RISC OS Audio input and output

Introduction and Overview

This document describes the new Audio interface module – SoundSystem – and how it relates to the existing APIs available in earlier versions of RISC OS.

Audio capabilities – a brief history

Audio output

The first versions of RISC OS had a 1-8 channel audio sound output system that supported an 8-bit µ-Law based audio playback.

Later on, a 2-channel (stereo) 16-bit linear PCM (LPCM) was added to the hardware capabilities, with existing audio supported via a mechanism to translate the µ-Law audio into the 16-bit linear PCM format.

At a later stage, a mechanism was created that allowed different audio playback applications to share the sound system by merging the audio playback data together.

Audio input

No version of RISC OS has ever supported a generic audio input capability. Independent vendors had their own solution, and some published their mechanisms to allow other vendors to use their hardware within audio capture software

Audio devices

Within the legacy ARM hardware, audio was part of the video hardware; the VIDC1 and VIDC20 had methods to read data (via the MEMC), and play it back via audio hardware.

With more modern system on a chip (SoC) hardware, audio is separated from the video, and has dedicated registers controlling the flow of the data. These are generally I²S-based, which is a simple digital mechanism for transferring audio data to a digital to analogue converter (DAC). They also have the capability of providing audio input, using an analogue to digital converter (ADC).

Vendors of RISC OS hardware have replaced the VIDC interfaces with on-device audio output – although the underlying mechanism has not changed.

Audio APIs

All existing APIs follow the same basic method – a buffer is provided for the audio playback software to populate with data that will be output via the hardware interface.

A double-buffer approach is taken, where the software is filling one sound buffer while the hardware is emptying another. When the hardware has emptied the buffer, an interrupt is used to tell the operating system that it needs new data, and the operating system gives the hardware the buffer the software has just filled.

The software then fills the second buffer with data in preparation for the hardware to play.

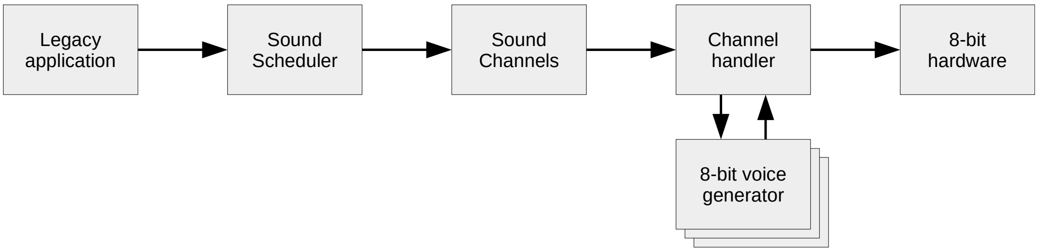
8-bit µ-Law audio

This API dates back to the original Archimedes computers. The number of sound channels is configured by software that is performing the playback.

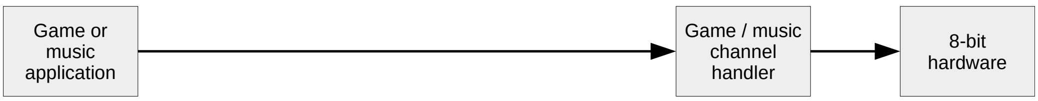
The sound system had the capability of providing up to 32 different ‘voices’, or instruments, and these could be scheduled to play with the BASIC SOUND command (or via SWI calls). These voices filled the audio buffers with the data that is needed to play the audio.

In order to maintain a degree of backwards compatibility with the BBC Micro series of computers, the 8-bit voice generators could be controlled by the SoundChannels module, which allows programs to instruct the voice generators to perform audio playback via SWI calls.

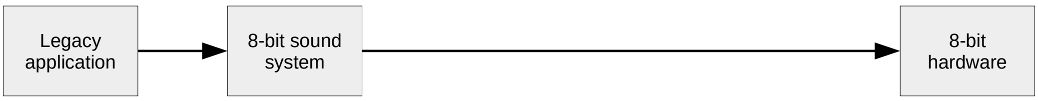
The playback of audio can be scheduled via the Sound Scheduler, which allowed the voice generators to be controlled during interrupt calls via a channel handler. Few programmes use this – Maestro is about the only one, although the system error “beep” still relies on this mechanism. Sonor performs its playback as an 8-bit voice generator, but does not use SoundScheduler to control it under interrupt:



Games and music programmes such as Tracker, Desktop Tracker and Digital Symphony replaced the SoundChannels (and therefore bypassed the voice generator scheme) with their own channel handling code that was faster and more controllable than the 8-bit voice generation and SoundScheduler mechanism:



For rest of this document, the SoundScheduler, SoundChannels and voice generators are collectively known as the “8-bit sound system”, as shown below:

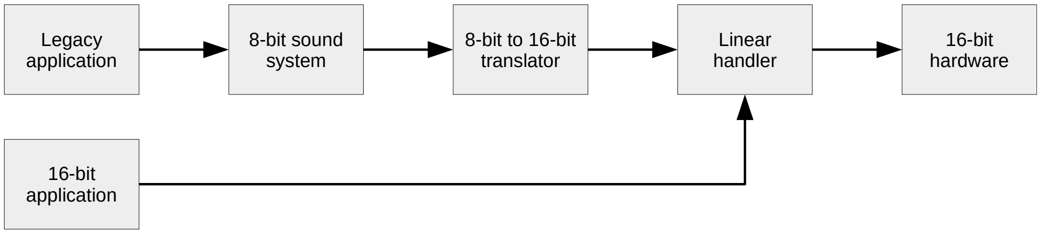


16-bit single application audio

With the later RiscPC hardware, 16-bit stereo audio capability was added. In order to support this, two changes were made:

- The 8-bit µ-Law audio worked as before – voices fill a buffer with the sound data. However, when complete, the data is translated to 16-bit stereo data.
- A 16-bit buffer fill API was provided that applications can provide the data with 16-bit stereo audio data.

Applications using the 16-bit API are informed if there is 8-bit data present in the buffer in order to merge the sound – however, some applications ignore this and overwrite their data all of the time.



These applications implement a *Linear Handler* that is called when more audio data is required. The sound system is triggered by a hardware event that indicates that data is required, and this is passed to the Linear Handler so that it can fill the buffers.

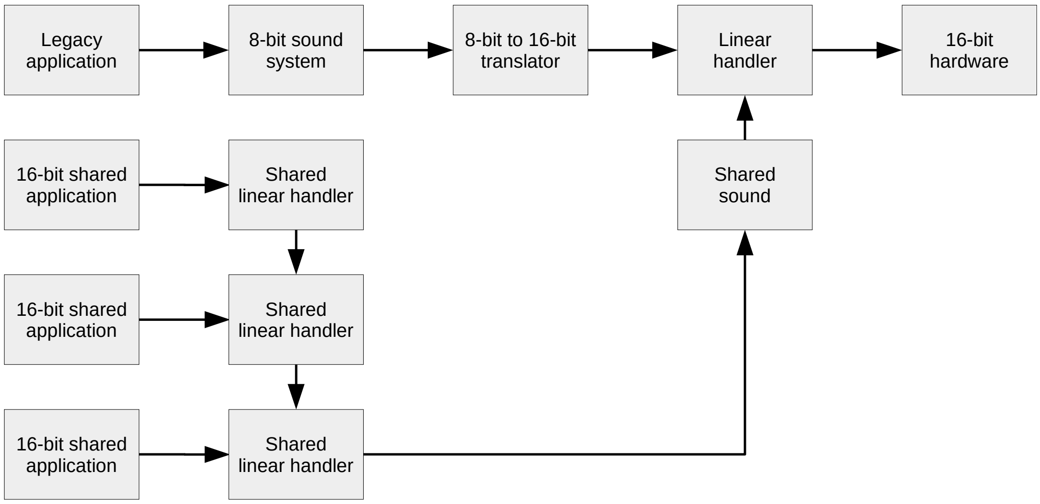
Shared audio

In order to accommodate multiple audio streams using the hardware at the same time, an API was developed to allow applications to share the audio hardware output.

