RadiSys.

Using the Sound Driver Interface

Version 3.2

www.radisys.com

World Headquarters
5445 NE Dawson Creek Drive • Hillsboro, OR 97124 USA
Phone: 503-615-1100 • Fax: 503-615-1121
Toll-Free: 800-950-0044

International Headquarters
Gebouw Flevopoort • Televisieweg 1A
NL-1322 AC • Almere, The Netherlands
Phone: 31 36 5365595 • Fax: 31 36 5365620

RadiSys Microware Communications Software Division, Inc.
1500 N.W. 118th Street
Des Moines, Iowa 50325
515-223-8000

Revision A
November 2001

Copyright and publication information

Disclaimer

The information contained herein is believed to be accurate as of the date of publication. However, RadiSys Corporation will not be liable for any damages including indirect or consequential, from use of the OS-9 operating system, Microware-provided software, or reliance on the accuracy of this documentation. The information contained herein is subject to change without notice.

Reproduction notice

The software described in this document is intended to be used on a single computer system. RadiSys Corporation expressly prohibits any reproduction of the software on tape, disk, or any other medium except for backup purposes. Distribution of this software, in part or whole, to any other party or on any other system may constitute copyright infringements and misappropriation of trade secrets and confidential processes which are the property of RadiSys Corporation and/or other parties. Unauthorized distribution of software may cause damages far in excess of the value of the copies involved.

November 2001

Table of Contents

Chapter 1: Sound Concepts

The Sound Driver Interface enables your MAUI application to play and record sound data. This chapter explains the concepts of using sound in your applications.

Overview

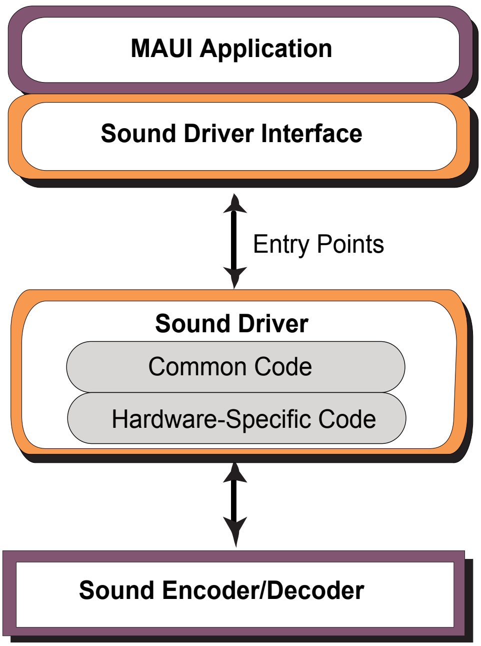
The Sound Driver Interface:

- provides a set of primary entry points, GetStat sub-functions, and SetStat sub-functions through which MAUI applications can control the sound driver.
- is a dual-ported I/O (DPIO) driver that uses the multimedia file manager (MFM). This allows the driver to work under both OS-9 and OS-9000.

The sound driver is sharable (multiple paths may be open to it at the same time). This enables multiple play and record paths, but not concurrent play and record.

The Sound Driver Interface is accessible by MAUI applications and directly controls the operation of the sound driver as shown in the following figure.

Figure 1-1 Sound Driver Interface Architecture



Entry Points

The primary entry points into the sound driver are those found in all DPIO drivers. They are `init`, `terminate`, `open`, `close`, `getstat`, and `setstat`. The getstat and setstat entry points are dispatch functions that call the proper sub-function based on the information in the parameter block passed to it.

Sound Functions and Data Types

The Sound Driver Interface provides the functions your application needs to play and record sound. The majority of functions can be loosely divided into two categories: get status functions and set status functions.

Get status functions are identified by the `_os_gs_*` name construction. These functions are used by the application to get information or status from the sound driver. Your application uses get status functions to determine sound device status and capabilities.

Set status functions are identified by the `_os_ss_*` name construction. They are used by the application to send control commands or set the status of the sound driver. Your application uses set status functions to instruct the sound driver to send signals when the device is ready, play sound data, record sound, and release the device when it is no longer needed.

Sound data types comprise constants, enumerated types, and data structures that provide information to the application and the sound driver. Data types control how sound drivers operate and are fundamental to hardware independence.

Sound Maps

Sound map objects are created by the application to control play and record operations. Sound maps contain all the information needed by the sound driver to play and record sound data. Your application may define one or more sound maps depending on the type of operations and type of sound data used.

Play and record operations can be synchronous or asynchronous. The driver generates signals for the application to keep the application informed about the operation. Since the sound device can be shared by multiple processes, these signals can also enable your application to manage multiple play and record requests.

Sound maps used for play operations are built by the application at run time with the information contained in the headers of the sound data files. Sound maps used for recording operations are initialized and defined by the application through standard C syntax.

Sound data is contained in buffers and played from buffers. Sound map looping enables your application to loop segments of sound data during play, or play a segment of sound data rather than an entire sequence from start to finish.

The sound map can also contain a pointer to another sound map. This pointer enables you to link sound maps together for continuous operation from a single play or record command or link sound maps of different formats.

The sound map is defined by the data structure `SND_SMAP` and contains the following information:

Triggers and status	Trigger signals and status information are sent to the application to indicate the current sound device conditions.
Encoding parameters	Sound data encoding method, number of channels, bits per sample, and sample rate of the sound data.
Sound Buffer	Pointer or address of the buffer, buffer size, and current offset.
Looping	Loop start offset, end offset, loop count, and current loop counter value.
Next sound map	Pointer to the next sound map to play or record.

Each play and record operation is associated with one or more sound maps. Your application may have a single sound map defined for play and a single sound map defined for record, or you may have a different sound map defined for each chunk of sound data your application may play or record. Since each sound map is associated with a single buffer, the size of the buffer determines how much sound data can be stored at any time. If your application uses sound data that is of the same general size, a single buffer and sound map may be appropriate. If your application uses sound data of various sizes, or if your application actively manipulates the sound data by using offsets and looping, you may want to define a larger number of small buffers.

Sound maps may be linked together to form a chain of sound maps to play or record. These sound maps do not have to contain data of the same encoding formats and parameters. When sound maps of different encoding formats and parameters are linked, you may notice some delay when transitioning from one encoding format to another as the driver calibrates to the new parameters. This depends on the characteristics of the hardware.

Triggers and Status

The sound map data structure defines the triggers and status used during the play and recording operations.

Trigger status	The variable `trig_status` indicates the sound map status. The trigger status is modified as the status changes. The data structure `SND_TRIGGER` describes each of the possible status values.
Trigger mask	The variable `trig_mask` indicates which activities prompt the sending of a trigger signal. If the trigger mask is set to `SND_TRIG_NONE`, no signals are sent.
Trigger signal	The variable `trig_signal` defines the signal to send when the sound operation completes an activity specified by the trigger mask. If the trigger signal is set to zero, no signals are sent.
Error code	The variable `err_code` contains the exit status for the sound map. If an error occurs, such as `EOS_MAUI_NOHWSUPPORT` or `EOS_MAUI_ABORT`, the error code is placed in `errno`. Otherwise, the error code is set to zero at the end of the sound operation.

Encoding Parameters

The encoding parameters in `SND_SMAP` define the specifics of the sound data.

Coding method	The variable `coding_method` defines the audio encoding method and format for the sound data attached to the sound map.
Number of Channels	The variable `num_channels` specifies the number of audio channels. Mono data requires one channel, stereo data requires two channels.
Sample size	The variable `sample_size` indicates the number of bits per sample. Typically, this value is 8 or 16 bits per sample.
Sample rate	The variable `sample_rate` indicates the number of samples per second. Typical sample rates include 4000, 8000, 11025, 22050, 44100, and 48000 samples per second.

Different sound devices support a variety of encoding parameters. Sound maps contain the information required by sound devices to properly calibrate to the parameters of various encoding parameters.

The two parameters that most affect sound quality are the sample size and the sample rate. Higher sample rates provide a higher density of information in the sound data. Higher sample sizes provide greater dynamic range and a more favorable signal to noise ratio. Alternative coding methods such as ADPCM provide a means to represent the same audio data in smaller sample sizes.

Sound Buffer

The actual sound data is stored in a simple buffer.

Buffer Pointer	The entry *buf points to the buffer containing the sound data. The starting point of the buffer should be on a multiple of cm_boundary_size. cm_boundary_size is defined in the structure SND_DEV_CM.
Buffer size	The variable buf_size indicates the size of the sound data buffer in bytes. The size of the buffer should be a multiple of cm_boundary_size.
Current offset	The variable cur_offset specifies the offset in bytes within the buffer for the current sound operation. The offset is updated throughout the I/O operation by the driver. The application should set the offset to the play or record starting point. This should be set to a value consistent with cm_boundary_size.

Looping

Looping entries are used for play operations to specify the area of the buffer to play and how many times to play the buffer area. These entries must be set to zero for recording operations.

Loop start	The variable `loop_start` specifies the start of the loop expressed as an offset in bytes from the start of the buffer. This value should be consistent with `cm_boundary_size`.
Loop end	The variable `loop_end` specifies the end of the loop expressed as an offset in bytes from the start of the buffer. This value should be consistent with `cm_boundary_size`.
Loop count	The variable `loop_count` specifies the number of times to execute the loop.
Loop counter	The variable `loop_counter` indicates the current number of times the loop has executed. This entry is set to zero when the sound map is accepted by the sound driver and incremented as the driver plays the sound data.

Next Sound Map

The final entry in the `SND_SMAP` data structure, `next`, is a pointer to another sound map. `next` identifies the next sound map that is used when the current sound map is finished. This entry is `NULL` if this sound map is not linked to another sound map.

Linked sound maps may have different encoding parameters from each other. Since most hardware requires some calibration time to switch to a different format, there may be some delay before a sound map of a different format begins to play or record.

Preparing to Use Sound

MAUI applications are likely to run on a variety of user devices, so the application must evaluate its environment and adjust its operating parameters appropriately before actually playing and recording sound. Three functions enable your application to evaluate the software and hardware environment. They are:

- Check device capabilities_os_gs_snd_devcap() determines the capabilities of the sound hardware and driver.
- Check software compatibility_os_gs_snd_compat() determines that the compatibility level of the sound driver and sound driver interface are functionally compatible.
- Check the status_os_gs_snd_status() determines the current status of the sound device.

Device Capabilities

The specific set of sound device capabilities is stored in a data structure called SND_DEV_CAP. The getstat sub-function _os_gs_snd_devcap() reads the data structure SND_DEV_CAP and returns the information to the application. Based on the information returned, your applications can make adjustments to run correctly on the target system. Following are two examples of how an application can deal with differing hardware capabilities at run time.

Example 1

When you design your application you must be flexible since you may not know which specific sound board and processor are installed in the user hardware. Consequently, you cannot know which audio file formats are supported. To deal with this unknown, you can encode your sound data in several file formats. Give each file the same base name, but store each set of files in a different directory, such as PCM8/intro.snd and ADPCM/intro.snd.

When your application plays sound data, it uses a variable as part of the path name (%s/intro.snd). At run time, in this example, your application reads the SND_DEVCAP and discovers that the user hardware supports 8-bit PCM audio. At that time, your application adjusts to the hardware by setting the %s variable to PCM8. When your application plays the sound data, it gets the data from the PCM8 directory.

You can use this same technique to adjust to mono and stereo options and sample rates.

Example 2

An application is designed to play a sound sample that was recorded at 22000 Hz, but that specific sampling rate is not supported on the playback device. The application can examine the sample rates section of the SND_DEV_CAP to determine the supported sampling rate that is closest to 22000 Hz and convert the original sound data to match the nearest supported sample rate or play the sample back at a supported rate.

Applications can provide built-in conversion utilities for mono and stereo options, gain control, sample rates, or coding method.

SND_DEV_CAP Data Structure

SND_DEV_CAP contains the following device capabilities information:

- Hardware type
- Hardware subtype
- Mask of supported triggers
- Gain controls for play operations
- Gain controls for record operations
- Mask of supported gain commands
- Number and pointer to a list of gain capabilities
-

- Number and pointer to a list of supported sample rates
- Number of channel entries and pointer to a list of supported channels
- Number and pointer to a list of supported coding methods

The hardware type and subtype describe the specific sound hardware installed in the user system. Hardware type indicates the general class of the hardware, such as the type of sound chip. Hardware subtype typically indicates the specific sound board installed in the system.

The sound triggers entry defines the trigger events in a sound operation. The following table describes the six trigger events and how they are generated.

Table 1-1 Trigger Signals

Trigger	Description
`SND_TRIG_NONE`	Mask value indicates to send no trigger signals to the application.
`SND_TRIG_ANY`	Mask value indicates to send all trigger signals to the application.
`SND_TRIG_START`	Trigger value indicates to send a trigger signal to the application when the output device is actively playing or has started accepting sound input.
`SND_TRIG_FINISH`	Trigger value indicates to send a trigger signal to the application when the output device has completed playing or stopped accepting sound input.

Table 1-1 Trigger Signals (continued)

Trigger	Description
SND_TRIG_BUSY	Trigger value indicates to send a trigger signal to the application when the driver has started consuming the data in the buffer during play or started filling the buffer during record operations.
SND_TRIG_READY	Trigger value indicates to send a trigger signal to the application when the driver has finished consuming the data in the buffer and is ready to accept the next buffer for play or has finished filling the buffer and is ready to use the next buffer for record operations.

Digital sound data can be classified by coding method and sample rate. The list that follows presents some commonly supported coding methods for sound data. For more specific information about these coding methods, see the data type reference section.

µ-Law	Pronounced mu-law. 8-bit companded format that is a telephony standard for the United States and Japan. (Pronounced mu-law)
A-Law	8-bit companded format that is a telephony standard for Europe.
PCM Linear	Pulse-Coded Modulation. 8-bit and 16-bit format that is the most common method used by sound cards for record and playback.
ADPCM	Adaptive Differential Pulse Code Modulation is a compressed form of PCM coding. There are several different ADPCM standards, each with their own type code.

Each coding method supports one or more sample rates. Some of the sample rates may employ different formatting options. For example, 16-bit signed linear PCM is formatted either MSB first (SND_CM_MSBYTE1ST, big endian) or LSB first (SND_CM_LSBYTE1ST, little endian). Some sound devices accept either format, while other sound devices support only one format.

Driver Compatibility Level

Typically, the compatibility interface is used by the MAUI APIs to compare the compatibility level of the driver to the managing API. The newer entity of the two determines if it can operate with the other entity. If they are compatible, the application operates without problem. If they are not compatible, the newer entity either copes with the differences, or simply rejects the operation by returning an error. Since the sound driver does not currently have a high-level MAUI Sound API, this call may be used by an application to cope with compatibility differences between the Sound Driver Interface and the driver. MAUI applications may determine their compatibility level by examining the value of MAUI_COMPAT_LEVEL, which is available when you include maui_com.h.

Chapter 2: Sound Operations

This chapter provides information about the functions that are available to your applications and how to play and record sound.

Preparing the Sound Device for Use

Before sending to or receiving sound data from the sound device, your application must get the name of the sound device, open the device for use, and then make needed adjustments to the specific capabilities of the sound device.

