize the comment for a specified marker in an AFfilesetup configuration structure. Not covered in text.
void **afInitInstIDs** (AFfilesetup *setup*, int *instids*[], int *ninsts*)	Specify a list of instrument parameter chunk identifiers to be stored in an AFfilesetup configuration structure.

Table 7-5 (continued) Audio File Creation and Configuration Functions

Function	Description
void **afInitLoopIDs** (AFfilesetup *setup*, int *inst*, int *loopids*[], int *nloops*)	Initialize a list of loop IDs for a given instrument in an AFfilesetup structure.
void **afInitMiscSize** (AFfilesetup *setup*, int *chunkid*, int *size*)	Initialize the number of data bytes for a given miscellaneous chunk in an AFfilesetup structure.
void **afInitMiscIDs** (AFfilesetup *setup*, int *miscids*[], int *nmisc*)	Initialize the list of miscellaneous data chunk IDs in an AFfilesetup file configuration structure.
void **afInitMiscType** (AFfilesetup *setup*, int *chunkid*, int *type)*	Initialize the chunk type for a given miscellaneous chunk in an AFfilesetup structure.
void **afInitTrackIDs** (AFfilesetup *setup*, int *trackids*[], int *ntracks*)	Initialize the list of audio track identifiers in an AFfilesetup structure. Not covered in text.
void **afInitPCMMapping** (AFfilesetup *setup*, int *track*, double *slope*, double *intercept*, double *minclip*, double *maxclip*)	Configure the PCM mapping for an audio track in an AFfilesetup structure.
void **afInitDataOffset** (AFfilesetup *setup*, int *track*, AFfileoffset *offset*)	Initialize the audio data byte offset in an AFfilesetup for a specified raw-format audio track. Not covered in text.
void **afInitFrameCount** (AFfilesetup *setup*, int *track*, AFframecount *count*)	Initialize the audio frame count in an AFfilesetup for a specified raw-format audio track. Not covered in text.

Opening, Closing, and Identifying Audio Files

Before opening a new audio file using **afOpenFile()**, create and configure an appropriate AFfilesetup structure (as described in "Creating and Configuring Audio Files"). Audio files can be opened either for reading or for writing (but not both simultaneously). In order to change an existing file, you must copy the contents of the file to a new file, writing edits as you go.

Opening an Audio File

afOpenFile() allocates and initializes an AFfilehandle structure for a named file. The audio track logical read/write pointer used by **afReadFrames()** and **afWriteFrames()** is initialized to point to the location of the first sample in the audio file. Its function prototype is:

```
AFfilehandle afOpenFile ( const char *name, const char *mode,
                          AFfilesetup setup )
```

where *name* is a character string that names the file to be opened, and *mode* identifies whether the file is being opened for read or write access. Valid values for *mode* are:

- "r"—read-only access
- "w"—write-only access

setup is an AFfilesetup structure previously created using **afNewFileSetup()** and configured using various AF Library initialization functions described in previous sections. *setup* is ignored when *mode* is set to "r".

afOpenFile() returns an AFfilehandle structure for the named file. If an error occurs, **afOpenFile()** returns the value AF_NULL_FILEHANDLE.

Getting an IRIX File Descriptor for an Audio File

Another way of opening a file is to call the IRIX system function **open()** to open the file, and then get a handle to the file descriptor from the AF Library.

afOpenFD() returns an AFfilehandle structure for a file that has already been opened. Its function prototype is:

```
AFfilehandle afOpenFD ( int fd, char *mode, AFfilesetup setup )
```

where *fd* is an IRIX file descriptor previously returned by **open()**, *mode* identifies whether the file is being opened for read or write access (see **afOpenFile()**), and *setup* is an AFfilesetup structure previously created using **afNewFileSetup()** and configured using various AF Library initialization functions described in previous sections. *setup* is ignored when *mode* is set to "r".

afOpenFD() returns an AFfilehandle structure for the named file. If an error occurs, **afOpenFD()** returns the value AF_NULL_FILEHANDLE.

afGetFD() returns the IRIX file descriptor associated with the audio file referred to by the given AFfilehandle structure. Its function prototype is:

```
int afGetFD ( AFfilehandle file )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()**.

The file descriptor returned by **afGetFD()** is intended for use in a **select()** loop. It is not intended to allow reading, writing, and seeking in an audio file without the knowledge of the Audio File Library. Doing so causes unpredictable results unless you save and restore the file position whenever you modify it.

The AF does not reposition the file to the correct place before reading from (using **afReadFrames()**) or writing to (using **afWriteFrames()**) it. If you modify the file position of the file descriptor given by **afGetFD()**, you should save the file position and restore it to its previous position before reading or writing data to the file. Alternately, you can use one of two different file descriptors opened to the same file. The file must be re-opened in order to get a separate file descriptor (*dup*(2) will not work because it gives you two file descriptors that share the same file offset).

In addition, if you attempt to write to the file, no matter how the AFfilehandle was opened, the results are undefined.

Getting Audio File Format

This section describes functions that query the file format from either a file handle or from an IRIX file descriptor of an opened audio file.

afGetFileFormat() returns an integer value indicating the format of the file and returns a separate version number for AIFF-C files. Its function prototype is:

```
int afGetFileFormat ( AFfilehandle file, int *vers )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, and *vers* is used to return a file format version number in the form of a non-negative integer. AIFF files do not use version numbers, so a value of 0 is always returned as the AIFF version number.

afGetFileFormat() returns a non-negative integer indicating the format of the file. For a list of these format see **afInitFileFormat()** under "Initializing the Audio File Format."

afIdentifyFD() returns the file format of a given IRIX file descriptor. Its function prototype is:

```
int afIdentifyFD ( int fd )
```

where *fd* is an IRIX file descriptor previously returned by **open()**.

afIdentifyFD() returns a integer value representing the audio file format (see **afGetFileFormat()** for the return values for supported formats). If **afIdentifyFD()** does not recognize the format, AF_FILE_UNKNOWN is returned. If the format is not one supported by the AF Library, AF_FILE_UNSUPPORTED is returned.

To determine whether a file is a sound file that can be opened by the AF, check for an unrecognizable format rather than a recognizable format. For example, rather than testing whether the file format is explicitly known, use this code:

```
if (filefmt == AF_FILE_UNSUPPORTED ||
    filefmt == AF_FIILE_UNKNOWN)
        {
        printf("file is not supported by the AF library!");
    exit(0);
}
```

Applications that branch depending on the file format should still check for unrecognized formats:

```
switch (afIdentifyFD(fd))
{
case AF_FILE_AIFF: do_aiff_thing(); break;
case AF_FILE_AIFC: do_aiffc_thing(); break;
case AF_FILE_UNKNOWN:
case AF_FILE_UNSUPPORTED:
        printf("this file is not supported by AF library!!");
        exit(0);
```

```
default:
        printf("program cannot handle this file format!");
        exit(0);
}
```

Tip: Sometimes, instead of checking the file format, you should check the sampling format and other track parameters from the audio file track, as described in "Getting and Setting Audio Parameters." For example, a program that simply reads 16-bit AF_SAMPFMT_TWOSCOMP audio data out of an AIFF file should be able to correctly read that type of data out of a file whose file format is not AIFF, as long as it does not also intend to read AIFF-specific chunks from the data (for example, certain MISC and INST chunks). Such a program has no need to call **afIdentifyFD()** or **afGetFileFormat()** to get the file format.

Closing and Updating Files

afCloseFile() releases a file's resources back to the system. It also updates the headers of files opened for write access. The AFfilehandle structure deallocated by **afCloseFile()** should not be used by any subsequent AF Library function calls. Its function prototype is:

```
int afCloseFile ( AFfilehandle file )
```

where *file* is the AFfilehandle structure to be deallocated. This structure was returned by **afOpenFile()** when the file being closed was created.

afCloseFile() returns a negative value if an error occurs while closing a file and updating the header fields. If compression was used to write a file, a negative value indicates that some sample frames were lost due to the filter delay of the compressor. If no error occurs, the return value is 0.

afSyncFile() updates the complete contents of an audio file opened for writing without actually closing the file. This is useful for maintaining consistent header information between writing samples to the file's audio track. Its function prototype is:

```
int afSyncFile ( AFfilehandle file )
```

where *file* is the AFfilehandle structure to be updated. This structure was returned by **afOpenFile()** when the file being closed was created.

afSyncFile() returns a negative value if an error occurs while trying to update *file*. If the update is successful, or if *file* was opened as read-only, **afSyncFile()** returns 0**.**

Summary of the Opening, Closing, and Identifying Functions

The following table is a summary of the Audio File Library functions for opening, closing, and identifying files. Functions that are not covered in the preceding text have a notation in the Description column. Please refer to a function's reference page if a more detailed explanation is needed. For example, the reference page for **afSaveFilePosition()** is afSaveFilePosition(3dm)

Table 7-6 Audio File Opening, Closing and Identifying Functions

Function	Description
AFfilehandle **afOpenFile** (const char **name*, const char **mode*, AFfilesetup *setup*)	Allocate and initialize an AFfilehandle structure for an audio file identified by name.
AFfilehandle **afOpenFD** (int *fd*, const char **mode*, AFfilesetup *setup*)	Allocate and initialize an AFfilehandle structure for an audio file identified by a Unix file descriptor.
AFfilehandle **afOpenNamedFD** (int *fd*, const char **mode*, AFfilesetup *setup*, const char **name*)	Use instead of **afOpenFD()** for Sound Designer II files. Not covered in text.
int **afIdentifyFD** (int *fd*)	Retrieve the audio file format associated with a file descriptor.
int **afIdentifyNamedFD** (int *fd*, const char* *filename*, int* *implemented*)	Use instead of **afIdentifyFD()** for Sound Designer II files. Not covered in text.
int **afGetFD** (AFfilehandle *file*)	Get the UNIX file descriptor for the file associated with an AFfilehandle structure.
int **afGetFileFormat** (AFfilehandle *file*, int **vers*)	Retrieve the audio file format associated with an open AFfilehandle.

Table 7-6 (continued) Audio File Opening, Closing and Identifying Functions

Function	Description
int **afSyncFile** (AFfilehandle *file*)	Write out a snapshot of an audio file without closing the file.
void **afSaveFilePosition** (AFfilehandle *file*)	Save a logical audio sample read pointer to allow UNIX operations. Not covered in text.
void **afRestoreFilePosition** (AFfilehandle *file*)	Retrieve a logical audio sample read pointer after UNIX operations. Not covered in text.
int **afCloseFile** (AFfilehandle *file*)	Close an audio file; update the file header if the file was opened for write access.

Reading and Writing Audio Track Information

This section describes functions that read and manipulate audio track data and parameters in an audio file. Your application should query for audio file characteristics before opening a file and reading and writing data.