This has similar concepts to a Linear Handler, except that applications must:

- Cope with other applications using the audio system
- Work at (potentially) a different sample rate to what it is expecting

The Shared Audio system is implemented as a Linear Handler, and can be replaced by a different Linear Handler if an application desires it.



An application that has been written as a standard linear handler, rather than a shared linear handler, can hook into the sound system, and this stops all shared applications from being able to use the sound system.

SoundSystem API

The SoundSystem API is designed to provide a future-resistant mechanism for audio playback and capture whilst providing a degree of backward compatibility for earlier applications. The level of backward compatibility may change in the future as more applications support the SoundSystem mechanisms.

It has a number of design goals, including:

- Higher audio precision
- More than one output device
- Multi-channel audio devices
- Non-LPCM encoding

Higher audio precision

When the sound system was changed to allow 16-bit digital audio, this was the quality that was expected – 24-bit and 32-bit was in its infancy (and expensive).

With more modern technology, inexpensive audio devices are available that support 24-bit audio and the higher resolution available cannot be used by RISC OS applications.

The SoundSystem API uses 32-bit audio data paths for LPCM formatted data. This is higher than most audio devices can cope with, but has been chosen for the following reasons:

- It is unlikely to need to go to a higher resolution as the human ear has its limitations
- It is the ARM processor’s native word size, so is a programmatically efficient encoding

More than one output device

With the advent of USB and Bluetooth, computers can often have several devices that are capable of audio playback and capture.

Users would tend to use a single audio device, and the user would configure which of their devices they want to use.

Some users have multiple audio devices, and would switch between them depending on circumstances. For example:

- Using a USB headset when making voice calls
- Using a Bluetooth headset for privacy
- Using a high quality, multi-channel device for music composition

Multi-channel audio devices

While 2-channel (stereo) is “multi-channel”, multi-channel audio devices refer to devices that have more than 2 channels – for example, surround sound devices, and professional audio devices whose outputs are destined for a mixing desk.

Non-LPCM encoding

Previous RISC OS sound system implementations required the audio to be in a linear PCM encoded format – however, some devices now support (or even require) non-LPCM formatted data, which offloads the conversion of compressed audio files from the CPU to the audio device.

Procedural interfaces

The SoundSystem API defines four interfaces that applications and hardware developers can use to provide and consume audio resources.

Discovery interfaces

This interface is used by both applications and hardware.

Applications use the discovery interface to find out about the audio capabilities of the system; hardware uses the discovery interface to inform SoundSystem that audio hardware is available, and its capabilities.

Control interfaces

This interface is used by both applications and hardware.

Applications use the control interface to configure aspects of the audio system (such as mixer volumes and input selectors); hardware uses the interface to receive configuration requests from applications.

Buffer manager

This interface is used by both applications and hardware.

Applications use the buffer manager to provide and retrieve data for audio playback and capture; hardware uses the buffer manager to request data from the applications for playback, and inform applications that data is available for capture.

Codec interface

The codec interface is primarily used by applications, although it is envisaged that hardware may use it at a later date.

The codec interface provides a method of converting one audio format into another audio format – for example, a frame of MP3 data into LPCM data. It can even be used for more basic LPCM format conversions (such as sample rate conversion, audio mixing or channel selection),

Hardware may use this interface to convert LPCM data into the hardware-native format.

Device identifiers and names

In order for applications to direct their audio to a particular sound device, or to receive data, then a device identifier is used. This is a sequence of up to 63 characters that allows the device drivers to identify its location within the hardware. This is guaranteed to be unique and consistent on a given computer and hardware configuration.

A secondary device name is a user-orientated name that an application should present to the user when prompting for the audio device to use with that application. This can be up to 31 characters in length.

For example, a device identifier may look like “USB#0001.0044.23E7”, and the device name “Focusrite Scarlett 2i2”.

The device driver provides both the device identifier and the device name to SoundSystem – the device name should be internationalised where possible.

Audio format descriptors

With many different audio formats available, a method to describe the format is required. The simplest format – and the one that is currently supported by the sound system – is LPCM. Any audio device driver should support 32-bit 2-channel LPCM format in order to maximise support for applications, but other formats can be supported, if the applications support them.

The sample rate is not considered to be part of the format descriptor, although it is part of the overall device configuration.

Format lists

When registering, a device informs SoundSystem what formats it is capable of supporting. This is done via a pointer to a list of formats, each notionally 1 word in length.

If the word in the list is zero, then this denotes the end of the list.

If the word in the list has either of the bottom two bits set, then it is a simple format that consists of a single word.

If the word in the list has both of the bottom two bits clear, then it is a pointer (or an offset) to a value held in memory for an extended format.

Simple formats

The simple format descriptor consists of single 32-bit word, with one, or both, of the bottom two bits set.

The format of this word is &nnnnnnxx where “nnnnnn” are bits used to describe the format, and &xx is the format.

The values for &xx are as follows:

01	2-channel 32-bit LPCM format
02	2-channel 24-bit LPCM format
03	2-channel 16-bit LPCM format
05	2-channel 16-bit SoundDMA LPCM format
06	*n*-channel, *m*-bit LPCM format
07	*n*-channel, 8-bit µ-Law format
Others	Reserved

Note that the format descriptor of &FFFFFFFF is reserved, and used to indicate the end of a list of formats.

2-channel 32-bit LPCM format (format &01)

This format should be supported by all SoundSystem devices, as this is the native format that applications must provide.

The “nnnnnn” bits are reserved and set to zero.

The format of a the data itself consists of two signed 32-bit integers per sample, with the first integer being the left channel's data, and the second integer being the right channel's data.

Each integer is stored in little endian format, so a sample value of &12345678 is stored as &78, &56, &34, &12.

Any buffer **must** be word aligned, and have a length of a multiple of words.

2-channel 24-bit LPCM format (format &02)

This format can only be used by applications registering exclusive access to the audio device where the audio device supports it.

The "nnnnnn" bits are reserved and set to zero.

The format of the data itself consists of a pair of 3-byte signed integers per sample, with the first of the three bytes giving the left channel's data, and the last of the three bytes giving the right channel's data.

Each 3-byte value is formatted in little endian format, which means a sample of &123456 is encoded as the bytes &56, &34, &12.

This format does not require the buffers to be word aligned.

2-channel 16-bit LPCM format (format &03)

This format can only be used by applications registering exclusive access to the audio device where the audio device supports it.

The "nnnnnn" bits are reserved and set to zero.

The format of the data itself consists of a pair of 2-byte signed integers per sample, with the first of the two bytes giving the left channel's data, and the last of the two bytes giving the right channel's data.

Each integer is stored in little endian format, which means a sample of &1234 (left) paired with &5678 (right) is encoded as the byte sequence &34, &12, &78, &56.

This format requires the buffers are word aligned.

2-channel 16-bit SoundDMA format (format &05)

This format can only be used by applications registering exclusive access to the audio device where the audio device supports it. It is also the format that is supported by SharedSound and SoundDMA interfaces.

The "nnnnnn" bits are reserved and set to zero.

The format of the data itself consists of a pair of 2-byte signed integers per sample, with the first of the two bytes giving the right channel's data, and the last of the two bytes giving the left channel's data.

Each integer is stored in little endian format, which means a sample of &1234 (left) paired with &5678 (right) is encoded as the byte sequence &78, &56, &34, &12.

This format requires the buffers are word aligned.

N-channel, m-bit LPCM format (format &06)

This format can only be used by applications registering exclusive access to the audio device where the audio device supports it.

The top-24 bits are split into the top 16-bits giving a bit field indicating the channels present, the next 4 bits the number of channels, and the last 2 bits the number of bytes per sample:

Bit(s)	Description
0 – 7	&06 (format descriptor)
8 – 11	Number of bytes per sample 0 = reserved, 1 = 8-bit, 2 = 16-bit, 3 = 24-bit, 4 = 32-bit, 5 – 15 = reserved
12 – 15	Number of channels (1-15, 0 = reserved)
16	If set, left front data is present
17	If set, right front data is present
18	If set, centre front data is present
19	If set, low frequency enhancement data is present
20	If set, left surround data is present
21	If set, right surround data is present
22	If set, left of centre data is present
23	If set, right of centre data is present
24	If set, surround data is present
25	If set, side left data is present
26	If set, side right data is present
27	If set, top data is present
28 – 31	Reserved (0)

Note that the number of bits set in 16 – 31 does not need to equal the number of channels – although it must not exceed the number of channels. This permits data for spacial locations that are not defined in the above list to be present. The interpretation of such data depends on the audio device.

