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SEGA OF AMERICA, INC.  
Consumer Products Division

# **SEGA SATURN Sound Driver Implementation Manual**

Doc. # ST-241-042795

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### ***Corrections***

Chapter	Page	Correction

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**SATURN Sound Driver**  
**Sound Incorporation Manual**

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## Preface

The purpose of this manual is to aid SATURN game programmers in incorporating sound drivers into games. This manual also explains how to activate a sound system, the different sound types, and how to produce the sound.

Although this manual provides detailed instructions on how users can program all sound controls, a Sound Interface Library provided by SEGA is also available for incorporating sounds. Readers are urged to use the library when necessary. For information on the contents and usage of the Sound Interface Library, refer to its user's manual.

## Limitations

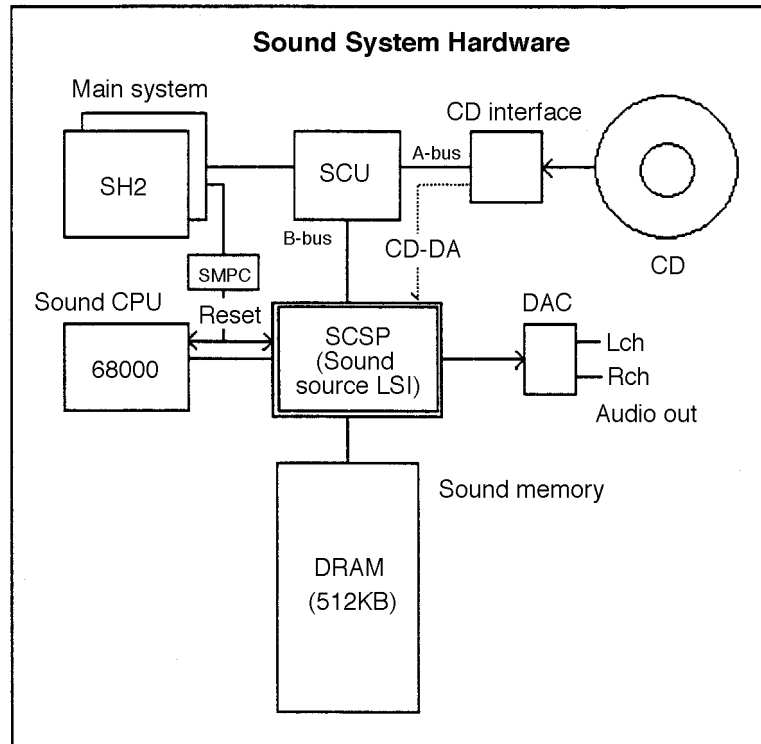
1. Sound Driver Ver. 1.29 does not support the timing flag handshake, which is a command issuing method described in section 6.1, *Executing Commands* in "Supplemental Information."
2. Sound Driver Ver. 1.29 does not support the Yamaha 3D Sound and the Q Sound, which are described in section 6.9, *3D Sounds*
3. Sound Driver Ver. 1.29 does not support P4 priority level (0-31) of the sequence start command (01h).

## Terminology

Term	Description
SCSP	Acronym for SATURN Custom Sound Processor. The sound generator LSI chip that forms the nucleus of the SATURN sound system.
Sound data	Data stored in the sound memory. The three types of sound data are tone data, sequence data, and the DSP program.
Tone data	The tone portion used in playing sequence data. Tone data consists of PCM wave data, sound parameters, and FM tones.
Sequence data	Music play data. Sequence data is MIDI-composed music data that has been compressed for SATURN and consists of play information such as sound strength, pitch, sound start, and sound stop. Effects are created with the same format.
DSP program	A DSP microprogram that operates the sound DSP. Effect programs such as reverb and surround are available.
DSP work RAM	Area for the DSP delay RAM and effect coefficients. The DSP work RAM is used for DSP internal processing. The size of this area changes according to the micro program.
Sound creator	Person in charge of creating sounds in a sound system. The sound creator creates tones and effects and composes music.
Main system	Game program in the SH unit. The sound driver in the 68000 is the sound subsystem. The game program, which controls the sound subsystem, is considered the main system.
Sample count	A sample depends on the data width of the PCM wave. For 8-bit PCM, a sample is 1 byte. For 16-bit PCM, a sample is 2 bytes. Thus a 4K sample contains 4,096 bytes for 8-bit PCM, and 8,192 bytes for 16-bit PCM.