Getting and Setting Audio Parameters

Most audio track parameters (except markers) must be initialized before a new audio file is opened and cannot be modified after that point, but you should query an audio file for its track parameters.

Getting the Audio Track Format

The function **afGetFormatParams()** uses a DMparams structure to get the audio data format in an AFfilehandle for a specified audio track.

```
DMstatus afGetFormatParams ( AFfilehandle file, int track,
                             DMparams *params )
```

The parameter *file* is an AFfilehandle structure, previously created by a call to **afOpenFile()** or **afOpenFD()**. The *track* parameter is an integer which identifies an audio track in *file*, and should be set to AF_DEFAULT_TRACK. Finally, *params* is a DMparams list previously created by a call to **dmParamsCreate()**.

afGetFormatParams() retrieves all the parameters associated with an audio track's data in an AFfilehandle structure. For a table of these parameters see **afInitFormatParams()** under "Initializing the Audio File Format."

Note: **afGetFormatParams()** replaces the individual routines **afGetSampleFormat()**, **afGetChannels()**, **afGetRate()**, **afGetCompression()**, and **afGetCompressionParams()**.

Getting and Setting the Virtual Audio Format

The functions **afGetVirtualFormatParams()** and **afSetVirtualFormatParams()** use a DMparams structure to get and the virtual audio data format in an AFfilehandle for a specified audio track.

```
DMstatus afGetFormatParams ( AFfilehandle file, int track,
                             DMparams *params )

DMstatus afSetFormatParams ( AFfilehandle file, int track,
                             DMparams *params )
```

The parameter *file* is an AFfilehandle structure, previously created by a call to **afOpenFile()** or **afOpenFD()**. The *track* parameter is an integer which identifies an audio track in *file*, and should be set to AF_DEFAULT_TRACK. Finally, *params* is a DMparams list previously created by a call to **dmParamsCreate()**.

afGetVirtualFormatParams() retrieves and **afSetVirtualFormatParams()** sets all the parameters associated with an audio track's virtual data in an AFfilehandle structure. For a table of these parameters see the function **afInitFormatParams()** under "Initializing the Audio File Format."

Note: **afSetVirtualFormatParams()** replaces the routines **afSetVirtualSampleFormat()**, **afSetVirtualChannels()**, **afSetVirtualRate()**, and **afSetVirtualPCMMapping()**.

Getting AES Data

afGetAESChannelData() retrieves AES channel status information from an opened audio file. Its function prototype is:

```
int afGetAESChannelData ( AFfilehandle file, int track,
                          unsigned char buf[24] )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *track* is the ID for the audio track and should always be AF_DEFAULT_TRACK, and *buf* is a 24-element array that receives the AES channel status bytes.

afGetAESChannelData() returns a 1 if there is AES channel data, or a 0 if there is no data.

Tip: There is no guarantee whether a given file format will contain AES data, so your application should call **afGetAESChannelData()** to determine whether AES channel bytes are encoded in an audio file.

Getting Audio Track Sample Frame Count

afGetFrameCount() returns the total number of sample frames in the audio track of an opened audio file. Its function prototype is:

```
AFframecount afGetFrameCount ( AFfilehandle file, int track )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**. *track* is the ID for the audio track and is always AF_DEFAULT_TRACK.

afGetFrameCount() returns an AFramecount value that is the current total of sample frames in the track.

Getting and Setting Audio Track Markers

This section describes functions that get information about the markers in a given audio track and explains how to set the position of those markers. Markers point to positions between adjacent sample frames. For a track containing *n* sample frames, position 0 is before the first sample frame, and position *n* is after the last sample frame in the track.

afGetMarkIDs() retrieves an array of marker IDs from a given audio track in an opened audio file. It returns the number of marker structures in the specified audio track. Its function prototype is:

```
int afGetMarkIDs ( AFfilehandle file, int trackID, int *markids )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *trackID* is the ID for the audio track and should always be AF_DEFAULT_TRACK, and *markids* is an array of integers that receives the marker IDs for the marker structures in the audio track.

afGetMarkIDs() returns a non-negative integer value specifying the number of marker structures in the given audio track.

Tip: Check for unrecognized mark return values rather than recognized values. Write your application so that it expects any number of marks and any type of mark (not just the currently defined types) and rejects files containing marks it does not support.

Typically, you call **afGetMarkIDs()** twice. The first time, you pass *markids* a null pointer and check the return value of the function. This value tells you how many locations to allocate in the *markids* array, which you pass back to **afGetMarkIDs()** to obtain the list of marker IDs.

afGetMarkName() returns the name string of a given marker within the audio track of an opened audio file. Its function prototype is:

```
char *afGetMarkName ( AFfilehandle file, int trackID, int markid )
```

where *file* is an AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *trackID* is the ID of the audio track and is always AF_DEFAULT_TRACK. *markid* is the ID of the marker whose name you want to retrieve.

afGetMarkName() returns a null-terminated character string that is the name associated with the given *markid*.

afGetMarkPosition() returns the frame location of a given marker in the audio track of an opened audio file. Its function prototype is:

```
AFframecount afGetMarkPosition ( AFfilehandle file, int trackID,
                                 int markid )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *trackID* is the ID for the audio track and is always AF_DEFAULT_TRACK. *markid* is the ID of the marker whose position you want to discover.

afGetMarkPosition() returns a non-negative AFramecount value indicating the position of the marker in the track.

afSetMarkPosition() sets the frame location of a given marker in the audio track of an audio file opened for write access. Its function prototype is:

```
void afSetMarkPosition ( AFfilehandle file, int trackID, int markid,
                         AFframecount markpos )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *trackID* is the ID for the audio track and should always be AF_DEFAULT_TRACK, *markid* is the ID of the marker whose position you want to move, and *markpos* describes the position to which you want to move the marker in the track.

Seeking, Reading, and Writing Audio Track Frames

This section describes functions that position the read pointer in a file's audio track and functions that read and write frames. You can read and seek only from a file opened for reading. Similarly, you can write frames only to a file opened for writing.

Seeking to a Position in an Audio File Track

When a file is opened for read access by **afOpenFile()** or **afOpenFD()**, the logical track pointer for the audio track is initialized to point to the first frame in the track. This location can be changed by calling **afSeekFrame()**. Before returning, **afReadFrames()** moves the logical track pointer so that it points to the frame following the one last copied into *frames*.

Caution: The logical track pointer is not the same thing as the IRIX file pointer which you position by calling the IRIX *lseek*(2) command.

afSeekFrame() moves the logical track pointer in the audio track of an audio file opened for read-only access to a specified frame. Its function prototype is:

```
AFframecount afSeekFrame ( const AFfilehandle file, int track,
                           AFframecount frameoffset )
```

where *file* is a AFfilehandle structure created by a call to **afOpenFile()** or **afOpenFD()**, *track* is the audio track ID and should always be AF_DEFAULT_TRACK, *frameoffset* is the number of frames from the beginning of the track that the pointer will be moved to. This value is between 0 and the total number of frames in the track, minus 1. The total number of frames in the track can be determined by calling **afGetFrameCount()**.

When **afSeekFrame()** succeeds, it returns the actual offset value; otherwise, it returns a negative value.

Reading Audio Frames from an Audio Track

afReadFrames() copies sample frames from an audio file opened for reading to a buffer. Its function prototype is:

```
int afReadFrames ( const AFfilehandle file, int track,
                   void *samples, const int count )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *track* is the ID for the audio track and should always be AF_DEFAULT_TRACK, *samples* is a pointer to a buffer into which you want to transfer copies of sample frames from *file*, and *count* is the number of sample frames you want to read from *file*.

afReadFrames() returns an integer value indicating the number of frames successfully read from the audio track.

Data copied into *frames* must be interpreted using the sample format, sample width and channel count parameters returned by **afGetFormatParams()**. For AF_SAMPFMT_TWOSCOMP, **afReadFrames()** copies the frames to the buffer using the smallest data type (char, short, or int) that will hold the data, and automatically decompresses data encoded with any of the supported compression algorithms.

Tip: Query for the sample format, sample width, and channels. Don't assume that a particular file format determines the sample format, sample width, or number of channels. Provide a mechanism for detecting and handling unsupported file configurations.

Writing Audio Frames to an Audio Track

When a file is opened for write access by **afOpenFile()** or **afOpenFD()**, the logical track pointer for the file's audio track is initialized to point to the first frame in the track. Before returning, **afWriteFrames()** moves the logical track pointer so that it points to the frame following the one last copied into *samples*.

Caution: The logical track pointer is not the same thing as the IRIX file pointer which you position by calling the IRIX *lseek*(2) command.

afWriteFrames() copies frames from a buffer to an audio file opened for writing. Its function prototype is:

```
int afWriteFrames ( const AFfilehandle file, int track,
                    void* samples, const int count )
```

file is the AFfilehandle structure created by a call to **afOpenFile()** or **afOpenFD()**, *track* is an integer that identifies the audio track and should always be AF_DEFAULT_TRACK, *samples* is a pointer to a buffer containing sample frames that you want to write to *file*, and *count* is the number of sample frames you want to write to *file*.

For AF_SAMPFMT_TWOSCOMP data, **afWriteFrames()** expects the frames to be buffered using the smallest data type (char, short, or int) capable of holding the data. **afWriteFrames()** automatically compresses data encoded using any of the supported compression algorithms.

afWriteFrames() returns an integer value indicating the number of frames successfully written to the audio track. The return value is normally greater than or equal to 0; however, when a codec is being used and buffered data cannot be written to disk, that data is lost. In such a case, **afWriteFrames()** returns a negative value, indicating the number of sample frames lost.

Reading and Writing Instrument Configurations

Use the functions in this section to retrieve and manipulate instrument configuration data and parameters.

Getting and Setting Instrument Parameters

Use the functions described in this section to retrieve and set the instrument configuration parameters of an audio file. The parameters can be read from any opened audio file and written to any audio file opened as write-only.

afGetInstIDs() retrieves an array of instrument IDs corresponding to the instrument chunks in a given audio file. It returns the number of instrument chunks in the file. Its function prototype is:

```
int afGetInstIDs ( AFfilehandle file, int *instids )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, and *instids* is a pointer to the first member of an array of integer instrument IDs that reference instrument chunks within the file.

Typically, you call **afGetInstIDs()** twice. The first time, you pass *instids* a null pointer and check the return value of the function. This value tells you how many locations to allocate in the *instids* array, which you pass back to **afGetInstIDs()** to obtain the list of instrument IDs.

Tip: Write your application so that it checks for and rejects instrument configurations that you don't want to support.

afGetInstParamLong() retrieves a long instrument configuration parameter value from an instrument configuration in an open audio file. Its function prototype is:

```
long afGetInstParamLong ( AFfilehandle file, int instid, int param )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**. *instid* is the instrument ID for the instrument configuration chunk (for AIFF and AIFF-C files, this value should always be AF_DEFAULT_INST). *param* is a symbolic constant that identifies an instrument parameter. See Table 7-7 for a list of valid parameter constants and values associated with them.

afGetInstParamLong() returns the long integer value associated with the parameter specified in *param*. If *instid* or *param* is not valid, the value returned is 0.