`cdb_get_ddr()`	Returns the name of the sound device.
`_os_open()`	Opens a path to the sound device.
`_os_gs_snd_devcap()`	Returns the device capabilities of the sound device.

Get the Sound Device Name

Use the CDB function `cdb_get_ddr()` with `CDB_TYPE_SOUND` to get the name of the sound device.

For More Information

cdb_get_ddr() and CDB_TYPE_SOUND are explained in detail in the MAUI Programming Reference Manual.

Open

The function `_os_open()` establishes a path ID to the named sound device for the process that opened the device. All subsequent sound control commands issued by that process use the path ID to address the sound device.

The `mode` parameter in the `_os_open()` command indicates the operational mode of the device for this process. For play operations, the device must be opened for write (mode `S_IWRITE`). For record operations, the device must be opened for read (mode `S_IREAD`).

Get Device Capabilities

The function `_os_gs_snd_devcap()` returns the sound device capabilities to the application. The device capabilities enable the application to know which sound operations are supported and what sound data encoding parameters are valid. With this information the application can make adjustments to the hardware and software environment of the target system.

Keeping Track of the Sound Device

Before calling any sound operations, your application should set up the signals and triggers that it needs to manage sound operations effectively. There are three functions your application can use to monitor and control the sound device operational status:

`_os_gs_snd_status()`	Gets the sound device status.
`_os_ss_sendsig()`	Sets up a signal for the driver to send to a process when the driver is ready for a new operation.
`_os_ss_relea()`	Removes the signal set by `_os_ss_sendsig()`.

Sound Device Status

The status of the sound device can be checked at any time by using the function `_os_gs_snd_status()`. This function returns a pointer to the `SND_DEV_STATUS` structure. The `SND_DEV_STATUS` structure provides information about what the sound device is doing and which process or processes are currently using the sound device.

Status	Indicates the general status of the sound device.
Play PID	Process ID of the current Play operation.
Record PID	Process ID of the current Record operation.
Gain	Pointer to an array of `num_gain SND_GAIN` structures. These describe the current settings of each gain control.

Status

The status member of SND_DEV_STATUS is of type SND_STATUS. Six status types are defined in the enumerated type SND_STATUS.

SND_STATUS_NONE	This status indicates the sound device is in the initial state.
SND_STATUS_PLAY	This status indicates a play operation is active.
SND_STATUS_PLAY_PAUSED	This status indicates the current play operation is paused.
SND_STATUS_RECORD	This status indicates a record operation is active.
SND_STATUS_RECORD_PAUSED	This status indicates a record operation is paused.
SND_STATUS_BUSY	This status indicates the driver is processing a sound operation.

Gain

The gain member of SND_DEV_STATUS points to an array of SND_GAIN structures that contain the current settings of the device's various mixer lines.

Send Signal

The function _os_ss_sendsig() sets up a signal to send to a process when the sound device is ready to accept a new sound operation (play or record). This function is especially useful when several processes share the sound device, so a process can be signalled when the sound device becomes idle.

Each time the signal is sent, this function must be reset. If a signal request is already set when this function is called, the error EOS_DEVBSY is returned.

Release the Device

When a process no longer needs the signal installed with `_os_ss_sendsig()`, the process should call the function `_os_ss_relea()`. This function releases the device from any `_os_ss_sendsig()` request made by the calling process.

Playing and Recording Sound Data

Several functions are available to play and record sound data:

`_os_ss_snd_play()`	Begins playing from a sound map. If the hardware is not capable of playing sound, such as a record-only device, the play function returns the error `EOS_UNKSVC` (unknown service request).
`_os_ss_snd_record()`	Begins recording to a sound map.If the hardware is not capable of recording sound, such as a play-only device, the function returns the error `EOS_UNKSVC` (unknown service request).
`_os_ss_snd_gain()`	Controls the output signal level of the sound decoder.
`_os_ss_snd_pause()`	Pauses the play operation.
`_os_ss_snd_cont()`	Continues the play operation from the point at which it was paused.
`_os_ss_snd_abort()`	Stops the play operation.

Play

The play command `_os_ss_snd_play()` sends a request to the sound decoder to begin playing from a sound map. The sound data and all parameters the decoder needs to decode the data are contained in the sound map. This function requires the path ID to the sound device (established when the device was opened), a pointer to the sound map, and the blocking type.

When the first sound map is accepted by the driver, the driver sets the play and busy triggers, clears the loop counter, and sets the current offset to a value equal to the loop start value. If multiple sound maps are linked together, these fields are modified in the subsequent sound maps when the driver begins processing each of these subsequent sound maps. This enables applications to modify the `next` pointers in the sound map even after calling the play function.

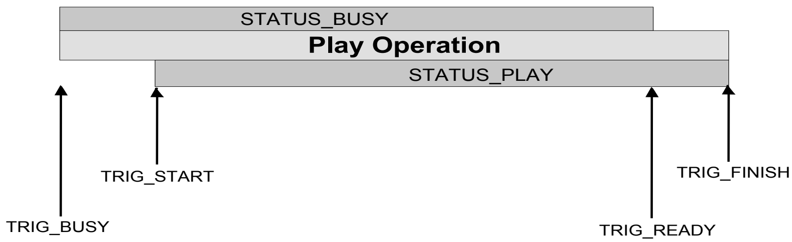
The blocking type determines how the play call functions. Blocking types are described in the data structure `SND_BLOCK_TYPE`. The play function may specify one of three blocking types:

`SND_NOBLOCK`	If the driver is busy, it immediately returns the error `EOS_MAUI_BUSY`. Otherwise, it starts the play and returns immediately.
`SND_BLOCK_START`	The play call waits while the device is busy and returns immediately when the sound device begins to play.
`SND_BLOCK_FINISH`	The play call waits to return until all linked sound maps have been consumed or the play operation is aborted.

If a fatal signal is received while the call is blocked waiting for access to the sound device (the play has not started yet and `block_type` is equal to `SND_BLOCK_START` or `SND_BLOCK_FINISHED`), the call returns `EOS_SIGNAL` and the play request is cancelled. If a fatal signal is received after the play has started and the call is blocked waiting for the play to finish (`block_type` is equal to `SND_BLOCK_FINISH`), the call returns `EOS_SIGNAL`, but the play operation continues. Examine `smap->trig_status` to determine if the sound map is still being used by the driver.

When the sound decoder completes an activity that satisfies a trigger in the sound map trigger mask and the sound map trigger signal is not set to zero, the driver sends the signal `smap->trig_signal` to the calling process. The following figure shows the status and trigger points of a typical play operation

Figure 2-1 Play Operation Status and Trigger Points.



Note

Not all drivers support all triggers. Check the device capabilities to determine which triggers are supported by your hardware.

Record

The record command `_os_ss_snd_record()` sends a request to the sound encoder to begin recording to a sound map. The sound data and all parameters the encoder needs to encode the data are contained in the sound map. This function requires the path ID to the sound device (established when the device was opened), a pointer to the sound map, and the block type.

When the first sound map is accepted by the driver, the driver sets the record and busy triggers. If multiple sound maps are linked together, these fields are modified in the subsequent sound maps when the driver begins processing each of these subsequent sound maps. This enables applications to modify the `next` pointers in the sound map even after calling the record function.

Record operations do not support loop operations. The `SND_SMAP` fields `loop_start`, `loop_end`, `loop_count`, and `loop_counter` must be set to zero by the application before the sound map is accepted by the driver.

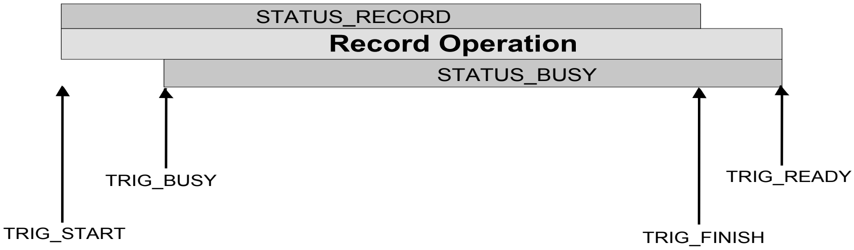
The blocking type determines how the record call functions. Blocking types are described in the data structure `SND_BLOCK_TYPE`. The record function may specify one of three blocking types:

`SND_NOBLOCK`	If the driver is busy, it immediately returns the error `EOS_MAUI_BUSY`. Otherwise, it starts the record and returns immediately.
`SND_BLOCK_START`	The record call waits while the device is busy and returns immediately when the sound device begins to record.
`SND_BLOCK_FINISH`	The record call waits to return until all linked sound maps have been consumed or the record operation is aborted.

If a fatal signal is received while the call is blocked waiting for access to the sound device (the record has not started yet and `block_type` is equal to `SND_BLOCK_START` or `SND_BLOCK_FINISH`), the call returns `EOS_SIGNAL` and the record request is cancelled. If a fatal signal is received after the record has started and the call is blocked waiting for the record to finish (`block_type` is equal to `SND_BLOCK_FINISH`), the call returns `EOS_SIGNAL`, but the record operation continues. Examine the `smap->trig_status` to determine if the sound map is still being used by the driver.

When the sound encoder completes an activity that satisfies a trigger in the sound map trigger mask and the sound map trigger signal is not set to zero, the driver sends the signal `smap->trig_signal` to the calling process. The following figure shows the status and trigger points of a typical recording operation.

Figure 2-2 Record Operation Status and Trigger Points



Note

Not all drivers support all triggers. Check the device capabilities to determine which triggers are supported by your hardware.

Gain Control

The output signal gain for play operations and the input signal gain for record operations is controlled with the function `_os_ss_snd_gain()`. This function requires the path ID of the sound device, and specifies the mix lines and the gain operation to perform.

The gain capabilities of the sound device are stored in a data structure named `SND_GAIN_CAP`. This data structure is returned as part of the `SND_DEV_CAP` structure when the application calls `_os_gs_snd_devcap()`. The `SND_GAIN_CAP` includes the following capabilities information:

Table 2-1 SND_GAIN_CAP Structure

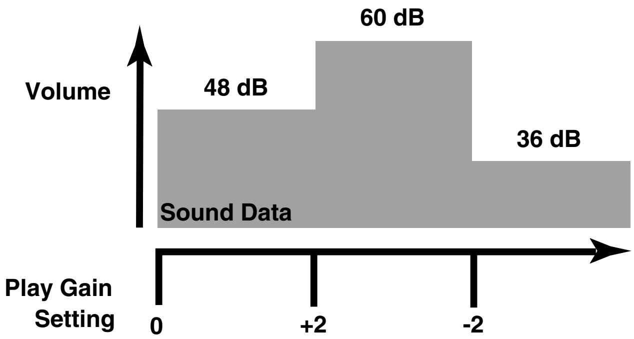
Types	Supported Gain Types
Lines	Mask of one or more mix lines that share a `SND_GAIN_CAP` description.
Support for Mute	Boolean `TRUE` if muting is supported.
Default Type	The driver's default gain type for the mix lines.
Default Level	The driver's initial gain value for the indicated mix lines.
Zero Level	Level value where the change in dB is zero.
Number of Step	Number of actual gain or attenuation values supported by the hardware.
Step size	Average size of each step of gain, expressed in 100^{th} of a dB.
Minimum dB	Change in dB at level `SND_GAIN_MIN`.
Maximum dB	Change in dB at level `SND_GAIN_MAX`.

Gain is controlled by increasing or decreasing gain in increments. The step size gives you information about the average size of an increment. However, the incremental increase or decrease may not be linear. The zero level is the level at which all increases are gain and all decreases are attenuation. Zero gain does not imply zero dB.

Levels above zero are represented by positive gain values, and levels below the zero level are negative gain values. The gain levels are absolute levels, not relative to the current setting.

For example, if the zero setting correlates to a sound output that is 48 dB, and step size is 6.00 dB, increasing the play gain level to 2 increases the volume of the sound output to roughly 60 dB. To set the volume back to 48 dB, you set the play gain to zero. Setting the gain to -2 decreases the volume to 36 dB which is two steps below the 48 dB zero-level. The following figure illustrates how play gain works.

Figure 2-3 Effects of Play Gain Level on Volume



Note

The hardware specification for the particular driver that is installed provides more specific information regarding gain capabilities of the hardware. Contact the hardware manufacturer for the hardware specification for your driver.

The function `_os_ss_snd_gain()` sends a request to the sound driver to modify one or more mix lines. The following list describes each of the operations that can be applied with the `_os_ss_snd_gain()` function:

`SND_GAIN_CMD_UP`	Specifies the number of steps to increment the gain level of the mix line. This has the effect of increasing the volume or input signal level.
`SND_GAIN_CMD_DOWN`	Specifies the number of steps to decrement the gain level of the mix line. This has the effect of decreasing the volume or input signal level.
`SND_GAIN_CMD_RESET`	Indicates to reset the gain level to the default level found in `SND_GAIN_CAP`.
`SND_GAIN_CMD_MUTE`	Indicates to set or clear the mute bit according to `state` in `SND_GAIN_MUTE`.
`SND_GAIN_CMD_MONO`	Sets a specific mono gain level.
`SND_GAIN_CMD_STEREO`	Sets the specific left and right gain levels for stereo.
`SND_GAIN_TYPE_XSTEREO`	Sets a specific gain level for the X-stereo.

Mono is a single channel output. Stereo is a two-channel (left and right) output. X-stereo is a two-channel output that has the additional capability of switching the left channel to the right output and the right channel to the left output.

Pause

The pause command _os_ss_snd_pause() sends a request to the sound driver to pause the active operation. When the operation is paused, the SND_STATUS_PLAY_PAUSED or SND_STATUS_RECORD_PAUSED bit in the status field of SND_DEV_STATUS is set. Only the path that initiated the operation can pause the operation. If another path attempts to pause an operation, the error EOS_MAUI_NOTOWNER is returned. If a path is opened for both read and write, the pause command pauses the play and record operations that are running.

If the pause command is not supported by the hardware, this function returns the error EOS_MAUI_NOHWSUPPORT (no hardware support) and the driver continues to play.

Continue

The continue command _os_ss_snd_cont() sends a request to the sound driver to continue a currently paused operation. This function clears the SND_STATUS_PLAY_PAUSED or SND_STATUS_RECORD_PAUSED bit in the status field of SND_DEV_STATUS, then begins playing the sound data at the point at which it was paused.

Only the path that initiated the operation and issued the pause command can continue the operation. If another path attempts to continue a paused operation, the error EOS_MAUI_NOTOWNER is returned.

The sup_play_pause and sup_record_pause fields of SND_DEV_CAP indicates if the sound driver supports pause and continue commands. If the command is not supported, this function returns the error EOS_UNKSVC.

Abort

The abort command `_os_ss_snd_abort()` sends a request to the sound driver to synchronously abort the current play or record operation.

Only the path that initiated the operation can abort the operation. If another path attempts to abort an operation, the error `EOS_MAUI_NOTOWNER` is returned.

Completing Sound Operations

One entry point is available for completing sound operations.

`_os_close()` Closes a sound device.

Close

The entry point `_os_close()` closes the sound device to the process that calls the function. When a process has completed all sound operations, this function should be called.