## Chapter 1 System Overview

### 1.1 Hardware Overview



The sound system centers around a sound source LSI chip that connects to a sound CPU, a 512 KB sound memory, a main system, and a CD interface. The main system is able to access the sound memory and all SCSP I/O space through an SCU. The main system also controls the sound CPU stop (reset) and restart (reset release) operations through an SMPC.

An important component is the sound memory, which controls all sound system operations including sound driver execution, data management, storage of sound source data, and intersystem communication.

### 1.2 Sound Driver

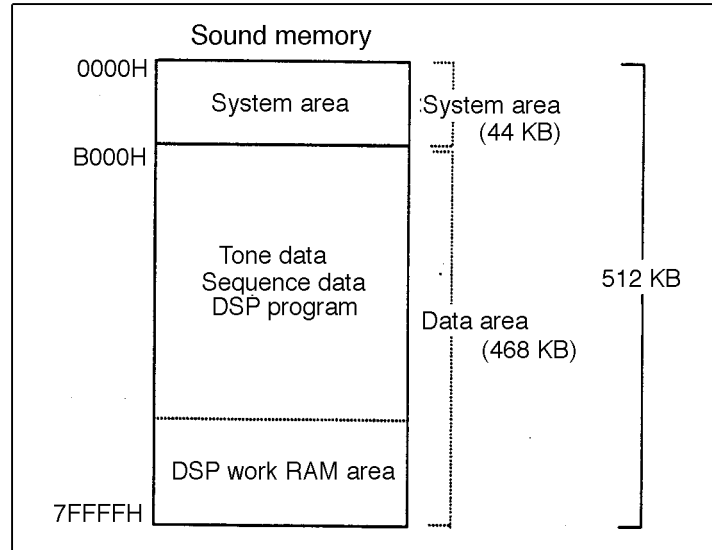
The sound driver is a sound control program that generates music and effects. The program is stored in the sound memory and operates independently from the main system. Therefore, to play music and effects, the main system need only execute commands.

Both the sound driver and the sound data are lost when the system is turned off, as they are stored in the sound memory. The main system must therefore activate the sound driver each time the power is turned on. To do so, the main system must first transfer the sound driver to the sound memory and then cancel the reset signal. This operation activates the sound driver and allows music and effects to be played.



### 1.3 Sound Memory

The sound memory includes a system area and a data area. The system area is a fixed area used for sound driver operation and system management. The data area is a variable area (variable mapping) that is used for sound source data storage and internal DSP processing.



#### System Area

The system area consists of a storage area for the sound driver operation program and a work area. The work area includes a program work area for the sound driver and a system interface area that interfaces with the main system and the sound development system. The size of the system area is fixed at 44 KB, and cannot be used for other purposes.

#### Data Area

The data area stores sound data. Three types of sound data--tone data, sequence data, and the DSP program--are stored in various combinations (mappings). If PCM stream play is performed, the data area is also used as a PCM stream play buffer. Depending on the game or the scene, the data stored in this data area and the mapping always change. In other words, the sound data is mapped for each scene in the 4 Mb range.

#### DSP Work RAM Area

The DSP work RAM area is for the DSP delay RAM and effect coefficients. If the DSP is not being used, this area is not necessary. Because of hardware constraints, the first address of this area must be set to a 2000h boundary. Also note that the size of the delay RAM and coefficient area changes according to the DSP microprogram.

## Chapter 2 Sound Driver Activation

### 2.1 Activating the Sound Driver

To activate the sound driver, transfer it to the sound memory and cancel the reset signal. However, because the memory contents are not guaranteed when the power is turned on, the memory must be initialized by zero-clear and other initializations.

Activate the sound driver according to steps 1 through 6 described below. This procedure activates and runs the program from the beginning. For information on the SCSP, reset, and other hardware details, refer to the SCSP and SMPC user's manuals.