Table 7-7 lists the instrument parameter constants and their valid values.

Table 7-7 Instrument Parameter Constants and Valid Values

Instrument Parameter Constant	Valid Values [Default]
AF_INST_MIDI_BASENOTE	MIDI base note for sample: 0-127. [60]
AF_INST_NUMCENTS_DETUNE	MIDI detune in cents: varies with file format (-50 to 50 for AIFC). [0]
AF_INST_MIDI LONOTE	Lowest MIDI note for sample: 0–127. [0]
AF_INST_MIDI_HINOTE	Highest MIDI note for sample: 0–127. [127]
AF_INST_MIDI_LOVELOCITY	Lowest MIDI velocity for sample: 1-127. [1]
AF_INST_MIDI_HIVELOCITY	Highest MIDI velocity for sample: 1–127. [127]
AF_INST_NUMDBS_GAIN	Gain in dB's for sample: -32768 to 32767. [0]
AF_INST_SUSLOOPID	ID for sustain loop (AIFF / AIFF-C only). [1]
AF_INST_RELLOOPID	ID for release loop (AIFF and AIFF-C only). [2]
AF_INST_SAMP_STARTFRAME	Inst's starting frame of sample: 0 or greater.
AF_INST_SAMP_ENDFRAME	Inst's ending frame of sample: 0 or greater.
AF_INST_SAMP_MODE	Inst's sample looping mode. If present, will be one of: AF_INST_LOOP_OFF, AF_INST_LOOP_CONTINUOUS, AF_INST_LOOP_SUSTAIN.
AF_INST_TRACKID	Track ID for inst sample data: AF_DEFAULT_TRACK.
AF_INST_NAME	Name string for instrument configuration. Type AU_PVTYPE_PTR.
AF_INST_SAMP_RATE	Sample rate for sample associated with the inst. Type AU_PVTYPE_DOUBLE.

Tip: Check for unrecognized instrument configuration and parameters rather than recognized types. Write your application so that it expects any type of instrument configuration (not just the currently defined types) and rejects files containing instruments it does not recognize.

afSetInstParamLong() writes a long instrument configuration parameter value to a given instrument configuration chunk in an audio file that has been opened for writing. Its function prototype is:

```
void afSetInstParamLong ( AFfilehandle file, int instid, int param,
                          long value )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *instid* is the instrument ID for the instrument configuration chunk (for AIFF and AIFF-C files, this value should always be AF_DEFAULT_INST), *param* is a symbolic constant that identifies an instrument parameter, and *value* is the long integer value you want to assign to parameter named by *param*. See Table 7-7 for a list of valid parameter constants and values associated with them.

Getting and Setting Loop Information

This section describes functions that retrieve and set the positions of instrument loops within an opened audio file. The loop information may be read from any opened audio file and written to any audio file opened as write-only. To get and set instrument loop IDs, use **afGetInstParamLong()** and **afSetInstParamLong()**, as described in "Reading and Writing Instrument Configurations."

afGetLoopMode() returns the loop mode of a given loop in the instrument configuration of an opened audio file. Its function prototype is:

```
int afGetLoopMode ( AFfilehandle file, int instid, int loopid )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *instid* is the instrument ID for the instrument configuration chunk (for AIFF and AIFF-C files, this value should always be AF_DEFAULT_INST), and *loopid* is the ID number associated with the loop whose mode you wish to read.

afGetLoopMode() returns an integer value representing the loop mode. Current valid values for loop mode are:

- AF_LOOP_MODE_NOLOOP (no loop)
- AF_LOOP_MODE_FORW (forward loop)
- AF_LOOP_MODE_FORWBAKW (alternating forward/backward)

afSetLoopMode() sets the loop mode of a given loop in the instrument configuration of an audio file opened as write-only. Its function prototype is:

```
void afSetLoopMode ( AFfilehandle file, int instid, int loopid,
                     int mode )
```

where file is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *instid* is the instrument ID for the instrument configuration chunk (for AIFF and AIFF-C files, this value should always be AF_DEFAULT_INST), *loopid* is the ID number associated with the loop whose mode you wish to write, and *mode* is the integer value you wish to set for the loop mode. See **afGetLoopMode()** for the list of valid *mode* values.

afGetLoopStart() returns an audio track marker ID associated with the starting point of a given instrument loop. Its function prototype is:

```
int afGetLoopStart ( AFfilehandle file, int instid, int loopid )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *instid* is the instrument ID for the instrument configuration chunk (for AIFF and AIFF-C files, this value should always be AF_DEFAULT_INST), and *loopid* is the ID number associated with the loop whose starting point you wish to read.

afGetLoopStart() returns an integer value, which is a marker ID in the audio track. See "Getting and Setting Audio Track Markers" in "Reading and Writing Audio Track Information" for information on how to manipulate the position of the markers referred to by the marker IDs.

afSetLoopStart() causes an audio track marker ID to be associated with the starting point of a given instrument loop. Its function prototype is:

```
void afSetLoopStart ( AFfilehandle file, int instid, int loopid,
                      int markid )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *instid* is the instrument ID for the instrument configuration chunk (for AIFF and AIFF-C files, this value should always be AF_DEFAULT_INST), *loopid* is the ID number associated with the loop whose starting point you wish to write, and *markid* is the audio track marker that you wish to assign as the starting point of the given loop.

afGetLoopEnd() returns an audio track marker ID associated with the ending point of a given instrument loop. Its function prototype is:

```
int afGetLoopEnd ( AFfilehandle file, int instid, int loopid )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *instid* is the instrument ID for the instrument configuration chunk (for AIFF and AIFF-C files, this value should always be AF_DEFAULT_INST), and *loopid* is the ID number associated with the loop whose ending point you wish to read.

afGetLoopEnd() returns an integer value which is a marker ID in the audio track. See "Getting and Setting Audio Track Markers" in "Reading and Writing Audio Track Information" for information on how to manipulate the position of the markers referred to by the marker IDs.

afSetLoopEnd() causes an audio track marker ID to be associated with the ending point of a given instrument loop. Its function prototype is:

```
void afSetLoopEnd ( AFfilehandle file, int instid, int loopid,
                    int markid )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *instid* is the instrument ID for the instrument configuration chunk (for AIFF and AIFF-C files, this value should always be AF_DEFAULT_INST), *loopid* is the ID number associated with the loop whose ending point you wish to write, and *markid* is the audio track marker that you wish to assign as the ending point of the given loop.

Tip: Loop queries can return any configuration of loops within an instrument, not just the fixed value of 2 in AIFF/AIFF-C files. Have your application check for and reject loop configurations it does not support.

Handling Miscellaneous Data Chunks

The following sections describe how to read to, write from, and get information about the miscellaneous data chunks in an audio file.

Getting Miscellaneous Data Parameters

This section describes functions that get information about the number, size and type of miscellaneous data chunks in an opened audio file.

afGetMiscIDs() returns the number of miscellaneous data chunks in a file and an array containing the IDs of each miscellaneous chunk. Its function prototype is:

```
int afGetMiscIDs ( AFfilehandle file, int miscids[] )
```

file is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**. *miscids* is an array of positive integers that contains the IDs for the miscellaneous data chunks in *file*.

afGetMiscIDs() returns a nonnegative integer value equal to the number of miscellaneous data chunks in *file*.

To fill the *miscids* array with the corresponding IDs, you first call **afGetMiscIDs()** with a null *miscids* pointer, and then allocate a *miscids* buffer according to the return value. You can then call **afGetMiscIDs()** again, passing the properly dimensioned *miscids* buffer to obtain the list of IDs.

afGetMiscType() returns the type of a given miscellaneous chunk. Its function prototype is:

```
int afGetMiscType ( AFfilehandle file, int chunkid )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, and *chunkid* is a positive integer miscellaneous chunk ID from the *miscids* array returned by **afGetMiscIDs()**.

afGetMiscType() returns a positive symbolic constant that describes the chunk type. See Table 7-4 for the list of valid chunk types and constants. If the chunk is not recognized, **afGetMiscType()** will return the value AF_MISC_UNRECOGNIZED.

Tip: The set of chunk types may expand at any time. Check for unrecognized chunk types rather than recognized chunk types. Write your application so that it expects any type of MISC chunk (not just the currently defined types) and rejects miscellaneous chunks it does not recognize.

afGetMiscSize() returns the size of a given miscellaneous data chunk in bytes. Its function prototype is:

```
int afGetMiscSize ( AFfilehandle file, int chunkid )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, and *chunkid* is a positive integer miscellaneous chunk ID from the *miscids* array returned by **afGetMiscIDs()**.

afGetMiscSize() returns an nonnegative integer value that describes the size of the data in the chunk in bytes. This number does not take into account null-terminators in strings,

so you will need to add one to the value returned when actually reading string data (see **afReadMisc()**).

Reading, Writing, and Seeking Miscellaneous Data

This section describes functions that read and write miscellaneous data and to position the read/write location pointer within the data portion of a miscellaneous chunk. The AFfilehandle structure maintains a logical read/write pointer for each miscellaneous data chunk in the file. Each pointer is initialized to point at the first data byte with the chunk when the AFfilehandle structure is created.

Tip: To avoid file corruption, don't copy MISC chunks from one file to another unless the content of those chunks is known. A chunk can contain references to other parts of the file that have been modified by the application, in which case attempting to copy it without properly modifying its contents would cause an error.

afReadMisc() reads data from a given miscellaneous chunk into a buffer, and returns the number of bytes read. Its function prototype is:

```
int afReadMisc ( const AFfilehandle file, int miscid, void *buf,
                 int nbytes )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *miscid* is a positive integer miscellaneous chunk ID from the *miscids* array returned by **afGetMiscIDs()**, *buf* is a pointer to a buffer that will receive the data from the miscellaneous chunk, and *nbytes* is the number of bytes you want to read from the audio file into *buf*, beginning at the current position of *file*'s logical read pointer for the data in *miscid*. **afReadMisc()** will not read past the end of the chunk's data area. After reading the data, **afReadMisc()** updates the position of the read/write pointer to point to the data byte following the last one read.

afWriteMisc() writes data from a buffer to a given miscellaneous chunk, and returns the number of bytes successfully written. Its function prototype is:

```
int afWriteMisc ( AFfilehandle file, int miscid, void *buf,
                  int nbytes )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *miscid* is a positive integer miscellaneous chunk ID from the *miscids* array returned by **afGetMiscIDs()**, *buf* is a pointer to a buffer that contains the data you want to write to the miscellaneous chunk, and *nbytes* is the number of bytes you want to write to the audio file from *buf*, beginning at the current position of *file*'s logical write pointer

for the data in *miscid*. **afWriteMisc()** will not write past the end of the chunk's data area. After writing the data, **afReadMisc()** updates the position of the read/write pointer to point to the data byte following the last one written.