Chapter 3: Function Reference

This chapter describes each sound driver function.

Function Reference

Include Files

All applications using the Sound Driver Interface must contain the following code:

```
#include <MAUI/mfm_snd.h>
```

Standard Driver Entry Points

From `os_lib.l`:

`_os_open()`	Open Sound Device
`_os_close()`	Close Sound Device
`_os_ss_sendsig()`	Send Signal Request
`_os_ss_relea()`	Release Signal Request

Sound Input and Output

From `mfm.l`:

`_os_ss_snd_play()`	Play Sound to Sound Decoder
`_os_ss_snd_record()`	Record Sound from Sound Encoder
`_os_ss_snd_pause()`	Pause Active Sound Operation
`_os_ss_snd_cont()`	Continue Active Sound Operation
`_os_ss_snd_abort()`	Abort Active Sound Operation
`_os_ss_snd_gain()`	Gain/Mixing Control

Device Compatibility, Capability, and Status

From `mfm.l`:

`_os_gs_snd_compat()`	Exchange Compatibility Level
`_os_gs_snd_devcap()`	Get Sound Device Capabilities
`_os_gs_snd_status()`	Get Sound Device Status

_os_close()

Close Path to Sound Device

Syntax

```
error_code
_os_close(path_id path)
```

Parameter Block

```
typedef struct i_close_pb {
   syscb cb;
   path_id path;
} i_close_pb, *I_close_pb;
```

Description

`_os_close()` is fully documented in ***Ultra C Library Reference***.

`path` specifies an open path ID to the sound device.

Direct Errors

`EOS_BPNUM`	Bad path number.

See Also

`_os_open()`

`_os_close` in the ***Ultra C Library Reference***
`I_CLOSE` in the ***OS-9000 Technical Manual***

_os_gs_snd_compat()

Exchange Compatibility Level

Syntax

```
error_code
_os_gs_snd_compat(path_id path, u_int32
*ret_sdv_compat, u_int32 api_compat)
```

Parameter Block

```
typedef struct {
   u_int16 func_code;   /* Must be FC_SND_COMPAT */
   u_int32 *ret_dev_compat;/* Driver compat level */
   u_int32 api_compat; /* API compat level */
} gs_snd_compat_pb, *Gs_snd_compat_pb;
```

Description

`_os_gs_snd_compat()` exchanges the compatibility level of the caller with the compatibility level of the sound driver.

`path` specifies an open path ID to the sound device.

The compatibility level of the driver must be set by assigning the value `MAUI_COMPAT_LEVEL` to the return parameter `ret_sdv_compat`.

`api_compat` is the compatibility level of the caller and should be set to `MAUI_COMPAT_LEVEL`. If the compatibility level of the caller is less than the driver level, then the driver must cope with the differences in the respective levels, or it must return `EOS_MAUI_INCOMPATVER` if it cannot cope.

If successful, this function returns `SUCCESS`. Otherwise, the returned value is an error code. Error codes unique to the driver are defined below.

Direct Errors

`EOS_BPNUM`	Bad path number.
`EOS_MAUI_BADPTR`	`ret_sdv_compat` is NULL.

`EOS_MAUI_INCOMPATVER`	The compatibility level of the caller is less than the driver and the driver cannot cope with the differences.

See Also

`MAUI_COMPAT_LEVEL` (See ***MAUI Programming Reference Manual***)

_os_gs_snd_devcap()

Get Sound Device Capabilities

Syntax

```
error_code
_os_gs_snd_devcap(path_id path, SND_DEV_CAP
**ret_dev_cap)
```

Parameter Block

```
typedef struct {
   u_int16 func_code;  /* Must be FC_SND_DEVCAP */
   SND_DEV_CAP **ret_dev_cap;
                           /* Device capabilities */
} gs_snd_devcap_pb, *Gs_snd_devcap_pb;
```

Description

`_os_gs_snd_devcap()` gets information about the capabilities of the sound device. This information may be used to adjust the operation of the application so that it runs properly on different hardware platforms.

`path` specifies an open path ID to the sound device.

A pointer to the buffer containing the device capabilities structure is returned in `ret_dev_cap`. A pointer to this variable should be passed to `_os_gs_snd_devcap()`.

Note

Because the memory pointed to by `ret_dev_cap` belongs to the driver, do not attempt to modify or free this memory. Also, do not attempt to access this memory after the `path` is closed.

If successful, this function returns `SUCCESS`. Otherwise, the returned value is an error code. Error codes unique to the driver are defined as follows.

Direct Errors

EOS_BPNUM	Bad path number.
EOS_MAUI_BADPTR	ret_sdv_compat is NULL.

See Also

SND_DEV_CAP

_os_gs_snd_status()

Get Sound Device Status

Syntax

```
error_code
_os_gs_snd_status(path_id path, SND_DEV_STATUS
**ret_status)
```

Parameter Block

```
typedef struct {
   u_int16 func_code;  /* Must be FC_SND_DEV_STATUS */
   SND_DEV_STATUS **ret_status;
                              /* Status of device*/
} gs_snd_status_pb, *Gs_snd_status_pb;
```

Description

`_os_gs_snd_status()` returns the current status of the sound hardware. This function may be called by any process at any time.

`path` specifies an open path ID to the sound device. The `SND_DEV_STATUS` for a device is unique for each logical unit, not necessarily unique per `path`.

A pointer to the buffer containing the device logical unit status structure is returned in `ret_status`. A pointer to this variable should be passed to `_os_gs_snd_status()`.

Note

Because the memory pointed to by `ret_status` belongs to the driver, do not attempt to modify or free this memory. Also do not attempt to access this memory after the `path` is closed.

If successful, this function returns `SUCCESS`. Otherwise, the returned value is an error code. Error codes unique to the driver are defined below.

Direct Errors

`EOS_BPNUM`	Bad path number.
`EOS_MAUI_BADPTR`	`ret_sdv_compat` is NULL.

See Also

`_os_open()`
`SND_DEV_STATUS`

_os_open()

Open Path to Sound Device

Syntax

```
#include <modes.h>
error_code
_os_open(char *name, u_int32 mode, path_id *path)
```

Parameter Block

```
#include <srvcb.h>
typedef struct i_open_pb {
   syscb cb;
   u_char *name;
   u_int16 mode;
   path_id path;        /* output */
} i_open_pb, *I_open_pb;
```

Description

`_os_open()` is fully documented in ***Ultra C Library Reference***.

`name` is a pointer to the path name of the sound device. On CDB equipped systems, use `CDB_TYPE_SOUND` in `cdb_get_ddr()`, to determine the name of the sound device.

`mode` must be set to either `S_IWRITE` to use `_os_ss_snd_play()`, or `S_IREAD` to use `_os_ss_snd_record()`. To prevent functions such as `_os_ss_snd_abort()`, `_os_ss_snd_cont()`, and `_os_ss_snd_pause()` from being ambiguous, both `S_IWRITE` and `S_IREAD` must not be set on the same path. To perform both plays and records on the same device, simply open another path to the same device with the appropriate `mode`. Only 16 bits of `mode` are used.

`path` is a pointer to the location where `_os_open()` stores the resulting path number. Multiple paths may be open to the same device. Use `_os_close` to return the resources to the system.

If successful, this function returns SUCCESS. Otherwise, the returned value is an error code. Error codes unique to the driver are defined below. This call uses the standard _os_open() binding found in os_lib.l.

Direct Errors

EOS_BMODE	Bad I/O Mode.
EOS_BPNUM	Bad path number.
EOS_FNA	File not accessible.
EOS_PNNF	Path name not found.
EOS_PTHFUL	The user's (or system) path table is full.
EOS_SHARE	Non-sharable file/device is busy.

See Also

_os_close()
_os_ss_snd_abort()
_os_ss_snd_cont()
cdb_get_ddr()

CDB_TYPE_SOUND ***MAUI Programming Reference Manual***

_os_open ***Ultra C Library Reference***

I_OPEN, S_IREAD, S_IWRITE ***OS-9000 Technical Manual***

_os_ss_relea()

Release Signal Request

Syntax

```
#include <sg_codes.h>
error_code
_os_ss_relea(path_id path)
```

Parameter Block

```
#include <srvcb.h>
typedef struct f_clrsigs_pb {
   syscb cb;
   process_id proc_id;
} f_clrsigs_pb, *F_clrsigs_pb;
```

Description

`_os_ss_relea()` releases the device from any `_os_ss_sendsig()` request made by the calling process.

`path` is the path number of the device to release.

If no signal request is active, `SUCCESS` is returned.

If successful, this function returns `SUCCESS`. Otherwise, the returned value is an error code. Error codes unique to the driver are defined below.

This call uses the standard `_os_ss_relea()` binding found in `os_lib.l`.

Direct Errors

`EOS_BPNUM`	Bad path number.
`EOS_PERMIT`	Caller is not the `path` that installed the signal request.

See Also

`_os_ss_sendsig()`

_os_ss_sendsig()

Send Signal Request

Syntax

```
#include <sg_codes.h>
error_code
_os_ss_sendsig(path_id path, signal_code signal)
```

Parameter Block

```
#include <srvcb.h>
typedef struct f_clrsigs_pb {
   syscb cb;
   process_id proc_id;
   signal_code signal;
} f_clrsigs_pb, *F_clrsigs_pb;
```

Description

`_os_ss_sendsig()` sets up a signal to send to a process when the sound device is ready to accept a new sound operation such as `_os_ss_snd_play()` and `_os_ss_snd_record()`. This function is useful to determine when the sound device becomes idle. `_os_ss_sendsig()` must be reset each time the signal is sent.

For example, process A is playing sound data. Process B could issue an `_os_ss_sendsig()` to receive a signal when it can submit it's sound data without blocking.

`path` specifies an open path ID to the sound device.

`signal` specifies the signal to send when the driver clears the `SND_STATUS_BUSY` bit in the device status (see `_os_gs_snd_status()` on page 47). If the device is ready when `_os_ss_sendsig()` is called, signal is sent immediately.

Receipt of `signal` by a process is not a guarantee that the process will get access to the device. Another process could take control of the device between when the signal is sent, and when the receiving process attempts to use the device.

If successful, this function returns `SUCCESS`. Otherwise, the returned value is an error code. Error codes unique to the driver are defined below.

This call uses the standard `_os_ss_sendsig()` binding found in `os_lib.l`.

Direct Errors

`EOS_BPNUM`	Bad path number.
`EOS_DEVBSY`	Signal request is already set.

See Also

`_os_gs_snd_status()`
`_os_ss_relea()`
`SND_DEV_STATUS`
`SND_STATUS`

_os_ss_snd_abort()

Abort Active Sound Operation

Syntax

```
error_code
_os_ss_snd_abort(path_id path)
```

Parameter Block

```
typedef struct {
  u_int16 func_code;/* Must be FC_SND_ABORT */
} ss_snd_abort_pb, *Ss_snd_abort_pb;
```

Description

`_os_ss_snd_abort()` requests the driver to synchronously abort the currently active sound operation such as `_os_ss_snd_play()` and `_os_ss_snd_record()`. If there is no sound operation active, this function returns `EOS_MAUI_NOTBUSY`.

`path` specifies an open path ID to the sound device.

If a sound operation is currently active, it is aborted, `EOS_ABORT` is placed in the `err_code` member of the `SND_SMAP` referenced by the operation and the functions returns when the device is ready for the next sound operation.

The `SND_STATUS_FINISH` bit in the `trig_status` member of `SND_SMAP` is set and the signal `trig_signal` specified in `SND_SMAP` is sent to the calling process. If the specified signal is zero, then no signal is sent.

If this function is called with a `path` other than the `path` that initiated the sound operation, this function returns `EOS_MAUI_NOTOWNER`. If successful, this function returns `SUCCESS`.

Direct Errors

`EOS_BPNUM`	Bad path number.
`EOS_MAUI_NOTBUSY`	The sound hardware is not busy playing or recording, so there is nothing to abort.

`EOS_MAUI_NOTOWNER`	Caller is not the `path` that started the current sound operation.

See Also

`_os_ss_snd_cont()`
`_os_ss_snd_pause()`
`_os_ss_snd_play()`
`_os_ss_snd_record()`
`SND_SMAP`

_os_ss_snd_cont()

Continue Active Sound Operation

Syntax

```
error_code
_os_ss_snd_cont(path_id path)
```

Parameter Block

```
typedef struct {
   u_int16 func_code;/* Must be FC_SND_CONT */
} ss_snd_cont_pb, *Ss_snd_cont_pb;
```

Description

_os_ss_snd_cont() requests the driver to continue the current paused sound operation such as _os_ss_snd_play() or _os_ss_snd_record(). If there is no sound operation paused, this function returns EOS_MAUI_NOTPAUSED.

path specifies an open path ID to the sound device.

If a play is currently paused, it is continued and the SND_STATUS_PLAY_PAUSED bit in status of SND_DEV_STATUS is cleared. If a record is currently paused, it is continued and the SND_STATUS_RECORD_PAUSED bit in *status* of SND_DEV_STATUS is cleared. No other status bits are affected by the pause.

If the hardware does not support pause or continue, _os_ss_snd_cont() returns EOS_MAUI_NOHWSUPPORT.

If this function is called with a path other than the path that initiated the sound operation, this function returns EOS_MAUI_NOTOWNER. If successful, this function returns SUCCESS.

Direct Errors

EOS_BPNUM	Bad path number.
EOS_MAUI_NOHWSUPPORT	The hardware does not support pause and continue.

EOS_MAUI_NOTBUSY	The sound hardware is not busy so there is no play or record to act on.
EOS_MAUI_NOTOWNER	Caller is not the `path` that started the current sound operation.
EOS_MAUI_NOTPAUSED	The sound hardware is not paused so there is no play or record to act on.

See Also

_os_ss_snd_abort()
_os_ss_snd_pause()
_os_ss_snd_play()
_os_ss_snd_record()
SND_SMAP

_os_ss_snd_gain()

Set Gain

Syntax

```
error_code
_os_ss_snd_gain(path_id path, SND_GAIN *gain)
```

Parameter Block

```
typedef struct {
   u_int16 func_code;/* Must be FC_SND_GAIN */
   SND_GAIN *gain;/* Gain Setting */
} ss_snd_gain_pb, *Ss_snd_gain_pb;
```

Description

`_os_ss_snd_gain()` requests the driver to change the gain level.

`path` specifies an open path ID to the sound device.

`gain` is a pointer to a data structure that describes how to change the gain level and what lines are affected. (See `SND_GAIN` on page 92.) If this function is called with a `path` other than the `path` that initiated the current sound operation, this function returns `EOS_MAUI_NOTOWNER`.

`_os_gs_snd_devcap()` may be used to determine which mix lines are supported by the hardware. `_os_gs_snd_status()` may be used to determine current gain levels.

If successful, this function returns `SUCCESS`.

Direct Errors

`EOS_BPNUM`	Bad path number.
`EOS_MAUI_BADPTR`	The pointer `gain` is NULL.
`EOS_MAUI_BADVALUE`	Unknown gain command in `gain->cmd`.
`EOS_MAUI_NOHWSUPPORT`	The value passed in `play->cmd` is not supported by the hardware.