The data is presented in the order of the bits set in 16 – 31, with any non-spatially aware data after the spatially aware data. This means that if there are 5 channels present, with bits 16, 17 and 18 set, then the first data will be the left front channel, the second data will be the right front channel, the third data will be the centre front data. The remaining two pieces of data are for the non-spatially aware channels.

The buffer that holds the data does not need to be word aligned for 8-bit and 24-bit formats, but must be for 16-bit and 32-bit formats.

8-bit μ-Law format (format &07)

This format is only aimed at backward compatibility with earlier RISC OS systems.

Bits 8 – 11 of the format identifier are the number of channels (1, 2, 4 and 8 – the remainder are reserved). The remaining bits (12 – 31) are reserved.

Stereo positioning of the data is not included in the channel data, but is included as part of the 8-bit µ-Law configuration.

Each channel is presented as a single byte, as per the VIDC1 format (bit 0 = sign, bits 2 – 7 = logarithmic value).

The buffer that holds the data does not need to be word aligned.

Other formats

Other format descriptors are yet to be defined, but are anticipated to be non-LPCM data, such as MP3, or Bluetooth’s SBC.

These formats tend to be compressed, and will only be useable when a device is being used in exclusive access – merging with existing data will not be possible.

For these formats, the data is likely to consist of frames, and buffers will be large enough to contain one or more whole frames. The amount of data may be variable.

Audio handlers

In order for applications to send and receive data to an audio device, audio buffer handlers are registered by the applications with SoundSystem.

Data is transferred to the audio device with a playback handler, and received from the device with a recording handler.

Applications can either register for shared access to the audio devices, or they can register for exclusive access.

Shared access

Shared access allow multiple applications to share an audio device, by mixing the audio playback streams together. Audio capture data is passed to all applications interested in recording audio.

All applications implementing shared audio playback must support 2-channel (or higher), 32-bit linear PCM formatted data at any sample rate (that can change at any point) and mixing the sounds from other applications.

Audio devices must also support this format, converting the data to or from the device’s native format where necessary.

Exclusive access

Exclusive access precludes the ability for applications to share an audio device – but an application can reconfigure the audio device to support different audio formats.

As there is no need for applications to share the audio device, mixing and variable sample rates do not need to be considered for the audio playback handler.

If an application already has exclusive access to an audio device, and another application requests exclusive access, then the new application will have exclusive access. When the new application relinquishes its exclusive access, then the previous application will regain the access.

When the last exclusive application relinquishes its access, then the shared system will be given control back again.

If a shared application joins the shared system while exclusive access is in use by another application, then it will lie dormant until all exclusive applications have relinquished their access.

In order to maintain the exclusive access, when an application registers for exclusive access on a device, it is allocated a unique access token. In order to perform operations on the device on an exclusive basis, the application must pass this through to the relevant SWI calls. This also prevents one application from overriding another application's settings, which could cause unexpected results for both the user and the application.

Playback handler code

The audio playback handler's function is to provide data for the sound output. The code for all formats is essentially the same.

On entry

R0 = pointer to buffer to fill
R1 = length of buffer (in bytes)
R2 = bit field (undefined for exclusive access)

Bit	Description when set
0	Buffer contains existing data to be merged
1	Buffer will be merged down to a mono-channel output
2 – 23	Reserved
24 – 31	Volume (0 = silent, 128 = 1:1, 255 = approximately 2:1)

R3 = 0
R12 = value given when the playback handler was registered

On exit

R0 = pointer to byte after buffer was filled
R3 = maximum amplitude (pre-volume level), from 0 to 127 (if in shared playback mode)
Other registers must be preserved

Processor mode

Processor is in SVC or interrupt mode

Use

The current format and sample rate are not passed into this handler, as they can be queried via a SWI call, or via the SoundSystem Service Call.

If bit 0 or R2 is set on entry, then the application can stop at its last sample, and does not need to fill in the rest.

Shared playback

If the shared playback handler is currently silent, or has finished with its playback, then it can simply return, leaving R0 pointing to the word after the last one written.

The volume provided in R2 is set so that a value of 128 will result in no change between the source amplitude and the output amplitude. The use of long-multiply and saturated add instructions is recommended for performance.

R3 can be used by mixer applications to show the current amplitude of signals coming from each playback handler before they are affected by the volume.

Exclusive playback

The whole of the buffer must always be filled for exclusive playback handlers.

Example shared playback code

The following code is an example where data needs to be merged (i.e. bit 0 is set in R2 on entry). The source data is pointed to in R4, and is already 32-bit stereo, at the correct sample rate.

```
   AND      R2, R2, #&FF000000       ; Remove everything except volume
.fillmerge_loop
   LDR      R5, [R4], #4             ; Read the sample word in
   MOVS     R6, R5, ASR#24           ; Get the high byte
   RSBMI    R6, R6, #0               ; If it is negative, invert it
   CMP      R6, R3                   ; Compare with previous maximum
   MOVGT    R3, R6                   ; Replace if greater
   UMULL    R5, R6, R2, R5           ; Multiply by the volume
   LDR      R7, [R0]                 ; Read in the previous word
   QDADD    R7, R7, R6               ; Add the two (with saturation)
   STR      R7, [R0], #4             ; Store the new value in place
   SUBS     R1, R1, #4               ; See if we have more words
   BGT      fillmerge_loop           ; Yes, so loop back
```

This code does not differentiate between left and right samples – it treats the data as one contiguous block. Optimisations can be performed, such as performing multiple reads at the same time (to take advantage of the cache).

Note that the QDADD instruction is performing the conversion of a volume level of 128 into a 1:1 multiplication – but it will effectively lose the bottom bit from the volume-controlled sample. In almost all circumstances, this loss of precision is acceptable.

For the case where there is no previous data (bit 0 of R2 is clear), then it can be simplified:

```
   AND      R2, R2, #&FF000000       ; Remove everything except volume
.fill_loop
   LDR      R5, [R4], #4             ; Read the sample word in
   MOVS     R6, R5, ASR#24           ; Get the high byte
   RSBMI    R6, R6, #0               ; If it is negative, invert it
   CMP      R6, R3                   ; Compare with previous maximum
   MOVGT    R3, R6                   ; Replace if greater
   UMULL    R5, R6, R2, R5           ; Multiply by the volume
   QADD     R6, R6, R6               ; Double the value, and saturate
   STR      R6, [R0], #4             ; Store the new value in place
   SUBS     R1, R1, #4               ; See if we have more words
   BGT      fill_loop                ; Yes, so loop back
```

Again, optimisations can be made.

Recording handler code

The audio recording handler is called when there is data received from the audio device. The code for all formats is essentially the same.

On entry

R0 = pointer to buffer containing data
R1 = length of data in buffer (in bytes)
R3 = maximum amplitude in the sample, from 0 to 127
R12 = value given when the recording handler was registered

On exit

Registers must be preserved

Processor mode

Processor is in SVC or interrupt mode

Use

The current format and sample rate are not passed into this handler, as they can be queried via a SWI call, or via the SoundSystem Service Call.

Simple shared playback and recording

For convenience, SoundSystem provides an application that does not need low-latency audio playback and / or recording. This is performed via the SWI SoundSystem_LargeSharedBuffer, and is particularly suitable for applications such as CD players, and audio recorders.

Applications need to allocate an area of memory that will be used to play and / or record audio, and register this area with SoundSystem, and the application can query the buffer status so that it can provide new audio or deal with the recorded data.

Applications that need low-latency or exclusive access will need to implement their own playback and / or recording handler.