It is up to the application to fill the data area of a chunk with consistent information (for example, if you don't use all the bytes you allocated in a MIDI data chunk, you need to fill the remaining bytes with no-ops).

afSeekMisc() moves the logical read/write pointer for a miscellaneous chunk to a specified offset from the beginning of the chunk's data area. Its function prototype is:

```
void afSeekMisc ( AFfilehandle file, int chunkid, int offbytes )
```

where *file* is the AFfilehandle structure previously created by a call to **afOpenFile()** or **afOpenFD()**, *chunkid* is a positive integer miscellaneous chunk ID from the *miscids* array returned by **afGetMiscIDs()**, *offbytes* is a non-negative integer specifying the number of bytes past the start of the data area the read/write pointer should be moved, and *offbytes* should always be less than the size of the total data area (in bytes).

afSeekMisc() returns the new location of the logical read/write pointer, measured as the number of bytes from the beginning of the chunk data area.

Handling PCM Data

This section discusses the functions that get and set PCM mapping values for tracks and buffers. PCM mapping is useful for modifying frames only as they are read into or written out of a buffer with **afReadFrames()** or **afWriteFrames()**. None of the supported file formats can store this information, even though the PCM mapping can be applied to frames stored in the files. More information about PCM mapping can be found in the "PCM Mapping" and "Initializing PCM Mapping" sections. See also the reference page afIntro(3dm).

The functions **afGetPCMMapping()** and **afGetVirtualPCMMapping()** get the track and virtual PCM mapping values respectively from an AFfilehandle structure for a specified audio track.

```
void afGetPCMMapping ( AFfilehandle file, int track,
                       double *slope, double *intercept,
                       double *minclip, double *maxclip )
```

```
void afGetVirtualPCMMapping ( AFfilehandle file, int track,
                              double *slope, double *intercept,
                              double *minclip, double *maxclip )
```

The *file* parameter is an AFfilehandle structure previously created by a call to **afOpenFile()**. *track* is an integer that identifies an audio track in *file*. Since all supported file formats contain one audio track per file, *track* is the constant AF_DEFAULT_TRACK.

slope is a pointer to a double precision floating point value that specifies the amplitude scaling factor for the audio waveform associated with *track*. *intercept* is a pointer to a double precision floating point value that specifies the audio waveform's vertical midpoint (zero-crossing) value.

minclip is a pointer to a double precision floating point value that indicates the minimum audio data sample value to be returned. Any value less than this is reset to *minclip*. *maxclip* is a pointer to a double precision floating point value that indicates the maximum audio data sample value to be returned. Any value greater than this is reset to *maxclip*. If *maxclip* is less than or equal to *minclip*, no clipping will be done. This means all PCM values are legal, even if they are outside the full-voltage range (see "PCM Mapping").

Using the double precision pointer arguments, **afGetPCMMapping()** returns the four values associated with the PCM (Pulse Code Modulation) mapping in the audio track of *file*. **afGetVirtualPCMMapping()** does the same for the virtual format of the given track. The track's virtual values were set by calls to **afSetVirtualFormatParams()** and **afSetTrackPCMMapping()**.

The function **afSetTrackPCMMapping()** modifies the current PCM mapping values associated with a given track in an AFfilehandle.

```
int afSetTrackPCMMapping ( AFfilehandle file, int track,
                           double slope, double intercept,
                           double minclip, double maxclip )
```

The parameters are the same as for **afGetPCMMapping()**. Because none of the supported file formats can store PCM mapping information, it is important to specify the parameter values carefully. In general, all two's complement and floating point sample formats are expected to be symmetrical about zero. The *intercept* value should be 0.0, and *minclip* and *maxclip* should be negative and positive N, where N is some non-zero positive value.

An application specifies a PCM mapping for the virtual format and optionally for the track format. The following pseudo-code shows how the AF maps each input sample

value, “in_pcm,” to an output sample value, “out_pcm.” For an AFfilehandle opened for input, “in_pcm” is the track and “out_pcm” is the buffer. For an AFfilehandle opened for output, “in_pcm” is the buffer and “out_pcm” is the track.

Example 7-2 Audio File PCM Mapping Pseudo-Code

```
/* transform in_pcm to volts */
if (in_maxclip > in_minclip) {
    if (in_pcm < in_minclip) in_pcm = in_minclip;
    if (in_pcm > in_maxclip) in_pcm = in_maxclip;
}
volts = (in_pcm - in_intercept) / in_slope;

/* transform volts to out_pcm */
out_pcm = out_intercept + out_slope * volts;
if (out_maxclip > out_minclip) {
   if (out_pcm < out_minclip) out_pcm = out_minclip;
   if (out_pcm > out_maxclip) out_pcm = out_maxclip;
}
```

Summary of the Audio Track Reading and Writing Functions

The following table is a summary of the Audio File Library functions for reading and writing audio tracks. Functions not covered in the preceding text have a notation in the Description column. Refer to a function’s reference page for a more detailed explanation. For example, the reference page for **afGetPCMMapping()** is afGetPCMMapping(3dm)

Table 7-8 Audio File Track Reading and Writing Functions

Function	Description
DMstatus **afGetFormatParams** (AFfilehandle *file*, int *track*, DMparams **params*)	Get the audio data format in an AFfilehandle for a specified audio track via DMparams.
double **afGetRate** (AFfilehandle *file*, int *track*)	Obsolete. See **afGetFormatParams()**.
double **afGetVirtualRate** (AFfilehandle *file*, int *track*)	Get the virtual sample rate for a specified audio track from an AFfilehandle structure. Not covered in text.

Table 7-8 (continued) Audio File Track Reading and Writing Functions

Function	Description
void **afGetSampleFormat** (AFfilehandle *file*, int *track*, int **sampfmt*, int **sampwidth*)	Obsolete. See **afGetFormatParams()**.
void **afGetVirtualSampleFormat** (AFfilehandle *file*, int *track*, int **sampfmt*, int **sampwidth*)	Get the virtual sample format for a specified audio track from an AFfilehandle structure. Not covered in text.
int **afGetByteOrder** (AFfilehandle *file*, int *track*)	Get the byte order for a specified audio track from an AFfilehandle structure. Not covered in text.
int **afGetVirtualByteOrder** (AFfilehandle *file*, int *track*)	Get the virtual byte order for a specified audio track from an AFfilehandle structure. Not covered in text.
int **afGetChannels** (AFfilehandle *file*, int *track*)	Obsolete. See **afGetFormatParams()**.
int **afGetVirtualChannels** (AFfilehandle *file*, int *track*)	Get the number of interleaved virtual channels from an AFfilehandle structure for an audio track. Not covered in text.
int **afGetAESChannelData** (AFfilehandle *file*, int *track*, unsigned char *buf*[24])	Get AES channel status information in an AFfilehandle structure for an audio track.
void **afSetAESChannelData** (AFfilehandle *file*, int *track*, unsigned char *buf*[24])	Set AES channel status information in an AFfilehandle structure for an audio track. Not covered in text.
int **afGetCompression** (AFfilehandle *file*, int *trackid*)	Obsolete. See **afGetFormatParams()**.
int **afGetCompressionParams** (AFfilehandle *file*, int *trackid*, int **compression*, AUpvlist *pvlist*, int *numitems*)	Obsolete. See **afGetFormatParams()**.

Table 7-8 (continued) Audio File Track Reading and Writing Functions

Function	Description
AFframecount **afGetFrameCount** (AFfilehandle *file*, int *track*)	Get the total sample frame count for a specified audio track from an AFfilehandle structure.
AFfileoffset **afGetTrackBytes** (AFfilehandle *file*, int *track*)	Get the total data bytes for a specified audio track from an AFfilehandle structure. Not covered in text.
int **afGetDataOffset** (AFfilehandle *file*, int *track*)	Get the total data offset for a specified audio track from an AFfilehandle structure. Not covered in text.
int **afGetMarkIDs** (AFfilehandle *file*, int *trackID*, int **markids*)	Get the number and list of marker IDs for an audio track.
char* **afGetMarkName** (AFfilehandle *file*, int *trackID*, int *markid*)	Get the name string for a given marker ID in an audio track.
char* **afGetMarkComment** (AFfilehandle *file*, int *trackID*, int *markid*)	Get the comment string for a given marker ID in an audio track. Not covered in text.
AFframecount **afGetMarkPosition** (AFfilehandle *file*, int *trackID*, int *markid*)	Get the position of a marker in an audio track.
void **afSetMarkPosition** (AFfilehandle *file*, int *trackID*, int *markid*, AFframecount *markpos*)	Set the position of a marker in an audio track.
AFframecount **afSeekFrame** (const AFfilehandle *file*, int *track*, AFframecount *frameoffset*)	Move the logical file read pointer for a specified audio track to a desired sample frame location.
AFframecount **afTellFrame** (const AFfilehandle *file*, int *track*)	Retrieve current value of file read or write pointer. Not covered in text.

Table 7-8 (continued) Audio File Track Reading and Writing Functions

Function	Description
int **afReadFrames** (const AFfilehandle *file*, int *track*, void **samples*, const int *count*)	Read sample frames from a specified audio track in an audio file.
int **afWriteFrames** (const AFfilehandle *file*, int *track*, void **samples*, const int *count*)	Write audio sample frames to a specified track in an audio file.
int **afGetTrackIDs** (AFfilehandle *file*, int *trackids*[])	Get the list of track descriptor IDs for the given AFfilehandle. Not covered in text.
int **afGetInstIDs** (AFfilehandle *file*, int **instids*)	Get a list of instrument configurations from an AFfilehandle.
long **afGetInstParamLong** (AFfilehandle *file*, int *instid*, int *param*)	Get long parameter value for an instrument configuration in an AFfilehandle structure.
void **afGetInstParams** (AFfilehandle *file*, int *instid*, AUpvlist *pvlist*, int *nparams*)	Get a parameter list for an instrument configuration in an AFfilehandle structure. Not covered in text.
void **afSetInstParams** (AFfilehandle *file*, int *instid*, AUpvlist *pvlist*, int *nparams*)	Set a parameter list for an instrument configuration in an AFfilehandle structure. Not covered in text.
void **afSetInstParamLong** (AFfilehandle *file*, int *instID*, int *param*, long *value*)	Set a long parameter value for an instrument configuration in an AFfilehandle structure.
int **afGetLoopStart** (AFfilehandle *file*, int *instID*, int *loopid*)	Get the start marker from an AFfilehandle structure for a specified loop.
int **afGetLoopEnd** (AFfilehandle *file*, int *instID*, int *loopid*)	Get the end marker from an AFfilehandle structure for a specified loop.