EOS_MAUI_NOTOWNER	Caller is not the path that started the current sound operation.
EOS_UNKSVC	The hardware is incapable of supporting gain control for the specified line.

See Also

_os_gs_snd_devcap()
_os_gs_snd_status()
SND_DEV_CAP
SND_DEV_STATUS
SND_GAIN

_os_ss_snd_pause()

Pause Active Sound Operation

Syntax

```
error_code
_os_ss_snd_pause(path_id path)
```

Parameter Block

```
typedef struct {
  u_int16 func_code;/* Must be FC_SND_PAUSE */
} ss_snd_pause_pb, *Ss_snd_pause_pb;
```

Description

`_os_ss_snd_pause()` requests the driver to pause the current active sound operation such as `_os_ss_snd_play()` or `_os_ss_snd_record()`. If there is no sound operation active, this function returns `EOS_MAUI_NOTBUSY`.

`path` specifies an open path ID to the sound device.

If a play is currently active, it is paused and the `SND_STATUS_PLAY_PAUSED` bit in `status` of `SND_DEV_STATUS` is set. If a record is currently active, it is paused and the `SND_STATUS_RECORD_PAUSED` bit in `status` of `SND_DEV_STATUS` is set. No other status bits are affected by the pause.

If the hardware does not support pause and continue, `sup_play_pause` returns `EOS_MAUI_NOHWSUPPORT`.

If this function is called with a `path` other than the `path` that initiated the sound operation, this function returns `EOS_MAUI_NOTOWNER`.

If successful, this function returns `SUCCESS`. Otherwise, the returned value is an error code. Error codes unique to the driver are defined below.

Direct Errors

`EOS_BPNUM`	Bad path number.

EOS_MAUI_NOHWSUPPORT	The hardware does not support pause and continue.
EOS_MAUI_NOTOWNER	Caller is not the `path` that started the current sound operation.
EOS_MAUI_NOTBUSY	The sound hardware is not busy playing or recording so there is no play or record to act on.
EOS_MAUI_PAUSED	The sound hardware is already paused.

See Also

_os_ss_snd_abort()
_os_ss_snd_cont()
_os_ss_snd_play()
_os_ss_snd_record()
SND_DEV_CAP
SND_DEV_STATUS
SND_SMAP

_os_ss_snd_play()

Play Sound to Sound Decoder

Syntax

```
error_code
_os_ss_snd_play(path_id path, SND_SMAP *smap,
SND_BLOCK_TYPE block_type)
```

Parameter Block

```
typedef struct {
   u_int16 func_code;/* Must be FC_SND_PLAY */
   SND_SMAP *smap;/* Sound map */
   SND_BLOCK_TYPE block_type;
      /* Type of blocking to */
      /* perform */
} ss_snd_play_pb, *Ss_snd_play_pb;
```

Description

`_os_ss_snd_play()` requests the sound decoder to decode and play sound data. The sound data and all parameters necessary to decode it, are stored in the sound map referenced by the pointer `smap`.

`path` specifies an open path ID to the sound device. `path` must be opened for write access.

The driver only processes one `SND_SMAP` at a time, following the `smap->next` pointers as the sound data is consumed. As each `SND_SMAP` is accepted by the driver, it sets the `SND_TRIG_BUSY` bit in `smap->trig_status`, clears `smap->loop_counter`, and sets `smap->cur_offset` equal to `smap->loop_start`.

The blocking type determines how the driver functions. One of three blocking types must be specified:

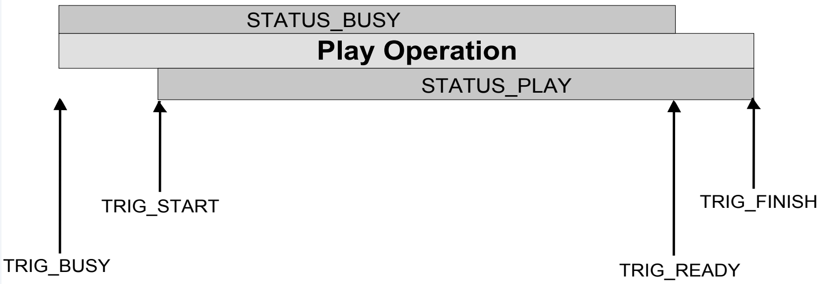
SND_NOBLOCK	prevents the driver from blocking. If the driver is busy when called, it immediately returns the error EOS_MAUI_BUSY. Otherwise, it starts the play and returns immediately.
SND_BLOCK_START	returns when the play starts (when SND_TRIG_START is set). The play call waits while the device is busy and returns immediately when the sound device begins to play.
SND_BLOCK_FINISH	returns when the play is finished (when SND_TRIG_FINISHED is set). The play call waits to return until all linked sound maps are consumed or the play operation is aborted.

If a fatal signal is received while the call is blocked waiting for access to the sound device (the play has not started yet and block_type is equal to SND_BLOCK_START or SND_BLOCK_FINISH) the call returns EOS_SIGNAL and the play request is canceled. If a fatal signal is received after the play has started, and the call is blocked waiting for the play to finish (block_type is equal to SND_BLOCK_FINISH) the call returns EOS_SIGNAL, but the play operation continues. Examine smap->trig_status to determine if the smap is still being used by the driver.

When the sound decoder completes an activity that satisfies a trigger in the smap->trig_mask and smap->trig_signal is not zero, the driver sends the signal smap->trig_signal to the calling process.

The following figure shows the status and trigger points of a typical play operation

Figure 3-1 os_ss_snd_play() Status and Trigger Points



After consuming the data in `smap->buf`, if the `smap->next` field is not `NULL`, the driver sets the internal `smap` pointer to `smap->next` and continues.

If successful, this function returns `SUCCESS`. Otherwise, the returned value is an error code. Error codes unique to the driver are defined below.

Direct Errors

`EOS_BMODE`	The path is not open for write access.
`EOS_BPNUM`	Bad path number.
`EOS_MAUI_BADNUMCHAN`	The specified `smap->num_channels` is not supported by the hardware.
`EOS_MAUI_BADPTR`	The pointer for `smap` or `smap->buf` is NULL.
`EOS_MAUI_BADRATE`	The specified `smap->sample_rate` is not supported by the hardware.
`EOS_MAUI_BADSIZE`	The specified `smap->sample_size` is not appropriate for the `smap->coding_method` or is not supported by the hardware.

`EOS_MAUI_BADVALUE`	The value passed for `block_type` is invalid or one or more of the loop fields are invalid*.
`EOS_MAUI_BUSY`	The sound decoder is busy.
`EOS_MAUI_NOHWSUPPORT`	The specified `smap->coding_method` is not supported by the hardware.
`EOS_SIGNAL`	A fatal signal was received while blocked.
`EOS_UNKSVC`	This function is not supported because the sound hardware does not contain a decoder.

```
* if (smap->loop_count > 0) {
  if (smap->loop_start >= smap->loop_end
  || smap->loop_end > smap->buf_size)}
  return (smap->err_code = EOS_MAUI_BADVALUE);
  }
  }
```

Indirect Errors

`_os_ev_anyclr()`
`_os_send()`
`_os_ev_setand()`
`_os_ev_tstset()` ***OS-9 Technical Reference***

See Also

`_os_ss_snd_abort()`
`_os_ss_snd_gain()`
`_os_ss_snd_cont()`
`_os_ss_snd_pause()`
`SND_BLOCK_TYPE`
`SND_SMAP`
`SND_STATUS`
`SND_TRIGGER`

_os_ss_snd_record()

Record Sound

Syntax

```
error_code
_os_ss_snd_record(path_id path, SND_SMAP *smap,
SND_BLOCK_TYPE block_type)
```

Parameter Block

```
typedef struct {
    u_int16 func_code;   /* Must be FC_SND_RECORD */
    SND_SMAP *smap;      /* Sound map */
    SND_BLOCK_TYPE block_type;
                         /* Type of blocking to perform */
} ss_snd_record_pb, *Ss_snd_record_pb;
```

Description

`_os_ss_snd_record()` requests the sound encoder to record sound data. The buffer for encoded sound data and all parameters necessary to encode it, are stored in the sound map referenced by the pointer `smap`.

`path` specifies an open path ID to the sound device. `path` must be opened for read access.

The driver only processes one `SND_SMAP` at a time, following the `smap->next` pointers as the sound data fills each `smap->buf`. As each `SND_SMAP` is accepted by the driver, it sets the `SND_TRIG_BUSY` bit in `smap->trig_status`.

`_os_ss_snd_record()` does not support loop operations. The `SND_SMAP` fields `loop_start`, `loop_end`, `loop_count`, and `loop_counter` must be set to zero by the application before the sound map is accepted by the driver. The driver returns `EOS_MAUI_BADVALUE` if any of these fields are not zero.

The blocking type determines how the driver functions. One of three blocking types must be specified:

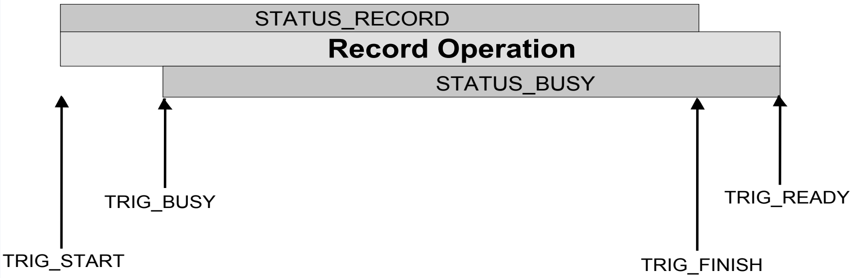
`SND_NOBLOCK`	prevents the driver from blocking. If the driver is busy when called, it immediately returns the error `EOS_MAUI_BUSY`. Otherwise, it starts the record and returns immediately.
`SND_BLOCK_START`	returns when the recording starts (when `SND_TRIG_STAR` is set). The record call waits while the device is busy and returns immediately when the sound device begins to record.
`SND_BLOCK_FINISH`	returns when the record has finished (when `SND_TRIG_FINISH` is set). The record call waits to return until all linked sound maps are consumed or the record operation is aborted.

If a fatal signal is received while the call is blocked waiting for access to the sound device (the record has not started yet and `block_type` is equal to `SND_BLOCK_START` or `SND_BLOCK_FINISH`) the call returns `EOS_SIGNAL` and the record request is canceled. If a fatal signal is received, after the record has started, and the call is blocked waiting for the record to finish (`block_type` is equal to `SND_BLOCK_FINISH`) the call returns `EOS_SIGNAL`, but the record operation continues. Examine `smap->trig_status` to determined if the `smap` is still being used by the driver.

When the sound encoder completes an activity that satisfies a trigger in the `smap->trig_mask`, and `smap->trig_signal` is not zero, the driver sends the signal `smap->trig_signal` to the calling process.

The following figure illustrates the status and trigger points of a typical record operation.

Figure 3-2 _os_ss_snd_record() Status and Trigger Points



After filling the data in `smap->buf`, if the `smap->next` field is not `NULL`, the driver sets the internal `smap` pointer to `smap->next` and continues.

If successful, this function returns `SUCCESS`. Otherwise, the returned value is an error code. Error codes unique to the driver are defined below.

Direct Errors

`EOS_BMODE`	The path is not open for read access.
`EOS_BPNUM`	Bad path number.
`EOS_MAUI_BADNUMCHAN`	The specified `smap->num_channels` is not supported by the hardware.
`EOS_MAUI_BADPTR`	The pointer for `smap` or `smap->buf` is NULL.
`EOS_MAUI_BADRATE`	The specified `smap->sample_rate` is not supported by the hardware.
`EOS_MAUI_BADSIZE`	The specified `smap->sample_size` is not appropriate for the `smap->coding_method` or is not supported by the hardware.

EOS_MAUI_BADVALUE	The value passed for block_type is invalid, *or* one or more of the following smap entries are not equal to zero: loop_start, loop_end, loop_count, loop_counter.
EOS_MAUI_BUSY	The sound encoder is busy.
EOS_MAUI_NOHWSUPPORT	The coding_method is not supported by the hardware.
EOS_SIGNAL	A fatal signal was received while blocked.
EOS_UNKSVC	This function is not supported because the sound hardware does not contain an encoder.

Indirect Errors

_os_ev_anyclr()
_os_send()
_os_ev_setand()
_os_ev_tstset() ***OS-9 Reference Manual***

See Also

_os_ss_snd_abort()
_os_ss_snd_cont()
_os_ss_snd_gain()
_os_ss_snd_pause()
SND_BLOCK_TYPE
SND_SMAP
SND_STATUS
SND_TRIGGER

Chapter 4: Data Type Reference

This chapter provides a detailed reference for each data type defined in the Sound Driver Interface.

Data Type Reference

Defined Constants

SND_LEVEL_*	Gain Level Constants

Enumerated Types

SND_BLOCK_TYPE	Blocking Types

Data Types

SND_CM	Sound Coding Methods

Integers

SND_GAIN_CMD	Gain Commands
SND_LINE	Gain/Mixer Line Types
SND_STATUS	Status
SND_TRIGGER	Triggers

Data Structures

SND_DEV_CAP	Sound Device Capabilities
SND_DEV_CM	Sound Device Coding Method
SND_DEV_STATUS	Sound Device Status
SND_GAIN	Gain Control
SND_GAIN_UP	Gain Up Parameters

SND_GAIN_DOWN	Gain Down Parameters
SND_GAIN_MUTE	Mute Gain Parameters
SND_GAIN_MONO	Mono Gain Parameters
SND_GAIN_STEREO	Stereo Gain Parameters
SND_GAIN_XSTEREO	Cross-Stereo Gain Parameters
SND_GAIN_CAP	Sound Device Gain Capabilities
SND_SMAP	Sound Map

SND_BLOCK_TYPE

Blocking Types

Syntax

```
typedef enum {
   SND_BLOCK_START,     /* Block until sound */
                        /* operation starts */
   SND_BLOCK_FINISH,    /* Block until sound */
                        /* operation ends */
   SND_NOBLOCK          /* Do not block */
} SND_BLOCK_TYPE;
```

Description

This enumerated type defines the blocking mechanisms that are available when playing or recording sound samples.

The blocking type determines the behavior of the sound operation functions `_os_ss_snd_play()` and `_os_ss_snd_record()`. These functions are called with one of three blocking types:

`SND_BLOCK_START`	returns when the sound operation starts. The function call waits while the device is busy and returns immediately when the sound device begins to play or record.
`SND_BLOCK_FINISH`	returns when the sound operation is finished. The function call waits to return until all linked sound maps are consumed or the sound operation is aborted.
`SND_NOBLOCK`	prevents the driver from blocking. If the driver is busy when called, it immediately returns the error `EOS_MAUI_BUSY`. Otherwise, it starts the sound operation and returns immediately.

See Also

`_os_ss_snd_play()`
`_os_ss_snd_record()`

SND_CM

Sound Coding Methods

Syntax

```
typedef u_int32 SND_CM;
```

Description

This data type specifies a sound coding method. In a `SND_SMAP`, the `SND_CM` is used to tell the driver the type of sound data being passed to it for play operations, or what is required from it during record operations.