Codec interface

Codecs can be provided by relocatable modules that have registered the source and destination format with SoundSystem.

The codec interface is performed exclusively via Service Calls. Modules that implement codecs need to handle the Service Calls and provide pointers to code accordingly.

Codec configuration handler

The codec initialisation handler is called to prepare the codec for data it needs to convert, or to reconfigure itself for a different sample rate.

On entry

R0 = reason code

Other registers depend on reason code

On exit

R0 preserved
Other registers depend on reason code

Use

This call is used to configure, reconfigure and terminate the codec.

Codec configuration 0

Get the size of the contextual data.

On entry

R1 = pointer to codec configuration

+0	Bit field for codec configuration
+4	Source format descriptor
+8	Source sample rate (in Hz, multiplied by 1024)
+12	Destination format descriptor
+16	Destination sample rate (in Hz, multiplied by 1024)

R2 = length of the destination buffer (in bytes), or 0 for undefined

The bit field is as follows:

Bit	**Value when set**
0 – 3	Quality indicator (0 = low quality, 15 = highest quality)
4	Codec will need to merge with existing data
5	Source sample rate is the maximum sample rate (0 is average sample rate)
6	Destination sample rate is the maximum sample rate (0 is the average sample rate)
9 – 23	Reserved (0)
24 – 31	Volume (0 = silent, 255 = maximum)

On exit

R1 = the length of data required for the codec to store its contextual data

Use

This call is used to get the length of data the codec needs for its contextual data. The codec will return the number of bytes that the codec user will need to allocate in order for it to function.

Codec configuration 1

Configures and initialises the codec.

On entry

R1 = pointer to codec configuration (as per Codec configuration 0)
R2 = length of the destination buffer (in bytes), or 0 if undefined
R3 = pointer to codec contextual data (if any was needed)

On exit

R1 = pointer to codec conversion code

Use

This is used to initialise a codec for the given source and destination configuration

A codec may offer more than one ‘quality’ for its conversion, for example a codec that simply converts from one sample rate to another may have a simple conversion where samples are repeated or skipped, and another where some filtering is applied pre- or post- conversion to reduce aliasing artefacts that can occur with the simplistic approach. The quality would normally increase the amount of CPU activity required to perform the conversion.

Note that for non-LPCM formatted data, the sample rate may be variable – in which case the sample rate will be the average or maximum sample rate depending on bits 5 and 6 of the bit field.

Codec configuration 2

Reconfigures the codec.

On entry

Registers as per Codec configuration 1

On exit

R0 = 0 for codec needs to be reinitialised from the beginning
    = 1 for codec has been reconfigured
    Other values reserved
R1 = pointer to codec conversion code

Use

This is used by the codec user to indicate that some of the parameters have changed, and the codec needs to work differently.

If R0 is 0 on exit, then the codec is no longer relevant at all, and the codec needs to be destroyed, and a new codec created (with memory allocated).

Note that the codec conversion code pointer may change when the codec is reconfigured.

Codec conversion handler

The codec conversion handler is called to perform the conversion from one format to another format.

On entry

R0 = the pointer to the source buffer
R1 = the length of the data in the source buffer
R2 = the pointer to the destination buffer
R3 = the length available in the destination buffer
R12 = the pointer for the codec contextual data

On exit

R0 = the pointer to the byte after the source buffer that has been used
R1 is bit field
R2 = the pointer to the byte after the destination data that has been filled
R3 can be corrupted
Other registers preserved

Use

The codec conversion handler takes data passed into it, and writes it to the destination buffer. The bit field is as follows:

Bit	Value when set
0	Not all of the source data was used
1	Not all of the destination data was filled
2 – 31	Reserved (0)

If the codec does not use all of the source buffer, then the caller must add new data to the end of the previous data (or copy the previous data and then append it) ready for the next call. R1 will have bit 0 set on return.

If the codec requires the destination buffer to be completely filled, then the caller must add more data to the source and call the codec again until bit 1 of R1 is set on return.

If there is not enough data in the source or the destination for the codec to function, then R0 and R2 will be preserved. The caller will need to increase the amount of source data, and/or the output buffer size.

The contextual data is used to maintain information required by the codec between calls.

Codec finalisation handler

This is an optional handler that allows a codec to tidy up any data it may have held outside its contextual data.

On entry

R12 is the pointer to the codec contextual data

On exit

Registers preserved

Processor mode

Processor is in SVC or interrupt mode

Use

Not all codecs will require special finalisation code. Ordinarily, all that needs to happen is the codec is not called any more – and SoundSystem simply deallocates the memory that had been assigned to its contextual data.

Codecs should not store anything that is not in its contextual block, but if they do, then this handler is used to inform them that this instance is about to be terminated.

Hardware interface

The hardware interface is used by hardware drivers to interact with SoundSystem. Hardware drivers need to ask SoundSystem to get data to play back, and tell SoundSystem that there is data that has been recorded.

During registration, a hardware driver will pass in a control block, with the data at offset +12 being a pointer (or an offset from the module start) being “control code” that is called by SoundSystem to provide the driver with configuration details, and the data at offset +16 being a pointer to the driver's private word.

Hardware interfaces must support 32-bit 2-channel audio if they want to support the shared access system. Otherwise, they can only be used by applications that support their audio formats.

Control code

This performs various functions depending on the reason code passed in to R0. Unused and unsupported reason codes are to be ignored.

Control code 0

Sets up the addresses that the hardware driver needs to call in order to request or provide sample data

On entry

R0 = 0 (reason code)
R1 = pointer to playback data request
R2 = pointer to recording data provision
R3 = value that must be passed into R12 on entry to either routine above

On exit

Registers preserved

Use

The hardware driver is provided with the addresses that it must call in order to request the playback data, or to provide the recorded data.

The code that is called is essentially the same for both playback and recording, and is detailed later.

Note that both pointers will be provided even if the device does not support playback or recording.

Control code 1

Sets or queries the sample format for use for playback and recording.

On entry

R0 = 1 (reason code)
R1 = new sample playback format descriptor (or -1 to query)
R2 = new sample recording format descriptor (or -1 to query)

On exit

R1 = new or queried sample playback format descriptor
R2 = new or queried sample recording format descriptor

Use

This call is used by SoundSystem to inform the hardware driver that the sample format needs to be changed.

If the device does not support the given format descriptor, it should return R1 as 0.

Control code 2

Sets or queries the sample rate for playback and recording.

On entry

R0 = 2 (reason code)
R1 = new sample rate or -1 to query (in Hz, multiplied by 1024)

On exit

R1 = new or queried sample rate (in Hz, multiplied by 1024)

Use

This call is used by SoundSystem to get or set the sample rate for use in playback or recording.

If the device does not support the given sample rate for the current sample format, then it should return R1 as the nearest sample rate that is supported.

Control code 3

Sets or queries the overall output volume or output mixer volume level.

On entry

R0 = 3 (reason code)
R1 = mixer channel number (0 for overall)
R2 = new output volume level (0 = silent, 255 = maximum volume, -1 to query)

On exit

R2 = previous output volume level

Use

This call is used by SoundSystem to get or set the overall output volume level, or the output mixer level for the given channel number.

If R1 is 0 on entry, then the overall output volume level is being set (or queried).

Otherwise, R1 is set to the mixer channel number. The channels are defined by the hardware capability bit flags, with mixer channel 1 indicating the channel with the lowest bit set in bits 16 – 27, mixer channel 2 indicating the channel with the second lowest bit and so forth.

Control code 4

Sets or queries the overall input volume or input mixer volume level.

On entry

R0 = 4 (reason code)
R1 = mixer channel number (or 0 for overall)
R2 = new input volume level (0 = silent, 255 = maximum volume, -1 to query)

On exit

R2 = previous input volume level

Use

This call is used by SoundSystem to get or set the overall input volume level, or the input mixer level for the given channel number.

If R1 is 0 on entry, then the overall input volume level is being set (or queried).

Otherwise, R1 is set to the input mixer channel number. The channels are defined by the hardware capability bit flags, with mixer 1 indicating the channel with the lowest bit set in bits 8 – 11, mixer 2 indicating the channel with the second lowest bit and so forth.