Table 7-8 (continued) Audio File Track Reading and Writing Functions

Function	Description
int **afGetLoopTrack** (AFfilehandle *file*, int *instID*, int *loopid*)	Get the track from an AFfilehandle structure for a specified loop. Not covered in text.
int **afGetLoopMode** (AFfilehandle *file*, int *instID*, int *loopid*)	Get the play mode from an AFfilehandle structure for a specified loop.
int **afGetLoopIDs** (AFfilehandle file, int instID, int loopids[])	Get a number and list of loop IDs for an instrument configuration. Not covered in text.
AFframecount **afGetLoopStartFrame** (AFfilehandle *file*, int *instID*, int *loopid*)	Get the start frame from an AFfilehandle structure for a specified loop. Not covered in text.
AFframecount **afGetLoopEndFrame** (AFfilehandle *file*, int *instID*, int *loopid*)	Get the end frame from an AFfilehandle structure for a specified loop. Not covered in text.
int **afGetLoopCount** (AFfilehandle *file*, int *instID*, int *loopid*)	Get the loop count in an AFfilehandle structure for a specified loop. Not covered in text.
int **afSetLoopStartFrame** (AFfilehandle *file*, int *instID*, int *loopid*, AFframecount *startframe*)	Set the start frame in an AFfilehandle structure for a specified loop. Not covered in text.
int **afSetLoopEndFrame** (AFfilehandle *file*, int *instID*, int *loopid*, AFframecount *endframe*)	Set the end frame in an AFfilehandle structure for a specified loop. Not covered in text.
int **afSetLoopCount** (AFfilehandle *file*, int *instID*, int *loopid*, int *count*)	Set the loop count in an AFfilehandle structure for a specified loop. Not covered in text.
void **afSetLoopStart** (AFfilehandle *file*, int *instID*, int *loopid*, int *markid*)	Set the start marker in an AFfilehandle structure for a specified loop.

Table 7-8 (continued) Audio File Track Reading and Writing Functions

Function	Description
void **afSetLoopEnd** (AFfilehandle *file*, int *instID*, int *loopid*, int *markid*)	Set the end marker in an AFfilehandle structure for a specified loop.
void **afSetLoopTrack** (AFfilehandle *file*, int *instID*, int *loopid*, int *trackID*)	Set the track in an AFfilehandle structure for a specified loop. Not covered in text.
void **afSetLoopMode** (AFfilehandle *file*, int *instID*, int *loopid*, int *mode*)	Set the play mode in an AFfilehandle structure for a specified loop.
int **afGetMiscIDs** (AFfilehandle *file*, int *miscids*[])	Get the number and a list of miscellaneous chunk IDs for a file.
int **afGetMiscType** (AFfilehandle *file*, int *chunkid*)	Get the data type for a miscellaneous data chunk.
int **afGetMiscSize** (AFfilehandle *file*, int *chunkid*)	Get the size for a miscellaneous data chunk.
int **afReadMisc** (const AFfilehandle *file*, int *miscid*, void **buf*, int *nbytes*)	Read data from a miscellaneous chunk in an audio file.
int **afWriteMisc** (const AFfilehandle *file*, int *miscid*, void **buf*, int *nbytes*)	Write data to a miscellaneous chunk in an audio file.
int **afSeekMisc** (const AFfilehandle *file*, int *chunkid*, int *offbytes*)	Move logical read/write pointer for data in a miscellaneous chunk in an audio file.
int **afGetFrameSize** (AFfilehandle *file*, int *track*, int *extend3to4*)	Get the track frame size in bytes for a specified audio track from an AFfilehandle structure. Not covered in text.

Table 7-8 (continued) Audio File Track Reading and Writing Functions

Function	Description
int **afGetVirtualFrameSize** (AFfilehandle *file*, int *track*, int *extend3to4*)	Get the virtual frame size in bytes for a specified audio track from an AFfilehandle structure. Not covered in text.
int **afSetChannelMatrix** (AFfilehandle *file*, int *track*, double* *matrix*)	Set the channel mix matrix associated with a given track in an AFfilehandle. Not covered in text.
DMstatus **afGetVirtualFormatParams** (AFfilehandle *file*, int *track*, DMparams **params*)	Get the virtual audio data format in an AFfilehandle for a specified audio track with a DMparams structure.
DMstatus **afSetVirtualFormatParams** (AFfilehandle *file*, int *track*, DMparams **params*)	Set the virtual audio data format in an AFfilehandle for a specified audio track with a DMparams structure.
int **afSetVirtualSampleFormat** (AFfilehandle *file*, int *track*, int *sampfmt*, int *sampwidth*)	Obsolete. See **afSetVirtualFormatParams().**
int **afSetVirtualChannels** (AFfilehandle *file*, int *track*, int *channels*)	Obsolete. See **afSetVirtualFormatParams().**
int **afSetVirtualPCMMapping** (AFfilehandle *file*, int *track*, double *slope*, double *intercept*, double *minclip*, double *maxclip*)	Obsolete. See **afSetVirtualFormatParams().**
int **afSetVirtualRate** (AFfilehandle *file*, int *track*, double *rate*)	Obsolete. See **afSetVirtualFormatParams().**
int **afSetVirtualByteOrder** (AFfilehandle *file*, int *track*, int *byteorder*)	Set the virtual data format for a specified audio track. Not covered in text.
DMstatus **afSetConversionParams** (AFfilehandle *file*, int *track*, DMparams **params*)	Set the parameters associated with format conversion for a specified audio track with a DMparams structure. Not covered in text.

Table 7-8 (continued) Audio File Track Reading and Writing Functions

Function	Description
DMstatus **afGetConversionParams** (AFfilehandle *file*, int *track*, DMparams **params*)	Get the parameters associated with format conversion for a specified audio track with a DMparams structure. Not covered in text.
int **afSetTrackPCMMapping** (AFfilehandle *file*, int *track*, double *slope*, double *intercept*, double *minclip*, double *maxclip*)	Override the current PCM mapping values associated with a given track in an AFfilehandle.
void **afGetPCMMapping** (AFfilehandle *file*, int *track*, double **slope*, double **intercept*, double **minclip*, double **maxclip*)	Get the track PCM mapping values for a specified audio track from an AFfilehandle structure.
void **afGetVirtualPCMMapping** (AFfilehandle *file*, int *track*, double **slope*, double **intercept*, double **minclip*, double **maxclip*)	Get the virtual PCM mapping values for a specified audio track from an AFfilehandle structure.

Audio File Library Programming Tips

This section describes important Audio File Library programming tips:

- "Minimizing Data and File Format Dependence" describes how to maximize application compatibility by minimizing format dependence.
- "Preventing Concurrent Access from Multiple Threads" explains how to write a multithreaded AF application in order to prevent simultaneous access to an AFfilehandle from multiple threads.
- "Handling Errors in Multithreaded Applications" explains how to prevent an error handler from reporting simultaneous errors from a multithreaded application.

Minimizing Data and File Format Dependence

As the AF Library evolves to support new file formats and new data formats, file-format dependent applications will require more modifications to maintain compatibility than file-format independent programs. Making your application file format independent decreases the likelihood of compatibility problems with future releases of the library and

minimizes future modifications. Programming tips presented throughout this chapter call attention to methods you can use to make your application format independent.

Preventing Concurrent Access from Multiple Threads

The AF is not multithread/multiprocessor safe. Making multiple, simultaneous, uncoordinated AF calls on *different* AFfilehandles from different threads is possible and correct. Each AFfilehandle completely encapsulates the state (except for error handling, which is global) needed to perform operations on that AFfilehandle. In contrast, making multiple, simultaneous, uncoordinated AF calls on *the same* AFfilehandle from different threads is currently possible, but it is *not* proper programming practice.

In the following code, two threads are using one AFfilehandle:

Thread 1

- Some amount of time
- No semaphore locking
-

```
afSeekFrame(h,track,place1);
afReadFrames(h,track,...);
```

- Some amount of time
- No semaphore locking
-

Thread2

- Some amount of time
- No semaphore locking
-

```
afSeekFrame(h,track,place2);
afReadFrames(h,track,...);
```

- Some amount of time
- No semaphore locking
-

It is possible that these calls would be executed in the following order, in which case both threads would read the wrong data:

```
afSeekFrame(h,track,place1);  ||
afReadFrames(h,track,...);    ||  afSeekFrame(h,track,place2);
                              ||  afReadFrames(h,track,...);
```

The only way to ensure that concurrent operations take place in the correct order is to use a process coordination facility such as semaphore locking.

Proper multithreading looks like this:

Thread 1	Thread 2
•	•
• Some amount of time	• Some amount of time
•	•
Lock Semaphore that guards h	Lock Semaphore that guards h
`afSeekFrame(h,track,place1);`	`afSeekFrame(h,track,place2);`
`afReadFrames(h,track,...);`	`afReadFrames(h,track,...);`
Unlock Semaphore that guards h	Unlock Semaphore that guards h
•	•
• Some amount of time	• Some amount of time
•	•

IRIX guarantees that only one of the Lock Semaphore calls will succeed immediately. The thread whose lock does not succeed waits in the Lock Semaphore call (and thus does not proceed to the **afSeekFrame()** call) until the other thread has unlocked the semaphore (after it has finished seeking and reading). When the first thread unlocks the semaphore, the thread that is waiting can now proceed.

Follow these steps to add semaphore locking to a multithreaded application:

1. Use *usnewsema*(3P) to code to create a semaphore whose value is 1.
2. Use *uspsema*(3P) to lock the semaphore.
3. Use *usvsema*(3P) to unlock the semaphore.

Example 7-3 is a code fragment that demonstrates how to create a semaphore for protecting critical regions.