Be aware that not all sound coding methods can be encoded or decoded by all hardware. Use `_os_ss_snd_devcap()` to determine which sound coding methods the driver and hardware support.

The following diagram shows the bit-fields contained in the sound coding method:

Figure 4-1 Coding Method

31	30	29	28	27	26	25	24	23	22	21	20	19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
OEM							Reserved					BE	DE	Reserved								Name									

OEM

Bits 31 through 25 may be used by OEMs to indicate implementation- or driver-specific coding method modifiers.

BE

Bit 19 is a coding method modifier that indicates the **byte** endianess of the sound data. If this bit is set, the sound data is formatted as least significant byte first. If this bit is not set, the sound data is formatted as most significant byte first. This bit is considered a modifier to the sound coding method name.

Use the macro `snd_get_cm_byte_order(cm)` to get the byte endianess in a `SND_CM`. For example:

```
byteorder=snd_get_cm_byte_order(cm);
```

The `byteorder` variable is equal to `MSBFIRST` or `LSBFIRST`.

Use the macro `snd_set_cm_byte_order(order)` with `order` equal to `MSBFIRST` or `LSBFIRST`, or use the macros `SND_CM_MSBYTE1ST` or `SND_CM_LSBYTE1ST` to set the byte endianess in a `SND_CM`.

The following two code segments set least significant byte ordering to the variable `cm`:

```
cm=SND_CM_LSBYTE1ST |
   snd_get_cm_name(SND_CM_PCM_SLINEAR);
cm=snd_set_cm_byte_order(LSBFIRST) |
   snd_get_cm_name(SND_CM_PCM_SLINEAR);
```

Most significant byte ordering is the default. You do not need to use either of these macros, but they are provided for completeness. The following two code segments set most significant byte ordering to the variable `cm`:

```
cm=SND_CM_MSBYTE1ST |
   snd_get_cm_name(SND_CM_PCM_SLINEAR);
cm=snd_set_cm_byte_order(LSBFIRST) |
   snd_get_cm_name(SND_CM_PCM_SLINEAR);
```

bE

Bit 18 is a coding method modifier that indicates the **bit** endianess of the sound data. If this bit is set, the sound data is formatted as least significant bit first. If this bit is not set, the sound data is formatted as most significant bit first. This bit is considered a modifier to the sound coding method name.

Use the macro `snd_get_cm_bit_order(cm)` to get the bit endianess in a `SND_CM`. For example:

```
bitorder=snd_get_cm_bit_order(cm);
```

The `bitorder` variable is equal to `MSBFIRST` or `LSBFIRST`.

Use the macro `snd_set_cm_bit_order(order)` with `order` equal to `MSBFIRST` or `LSBFIRST`, or use the macros `SND_CM_MSBIT1ST` or `SND_CM_LSBIT1ST` to set the bit endianess in a `SND_CM`.

The following two code segments set least significant bit ordering to the variable `cm`:

```
cm=SND_CM_LSBIT1ST |
    snd_get_cm_name(SND_CM_PCM_SLINEAR);
cm=snd_set_cm_bit_order(LSBFIRST) |
    snd_get_cm_name(SND_CM_PCM_SLINEAR);
```

Most significant bit ordering is the default. You do not need to use either of these macros, but they are provided for completeness. The following two code segments set most significant bit ordering to the variable `cm`:

```
cm=SND_CM_MSBIT1ST |
    snd_get_cm_name(SND_CM_PCM_SLINEAR);
cm=snd_set_cm_bit_order(LSBFIRST) |
    snd_get_cm_name(SND_CM_PCM_SLINEAR);
```

Name

Bits 0 through 9 define the name of the coding method. This field is segmented into the following numeric ranges to indicate the class of coding method as defined in the following table:

Table 4-1 Coding Method Classes

Numeric Range	Description
`0-255`	Standard Microware-defined coding methods.
`256-767`	Reserved.
`768-1023`	Defined by OEMs.

The following standard coding method names are defined:

Table 4-2 Standard Coding Method Names

Value	Name	Description
0	`SND_CM_UNKNOWN`	Unknown or not yet determined.
1	`SND_CM_PCM_ULAW`	µLAW encoded PCM.
2	`SND_CM_PCM_ALAW`	ALAW encoded PCM.
3	`SND_CM_PCM_SLINEAR`	Signed Linear encoded PCM.
4	`SND_CM_PCM_ULINEAR`	Unsigned Linear encoded PCM.
5	`SND_CM_ADPCM_G721`	CCITT G.721 ADPCM.

Table 4-2 Standard Coding Method Names

Value	Name	Description
6	SND_CM_ADPCM_G723	CCITT G.723 ADPCM.
7	SND_CM_ADPCM_IMA	IMA ADPCM.

Use the macro `snd_get_cm_name(cm)` to extract the coding method name from a `SND_CM`. For example:

```
name=snd_get_cm_name(cm);
```

Use the macro `snd_set_cm_name(name)` to set the coding method name from a `SND_CM`. For example:

```
cm=snd_set_cm_name(name);
```

SND_CM_UNKNOWN

This type value indicates that the sound coding method is unknown or unspecified.

SND_CM_PCM_SLINEAR, SND_CM_PCM_SLINEAR | SND_CM_LSBYTE1ST SND_CM_PCM_ULINEAR SND_CM_PCM_ULINEAR | SND_CM_LSBYTE1ST

Pulse Coded Modulation (PCM) is one of the most common methods used by sound cards to record and play back recorded sound. `SND_CM_PCM_SLINEAR` is defined as signed (2's complement) linear PCM samples. `SND_CM_PCM_ULINEAR` represents unsigned PCM data.

Multi-channel samples under `SND_CM_PCM_SLINEAR` and `SND_CM_PCM_ULINEAR` are defined to be interleaved on a frame-by-frame basis: if there are N channels, the data is a sequence of frames, where each frame contains N samples, one from each channel (thus, the sampling rate is really the number of frames per second). For stereo, the left channel comes first, followed by the right channel.

Mono and stereo 8-bit linear PCM data is depicted in the following two figures.

Figure 4-2 Mono, 8-bit Data SND_CM_PCM_SLINEAR, SND_CM_PCM_ULINEAR, SND_CM_PCM_ULAW, SND_CM_PCM_ALAW

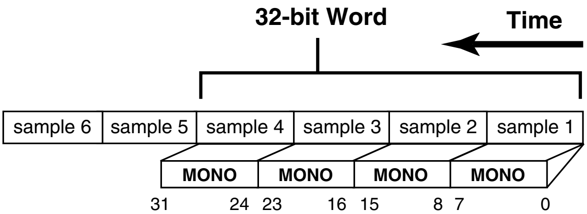
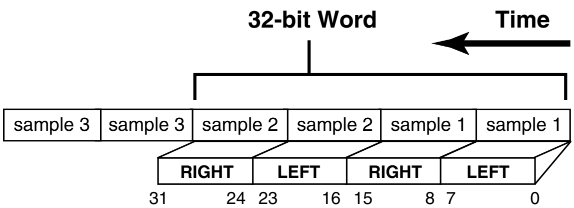


Figure 4-3 Stereo 8-bit, Data SND_CM_PCM_SLINEAR, SND_CM_PCM_ULINEAR, SND_CM_PCM_ULAW, SND_CM_PCM_ALAW



For 16-bit or larger linear PCM data, either the `SND_CM_MSBYTE1ST` or the `SND_CM_LSBYTE1ST` modifiers may be used. Mono and stereo 16-bit `SND_CM_PCM_SLINEAR` and `SND_CM_PCM_ULINEAR` are depicted in the following two figures.

Figure 4-5 Mono 16-bit, Big Endian Data SND_CM_PCM_SLINEAR SND_CM_PCM_ULINEAR

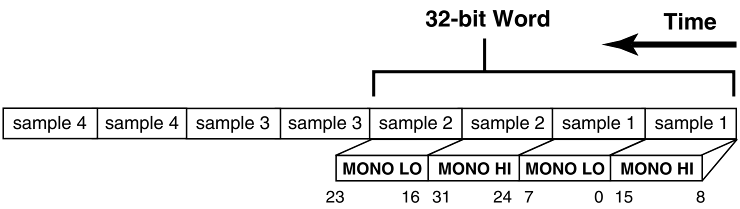
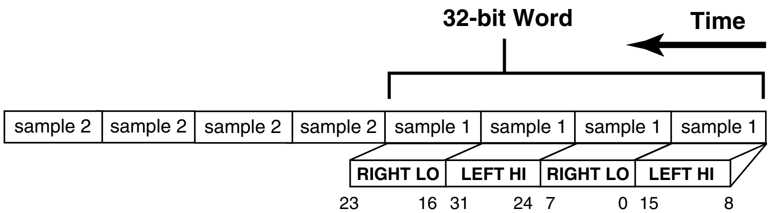


Figure 4-6 Stereo 16-bit, Big Endian Data SND_CM_PCM_SLINEAR SND_CM_PCM_ULINEAR



Mono and stereo 16-bit `SND_CM_PCM_SLINEAR` | `SND_CM_LSBYTE1ST` and `SND_CM_PCM_ULINEAR` | `SND_CM_LSBYTE1ST` are depicted in the following two figures.

Figure 4-7 Mono 16-bit, Little Endian Data
SND_CM_PCM_SLINEAR|SND_CM_LSBYTE1ST
SND_CM_PCM_ULINEAR|SND_CM_LSBYTE1ST

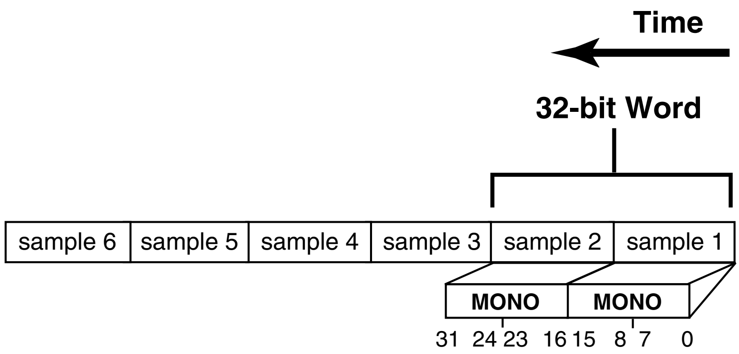
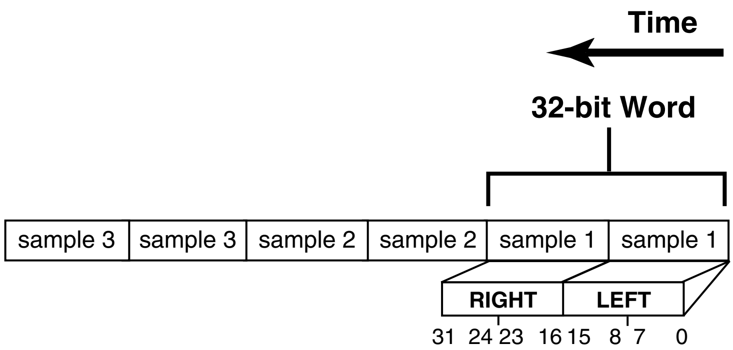


Figure 4-8 16-bit, Little Endian Data
SND_CM_PCM_SLINEAR|SND_CM_LSBYTE1ST
SND_CM_PCM_ULINEAR|SND_CM_LSBYTE1ST



SND_CM_PCM_ULAW
SND_CM_PCM_ALAW

The µ-Law (pronounced mu-law) format is an 8-bit companded format that is a telephony standard used in the United States and Japan. The A-Law format is an 8-bit companded format that is a telephony standard used in Europe. Under these encoding methods, the samples are logarithmically encoded into 8 bits. The quantization is not uniform (as in `SND_CM_PCM_SLINEAR` and `SND_CM_PCM_ULINEAR`), but is skewed so that the signal is sampled with greater resolution when the amplitude is less.

The official definition for µ-Law is contained in the CCITT standard G.711. `SND_CM_PCM_ULAW` and `SND_CM_PCM_ALAW` are depicted in the two following figures.

Figure 4-9 Stereo 16-bit, Big Endian Data SND_CM_PCM_SLINEAR SND_CM_PCM_ULINEAR

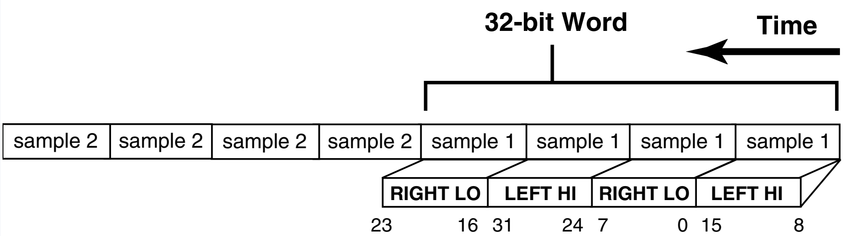
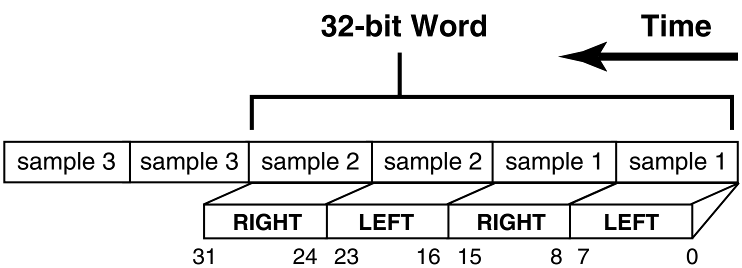


Figure 4-9 Stereo 8-bit, Data SND_CM_PCM_SLINEAR, SND_CM_PCM_ULINEAR, SND_CM_PCM_ULAW, SND_CM_PCM_ALAW



SND_CM_PCM_ALAW is depicted in the following two figures:

**Figure 4-10 Stereo 16-bit, Big Endian Data
SND_CM_PCM_SLINEAR
SND_CM_PCM_ULINEAR**

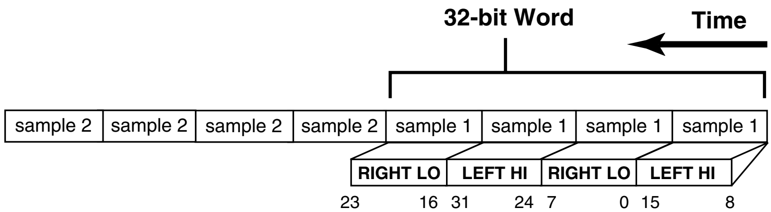
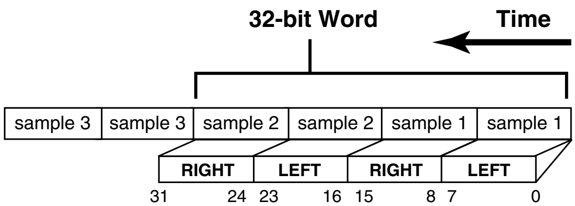


**Figure 4-10 Stereo 8-bit, Data
SND_CM_PCM_SLINEAR, SND_CM_PCM_ULINEAR,
SND_CM_PCM_ULAW, SND_CM_PCM_ALAW**



No modifiers are appropriate for SND_CM_PCM_ULAW or SND_CM_PCM_ALAW.