Control code 5

Enables or disables the playback and / or recording

On entry

R0 = 5 (reason code)
R1 = bit field

Bit	Value when set
0	Enable playback
1	Enable recording
2 – 31	Reserved (0)

On exit

Registers preserved

Use

This call is used by SoundSystem to tell the hardware driver if it needs to perform a request for playback data, or to provide recording data.

For example, if there are no playback or recording handlers attached to a device, then the device does not need to request or provide any data.

Control code 6

Request the list of format descriptors that the device supports

On entry

R0 = 6 (reason code)
R2 = pointer to block of sample format descriptors to fill in, or 0 to query length
R3 = length of block of data (in bytes)

On exit

The block pointed to by R2 is filled with the sample format descriptors, if non-zero on entry.

R3 is set to the number of format descriptors available.

Use

See SoundSystem_GetDeviceFormats for details on the returned data.

Control code 7

Request the list of sample rates for the given format indicator

On entry

R0 = 7 (reason code)
R1 = sample format descriptor
R2 = pointer to block of sample rates to fill in, or 0 to query length
R3 = length of block of data (in bytes)

On exit

The block pointed to by R2 is filled with the available sample rates (in Hz, multiplied by 1024), if non-zero on entry.

R3 is set to the number of sample rates available for the given sample format.

Use

This is used to query the list of sample rates that a device can support for a given sample format.

Some devices may have a bandwidth threshold which means that a higher number of channels or bits per channel may reduce the maximum sample rate supported by the device.

If all sample rates are supported by all sample formats, then the same data will be returned.

See SoundSystem_GetDeviceFormatSampleRates for details on the returned data (including the divisor-based mechanism).

Control code 8

Gets or sets the buffer size for playback and recording

On entry

R0 = 8 (reason code)

R1 = requested buffer size in bytes (must be a multiple of 4 * channel count for 32-bit LPCM samples), or:

0 = Query the current buffer size for the current sample format and rate
-1 = Query the default buffer size for the given sample format and rate
-2 = Query the minimum buffer size for the given sample format and rate
-3 = Query the maximum buffer size for the given sample format and rate

R2 = sample format descriptor (if R1 = -1, -2 or -3 on entry)
R3 = sample rate (in Hz, multiplied by 1024 – if R1 = -1, -2 or -3 on entry)

On exit

R1 = buffer size (in bytes)

Use

This is used to get or set the current buffer size in bytes. Setting the buffer size can only occur if exclusive access has been granted on the device.

Note that the buffer size for non-LPCM formats relates the amount of data that can be potentially used – a variable bit-rate format may result in less data being passed (the buffer can never be overfilled).

If the buffer size is outside its bounds, then the device driver can override the requested buffer size with values that are in the range. For example, if the requested buffer size is larger than the hardware buffers.

Playback data request

The hardware driver calls the playback data request to get data.

On entry

R0 = pointer to playback buffer to fill in
R1 = length of playback buffer (in bytes)
R12 = value of R3 when given the playback request code pointer in Control code 0.

On exit

R0 = pointer to byte after filled data
Other registers preserved

Use

When the hardware device needs some data, the driver makes a call to the playback data request passing in the pointer to a buffer in R0, and the length of the buffer in R1.

SoundSystem then goes through the list of audio playback handlers for them to fill the data, merging with the previous data if necessary.

LPCM formats

It is guaranteed that the whole buffer will be filled in by the playback handler(s), so the return value of R0 can be ignored by the hardware driver.

Non-LPCM formats

Not all of the data may be filled in when dealing with non-LPCM formats, and the value of R0 allows the hardware driver to determine how much of the buffer has actually been filled.

If the hardware driver requires more data, then it can call the playback data request code repeatedly until it has enough data – although if the value of R0 does not change, then it must assume that there is no more data ready, and cope with this accordingly.

Recording data provision

The hardware driver informs SoundSystem that there is data that has been captured.

On entry

R0 = pointer to buffer containing data
R1 = length of buffer (in bytes)
R12 = value of R3 when given the recording provision code pointer in Control code 0

On exit

Registers preserved

Use

When the hardware driver has recorded some data, it calls the recording data provision code to inform SoundSystem that new data is present, with R0 pointing to the start of the data, and R1 being the length of the data.

Any applications that are receiving this data must be able to handle all of this data, and when the call returns, the hardware driver can assume that the data is no longer required.

This is true for both LPCM and non-LPCM formats.

Service calls

Service_SoundSystemInitialised
(Service Call &81140)

The SoundSystem module is initialising, and applications and hardware devices can start to register with it.

On entry

R0 = 0 for initialisation; 1 for finalisation; all other values reserved (these must be ignored)

On exit

All registers preserved

Use

Hardware drivers can use this service call to register their hardware devices with SoundSystem so that applications can start to use them.

Finalisation is called so that hardware devices can stop providing their interrupts for audio playback and recording – hardware drivers and codec providers do not need to deregister the services they are supporting.

This service call must not be claimed.

Service_SoundSystemDeviceAdded
(Service Call &81141)

A new device has been registered with SoundSystem.

On entry

R0 is the pointer to the device identifier that has been added

On exit

All registers preserved

Use

This service call is issued when a new device has been registered with SoundSystem_RegisterDevice.

This service call must not be claimed.

Service_SoundSystemDeviceRemoved
(Service Call &81142)

Hardware has been removed from the system.

On entry

R0 is the device identifier that has been removed

On exit

All registers preserved

Use

This service call is issued when has been removed with SoundSystem_DeregisterDevice.

All playback and recording handlers associated with the given device will automatically be deregistered.

This service call must not be claimed.

Service_SoundSystemExclusiveAccessChanged
(Service Call &81143)

Exclusive access to a device has been changed.

On entry

R0 = the device identifier that exclusive access has been registered
R2 = the new identifier (nominally WIMP handle) for the exclusive access (as passed into SoundSystem_RegisterExclusiveAccess), or -1 if shared access is returned

On exit

All registers preserved

Use

This service call is issued when an application has requested exclusive access to an audio device, and allows applications to indicate that they no longer need to play their audio.

This call is also issued when an application deregisters exclusive access, and control is passed back to the previous application.

Any application that has regained its exclusive access to the device will need to reconfigure the device to its requirements, and re-attach its playback and/or recording handlers.

If an application deregisters exclusive access, and there were no previous exclusive registrations, then -1 is returned to indicate that shared access is returned.

This service call must not be claimed.

Service_SoundSystemRequestCodec
(Service Call &81144)

Used by applications and/or drivers to request a codec.

On entry

R0 = source format descriptor
R2 = source sample rate (in Hz, multiplied by 1024)
R3 = destination format descriptor
R4 = destination sample rate (in Hz, multiplied by 1024)
R5 = bit flags

Bit	Value when set
0 – 3	Quality indicator (0 = low quality, 15 = highest quality)
4	Codec will need to merge with existing data
5	Source sample rate is the maximum sample rate (0 is average sample rate)
6	Destination sample rate is the maximum sample rate (0 is the average sample rate)
7	Codec must support volume levels
8 – 31	Reserved (0)

On exit

If the codec is supported, then this service call is claimed, with the following registers set:

R0 = bit flags

Bit	Value when set
0 – 1	Reserved (0)
2	Codec can support changing of sample rates without reconfiguring
3 – 31	Reserved (0)

R2 = pointer to codec provider name
R3 = pointer to codec initialisation code
R4 = pointer to codec finalisation code (zero if none)
R5 = size of contextual data needed

Use

This service call is used by an application or device driver to request code to convert from one format to another format, and/or one sample rate to another sample rate. For example:

- Applications can use this to convert a format that is present on a hard disk to another format before passing it through to SoundSystem.
- Hardware devices can use this to convert the standard 32-bit, 2-channel LPCM format to a format the device itself supports

If the codec is needed, then the application needs to allocate the amount of memory as indicated by R5 (in an area of memory that will be available when the codec conversion code is called).