Example 7-3 Creating a Semaphore

```
#include <ulocks.h>

AFfilehandle h;      /* global file handle */
usema_t *HSema;      /* global semaphore to protect h */

/* Initialize semaphore support -- do this once. */
```

```
{
usptr_t *usptr;
char *arenafile;

/* Use the fastest type (nondebugging) semaphores. */
usconfig(CONF_LOCKTYPE, US_NODEBUG);

/* Create a shared arena to hold the semaphore.   */

arenafile = tmpnam(NULL);
usptr = usinit(arenafile);

/*
Create the semaphore with count 1 in that arena.
There is 1 resource (h) initially available.     */

HSema = usnewsema(usptr,1);

/* No need to refer to arena again, so unlink file */

unlink(arenafile);
}
```

To use the semaphore created in Example 7-3 do this:

Thread 1

-
- Some amount of time
-

```
uspsema(HSema); /* lock */
afSeekFrame(h,track,place1);
afReadFrames(h,track,...);
usvsema(HSema); /* unlock */
```

-
- Some amount of time
-

Thread 2

-
- Some amount of time
-

```
uspsema(HSema); /* lock */
afSeekFrame(h,track,place2);
afReadFrames(h,track,...);
usvsema(HSema); /* unlock */
```

-
- Some amount of time
-

Semaphore locking can prevent a worst-case scenario such as seeking from the second thread before the first thread has finished reading. Currently, an AF application without semaphores might not cause any problems when making simultaneous, uncoordinated AF calls on the same AFfilehandle from different threads. But this is because—by chance—the CPU scheduler timing has arranged the process timing so that both threads don't use the handle at the same time. Another time, the CPU scheduling might not be favorable, so it's best to protect the critical regions with semaphores.

In summary, you cannot make multiple, simultaneous, uncoordinated AF calls on the same AFfilehandle from different threads, even if the order of execution of those calls does not matter. Doing so is likely to cause a core dump, or at least corruption of the AFfilehandle. The application is responsible for implementing any semaphore protection that is needed; such protection is not built in to the AF calls themselves.

Handling Errors in Multithreaded Applications

You cannot make multiple, simultaneous, uncoordinated AF calls from different threads that affect the library's global state—namely, the error handler function. If two threads simultaneously try to set the error handler (even if it is the same error handler), the behavior is undefined.

If you write your own error handler and then make multiple, simultaneous, uncoordinated AF calls on different file handles from different threads (and both AF calls issue an error simultaneously), then two instances of your error handler are called in a simultaneous, uncoordinated manner in both threads. If this situation is possible in your program, you should use semaphores in your error handler (in addition to the semaphores in your main program) to prevent simultaneous error reporting or handling.

Handling Audio File Library Errors

The AF Library provides an error handling mechanism that directs error messages to *stderr*. You can replace the default AF Library error handler with one of your own.

afSetErrorHandler() lets you replace the default error handler function with one of your own. Its function prototype is:

```
AFerrfunc afSetErrorHandler ( AFerrfunc errfunc )
```

where *errfunc* is a pointer to an alternate error handling routine of type AFerrfunc that is declared as:

```
void errfunc ( long arg1, const char* arg2 )
```

The AF library error handler function pointer is declared as a global variable, and therefore is not safe for use in multi-threaded applications. Specifically, a core dump may result if more than one thread attempts to use the error handler simultaneously. This can be avoided by calling **afSetErrorHandler(**NULL**)** to disable the feature entirely. The current version of the AF library has a MT-safe alternative to the AFerrorhandler:

After disabling the internal error handler as shown above, an application should handle errors as in the example below:

```
#include <dmedia/dmedia.h>

AFfilehandle handle;

handle = afOpenFile ( “a_filename”, “r”, NULL ); /* attempt to open */
if(handle == NULL) {
    char detail[DM_MAX_ERROR_DETAIL]; /* storage for error detail */
    int errorNumber;                  /* error token storage */
    char *msg;                        /* short error message */

    msg = dmGetError ( &errorNumber, detail );
    if(msg != NULL)                   /* error was reported */
        fprintf ( stderr, “%s [error number %d]\\n”,
                  detail, errorNumber );
    exit(1); /* or whatever */
}
```

The application must add **-ldmedia** to its link list if it calls **dmGetError()**.

Appendix A

Digital Media Conversion Libraries

This appendix contains the APIs of the individual image and audio conversion libraries. These libraries are not discussed in detail. As discussed in Chapter 6, "Digital Media Data Conversion," most developers need only use the Image Conversion Library and the Audio Conversion Library. Those two libraries call the libraries described in this appendix as needed.

The Color Space Library

The Color Space Library (CSL) provides developers with the ability to convert the color spaces, packings, subsamplings, and data types of images. It also enables them to perform operations on image data, such as adjusting contrast. Like the Image Conversion Library, the CSL uses parameters describing the source and destination images as well as the conversion settings to create a *color converter* The image parameters are set using a **DMparams** structure using **dmColorSetSrcParams()** and **dmColorSetDstParams()**. The conversion parameters are set with **dmColorSetConvParams()**. In most cases, only the source and destination packings need to be specified; all other values will default to appropriate values. The CSL supports the RGB, YCrCb, and Y (Luminance) color spaces.

Table A-1 The Color Space Library API

Function	Description
DMstatus **dmColorConvert** (const DMcolorconverter *converter*, void **srcImage*, void **dstImage*)	Perform the image conversion See also dmColorConvert(3dm).
DMstatus **dmColorCreate** (DMcolorconverter **converter*)	Create and initialize the color converter. See also dmColorCreate(3dm).
DMstatus **dmColorDestroy** (const DMcolorconverter *converter*)	Destroy the color converter. See also dmColorDestroy(3dm).

Table A-1 (continued) The Color Space Library API

Function	Description
DMstatus **dmColorGetError** (const DMcolorconverter *converter*, int **error*)	Return the value of the error flag. See also dmColorGetError(3dm).
const char * **dmColorGetErrorString** (const int *error*)	Returns a text error message. See also dmColorGetErrorString(3dm).
DMstatus **dmColorGetSrcSize** (const DMcolorconverter *converter*, int **size*)	Get the source image size in bytes. See also dmColorGetSrcSize(3dm).
DMstatus **dmColorGetDstSize** (const DMcolorconverter *converter*, int **size*)	Get the destination image size in bytes. See also dmColorGetSrcSize(3dm).
DMstatus **dmColorPrecompute** (const DMcolorconverter *converter*)	Perform any required early computations. See also dmColorPrecompute(3dm).
DMstatus **dmColorSetBrightness** (const DMcolorconverter *converter*, const float *brightness*)	Set the brightness delta value. See also dmColorSetBrightness(3dm).
DMstatus **dmColorGetBrightness** (const DMcolorconverter *converter*, float **brightness*)	Get brightness delta value. See also dmColorSetBrightness(3dm).
DMstatus **dmColorSetContrast** (const DMcolorconverter *converter*, const float *contrast*)	Set the contrast multiplier. See also dmColorSetContrast(3dm).
DMstatus **dmColorGetContrast** (const DMcolorconverter *converter*, float **contrast*)	Get the contrast multiplier. See also dmColorSetContrast(3dm).

Table A-1 (continued) The Color Space Library API

Function	Description
DMstatus **dmColorSetDefaultAlpha** (const DMcolorconverter *converter*, const float *defaultAlpha*)	Set the default alpha value of the source image. See also dmColorSetDefaultAlpha(3dm).
DMstatus **dmColorGetDefaultAlpha** (const DMcolorconverter *converter*, float **defaultAlpha*)	Get the default alpha value of the source image. See also dmColorSetDefaultAlpha(3dm).
DMstatus **dmColorSetHue** (const DMcolorconverter *converter*, const float *hue*)	Set the hue rotation. See also dmColorSetHue(3dm).
DMstatus **dmColorGetHue** (const DMcolorconverter *converter*, float **hue*)	Get the hue rotation. See also dmColorSetHue(3dm).
DMstatus **dmColorSetSaturation** (const DMcolorconverter *converter*, const float *saturation*)	Set the saturation multiplier. See also dmColorSetSaturation(3dm).
DMstatus **dmColorGetSaturation** (const DMcolorconverter *converter*, float **saturation*)	Get the saturation multiplier. See also dmColorGetSaturation(3dm).
DMstatus **dmColorSetSrcParams** (const DMcolorconverter *converter*, DMparams **scrParams*)	Set the source image parameters. See also dmColorSetSrcParams(3dm).
DMstatus **dmColorSetConvParams** (const DMcolorconverter *converter*, DMparams **convParams*)	Set the image conversion parameters. See also dmColorSetConvParams(3dm).
DMstatus **dmColorSetDstParams** (const DMcolorconverter *converter*, DMparams **dstParams*)	Set the destination image parameters. See also dmColorSetSrcParams(3dm).

Table A-1 (continued) The Color Space Library API

Function	Description
DMstatus **dmColorGetSrcParams** (const DMcolorconverter *converter*, DMparams **scrParams*)	Get the source image parameters. See also dmColorSetSrcParams(3dm).
DMstatus **dmColorGetConvParams** (const DMcolorconverter *converter*, DMparams **convParams*)	Get the image conversion parameters. See also dmColorGetConvParams(3dm).
DMstatus **dmColorGetDstParams** (const DMcolorconverter *converter*, DMparams **dstParams*)	Get the destination image parameters. See also dmColorSetSrcParams(3dm).
DMstatus **dmColorSetSubsamplingFilter** (const DMcolorconverter *converter*, const int *subsamplingFilter*)	Set the subsampling filter type. See also dmColorSetSubsamplingFilter(3dm).
DMstatus **dmColorGetSubsamplingFilter** (const DMcolorconverter *converter*, int **subsamplingFilter*)	Get the subsampling filter type. See also dmColorSetSubsamplingFilter(3dm).

The DVI Audio Compression Library

The DVI Audio Compression Library is based on Intel's Digital Video Interactive audio compression technology for multimedia applications. It implements the IMA (Interactive Multimedia Association) recommendations for ADPCM compression and decompression based on Intel's DVI algorithm.

Table A-2 The DVI Audio Library API

Function	Description
DMstatus **dmDVIAudioDecode** (DMDVIaudiodecoder *handle*, unsigned char **ibuf*, short **obuf*, int *nsamples*)	Do ADPCM decompression based on Intel's DVI algorithm. See also dmDVIAudioDecode(3dm).
DMstatus **dmDVIAudioDecoderCreate** (DMDVIaudiodecoder **decoder*)	Allocate a new DMDVIaudiodecoder structure. See also dmDVIAudioDecoderCreate(3dm).
DMstatus **dmDVIAudioDecoderDestroy** (DMDVIaudiodecoder *handle*)	Deallocate a DMDVIaudiodecoder. See also dmDVIAudioDecoderDestroy(3dm).
DMstatus **dmDVIAudioDecoderSetParams** (DMDVIaudiodecoder *handle*, DMparams **params*)	Set parameter values for a DMDVIaudiodecoder structure. See also dmDVIAudioDecoderSetParams(3dm).
DMstatus **dmDVIAudioDecoderGetParams** (DMDVIaudiodecoder *handle*, DMparams **params*)	Get parameter values for a DMDVIaudiodecoder structure. See also dmDVIAudioDecoderGetParams(3dm).
DMstatus **dmDVIAudioDecoderReset** (DMDVIaudiodecoder *handle*)	Fill buffers of a DMDVIaudiodecoder structure with zeroes. See also dmDVIAudioDecoderReset(3dm).
DMstatus **dmDVIAudioEncode** (DMDVIaudioencoder *handle*, short **ibuf*, unsigned char **obuf*, int *nsamples*)	Do ADPCM compression based on Intel's DVI algorithm. See also dmDVIAudioEncode(3dm).
DMstatus **dmDVIAudioEncoderCreate** (DMDVIaudioencoder **encoder*)	Allocate a new DMDVIaudioencoder structure. See also dmDVIAudioEncoderCreate(3dm).