SND_CM_ADPCM_IMA

Adaptive Differential Pulse Code Modulation (ADPCM) is used for improved performance and compression ratios over µ-Law and A-Law. The IMA ADPCM format uses the DVI® ADPCM algorithm. It provides a 4-to-1 compression ratio (4 bits are saved for each 16-bit sample capture). *

For more detailed information on the IMA ADPCM format contact IMA at (410) 626-1380.

The SND_CM_ADPCM_IMA format is depicted in the following two figures.

Figure 4-11 4-bit Mono, IMA ADPCM Data SND_CM_ADPCM_IMA

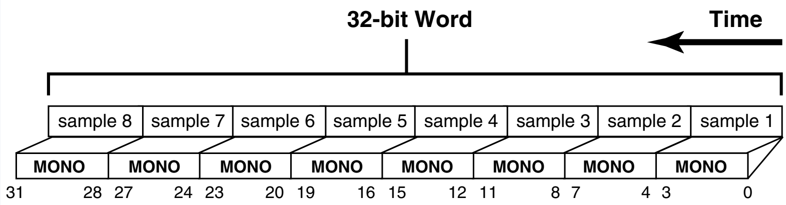
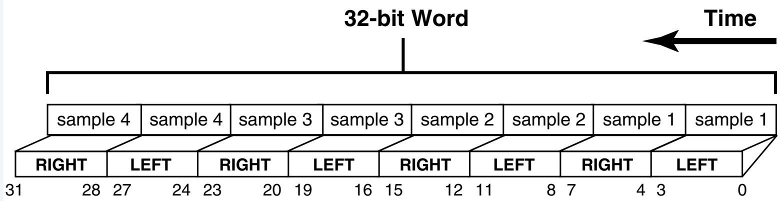


Figure 4-12 4-bit Stereo, IMA ADPCM Data SND_CM_ADPCM_IMA



SND_CM_ADPCM_G721

ADPCM stands for Adaptive Delta Pulse Code Modulation. G.721 is a CCITT standard for ADPCM at 32 kbits/second.

*. There are two forms of IMA ADPCM in common use. The most common form found in WAV audio files is a WAV type 0x11. This is an enhanced definition of the IMA ADPCM specification that divides the data into autonomous blocks. This form is usually encoded/decoded in software from/to 16-bit PCM. The second form of IMA ADPCM found in WAV audio files is a WAV type 0x39. This form follows the original IMA specification and is not “blocked”. This latter form can be encoded and decoded directly by several popular CODECs. This second form is represented by the `SND_CM` type `SND_CM_ADPCM_IMA`.

SND_CM_ADPCM_G723

G.723 is a CCITT standard for ADPCM at 24 and 40 kbits/second.

SND_CM_OEM_*

We recommend that if an OEM defines a coding method, that it be prefixed with `SND_CM_OEM_` and that it be in the range 768 through 1023. We also request that OEMs submit a detailed description of their coding methods to Microware. These coding methods may become part of the Standard Coding Method Names in future releases.

See Also

`_os_gs_snd_devcap()`
`SND_DEV_CAP`
`SND_DEV_CM`
`SND_SMAP`

Note

CCITT G.711, G.721, G.723: Sun and Microsoft have placed the source code of a portable implementation of these algorithms in the public domain. One place to ftp this source code from is ftp.cwi.nl:/pub/audio/ccitt-adpcm.tar.Z.

SND_DEV_CAP

Sound Device Capabilities

Syntax

```
typedef struct _SND_DEV_CAP {
   char *hw_type;            /* Hardware type */
   char *hw_subtype;         /* Hardware subtype */
   SND_TRIGGER sup_triggers;/* Supported triggers */
   SND_LINE play_lines;     /* Play Gain/mix lines */
   SND_LINE record_lines;   /* Record Gain/mix lines */
   SND_GAIN_CMD sup_gain_cmds;
                             /* Gain cmd mask */
   u_int16 num_gain_caps;   /* Num gain caps */
   SND_GAIN_CAP *gain_caps;/* Ptr to gain cap array */
   u_int16 num_rates;       /* Num sample rates
   u_int32 *sample_rates;  /* Ptr to sample rate array */
   u_int16 num_chan_info;  /* Num channel infos */
   u_int16 *channel_info;/* Ptr to channel info array */
   u_int16 num_cm;          /* Num coding methods */
   SND_DEV_CM *cm_info;     /* Ptr to coding method array
*/
}  SND_DEV_CAP;
```

Description

This data structure defines the capabilities supported by a sound device. Appendix A: Sound Hardware Specifications gives detailed information about the capabilities of some specific MAUI sound drivers. Use `_os_gs_snd_devcap()` to retrieve this information from the driver.

`hw_type` string indicates the class of hardware.

`hw_subtype` string indicates the sub-class of hardware.

`sup_triggers` indicates which triggers are supported by this device.

`play_lines` indicates which mix lines control the gain level for `_os_ss_snd_play()`.

`record_lines` indicates which mix lines control the gain level for `_os_ss_snd_record()`.

`sup_gain_cmds` is a mask of the supported gain commands.

`gain_caps` is a pointer to an array of `SND_GAIN_CAP` structures, with `num_gain_caps` entries. This array has entries for each mixing line supported by the hardware.

`sample_rates` is a pointer to an array of the sample rates supported by the sound hardware, with `num_rates` entries.

`channel_info` is a pointer to an array of the various number of channels supported by the sound hardware, with `num_chan_info` entries. The `channel_info` array is typically one of the following:

mono only	one element containing the value 1
stereo only	one element containing the value 2
stereo and mono	two elements, one containing the value 1 and the other containing the value 2

`cm_info` is a pointer to an array of `SND_DEV_CM` structures that describe coding methods supported by the sound hardware. The array has `num_cm` entries.

See Also

`_os_gs_snd_devcap()`
`_os_ss_snd_cont()`
`_os_ss_snd_pause()`
`_os_ss_snd_play()`
`_os_ss_snd_record()`
`SND_DEV_CM`
`SND_GAIN_CAP`
`SND_LINE`
`SND_TRIGGER`

SND_DEV_CM

Sound Device Coding Method

Syntax

```
typedef struct _SND_DEV_CM {
   SND_CM coding_method; /* Coding method */
   u_int32 sample_size;/* Number of bits per sample */
   u_int16 boundary_size;/* Boundary limitations */
} SND_DEV_CM;
```

Description

This data structure defines a coding method supported by a sound device. An array of sound coding methods supported by a driver is available via `_os_gs_snd_devcap()`. Appendix A: Sound Hardware Specifications gives detailed information about the capabilities of some specific MAUI drivers.

`coding_method` is the coding method of the sound data.

`sample_size` defines the size of each sample in bits.

`boundary_size` indicates the hardware memory boundary limitations in bytes. Memory submitted to the driver, via the `buf` field of a `SND_SMAP`, must start on the appropriate memory boundary and be a multiple of `boundary_size`. For example, if `boundary_size` is equal to 2, then `buf` should begin on an even byte boundary and `buf_size` should also be an even number of bytes. If `boundary_size` is equal to zero or 1, then there are no hardware limitations.

See Also

`_os_gs_snd_devcap()`
`SND_CM`
`SND_DEV_CAP`
`SND_SMAP`

SND_DEV_STATUS

Sound Device Status

Syntax

```
typedef struct _SND_DEV_STATUS {
   SND_STATUS status;   /* Current status*/
   process_id play_pid;/* Current play PID*/
   process_id record_pid;/* Current record PID*/
   u_int16 num_gain;    /* Num SND_GAIN structures */
   SND_GAIN *gain;      /* Ptr to SND_GAIN array */
} SND_DEV_STATUS;
```

Description

This structure returns the current status of the sound device. This structure indicates what is being performed by which process. A pointer to this data structure is returned by `_os_gs_snd_status()`.

`status` indicates the current status of the sound device.

If a play operation is active, the `play_pid` field contains the process id of the process that initiated the play. Otherwise, it contains zero.

If a record operation is active, the `record_pid` field contains the process id of the process that initiated the record. Otherwise it contains zero.

`gain` is a pointer to an array of `SND_GAIN` structures with `num_gain` entries. This array has entries for each mixing line that can be controlled on this hardware. Each entry indicates the current settings for each line.

See Also

`_os_gs_snd_status()`
`SND_GAIN`
`SND_STATUS`

SND_GAIN

Gain Control

Syntax

```
typedef struct _SND_GAIN {
   SND_LINE lines;        /* Mixer line mask */
   SND_GAIN_CMD cmd;      /* Type of gain */
   union {                /* cmd == */
   SND_GAIN_UP up;        /* SND_GAIN_CMD_UP */
      SND_GAIN_DOWN down;/* SND_GAIN_CMD_DOWN */
      SND_GAIN_MUTE mute;/* SND_GAIN_CMD_MUTE */
      SND_GAIN_MONO mono;/* SND_GAIN_CMD_MONO */
      SND_GAIN_STEREO stereo;/* SND_GAIN_CMD_STEREO */
      SND_GAIN_XSTEREO xstereo;
                          /* SND_GAIN_CMD_XSTEREO*/
   } param;
} SND_GAIN;
```

Description

This data structure is passed in `_os_ss_snd_gain()` to control input and output gain. The gain information specifies how much of each signal from the sound decoder/encoder contributes to the final channel outputs/inputs (such as speakers, headphones, microphones).

`lines` specifies which mix lines are affected.

`cmd` specifies how to modify the gain level.

The union `param` supplies the various parameters dependent on the value of `cmd`. The following table identifies which `cmd` values use which `params`:

Table 4-3 Relationship Between cmd and param in SND_GAIN

cmd	params
SND_GAIN_CMD_UP	param.up.levels
SND_GAIN_CMD_DOWN	param.down.levels
SND_GAIN_CMD_RESET	No params
SND_GAIN_CMD_MUTE	para.mute.state
SND_GAIN_CMD_MONO	param.mono.m
SND_GAIN_CMD_STEREO	param.stereo.ll, param.stereo.rr
SND_GAIN_CMD_XSTEREO	param.xstereo.ll param.xstereo.rr param.xstereo.rl param.xstereo.lr

See Also

_os_ss_snd_gain()
SND_GAIN_CMD
SND_GAIN_DOWN
SND_GAIN_MONO
SND_GAIN_MUTE
SND_GAIN_STEREO
SND_GAIN_UP
SND_GAIN_XSTEREO
SND_LINE

SND_GAIN_CAP

Sound Device Gain Capabilities

Syntax

```
typedef struct _SND_GAIN_CAP {
   SND_LINE lines;/* Mask of mix lines */
   BOOLEAN sup_mute;/* Supports mute if TRUE */
   SND_GAIN_CMD default_type;
      /* Default gain type */
   u_int16 default_level;/* Default gain level */
   u_int16 zero_level;/* Gain setting where dB is */
      /* zero */
   u_int16 num_steps;/* Number of gain steps*/
   int16 step_size;/* Average size of each */
      /* step */
   int32 mindb;/* dB at SND_GAIN_MIN */
   int32 maxdb;/* dB at SND_GAIN_MAX */
} SND_GAIN_CAP;
```

Description

This data structure defines the gain capabilities of a set of mixer lines of a sound device. Appendix A: Sound Hardware Specifications gives detailed information about the capabilities of each specific MAUI sound driver. A pointer to an array of these data structures is returned as part of the `SND_DEV_CAP` structure returned in `_os_gs_snd_devcap()`.

`lines` is a mask of one or more mix lines that share a `SND_DEV_CAP` description. A single `SND_GAIN_CAP` entry may describe more than one mixing line.

`sup_mute` is `TRUE` if the indicated mix lines support muting.

`default_type` specifies the default gain type for the indicated mix lines. This is the `SND_GAIN_CMD` that the mix lines revert to when `SND_GAIN_CMD_RESET` is specified in `_os_ss_snd_gain()`.

`default_level` contains the driver's initial gain value for the indicated mix lines.

zero_level contains the level value where delta dB is zero (no gain or attenuation is applied to the line). Steps above this step are positive gain. Steps below this step are negative gain (attenuation).

num_steps is the number of actual gain or attenuation values supported by the hardware. Applications specify gain in levels between 0 and 127. The driver scales the requested level into the range of steps supported by the hardware.

step_size is the average size of each step in 100^{ths} of a dB.

mindb is the delta dB in 100^{ths} of a dB at level SND_GAIN_MIN.

maxdb is the delta dB in 100^{ths} of a dB at level SND_GAIN_MAX.

The following two tables show example SND_GAIN_CAPs:

Table 4-4 Example A: SND_GAIN_CAP

Name	Value	Description
lines	SND_LINE_PCM	Primary PCM CODEC.
sup_mute	TRUE	Mute is supported.
default_type	SND_GAIN_CMD_STEREO	Left and Right channels are supported.
default_level	SND_LEVEL_MAX	These lines start at and reset to level 127.
zero_level	SND_LEVEL_MIN	No gain applied when level is 0.
num_steps	64	The hardware supports 64 steps.
step_size	150	The average step size is 1.5 dB.

Table 4-4 Example A: SND_GAIN_CAP (continued)

Name	Value	Description
mindb	-9450	At level 0, an attenuation of 94.5 dB is applied.
maxdb	0	At level 127, no attenuation is applied.

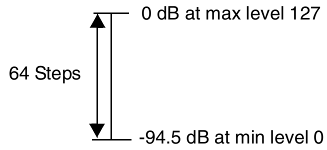
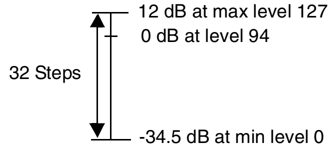


Table 4-5 Example B: SND_GAIN_CAP

Name	Value	Description
lines	SND_LINE_CD | SND_LINE_LINE	The CD and LINE inputs have the same gain capabilities.
sup_mute	TRUE	Mute is supported.
default_type	SND_GAIN_CMD_STEREO	Left and Right channels are supported.
default_level	94	These lines start at and reset to level 94.

Table 4-5 Example B: SND_GAIN_CAP (continued)

Name	Value	Description
zero_level	94	No gain applied when level is 94.
num_steps	32	The hardware supports 32 steps.
step_size	150	The average step size is 1.5 dB.
mindb	-3450	At level 0, an attenuation of 34.5 dB is applied.
maxdb	120	At level 127, a gain of 12 dB is applied.



See Also

_os_gs_snd_devcap()

BOOLEAN ***MAUI Programming Reference Guide***

SND_DEV_CAP

SND_LINE

SND_GAIN_CMD

Gain Commands

Syntax

```
typedef int {
   SND_GAIN_CMD_NONE,   /* Gain command unknown */
   SND_GAIN_CMD_UP,     /* Increment gain n levels */
   SND_GAIN_CMD_DOWN,   /* Decrement gain n levels */
   SND_GAIN_CMD_RESET, /* Reset to default level */
   SND_GAIN_CMD_MUTE,   /* Set/Unset mute */
   SND_GAIN_CMD_MONO,   /* Mono gain control */
   SND_GAIN_CMD_STEREO,/* Stereo gain control */
   SND_GAIN_CMD_XSTEREO/* Cross-stereo gain control */
} SND_GAIN_CMD;
```

Description

This integer defines the gain commands for modifying the gain level of the mix lines. The values are powers of two to facilitate the specification of their use in the `sup_gain_cmds` field of `SND_GAIN_CAP`.