When the codec is no longer needed, the finalisation code needs to be called (if present)

Service_SoundSystemCodecRemoved
(Service Call &81145)

A codec has been removed from SoundSystem.

On entry

R0 = the source format descriptor
R2 = the destination format descriptor
R3 = the pointer to the codec provider name

On exit

All registers preserved

Use

This service call is issued when a codec is deregistered. Applications should check the values passed in – if R0 is 0, then all codecs supported by the provider are no longer valid; otherwise the given source and destination formats are no longer supported.

If an application finds that its codec is no longer supported, then it must stop using that codec, and either select a different codec, or stop any audio playback (or ignore incoming audio).

Applications do not need to deregister their codecs upon receipt of this service call.

Service_SoundSystemSampleRateChanged
(Service Call &81146)

Informs applications that the shared sample rate has been changed.

On entry

R0 = old sample rate (in Hz, multiplied by 1024)
R2 = new sample rate (in Hz, multiplied by 1024)
R3 = device identifier

On exit

All registers preserved

Use

This service call is issued when an application has requested the change of the sample rate of a shared access hardware device. The sample rate affects both playback and recording, so applications should adjust their playback and recording to accommodate the change.

Applications needing a codec to change the sample rate from one to another may need to change their codec.

This service call is not issued when exclusive access has been given on the hardware device, as only the application with exclusive access can change the sample rate.

SWI calls

SoundSystem_RegisterDevice (SWI &5A080)

Registers a hardware device with SoundSystem.

On entry

R0 = pointer to user-orientated device name
R1 = pointer to device identifier (zero-terminated string)
R2 = pointer to device information block

+0	Bit flags of capabilities
+4	Maximum number of channels
+8	Pointer or offset to control code
+12	Value to pass in as R12 to the device's Control Code

On exit

Registers preserved

Use

This call is used by hardware to register a device with SoundSystem, providing it with a user-orientated device name, and a device identifier (which must be unique across a system).

Applications should use the user-orientated device name when providing a list of options to the user, but the device identifier when registering to transfer data with the audio device.

The bit flags are:

Bit	Value when set
0	Device supports playback
1	Device supports recording
2	Device supports LPCM playback / recording
3	Device supports non-LPCM playback / recording
4 – 7	Reserved (0)
8	Device supports microphone input
9	Device supports line input
10	Device supports digital input
11 – 15	Reserved (0)
16	Left front mixer is present
17	Right front mixer is present
18	Centre front mixer is present
19	Low frequency enhancement mixer is present
20	Left surround mixer is present
21	Right surround mixer is present
22	Left of centre mixer is present
23	Right of centre mixer is present

24	Surround mixer is present
25	Side left mixer is present
26	Side right mixer is present
27	Top mixer is present
28	Reserved (0)
29	Microphone input mixer is present
30	Line input mixer is present
31	Digital input mixer is present

SoundSystem_DeregisterDevice
(SWI &5A081)

Deregisters a device that has been registered with SoundSystem.

On entry

R1 = pointer to device identifier (zero-terminated string)

On exit

Registers preserved

Use

This call is used to de-register a device with SoundSystem.

SoundSystem_EnumerateDevices
(SWI &5A082)

This call is used to find out the list of audio devices available on the system.

On entry

R1 = pointer to previous device identifier (or 0 to start)

On exit

R1 = pointer to device identifier (or 0 for end of list)
Other registers preserved

Use

This call is used by applications to determine the available audio devices that are available on the system.

The first call an application would make starts with R1 set to 0. Upon return, R1 is updated to point to the new device identifier name. A subsequent call to SoundSystem_GetDeviceInformation can be made to read the device's user-orientated name and capabilities

The application repeatedly calls this SWI until R1 is equal to zero.

Note that the value returned R1 is the copy of the value as passed into SoundSystem_RegisterDevice.

SoundSystem_GetDeviceInformation (SWI &5A083)

This call is used to find out the information about a particular device.

On entry

R1 = pointer to previous device identifier (or 0 to start)

On exit

R0 = pointer to user-orientated device name
R1 preserved
R2 = pointer to device information block (as per SoundSystem_RegisterDevice)

Use

This call is used to get the device name and device information block for a particular device identifier. It can be called with SoundSystem_EnumerateDevices to get a list of all of the devices and their capabilities.

Note that the values returned in R0 and R2 are copies of the values as passed into SoundSystem_RegisterDevice.

SoundSystem_GetDeviceFormats
(SWI &5A084)

Used to iterate over the list of device formats

On entry

R0 = pointer to block of memory to hold format descriptors, or 0 to query length
R1 = device identifier
R2 = length of data available in the block (if R0 is non-zero on entry)

On exit

R2 = length needed to hold all of the available sample rates (if R0 was zero on entry), or length filled in (if R0 was non-zero on entry)

Other registers preserved

The data block passed into R0 is filled in with the device formats supported by the device if R0 was non-zero on entry

Use

This call is used to iterate over the list of formats the device can support.

An application will typically call this with R0 set to zero to get the amount of memory that is needed to get the list of all of the available formats, as returned in R2.

With the memory allocated, the application can then make a second call to fill in the memory block with the list of available formats.

Note that if there is not enough memory in the block, then the memory will still be filled in – up to the number of available formats.

SoundSystem_GetDeviceFormatSampleRates (SWI &5A085)

Used to get the list of sample rates offered by a device for a particular format

On entry

R0 = pointer to block of data to hold sample rates, or 0 to query number of sample rates
R1 = device identifier
R2 = sample format indicator
R3 = length of the block (if R0 is non-zero on entry)

On exit

R3 = length needed to hold all of the available sample rates (if R0 was zero on entry), or length filled in (if R0 was non-zero on entry)

Other registers preserved

The data block passed into R0 is filled in with the available of sample rates for the given format indicator if R0 was non-zero on entry. Each sample rate is the rate given in Hz, multiplied by 1024.

Note that some devices can support an arbitrary divisor of a particular clock frequency or frequencies.

Use

An application can use this to query the list of sample rates available on a particular device for a given sample format indicator.

The application should initially call this with the required sample format indicator, and R0 set to zero. This will return with R3 set to the length of data required to fill in all of the information.

After allocating the required amount of memory, the application should then call this again with R0 pointing to the memory block, and R3 set to the size of the block.

If an application wants to use a fixed block, then it can simply pass R0 as pointing to the memory block, and R3 the size of the block.

The block will be filled in with as many sample rates as it can up to the available space.

Divisor-based sample rates

Some devices have the capability of generating a sample rate based on a particular clock frequency (or frequencies) with a divisor. As this can result in a large number of sample rates if there are a large number of divisors available, an alternative method is provided to support these devices.

The same mechanism to determine the size is used as before (R0 is set to zero on entry, and R3 is set to the data size on exit).

If the first sample rate is zero, then the divisor mechanism is active – the remaining data consists of a number of blocks consisting of 3 words:

+0	Clock frequency given in Hz (note: not multiplied by 1024).
+4	Minimum divisor value.
+8	Maximum divisor value.

This is repeated for each clock frequency.

For example, if a device supports two clock frequencies of 32MHz and 48MHz, with a divisor of between 128 and 1023, then the following block will be returned (R3 will be 28):

R3 = 28 (7 words of data)

+0	0 (divisor mechanism)
+4	32000000 (32MHz)
+8	128 (minimum divisor)
+12	1023 (maximum divisor)
+16	48000000 (48MHz)
+20	128 (minimum divisor)
+24	1023 (maximum divisor)

The available sample rates are 32,000,000 / 1023 to 32,000,000 / 128 (31.280KHz to 250.000KHz), and 48,000,000 / 1023 to 48,000,000 / 128 (46.920KHz to 375.000KHz).

A sample rate of 44.1KHz can be specified as a clock speed of 32MHz, and a divisor of 726 (this would actually give a sample rate of 44.077KHz). 48KHz can be represented as a clock speed of 48MHz, and a divisor of 1,000.

SoundSystem_RegisterExclusiveAccess (SWI &5A086)

This call is made by an application to register exclusive access to the audio device.