Table A-2 The DVI Audio Library API

Function	Description
DMstatus **dmDVIAudioEncoderDestroy** (DMDVIaudioencoder *handle*)	Deallocate a DMDVIaudioencoder structure. See also dmDVIAudioEncoderDestroy(3dm).
DMstatus **dmDVIAudioEncoderSetParams** (DMDVIaudioencoder *handle*, DMparams **params*)	Set parameter values for DMDVIaudioencoder structure. See also dmDVIAudioEncoderSetParams(3dm).
DMstatus **dmDVIAudioEncoderGetParams** (DMDVIaudioencoder *handle*, DMparams **params*)	Get parameter values for DMDVIaudioencoder structure. See also dmDVIAudioEncoderGetParams(3dm).
DMstatus **dmDVIAudioEncoderReset** (DMDVIaudioencoder *handle*)	Fill buffers of a DMDVIaudioencoder structure with zeroes. See also dmDVIAudioEncoderReset(3dm).

The G.711 Audio Compression Library

This library implements International Telecommunication Union Standard (ITU-T, formerly CCITT) G.711 for compression and decompression. The standard is for 64 Kb/s, 8 kHz, 16-bit pulse code modulation (PCM) audio encoding of voice frequencies.

Table A-3 The G.711 Audio Compression Library API

Function	Description
void **dmG711MulawEncode** (short **samples*, unsigned char **mulawdata*, int *numsamples*)	Convert a 16-bit linear PCM value to an 8-bit μ-law value. See also dmG711(3dm).
void **dmG711MulawDecode** (unsigned char **mulawdata*, short **samples*, int *numsamples*)	Convert an 8-bit μ-law value to a 16-bit linear PCM value. See also dmG711(3dm).
void **dmG711MulawZeroTrapEncode** (short **samples*, unsigned char **mulawdata*, int *numsamples*)	Do ITU G.711 μ-law compression with zero trap during compression. See also dmG711(3dm).
void **dmG711MulawZeroTrapDecode** (unsigned char **mulawdata*, short **samples*, int *numsamples*)	Same as **dmG711MulawDecode()**. See also dmG711(3dm).
void **dmG711AlawEncode** (short **samples*, unsigned char **Alawdata*, int *numsamples*)	Convert a 16-bit linear PCM value to an 8-bit A-law value. See also dmG711(3dm).
void **dmG711AlawDecode** (unsigned char **Alawdata*, short **samples*, int *numsamples*)	Convert an 8-bit A-law value to a 16-bit linear PCM value. See also dmG711(3dm).
void **dmG711MulawToAlaw** (unsigned char **mulawdata*, unsigned char **Alawdata*, int *numsamples*)	Convert μ -law data to A-law data. See also dmG711(3dm).

Table A-3 (continued) The G.711 Audio Compression Library API

Function	Description
void **dmG711AlawToMulaw** (unsigned char **Alawdata*, unsigned char **mulawdata*, int *numsamples*)	Convert A-law date to μ-law data. See also dmG711(3dm).
void **dmSunMulawEncode** (short **samples*, unsigned char **mulawdata*, int *numsamples*)	Convert a 16-bit linear PCM value to an 8-bit μ-law value using the conversion tables of Sun Microsystems. See also dmG711(3dm).
void **dmSunMulawDecode** (unsigned char **mulawdata*, short **samples*, int *numsamples*)	Convert an 8-bit μ-law value to a 16-bit linear PCM value using the conversion tables of Sun Microsystems. See also dmG711(3dm).
void **dmNeXTMulawEncode** (short **samples*, unsigned char **mulawdata*, int *numsamples*)	Convert a 16-bit linear PCM value to an 8-bit μ-law value using the conversion tables of NeXT Computers. See also dmG711(3dm).
void **dmNeXTMulawDecode** (unsigned char **mulawdata*, short **samples*, int *numsamples*)	Convert an 8-bit μ-law value to a 16-bit linear PCM value using the conversion tables of NeXT Computers. See also dmG711(3dm).

The G.722 Audio Compression Library

This library implements International Telecommunication Union Standard (ITU-T, formerly CCITT) G.722 for compression and decompression. The standard is for 7 kHz audio encoding within 64 Kb/s.

Table A-4 The G.722 Audio Compression Library API

Function	Description
DMstatus **dmG722Decode** (DMG722decoder *handle*, unsigned char **ibuf*, short **obuf*, int *nsamples*)	Do G.722 decompression. See also dmG722Decode(3dm).
DMstatus **dmG722DecoderCreate** (DMG722decoder **decoder*, int *maxsamples*, int *decodemode*)	Allocate a new DMG722decoder structure. See also dmG722DecoderCreate(3dm).
DMstatus **dmG722DecoderDestroy** (DMG722decoder *handle*)	Deallocate a DMG722decoder structure. See also dmG722DecoderDestroy(3dm).
DMstatus **dmG722DecoderGetParams** (DMG722decoder *handle*, DMparams **params*)	Get parameter values for a DMG722decoder structure. See also dmG722DecoderGetParams(3dm).
DMstatus **dmG722DecoderReset** (DMG722decoder *handle*)	Fill buffers of a DMG722decoder structure with zeroes. See also dmG722DecoderReset(3dm).
DMstatus **dmG722Encode** (DMG722encoder *handle*, short **ibuf*, unsigned char **obuf*, int *nsamples*)	Do G.722 compression. See also dmG722Encode(3dm).
DMstatus **dmG722EncoderCreate** (DMG722encoder **encoder*, int *maxsamples*)	Allocate a new DMG722encoder structure. See also dmG722EncoderCreate(3dm).
DMstatus **dmG722EncoderDestroy** (DMG722encoder *handle*)	Deallocate a DMG722Encoder structure. See also dmG722EncoderDestroy(3dm).

Table A-4 (continued) The G.722 Audio Compression Library API

Function	Description
DMstatus **dmG722EncoderGetParams** (DMG722encoder *handle*, DMparams **params*)	Get parameter values for aDMG722encoder structure. See also dmG722EncoderGetParams(3dm).
DMstatus **dmG722EncoderReset** (DMG722encoder *handle*)	Fill buffers of a DMG722encoder structure with zeroes. See also dmG722EncoderReset(3dm).

The G.726 Audio Compression Library

This library implements International Telecommunication Union Standard (ITU-T, formerly CCITT) G.726 for ADPCM compression and decompression. The standard is for a compressed data bit stream of 40, 32, 24, or 16 Kb/s and a decompressed A-law, μ-law, or linear PCM data stream of 64 Kb/s.

Table A-5 The G.726 Audio Compression Library API

Function	Description
DMstatus **dmG726Decode** (DMG726decoder *handle*, unsigned char **inBuffer*, short **outBuffer*, int *numSamples*)	Do G.726 ADPCM decompression. See also dmG726Decode(3dm).
DMstatus **dmG726DecoderCreate** (DMG726decoder **decoder*, int *bitRate*, int *outputMode*)	Allocate a new DMG726decoder structure. See also dmG726DecoderCreate(3dm).
DMstatus **dmG726DecoderDestroy** (DMG726decoder *handle*)	Deallocate a DMG726decoder structure. See also dmG726DecoderDestroy(3dm).
DMstatus **dmG726DecoderSetParams** (DMG726decoder *handle,* DMparams **params*)	Set a DMG726decoder structure's parameter values. See also dmG726DecoderSetParams(3dm).
DMstatus **dmG726DecoderGetParams** (DMG726decoder *handle,* DMparams **params*)	Get a DMG726decoder structure's parameter values. See also dmG726DecoderGetParams(3dm).
DMstatus **dmG726DecoderReset** (DMG726decoder *handle*)	Fill a DMG726decoder structure's internal buffers with zeroes. See also dmG726DecoderReset(3dm).
DMstatus **dmG726Encode** (DMG726encoder *handle*, short **ibuf*, unsigned char **obuf*, int *numSamples*)	Do G.726 ADPCM compression. See also dmG726Encode(3dm).

Table A-5 (continued) The G.726 Audio Compression Library API

Function	Description
DMstatus **dmG726EncoderCreate** (DMG726encoder **encoder*, int *bitRate*, int *outputMode*)	Allocate a new DMG726encoder structure. See also dmG726EncoderCreate(3dm).
DMstatus **dmG726EncoderDestroy** (DMG726encoder *handle*)	Deallocate a DMG726Encoder structure. See also dmG726EncoderDestroy(3dm).
DMstatus **dmG72EncoderSetParams** (DMG726encoder *handle*, DMparams **params*)	Set a DMG726encoder structure's parameter values. See also dmG726EncoderSetParams(3dm).
DMstatus **dmG726EncoderGetParams** (DMG726encoder *handle*, DMparams **params*)	Get a DMG726decoder structure's parameter values. See also dmG726EncoderGetParams(3dm).
DMstatus **dmG726EncoderReset** (DMG726encoder *handle*)	Fill a DMG726encoder structure's internal buffers with zeroes. See also dmG726EncoderReset(3dm).

The G.728 Audio Compression Library

This library implements International Telecommunication Union Standard (ITU-T, formerly CCITT) G.728 for the audio encoding used in videoconferencing. The standard controls the coding of speech at 16 Kb/s using Low-Delay Code Excited Linear Prediction (LD-CELP).