`SND_GAIN_CMD_NONE` usually indicates that the gain command is not set. A `SND_GAIN->cmd` of `SND_GAIN_UP` in `_os_ss_snd_gain()` indicates to increase the gain the number of `levels` specified in `SND_GAIN_UP`.

A `SND_GAIN->cmd` of `SND_GAIN_DOWN` in `_os_ss_snd_gain()` indicates to decrease the gain the number of `levels` specified in `SND_GAIN_DOWN`.

A `SND_GAIN->cmd` of `SND_GAIN_CMD_RESET` in `_os_ss_snd_gain()` indicates to reset the gain level to the `default_level` found in `SND_GAIN_CAP`.

A `SND_GAIN->cmd` of `SND_GAIN_CMD_MUTE` in `_os_ss_snd_gain()` indicates to set or clear the mute bit according to `state` specified in `SND_GAIN_MUTE`.

SND_GAIN_CMD_MONO specifies mono gain control. Parameters for SND_GAIN_CMD_MONO are specified by SND_GAIN_MONO.
SND_GAIN_STEREO specifies stereo gain control. Parameters for SND_GAIN_CMD_STEREO are specified by SND_GAIN_STEREO.

SND_GAIN_CMD_XTEREO is used for hardware that supports cross-gain or cross-attenuation. That is the ability to send the audio signal from the right channel to the left channel, and audio from the left channel to the right channel. Parameters for SND_GAIN_CMD_XSTEREO are specified by SND_GAIN_XSTEREO.

See Also

SND_GAIN
SND_GAIN_CAP
SND_GAIN_DOWN
SND_GAIN_MONO
SND_GAIN_MUTE
SND_GAIN_STEREO
SND_GAIN_UP
SND_GAIN_XSTEREO

SND_GAIN_DOWN

Gain Down Parameters

Syntax

```
typedef struct _SND_GAIN_DOWN {
   u_int8 levels;       /* Levels to decrement the */
                        /* gain */
} SND_GAIN_DOWN;
```

Description

This data structure specifies how much to decrement the gain level when `cmd` of `SND_GAIN` is `SND_GAIN_CMD_DOWN`.

`levels` is the number of levels to decrement the gain level. This has the effect of decreasing the volume.

If decrementing the gain level by `levels` causes the gain level to drop below `SND_LEVEL_MIN`, the gain level is set to `SND_LEVEL_MIN`.

See Also

`_os_ss_snd_gain()`
`SND_GAIN`
`SND_GAIN_CMD`
`SND_LEVEL_*`

SND_GAIN_MONO

Mono Gain Parameters

Syntax

```
typedef struct _SND_GAIN_MONO {
   u_int8 m;             /* Gain of mono signal */
} SND_GAIN_MONO;
```

Description

This data structure specifies a specific mono gain level when `cmd` of `SND_GAIN` is `SND_GAIN_CMD_MONO`.

The gain level specifies how much of the mono signal from the sound decoder/encoder contributes to the final output or input (such as speaker, headphone, microphone).

The range of `m` is from `SND_LEVEL_MIN` to `SND_LEVEL_MAX`, inclusive. In general, higher `m` values are louder than lower `m` values. The significant values and translation of `m` to dB varies with the capabilities of the hardware. Use `_os_gs_snd_devcap()` and examine the `SND_GAIN_CAP` to determine the capabilities of a specific sound device.

Values in `m` with `SND_LEVEL_MUTE` bit set are considered mute. This allows the setting and clearing of the mute bit to enable and disable muting, without affecting the gain level of `m`.

See Also

`_os_ss_snd_gain()`
`SND_GAIN`
`SND_GAIN_CAP`
`SND_GAIN_CMD`
`SND_LEVEL_*`

SND_GAIN_MUTE

Gain Mute Parameters

Syntax

```
typedef struct _SND_GAIN_MUTE {
   BOOLEAN state;      /*Mute state*/
} SND_GAIN_MUTE;
```

Description

This data structure specifies whether to mute or un-mute the line when `cmd` of `SND_GAIN` is `SND_GAIN_CMD_MUTE`.

If `state` is equal to `FALSE`, the line is un-muted. If `state` is equal to `TRUE`, the line is muted.

See Also

`_os_ss_snd_gain()`
`SND_GAIN`
`SND_GAIN_CMD`
`SND_LEVEL_*`

`BOOLEAN` (See the ***MAUI Programming Reference Manual***)

SND_GAIN_STEREO

Stereo Gain Parameters

Syntax

```
typedef struct _SND_GAIN_STEREO {
   u_int8 ll;            /* Left input to left output */
   u_int8 rr;            /* Right input to right output */
} SND_GAIN_STEREO;
```

Description

This data structure specifies a specific stereo gain level when `cmd` of `SND_GAIN` is `SND_GAIN_CMD_STEREO`.

The gain level specifies how much of each signal from the sound decoder/encoder contributes to the final left and right outputs or inputs (such as speaker, headphone, or microphone).

The range of each member (`ll` and `rr`) are from `SND_LEVEL_MIN` to `SND_LEVEL_MAX` inclusive. In general, higher values are louder than lower values. The significant values and translation to dB varies with the capabilities of the hardware. Use `_os_gs_snd_devcap()` to examine the `SND_GAIN_CAP` and determine the capabilities of a specific sound device.

Values in `ll` and `rr` with the `SND_LEVEL_MUTE` bit set are considered mute. This allows the setting and clearing of the mute bit to enable and disable muting without affecting the original values of `ll` and `rr`.

`ll` specifies the amount of audio to go from the left input to left output. `rr` specifies the amount of audio to go from the right input to right output. These fields allow control of the overall volume as well as adjusting balance.

See Also

`_os_ss_snd_gain()`
`SND_GAIN`
`SND_GAIN_CAP`
`SND_GAIN_CMD`
`SND_LEVEL_*`

SND_GAIN_UP

Gain Up Parameters

Syntax

```
typedef struct _SND_GAIN_UP {
   u_int8 levels;/*Levels to increment the gain */
} SND_GAIN_UP;
```

Description

This data structure specifies how much to increment the gain level when cmd of SND_GAIN is SND_GAIN_CMD_UP.

levels is the number of levels to increment the gain level. This has the effect of increasing the volume. If incrementing the gain level by levels causes the gain level rise above SND_LEVEL_MAX, the gain level is set to SND_LEVEL_MAX.

See Also

_os_ss_snd_gain()
SND_GAIN
SND_GAIN_CMD
SND_LEVEL_*

SND_GAIN_XSTEREO

Cross-Stereo Gain Parameters

Syntax

```
typedef struct _SND_GAIN_STEREO {
   u_int8 ll;          /* Left input to left output */
   u_int8 rr;          /* Right input to right output */
   u_int8 rl;          /* Right input to left output */
   u_int8 lr;          /* Left input to right output */
} SND_GAIN_STEREO;
```

Description

This data structure specifies a specific stereo gain level when `cmd` of `SND_GAIN` is equal to `SND_GAIN_CMD_XSTEREO`.

The gain level specifies how much of each signal from the sound decoder/encoder contributes to the final left and right outputs and inputs (such as speaker, headphone, and microphone).

The range of each member (`ll, rr, rl`, and `lr`) is from `SND_LEVEL_MIN` to `SND_LEVEL_MAX` inclusive. In general, higher values are louder than lower values. The significant values and translation to dB varies with the capabilities of the hardware. Use `_os_gs_snd_devcap()` and examine the `SND_GAIN_CAP` to determine the capabilities of a specific sound device.

Values in `ll`, `rr`, `rl`, and `lr` with the `SND_LEVEL_MUTE` bit set are considered mute. This allows the setting and clearing of the mute bit to enable and disable muting, without affecting the original values of `ll`, `rr`, `rl`, and `lr`.

`ll` specifies the amount of audio going from the left input to left output.

`rr` specifies the amount of audio going from the right input to the right output.

`rl` specifies the amount of audio going from the right input to left output.

`lr`	specifies the amount of audio going from the left input to right output.

These fields allow control of the overall volume as well as adjusting balance.

See Also

`_os_ss_snd_gain()`
`SND_GAIN`
`SND_GAIN_CAP`
`SND_GAIN_CMD`
`SND_LEVEL_*`

SND_LEVEL_*

Gain Level Constants

Syntax

SND_LEVEL_*

Description

SND_LEVEL_ is a prefix used to define a set of constant values that may be used to specify gain levels as shown in the following table. These constants are commonly used in SND_GAIN_DOWN, SND_GAIN_MONO, SND_GAIN_MUTE, SND_GAIN_STEREO, SND_GAIN_UP, and SND_GAIN_XSTEREO.

Table 4-6 SND_LEVEL_* Definitions

Define Name	Value	Description
SND_LEVEL_MIN	0	Defines the minimum (quietest) gain level.
SND_LEVEL_MAX	127	Defines the maximum (loudest) gain level.
SND_LEVEL_MUTE	0x80	Mask value used to mute a line.

SND_LEVEL_MUTE may be logically or'ed (|) with a gain level to mute the line without affecting the current gain level. For example:

```
SN_GAIN in_gain, out_gain;
in_gain.param.mono.m |= SND_LEVEL_MUTE;
out_gain.param.stereo.ll |= SND_LEVEL_MUTE;
out_gain.param.stereo.rr |= SND_LEVEL_MUTE;
```

The negation (~) of this mask may be logically and'ed (&) with gain to un-mute the line without affecting the current gain level. For example:

```
SND_GAIN in_gain, out_gain;
in_gain.param.mono.m &= ~SND_LEVEL_MUTE;
out_gain.param.stereo.ll &= ~SND_LEVEL_MUTE;
out_gain.param.stereo.rr &= ~SND_LEVEL_MUTE;
```

See Also

_os_ss_snd_gain()
SND_GAIN
SND_GAIN_DOWN
SND_GAIN_MONO
SND_GAIN_MUTE
SND_GAIN_STEREO
SND_GAIN_UP
SND_GAIN_XSTEREO

SND_LINE

Gain/Mixer Line Types

Syntax

```
typedef int {
   SND_LINE_VOLUME,       /* Master output */
   SND_LINE_BASS,         /* Bass */
   SND_LINE_TREBLE,       /* Treble */
   SND_LINE_SYNTH,        /* Synthesizer input */
   SND_LINE_PCM,          /* PCM/CODEC */
   SND_LINE_SPEAKER,      /* PC speaker */
   SND_LINE_LINE,         /* Line input */
   SND_LINE_MIC,          /* Microphone input */
   SND_LINE_CD,           /* CD input */
   SND_LINE_IMIX,         /* Recording monitor */
   SND_LINE_OMIX,         /* Loopback */
   SND_LINE_ALTPCM,       /* Alternative PCM/CODEC */
   SND_LINE_RECLEV,       /* Encoder level */
   SND_LINE_IGAIN,        /* Input gain */
   SND_LINE_OGAIN,        /* Output gain */
   SND_LINE_LINE1,        /* Lines 1-3 are generic mixer
                          lines */
   SND_LINE_LINE2,
   SND_LINE_LINE3,
   SND_NUM_LINES          /* Number of mixer lines */
   SND_MAX_LINES          /* Maximum mixer lines */
   SND_LINE_MASK_ALL      /* Mask to select all mixer
                          lines */
} SND_LINE;
```

Description

This integer defines the valid mixer line types. The values are powers of two. Therefore, you may combine them safely. It is used in `SND_DEV_CAP`, `SND_GAIN`, and `SND_GAIN_CAP`. These definitions are similar to LINUX's `SOUND_MIXER_*` definitions.

`SND_LINE_VOLUME` is the master output level (headphone or line out volume).

`SND_LINE_BASS` controls the bass level of all the output lines.

`SND_LINE_TREBLE` controls the treble level of all the output lines.

`SND_LINE_SYNTH` controls the synthesizer input (FM, wave table) of the sound card.

`SND_LINE_PCM` is the output level for the audio (CODEC, PCM, ADC) line.

`SND_LINE_SPEAKER` is the output level of the PC speaker signals. This is typically mono.

`SND_LINE_LINE` is the input level for the line-in jack.

`SND_LINE_MIC` is the input level for the signal coming from the microphone-in jack.

`SND_LINE_CD` is the level for signal connected to the CD audio input.

`SND_LINE_IMIX` is the recording monitor. For example, on PAS16 and some other cards, this controls the output gain (headphone jack) of the selected recording sources while recording. This line only has effect when recording.

`SND_LINE_OMIX` controls the loopback of output to input.

`SND_LINE_ALTPCM` controls the alternative CODEC line such as the SB emulation of the PAS16 board.

`SND_LINE_RECLEV` is the global record level setting. On the SB16 card, this controls the input gain, which has 4 possible levels.

`SND_LINE_IGAIN` is the input gain control.

`SND_LINE_OGAIN` is the output gain control.

`SND_LINE_LINE1`, `SND_LINE_LINE2`, and `SND_LINE_LINE3` are generic mixer lines that are used in cases when precise meaning of a physical mixer line is not known. Actual meaning of these signals are vendor-defined. Usually these lines are connected to synth, line-in, and CD inputs of the card, but the order of the assignment is not known to the driver.

`SND_LINE_MASK_ALL` is a mask to select all mixer lines.

See Also

SND_DEV_CAP
SND_GAIN
SND_GAIN_CAP

SND_SMAP

Sound Map

Syntax

```
typedef struct _SND_SMAP {
   SND_TRIGGER trig_status;
                           /* Current sound map status */
   SND_TRIGGER trig_mask;/* Signal trigger mask */
   signal_code trig_signal;
                           /* Signal to send on */
                           /* triggers */
   error_code err_code;/* Error code on termination */
   SND_CM coding_method;/* Coding method */
   u_int8 num_channels;/* Number of channels */
   u_int32 sample_size;/* Number of bits per sample */
   u_int32 sample_rate;/* Number of samples per sec */
   u_char *buf;         /* Sound data buffer */
   u_int32 buf_size;    /* Size of sound data buffer */
   u_int32 cur_offset;  /* Current offset into "buf" */
   u_int32 loop_start;  /* Offset to start of loop */
   u_int32 loop_end;    /* Offset to end of loop */
   u_int32 loop_count;  /* Num of times to play loop */
   u_int32 loop_counter;/* Num times loop has played */
   SND_SMAP *next;      /* Link to next SND_SMAP */
} SND_SMAP;
```

Description

This data structure defines the sound map. This object is created by the application and is used to control play and record operations.

`trig_status` indicates the current sound map status. It is modified as the status changes. See `SND_TRIGGER` for a description of each value.

The specified `trig_signal` is sent when the sound operation (`_os_ss_snd_play()` or `_os_ss_snd_record()`) completes an activity that satisfies a trigger specified by `trig_mask`. If the specified `trig_mask` is equal to `SND_TRIG_NONE` or `trig_signal` is equal to zero, then no signals are sent.