On entry

R1 = device identifier

On exit

R0 = assigned exclusive access token

Use

The value returned by R0 is a unique token that is allocated for this particular exclusive access request. Applications must use it when de-registering their exclusive access.

If more than one application registers for exclusive access to a device, the most recently registered application will have the access. When the application deregisters, the previous application will have exclusive access again.

SoundSystem_DeregisterExclusiveAccess (SWI &&5A087)

This call is made by an application to deregister exclusive access to the audio device.

On entry

R0 = exclusive access token (as given by SoundSystem_RegisterExclusiveAccess)

On exit

Registers preserved

Use

Applications will call this SWI when deregistering their exclusive access – for example, at application termination. Control will pass back to the previous application that had exclusive access – unless there are no exclusive applications remaining, in which case control passes back to the shared sound system.

Note that an application can deregister itself when another application has exclusive access, and it is removed from the list of applications having exclusive access to that device.

SoundSystem_BufferSize
(SWI &5A088)

Used to get or set the buffer size

On entry

R0 = exclusive access token (if setting the buffer size)
R1 = pointer to device identifier
R2 = requested buffer size (in bytes), or value used to query buffer sizes

0	Request current buffer size (for current sample format and rate)
-1	Request default buffer size (for given sample format and rate)
-2	Request minimum buffer size (for given sample format and rate)
-3	Request maximum buffer size (for given sample format and rate)

R3 = sample format indicator (if R0 < 0 on entry), 0 for current format
R4 = sample rate (if R0 < 0 on entry), 0 for current rate

On exit

R0, R1 preserved
R2 = buffer size (new or queried)
R3, R4 preserved

Use

This call can be used to query the available buffer sizes for the given sample format and rates, or set the buffer size.

Note that the buffer size can only be set if the device has been granted exclusive access to the application.

SoundSystem_SampleFormat
(SWI &5A089)

This call is used to set the audio playback and recording sample format.

On entry

R0 = exclusive access token (ignored if querying)
R1 = pointer to device identifier (zero terminated string)
R2 = playback sample format identifier, or -1 if querying the current playback format
R3 = recording sample format identifier, or 0 to use the same as playback, or -1 to query

On exit

R0, R1 preserved
R2 = new or current playback sample format identifier
R3 = new or current recording sample format identifier

Use

This call is used to set or query the audio playback and recording format.

If R2 = -1 on entry, then R0 is ignored on entry, and the return values refer to the current audio playback format. Otherwise, R2 and R3 specify the required audio playback and recording formats respectively.

This call can only be used to set the sample format when an exclusive access has been claimed on an audio device.

SoundSystem_SampleRate
(SWI &5A08A)

This call is used to get or set the current sample rate.

On entry

R0 = exclusive access token (ignored if querying)
R1 = pointer to device identifier (zero terminated string)
R2 = sample rate (in Hertz multiplied by 1024), or -1 to query

On exit

R0, R1 preserved
R2 = new or current sample rate

Use

If R2 = -1 on entry, then R2 is set to the current sample rate (in Hertz multiplied by 1024). Otherwise, R2 is the new sample rate, and is set to the previous sample rate on exit.

Note that this affects both playback and recording sample rate.

SoundSystem_AttachHandler
(SWI &5A08B)

This call is used to attach a playback or recording handler.

On entry

R0 = pointer to playback or record handler code
R1 = pointer to device identifier (zero terminated)
R2 = bit flags:

Bit	Value when set
0	0 = Playback handler, 1 = Recording handler
1 – 31	Reserved (0)

R3 = the unique access token, or 0 if registering as a shared playback or recording handler
R4 = value of R12 passed into recording handler code
R5 = task handle, of 0 for not applicable

On exit

R0 = handle of attached handler (if shared)
Registers preserved

Use

This call is used by an application to register a playback to send audio to a device, or a recording handler to receive audio from a device.

R2 is used to indicate whether a playback handler is to be attached, or a recording handler.

R3 is used if attaching when exclusive access has been granted to the device.

The task handle is used in case the task is terminated unexpectedly – SoundSystem will immediately attempt to remove all of the task's handlers if it receives the task termination service call.

SoundSystem_DetachHandler (SWI &5A08C)

This call is used to detach a playback or recording handler.

On entry

R0 = handle of attached handler
R1 = pointer to device identifier (zero terminated)
R3 = the unique access token, or 0 if detaching a shared playback or recording handler

On exit

Registers preserved

Use

This call is used by an application to deregister a playback or recording handler from a device.

SoundSystem_EnumerateSharedHandlers (SWI &5A08D)

This call is used to iterate over the attached shared handlers

On entry

R0 = previous handler (0 to start)
R1 = pointer to device identifier (zero terminated)
R2 = bit flags:

Bit	Value when set
0	0 = Playback handlers, 1 = Recording handlers
1 – 31	Reserved (0)

On exit

R0 = current handle
R1, R2 preserved
R3 = task handle
R4 = volume

Use

This call is used to iterate over the shared handlers for a given device. R2 specifies whether the playback handlers or the recording handlers are enumerated.

SoundSystem_Volume
(SWI &5A08E)

Gets or sets the mixer settings

On entry

R0 = bit field

Bit	Value when set
0 – 7	Mixer number, or 0 for overall volume
8 – 30	Reserved (0)
31	Change input mixer (if clear, change output mixer)

R1 = mixer value (0 = silent, 128 = 1:1, 255 = maximum, or -1 to read)
R2 = device identifier
R3 = exclusive access token (or 0 for shared)

On exit

R1 = previous mixer value
Other registers preserved

Processor mode

Processor is in SVC mode

Re-entrancy

SWI is re-entrant

Use

This SWI is used to get or set the mixer values. For shared access, only the left and right channels are available. For exclusive access, all channels can be changed.

SoundSystem_Configure
(SWI &5A08F)

Configures SoundSystem

On entry

R0 = reason code
Other registers depend on reason code

On exit

R0 preserved
Other registers depend on reason code

Processor mode

Processor is in SVC mode

Re-entrancy

SWI is re-entrant

Use

This SWI provides a number of configuration options for SoundSystem.

The reason code values are as follows:

R0	**Action**
0	Mono mode

All other values are reserved

SoundSystem_Configure 0
(SWI &5A08F)

Configures the shared access sound to be mono mode.

On entry

R0 = 0 (reason code)
R1 = device identifier
R2 = 0 => stereo mode, 1 => mono mode

On exit

R2 = previous stereo mode
Other registers preserved

Use

This is used to configure the shared sound system to output mono audio, by providing a stereo to mono mix-down after all the audio playback handlers have filled their data.

Audio playback handlers are informed if the data is going to be mixed down to a mono playback when they need to fill their buffers so they can use different routines for optimisation.

Mono mode would normally only be needed if there was a single speaker output, such as that provided by an on-board sound system.

SoundSystem_SetSharedVolume
(SWI &5A090)

Sets the playback volume for a shared handler

On entry

R0 = handle of shared playback handler
R2 = new volume (0 = silent, 128 = normal, 255 = maximum)

On exit

R2 = old volume
Other registers preserved

Use

This call is used to set the volume of a shared playback handler.

A volume value of 0 indicates silence, and value of 255 is the maximum (which is approximately twice the normal volume).

A value of 128 will give the normal volume of the playback handler.

SoundSystem_LargeSharedHandler (SWI &5A091)

This call is a convenience for WIMP applications that need to perform audio playback and / or recording using a shared handler.

On entry

R0 = reason code
Other registers depend on reason code

One exit

R0 preserved
Other registers depend on reason code

Use

This is used by WIMP applications that do not need low-latency access to play or record audio data from the shared sound system, such as the playback of an audio file.

SoundSystem_LargeSharedHandler 0
(SWI &5A091)

Register a pair of buffers for audio playback and / or recording.