Table A-6 The G.728 Audio Compression Library API

Function	Description
DMstatus **dmG728Decode** (DMG728decoder *handle*, unsigned char **ibuf*, short **obuf*, int *nsamples*)	Do ITU G.728 decompression (LD-CELP). See also dmG728Decode(3dm).
DMstatus **dmG728DecoderCreate** (DMG728decoder **decoder*)	Allocate a new DMG728decoder structure. See also dmG728DecoderCreate(3dm).
DMstatus **dmG728DecoderDestroy** (DMG728decoder *handle*)	Deallocate a DMG728decoder structure. See also dmG728DecoderDestroy(3dm).
DMstatus **dmG728DecoderGetParams** (DMG728decoder *handle*, DMparams **params*)	Get the parameter values of a DMG728decoder structure. See also dmG728DecoderGetParams(3dm).
DMstatus **dmG728DecoderSetParams** (DMG728decoder *handle*, DMparams **params*)	Set the parameter values of a DMG728decoder structure. See also dmG728DecoderSetParams(3dm).
DMstatus **dmG728DecoderReset** (DMG728decoder *handle*)	Fill internal buffers of a DMG728decoder structure with zeroes. See also dmG728DecoderReset(3dm).
DMstatus **dmG728Encode** (DMG728encoder *handle*, short **ibuf*, unsigned char **obuf*, int *nsamples*)	Do G.728 compression (LD-CELP). See also dmG728Encode(3dm).
DMstatus **dmG728EncoderCreate** (DMG728encoder **encoder*)	Allocate a new DMG728encoder structure. See also dmG728EncoderCreate(3dm).

Table A-6 (continued) The G.728 Audio Compression Library API

Function	Description
DMstatus **dmG728EncoderDestroy** (DMG728encoder *handle*)	Deallocate a DMG728Encoder structure. See also dmG728EncoderDestroy(3dm).
DMstatus **dmG728EncoderGetParams** (DMG728encoder *handle*, DMparams **params*)	Get the parameter values of a DMG728encoder structure. See also dmG728EncoderGetParams(3dm).
DMstatus **dmG728EncoderReset** (DMG728encoder *handle*)	Fill the internal buffers of a DMG728encoder structure with zeroes. See also dmG728EncoderReset(3dm).

The GSM Audio Compression Library

This library implements the European Global System for Mobile telecommunication (GSM) 06.10 provisional standard used for digital cellular phones. The standard, prI-ETS 300 036, describes full-rate speech transcoding which uses RPE–LTP (Regular–Pulse Excitation Long–Term Predictor)) coding at 13 Kb/s.

Table A-7 The GSM Audio Compression Library API

Function	Description
DMstatus **dmGSMDecode** (DMGSMdecoder *handle*, unsigned char **ibuf*, short **obuf*, int *numSamples*)	Do GSM decoding. See also dmGSMDecode(3dm).
DMstatus **dmGSMDecoderCreate** (DMGSMdecoder **decoder*)	Allocate a new DMGSMdecoder structure. See also dmGSMDecoderCreate(3dm).
DMstatus **dmGSMDecoderDestroy** (DMGSMdecoder *handle*)	Deallocate a DMGSMdecoder structure. See also dmGSMDecoderDestroy(3dm).
DMstatus **dmGSMDecoderGetParams** (DMGSMdecoder *handle*, DMparams **params*)	Get the parameter values of a DMGSMdecoder structure. See also dmGSMDecoderGetParams(3dm).
DMstatus **dmGSMDecoderReset** (DMGSMdecoder *handle*)	Fill internal buffers of a DMGSMdecoder structure with zeroes. See also dmGSMDecoderReset(3dm).
DMstatus **dmGSMEncode** (DMGSMencoder *handle*, short **ibuf*, unsigned char **obuf*, int *numSamples*)	Do GSM encoding. See also dmGSMEncode(3dm).
DMstatus **dmGSMEncoderCreate** (DMGSMencoder **encoder*)	Allocate a new DMGSMencoder structure. See also dmGSMEncoderCreate(3dm).
DMstatus **dmGSMEncoderDestroy** (DMGSMencoder *handle*)	Deallocate a DMGSMEncoder structure. See also dmGSMEncoderDestroy(3dm).

Table A-7 (continued) The GSM Audio Compression Library API

Function	Description
DMstatus **dmGSMEncoderGetParams** (DMGSMencoder *handle*, DMparams **params*)	Get the parameter values of a DMGSMencoder structure. See also dmGSMEncoderGetParams(3dm).
DMstatus **dmGSMEncoderReset** (DMGSMencoder *handle*)	Fill internal buffers of a DMGSMencoder structure with zeroes. See also dmGSMEncoderReset(3dm).

The MPEG-1 Audio Compression Library

This library implements the Moving Pictures Experts Group MPEG-1 audio standard. The standard is designed for encoding non-interlaced material and is optimized for single-speed CD-ROM bit rates (about 1.5 Mb/s). The compression is based on subband coding with adaptive quantization. Input data is divided into different frequency bands which are weighted by their perceptual importance. Mono and stereo sources are supported at sampling rates of 32, 44.1 and 48 kHz. Allowable bit rates range from 32 to 448 Kb/s.

Table A-8 The MPEG-1 Audio Compression Library API

Function	Description
DMstatus **dmMPEG1AudioDecode** (DMMPEG1audiodecoder *decoder*, unsigned char **cmpData*, short **output*, int *fmtBytes*)	Decode a single compressed block of data created by a call to **dmMPEG1AudioEncode()**. See also dmMPEG1AudioDecode(3dm).
DMstatus **dmMPEG1AudioDecoderCreate** (DMMPEG1audiodecoder **decoder*)	Allocate a new DMMPEG1audiodecoder structure. See also dmMPEG1AudioDecoderCreate(3dm).
DMstatus **dmMPEG1AudioDecoderDestroy** (DMMPEG1audiodecoder *decoder*)	Deallocate an DMMPEG1audiodecoder structure. See also dmMPEG1AudioDecoderDestroy(3dm).
DMstatus **dmMPEG1AudioDecoderGetParams** (DMMPEG1audiodecoder *decoder*, DMparams **params*)	Get the parameter values for a DMMPEG1audiodecoder structure. See also dmMPEG1AudioDecoderGetParams(3dm).
DMstatus **dmMPEG1AudioDecoderSetParams** (DMMPEG1audiodecoder *decoder*, DMparams **params*)	Set the parameter values for a DMMPEG1audiodecoder structure. See also dmMPEG1AudioDecoderSetParams(3dm).
DMstatus **dmMPEG1AudioDecoderReset** (DMMPEG1audiodecoder *handle*)	Fill the internal buffers of an DMMPEG1audiodecoder structure with zeros. See also dmMPEG1AudioDecoderReset(3dm).

Table A-8 (continued) The MPEG-1 Audio Compression Library API

Function	Description
DMstatus **dmMPEG1AudioEncode** (DMAudioRateConverter *encoder*, short **sampBuf*, unsigned char **output*, int *frameBytes*)	Compress a single block of audio data using MPEG-1 audio compression algorithm. See also dmMPEG1AudioEncode(3dm).
DMstatus **dmMPEG1AudioEncoderCreate** (DMMPEG1audioencoder **encoder*)	Allocate a new DMMPEG1audioencoder structure. See also dmMPEG1AudioEncoderCreate(3dm).
DMstatus **dmMPEG1AudioEncoderDestroy** (DMMPEG1audioencoder *encoder*)	Deallocate a DMMPEG1audioencoder structure. See also dmMPEG1AudioEncoderDestroy(3dm).
DMstatus **dmMPEG1AudioEncoderGetParams** (DMMPEG1audioencoder *encoder*, DMparams **params*)	Get the parameter values for an DMMPEG1audioencoder structure. See also dmMPEG1AudioEncoderGetParams(3dm).
DMstatus **dmMPEG1AudioEncoderSetParams** (DMMPEG1audioencoder *encoder*, DMparams **params*)	Set the parameter values for an DMMPEG1audioencoder structure. See also dmMPEG1AudioEncoderSetParams(3dm).
DMstatus **dmMPEG1AudioEncoderReset** (DMMPEG1audioencoder *handle*)	Fill the internal buffers of a DMMPEG1audioencoder with zeros. See also dmMPEG1AudioEncoderReset(3dm).
DMstatus **dmMPEG1AudioFilterStateCreate** (DMMPEG1audiofilterstate **filterState*)	Allocate a new DMMPEG1audiofilterstate structure. See also dmMPEG1AudioFilterStateCreate(3dm).
DMstatus **dmMPEG1AudioFilterStateDestroy** (DMMPEG1audiofilterstate *filterState*)	Free a DMMPEG1audiofilterstate structure. See also dmMPEG1AudioFilterStateDestroy(3dm).
DMstatus **dmMPEG1AudioFilterStateRestore** (void **coder*, DMMPEG1audiofilterstate *filterState*)	Restore a DMMPEG1audiofilterstate structure. See also dmMPEG1AudioFilterStateRestore(3dm).

Table A-8 (continued) The MPEG-1 Audio Compression Library API

Function	Description
DMstatus **dmMPEG1AudioFilterStateSave** (void **coder*, DMMPEG1audiofilterstate *filterState*)	Save a DMMPEG1audiofilterstate structure. See also dmMPEG1AudioFilterStateSave(3dm).
DMstatus **dmMPEG1AudioHeaderGetBlockBytes** (DMMPEG1audiodecoder *decoder*, unsigned char **cmpData*, int **blockSize*)	Get the expected length in bytes of a compressed data block. See also dmMPEG1AudioHeaderGetBlockBytes(3dm).
DMstatus **dmMPEG1AudioHeaderGetParams** (unsigned char **cmpData*, DMparams **params*)	Get decoder parameter information from the header of a compressed MPEG-1 audio data block. See also dmMPEG1AudioHeaderGetParams(3dm).

The Audio Rate Conversion Library

This library enables the sampling rate conversion of single-channel, 32-bit, floating point audio data.

Table A-9 The Audio Rate Conversion Library API

Function	Description
DMstatus **dmAudioRateConvert** (DMaudiorateconverter *handle*, float **inbuf*, float **outbuf*, int *inlen*, int **numout*)	Convert the data sampling rate. See also dmAudioRateConvert(3dm).
DMstatus **dmAudioRateConverterCreate** (DMaudiorateconverter **converter*)	Allocate a new DMaudiorateconverter structure. See also dmAudioRateConverterCreate(3dm).
DMstatus **dmAudioRateConverterDestroy** (DMaudiorateconverter *handle*)	Deallocate an DMaudiorateconverter structure. See also dmAudioRateConverterDestroy(3dm).
DMstatus **dmAudioRateConverterGetParams** (DMaudiorateconverter *handle*, DMparams **params*)	Get the parameter values of a DMaudiorateconverter structure. See also dmAudioRateConverterGetParams(3dm).
DMstatus **dmAudioRateConverterSetParams** (DMaudiorateconverter *handle*, DMparams **params*)	Set the parameter values of a DMaudiorateconverter structure. See also dmAudioRateConverterSetParams(3dm).
DMstatus **dmAudioRateConverterReset** (DMaudiorateconverter *handle*, float *resetval*)	Fill the internal buffers of a DMaudiorateconverter structure with a constant value. See also dmAudioRateConverterReset(3dm).

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