If an error occurs, err_code is set to an appropriate error code. If an operation is aborted, err_code is set to EOS_MAUI_ABORT. Otherwise, err_code is set to SUCCESS at the end of the sound operation.

coding_method defines the audio encoding format for the sound map.

num_channels is the number of channels. Mono data requires one channel. Stereo data requires two channels. Currently all coding methods are defined such that the left stereo channel is always ordered before the right stereo channel.

There are two main parameters that effect the quality of the sound. First, is the sample_rate which is the number of samples per second. Typical values include 4000, 8000, 11025, 22050, 44100 and 48000. The second parameter that affects the quality is sample_size. sample_size is the number of bits per sample. Typical values are 8 or 16 bits per sample. The number of bits affect the dynamic range of the sample and the signal to noise ratio.

buf is a pointer to a buffer containing data for a play operation or sufficient space for a record operation. The memory pointed to by buf should start on a multiple of SND_DEV_CM->boundary_size. The size of the buffer, in bytes, is set in buf_size and should also be a multiple of
SND_DEV_CM->boundary_size.

cur_offset specifies the offset in bytes within buf for the current sound operation. It is updated throughout the I/O operation by the sound driver.

loop_start, loop_end, loop_count and loop_counter are used to specify the area of the buffer to play and how many times to play it. They are not used in record operations and must be set to zero when submitted to _os_ss_snd_record().

loop_start specifies the start of the loop as an offset in bytes within buf.

loop_end specifies the end of the loop as an offset in bytes within buf.

loop_count indicates the desired number of times to execute the loop.

loop_counter indicates the current number of times the loop has been executed. This field is set to zero by the sound driver when the sound map is accepted by the sound driver.

next may point to another sound map. In this case, the sound operation continues with the next SND_SMAP when the first one is finished. The trig_mask and trig_signal are consulted regardless of whether next is set or not. If this SND_SMAP is not linked to another SND_SMAP, next must be NULL.

Linked sound maps may have different encoding parameters (coding_method, num_channels, sample_size, sample_rate), although most hardware requires some calibration time to switch to the new formats, which may result in a delay.

See Also

_os_ss_snd_play()
_os_ss_snd_record()
SND_CM
SND_DEV_CM
SND_TRIGGER

SND_STATUS

Device Status

Syntax

```
typedef int {
   SND_STATUS_IDLE,       /* Device is idle */
   SND_STATUS_PLAY,       /* Play operation is active */
   SND_STATUS_PLAY_PAUSED,/* Play is paused */
   SND_STATUS_RECORD,     /* Record operation is active */
   SND_STATUS_RECORD_PAUSED,/* Record is paused */
   SND_STATUS_BUSY        /* The buffer is busy */
} SND_STATUS;
```

Description

This integer defines the valid values for the `status` field in `SND_DEV_STATUS`. `SND_DEV_STATUS` is returned from `_os_ss_snd_status()` and indicates the current state of the sound hardware. The values are powers of two and may be combined safely.

`SND_STATUS_IDLE` (no status bits are set) indicates the sound device is not busy.

The `SND_STATUS_PLAY` bit is set when a play operation is active. This bit is set during a play operation from the `SND_TRIG_START` event to the `SND_TRIG_FINISH` event.

The `SND_STATUS_PLAY_PAUSED` bit is set when a play operation is paused. This bit is never set if the `SND_STATUS_PLAY` bit is not set.

The `SND_STATUS_RECORD` bit is set when a record operation is active. This bit is set during a record operation from the `SND_TRIG_START` event to the `SND_TRIG_FINISH` event.

The `SND_STATUS_RECORD_PAUSED` bit is set when a record operation is paused. This bit is never set if the `SND_STATUS_RECORD` bit is not set.

The `SND_STATUS_BUSY` bit is set when the driver is processing a sound operation.

For illustrations of SND_TRIGGER in sound operations see the figure "Play Operation Status and Trigger Points." on page 29 and "Record Operation Status and Trigger Points" on page 31, respectively.

See Also

_os_ss_snd_play()
_os_ss_snd_record()
SND_SMAP

SND_TRIGGER

Triggers

Syntax

```
typedef int {
   SND_TRIG_NONE,        /* Mask for no triggers */
   SND_TRIG_START,       /* Audible start */
   SND_TRIG_FINISH,      /* Audible finish */
   SND_TRIG_BUSY,        /* Buffer Busy */
   SND_TRIG_READY,       /* Buffer ready */
   SND_TRIG_ANY          /* Mask for all triggers */
} SND_TRIGGER;
```

Description

This integer defines the trigger events in a sound operation (`_os_ss_snd_play()` or `_os_ss_snd_record()`) capable of generating signals. The values are powers of two and may be combined safely.

`SND_TRIG_NONE` is a mask value indicating interest in no triggers.

If the `SND_TRIG_START` bit is set for a play operation, this indicates that the output device is actively playing (it can be heard). For a record operation, this bit indicates that the driver has started accepting sound input.

If the `SND_TRIG_FINISH` bit is set for a play operation, this indicates that the driver has completed the play and no more sound is being produced. For a record operation, this bit indicates that the driver has stopped accepting sound input.

If the `SND_TRIG_BUSY` bit is set for a play operation, this indicates that the driver has started consuming the data in the buffer. For a record operation, this bit indicates that the driver has started filling the buffer.

If the `SND_TRIG_READY` bit is set for a play operation, this indicates that the driver has finished consuming the data in the buffer and is ready to accept the next buffer. For a record operation, this bit indicates that the driver has finished filling the buffer and is ready to use the next buffer.

SND_TRIG_ANY is a mask value indicating interest in all triggers.

For illustrations of SND_TRIGGER in sound operations see the figure "Play Operation Status and Trigger Points." on page 29 and "Record Operation Status and Trigger Points" on page 31.

See Also

_os_ss_snd_play()
_os_ss_snd_record()
SND_SMAP

Appendix A: Sound Hardware Specifications

This appendix contains the hardware specifications for the following sound device:

Crystal Semiconductor CS4231A

Crystal Semiconductor CS4231A

Specification version: March 27, 1997

Overview

This document describes the hardware specifications for the Crystal Semiconductor CS4231A driver (named `sd_cs`). The hardware sub-type defines the board configuration. This specification should be used in conjunction with the **MAUI Sound Driver Interface**.

Device Capabilities

Information about the hardware capabilities is determined by calling `_os_gs_snd_devcap()`. This function returns a data structure formatted as shown in the following table.

Table A-1 Data Returned in SND_DEV_CAP

Member Name	Value	Description
hw_type	"CS4231"	Hardware type
hw_subtype	"CS4231A"	Hardware sub-type
sup_triggers	SND_TRIG_ANY	Supported triggers
play_lines	SND_LINE_SPEAKER| SND_LINE_VOLUME	Play gain/mix lines
record_lines	SND_LINE_MIC| SND_LINE_LINE3| SND_LINE_LINE| SND_LINE_PCM	Record gain/mix lines
sup_gain_cmds	SND_GAIN_CMD_UP| SND_GAIN_CMD_DOWN| SND_GAIN_CMD_RESET | SND_GAIN_CMD_MONO| SND_GAIN_CMD_STERE O	Mask of supported gain commands
num_gain_caps	7	Number of SND_GAIN_CAPs

Table A-1 Data Returned in SND_DEV_CAP (continued)

Member Name	Value	Description
gain_caps	See paragraph Gain Capabilities Array	Pointer to SAND_GAIN_CAP array
num_rates	14	Number of sample rates
sample_rates	See paragraph Sample Rates	Pointer to sample rate array
num_chan_info	2	Number of channel info entries
channel_info	See paragraph Number of Channels	Pointer to an array of supported num_channels
num_cm	6	Number of coding methods
cm_info	See paragraph Encoding and Decoding Formats	Pointer to coding method array

For More Information

See SND_DEV_CAP in chapter 4 for more information regarding this data structure.

Gain Capabilities Array

The preceding table shows the various gain capabilities for the Crystal Semiconductor CS4231A. This information is pointed to by the `gain_cap` member of the `SND_DEV_CAP` data structure. The following seven tables (Tables 2 through 8) describe the gain capabilities.

Table A-2 L/R DAC Attenuator (I6) L/R DAC Attenuator (I7)

Member Name	Value
`lines`	`SND_LINE_PCM`
`sup_mute`	`TRUE`
`default_type`	`SND_GAIN_CMD_STEREO`
`default_level`	`SND_LEVEL_MAX`
`zero_level`	`SND_LEVEL_MIN`
`num_steps`	`64`
`step_size`	`150`
`mindb`	`-9450`
`maxdb`	`0`

Step	HW	Level	Comments
0-1	63	-94.5 dB	`default_level`
2-3	62	-93.0 dB	
4-5	61	-91.5 dB	
6-7	60	-90.0 dB	
...	...	...	

Step	HW	Level	Comments
120-121	3	-4.5 dB	
122-123	2	-3.0 dB	
124-125	1	-1.5 dB	
126-127	0	0.0 dB	`zero_level`

Table A-3 L/R Auxiliary #1 (I2, I3), L/R Auxiliary #2 (I4, I5), L/R Line Mix Gain (I18, I19)

Member Name	Value
lines	SND_LINE_CD | SND_LINE_LINE | SND_LINE_SYNTH | SND_LINE_LINE2 | SND_LINE_LINE | SND_LINE_LINE3
sup_mute	TRUE
default_type	SND_GAIN_CMD_STEREO
default_level	94
zero_level	94
num_steps	32
step_size	150
mindb	-3450
maxdb	120

Step	HW	Level	Comments
0-3	31	-34.5 dB	
4-6	30	-33.0 dB	
7-10	29	-31.5 dB	
11-14	28	-30.5 dB	
...	...	...	
89-92	9	-1.5 dB	
93-96	8	0 dB	zero_level, default_level

Step	HW	Level	Comments
97-100	7	1.5 dB	
...	...	...	
113-116	3	7.5 dB	
117-120	2	9.0 dB	
121-124	1	10.5 dB	
125-127	0	12.0 dB	

Table A-4 L/R ADC Gain (I0, I1)

Member Name	Value
lines	SND_LINE_IGAIN
sup_mute	FALSE
default_type	SND_GAIN_CMD_STEREO
default_level	SND_LEVEL_MIN
zero_level	SND_LEVEL_MIN
num_steps	16
step_size	150
mindb	0
maxdb	2250

Step	HW	Level	Comments
0-4	0	0 dB	zero_level, default_level
5-12	1	1.5 dB	
13-21	2	3.0 dB	
22-30	3	4.5 dB	
...	...	...	

Step	HW	Level	Comments
98-105	12	18.0 dB	
106-114	13	19.5 dB	
115-122	14	21.0 dB	
124-127	15	22.5 dB	

Table A-5 L/R Mic Gain Enable (I0, I1)

Member Name	Value
lines	SND_LINE_MIC
sup_mute	FALSE
default_type	SND_GAIN_CMD_STEREO
default_level	SND_LEVEL_MAX
zero_level	SND_LEVEL_MIN
num_steps	2
step_size	200
mindb	0
maxdb	200

Step	HW	Level	Comments
0-63	0 dB	0	zero_level
64-127	20 dB	1	default_level

Table A-6 Mono In/Out Speaker (I26)

Member Name	Value
lines	SND_LINE_SPEAKER
sup_mute	TRUE
default_type	SND_GAIN_CMD_MONO
default_level	SND_LEVEL_MAX | SND_LEVEL_MUTE
zero_level	SND_LEVEL_MIN
num_steps	16
step_size	300
mindb	-4500
maxdb	0

Step	HW	Level	Comments
0-4	0	0 dB	zero_level, default_level
5-12	1	-3.0 dB	
13-21	2	-6.0 dB	
22-30	3	-9.0 dB	
...	...	...	

Step	HW	Level	Comments
98-105	12	-36.0 dB	
106-114	13	-39.0 dB	
115-122	14	-42.0 dB	
124-127	15	-45.0 dB	

Table A-7 Loopback Attenuation (I13)

Member Name	Value
lines	SND_LINE_IMIX
sup_mute	TRUE
default_type	SND_GAIN_CMD_MONO
default_level	SND_LEVEL_MAX | SND_LEVEL_MUTE
zero_level	SND_LEVEL_MIN
num_steps	64
step_size	150
mindb	-9450
maxdb	0

Step	HW	Level	Comments
0-1	63	-94.5 dB	default_level
2-3	62	-93.0 dB	
4-5	61	-91.5 dB	
6-7	60	-90.0 dB	
...	...	...	
120-121	3	-4.5 dB	
122-123	2	-3.0 dB	
124-125	1	-1.5 dB	
126-127	0	0.0 dB	zero_level

\

Table A-8 On some boards the I10 register can mute the line out

Member Name	Value
lines	SND_LINE_VOLUME
sup_mute	TRUE
default_type	SND_GAIN_CMD_MUTE
default_level	0
zero_level	0
num_steps	0
step_size	0
mindb	0
maxdb	0

Note

See SND_GAIN_CAP in Chapter 4: Data Type Reference for more information about this data structure.

Sample Rates

The following table shows the supported sample rates for the Crystal Semiconductor CS4231A. This information is pointed to by the `sample_rates` member of the `SND_DEV_CAP` data structure.

Table A-9 Supported Sample Rates

Sample Rate	Sample Rate	Sample Rate
5510 Hz	11025 Hz	32000 Hz
6620 Hz	16000 Hz	33075 Hz
8000 Hz	18900 Hz	37800 Hz
9600 Hz	22050 Hz	44100 Hz
	27420 Hz	48000 Hz

Number of Channels

Table 10 shows the different supported number of channels for the CS4231A. This information is pointed to by the `channel_info` member of the `SND_DEV_CAP` data structure.

Table A-10 Supported Number of Channels

Channels	Description
2	Stereo (default)
1	Mono

Encoding and Decoding Formats

Table 11 shows the supported encoding and decoding formats for the Crystal Semiconductor CS4231A. The first entry in the table is the default format. This information is pointed to by the `cm_info` member of the `SND_DEV_CAP` data structure.

Table A-11 Supported Encoding and Decoding Formats

Coding Method	Sample Size	Bndry Size	Description
`SND_CM_PCM_ULAW`	8	2	8-bit µLaw companded
`SND_CM_PCM_ALAW`	8	2	8-bit ALaw companded
`SND_CM_PCM_ULINEAR`	8	2	8-bit Linear unsigned
`SND_CM_PCM_SLINEAR | SND_CM_LSBYTE1ST`	16	4	16-bit Linear (two's complement) little endian
`SND_CM_PCM_SLINEAR`	16	4	16-bit Linear (two's complement) little endian
`SND_CM_ADPCM_IMA`	4	64	4-bit ADPCM IMA compatible

Index

Symbols

A

B

C

D

E

F

G

L

M

N

O

P

R

S

T

Product Discrepancy Report

To: Microware Customer Support

FAX: 515-224-1352

From:____

Company:_______________________________________

Phone:____

Fax:_____________________Email:__________________

Product Name: Sound Driver Interface

Description of Problem:

____

____

____

____

____

____

____

____

____

Host Platform_____________________________________

Target Platform____________________________________