On entry

R0 = 0 (reason code)
R1 = pointer to first playback buffer (or 0 if no playback needed)
R2 = pointer to second playback buffer
R3 = pointer to first recording buffer (or 0 if no recording needed)
R4 = pointer to second recording buffer
R5 = buffer length (in bytes)

On exit

R1 = pointer to the buffer status block

+00	Bit flags	
	0	1 = First playback buffer needs to be filled
	1	1 = Second playback buffer needs to be filled
	2	1 = First recording buffer needs to be emptied
	3	1 = Second recording buffer needs to be emptied
	4	1 = Playback has stopped
	5	1 = Recording has stopped
	6	1 = Playback buffer underflow has occurred
	7	1 = Recording buffer overflow has occurred
	8..30	Reserved (0)
	31	1 = Shared audio device has been removed
+04	Bit flags	
	0	1 = First buffer is being emptied
	1	1 = Second buffer is being emptied
	2	1 = First recording buffer being filled
	3	1 = Second recording buffer being filled
	4..31	Reserved (0)
+08	Current position in playback buffer being filled	
+12	Current position in recording buffer being filled (from hardware)	

Other registers preserved

Use

This call is used to register a pair of buffers for shared audio playback. All buffers must be the same size, and present in non-application workspace (for example, in a dynamic area).

The data must be provided by the application as 32-bit stereo format, as required by the shared handler, at the same sample rate as the shared audio system.

The return value in R1 is block of words, and R1 itself can be used with Wimp_PollIdle to allow the application to be called with a null WIMP reason code if the buffer needs to be serviced.

If the device has been removed, then the application will be informed by the setting of bit 5 in the first word in the buffer status block.

If the audio device stops recording (either due to the user requesting it be stopped, or if the device is removed), then bit 4 will be set in the first word in the buffer status block, and the current position will indicate the maximum byte that the current recording buffer.

If the buffer pointers are 0 on entry, then the particular operation will not be performed. If there are no audio playback buffers are present, then bit 4 will not be set. Likewise, if there are no recording buffers present, then bit 5 will not be set.

Attempts to start the playback and / or recording via SoundSystem_LargeSharedHandler 1 will result in an error.

Handling playback

If an application is performing playback, then it should look at bits 0, 1, 4, 6 and 31 of the first word of the buffer status block.

If bit 0 is set, then the application needs to fill its first buffer with the data for that buffer. The whole buffer must be filled in; if there is no more data to fill, then the buffer must be filled with zeros. When the buffer is filled, the application must call SoundSystem_LargeSharedHandler with the reason code of 1 to indicate that the buffer has been filled.

If bit 1 is set, then the application needs to fill its second buffer with the data for that buffer. As for the first buffer, the whole buffer must be filled in.

If bits 0 and 1 are set, then the either buffer can be filled in first, and then the other buffer. Two calls must be made to SoundSystem_LargeSharedHandler 1 for each buffer.

If either bit is set, and there is no more data to be populated, then SoundSystem_LargeSharedHandler 2 can be called to stop audio playback.

If bit 4 is set, then the playback has been stopped. The buffers will both be cleared when this occurs, and playback will only start again when SoundSystem_LargeSharedHandler 1 has been called with new data.

If bit 6 is set, then the application can use this to give an indication to the user that the playback was interrupted by another task. Bit 4 will be set at the same time.

If bit 31 is set, then the device has been removed, and the application can stop polling to see if needs to service the buffers. Bit 6 will automatically be set when this happens, and the buffers cleared. The application can also deregister itself, and then re-register if a new shared audio device is available. If the application does not deregister itself, then as soon as the new shared audio device is available, then the playback will not automatically resume until SoundSystem_LargeSharedHandler 1 has been called to provide new audio data.

Handling recording

If an application is recording the audio data, then it should look at bits 2, 3, 4 and 7 of the first word of the buffer status block.

If bit 2 is set, then the first recording buffer is full of data, and it needs to be emptied.

If bit 3 is set, then the second recording buffer is full of data, and it needs to be emptied.

The application should keep track of the buffers it has emptied, so if both bits are set, then the application can work out the order in which the buffered data has been received. For example, if the application had previously emptied buffers 1, 2, 1, 2 and 1, then if both bits are set, it should empty buffer 2 first, and then 1.

If bits 5 or 7 are set, then the recording is currently stopped. The application can use the data in words at offset +12 to determine how much of the data is present in the buffer in order to use it. An application can resume audio recording (if the device is still present) by calling SoundSystem_LargeSharedHandler 2 is called.

If bit 31 is set, then the device has been removed. Recording will be stopped (and bit 5 will automatically be set), and the application can deregister itself, and then re-register itself when a new shared audio device is available. If the application does not deregister itself, then when a new shared audio device becomes available, it will remain registered – although recording will not automatically resume until SoundSystem_LargeSharedHandler 2 is called.

SoundSystem_LargeSharedHandler 1
(SWI &5A091)

Instructs SoundSystem that a buffer has been filled or emptied

On entry

R0 = 1 (reason code)
R1 = bit field

0	1 = First playback buffer filled
1	1 = Second playback buffer filled
2	1 = First recording buffer emptied
3	1 = Second recording buffer emptied
4..31	reserved (0)

R4 = pointer to the buffer status block, as returned in SoundSystem_LargeSharedHandler 0

On exit

Registers preserved

Use

This call is used to inform SoundSystem that a large shared handler's buffer has been filled or emptied.

For playback, any buffer that is not already full can be filled. If a buffer is filled before it has been emptied, then the result is undefined.

For recording, only a full buffer can be emptied. If a buffer is emptied before it has been filled, then the result is undefined.

More that one bit may be set, but only one of the playback bits should be set – otherwise, SoundSystem may not know the correct order to play the audio.

As indicated earlier, the data is in the 32-bit stereo format that is used by all shared audio handlers. The sample rate must be the same as the current device's sample rate, but the volume is handled by SoundSystem.

SoundSystem_LargeSharedHandler 2
(SWI &5A091)

Instructs SoundSystem that audio playback or recording is no longer needed

On entry

R0 = 1 (reason code)
R1 = bit field

0	1 = Stop playback
1	1 = Stop recording
2..3	Reserved (0)
4	1 = Acknowledge playback stopped
5	1 = Acknowledge recording stopped
4..31	reserved (0)

R4 = pointer to the buffer status block, as returned in SoundSystem_LargeSharedHandler 0

On exit

Registers preserved

Use

This call is used to inform SoundSystem that playback and / or recording is no longer required.

If the playback is stopped, then bit 4 of the first word of the playback buffer will be set so that the application is aware on its next poll cycle.

If the recording is stopped, then bit 5 of the first word of the buffer status block will be set so that the application is aware on its next poll cycle. The application can use the value indicated at offset +12 to identify how much data was placed in the recording buffer.

If bits 4 and 5 are set, then bits 4 and 5 of the first word of the buffer status block are cleared so that calls to Wimp_PollIdle will not result in a null poll event to be raised if playback or recording is not needed for a while. When playback and / or recording is resumed, then these bits will be set when the audio playback and / or recording is stopped.

Error messages

SoundSystem_ErrorNoSlots (Error &821500)

This error is reported when there are too many devices already registered with SoundSystem.

The maximum number of devices permitted in SoundSystem is configured at compile time, and defaults to 64.

SoundSystem_ErrorNameTooLong (Error &821501)

This error is reported when the device name is too long to copy into internal memory.

The maximum length for a device name is configured at compile time, and defaults to 32 characters (including the terminating zero).

SoundSystem_ErrorIdentifierTooLong (Error &821502)

This error is reported when the device identifier is too long to copy into internal memory.

The maximum length for a device identifier is configured at compile time, and defaults to 64 characters (including the terminating zero).

SoundSystem_ErrorDeviceNotFound (Error &821503)

This error is reported when the device identifier could not be found in the list of registered devices.

Note that device identifiers are case insensitive.

SoundSystem_ErrorExclusiveAccessRequired (Error &821504)

Certain calls to change a device configuration can only be performed if exclusive access has been granted.

This error will be reported if a call is made while a device is being used in shared access.

SoundSystem_ErrorTooManySharedHandlers (Error &821505)

This error is reported if too many shared handlers have been added.

SoundSystem_ErrorHandlerNotFound (Error &821506)

This error is reported if the handler could not be found to be detached.

Index

Table of Contents