#### 2.1.1 Activation Procedure

Sequence	Operation	Description
1	Stop the 68000. (Reset the sound CPU.)	Execute Sound OFF of the SMPC Interface Library (software library).
2	Initialize the SCSP registers.	Write 02h (1 byte) to SH address 25B00400. (Set: DAC18B = 0 and MEM4MB =1.)
3	Zero-clear the system area.	Zero-clear all SH addresses from 25A00000 to 25A0AFFF.
4	Transfer the sound program to the sound memory.	Transfer the sound program (SDDRVS.TSK) to the SH addresses beginning with 25A00000.
5	Transfer the sound area map to the sound memory.	Transfer the sound area map to the SH addresses beginning with 25A0A000.
6	Start the 68000. (Release the sound CPU reset.)	Execute Sound ON of the SMPC Interface Library (software library).

**Note:** The size of the sound area map can be up to 4,096 bytes. If the sound area map exceeds this size, divide the data and create multiple sound area maps. In this case, replace the sound area map before changing the map (see the explanation in Chapter 3, *Sound Play Procedure*), if necessary.

### 2.1.2 Necessary Data Files

The activation of the sound driver requires only the main sound driver program. The sound area map is also transferred during activation as it only has to be transferred once. During game operation, all area maps are stored in the sound area map area. Prepare the following two data files when activating the sound driver according to this procedure:

1. Sound driver (SDDRV.S) **(SDDRV.S)**  
The sound driver forms the body of the program and is an executable binary data file. SEGA provides the sound driver as a sound tool.
2. Sound area map (user-specified name)  
**The sound area map is a binary data file formed by combining all area maps in a game. This file is different for each game and is created by the sound creator. For details on the sound area map, see section 6.8, *Sound Area Map*.**

## Chapter 3 Sound Play

### 3.1 Sound Types

There are three different methods to producing sounds. The following tables describe the advantages and disadvantages of each method. Select the appropriate play method according to the content of the game and the particular scene. The three methods are:

1. Sequence play
2. PCM stream play
3. CD-DA play

#### Sequence Play

Sound generation method	Uses tone data (wave data) in the sound memory as the sound source and plays sounds while decompressing the sequence data.
Characteristics	Uses wave data as the sound source (an instrument) and treats the SATURN unit as a multi-sound source instrument.
Advantages	<ul style="list-style-type: none"> <li>• Many melodies can be included because the data size of each melody is small.</li> <li>• Music can be modified and corrected easily.</li> <li>• Tempo can be changed in real-time.</li> <li>• CD access is not necessary.</li> <li>• The main system only issues requests. (The sound driver controls all of the above.)</li> </ul>
Disadvantages	The wave capacity is limited to 4 Mb (512 KB) at any one time.

#### PCM Stream Play

Sound generation method	Loop-plays wave data. After sounds in the wave data are generated, the next wave data is transferred. This procedure is repeated to allow continuous play of long wave data.
Characteristics	Allows long, prerecorded wave data to be played without modification. However, the wave data must be transferred continuously to ensure uninterrupted sound generation.
Advantages	<ul style="list-style-type: none"> <li>• Wave data longer than 4 Mb can be played.</li> <li>• Various play patterns can be edited by modifying the transfer. (Edit synthesis is possible.)</li> <li>• The data width can be set to 8 or 16 bits. The frequency can also be changed freely. (Memory efficiency is good. The pitch can also be changed during record and play operations.)</li> <li>• Several channels can be played concurrently.</li> </ul>
Disadvantages	The wave data transfer adds a large overhead on the main system and the sound CPU.

**CD-DA Play**

Sound generation method	Uses hardware to play audio sounds recorded on a CD audio track. This is the same method used to play commercial music CDs.
Characteristics	Plays long, prerecorded wave data without modification. The audio sounds are automatically output by hardware.
Advantages	<ul style="list-style-type: none"><li>• Realistic and highly expressive music can be played because sounds can be recorded live.</li><li>• No overhead is placed on either the main system or the sound CPU.</li></ul>
Disadvantages	<ul style="list-style-type: none"><li>• The music cannot be modified or corrected easily, and must instead be re-recorded.</li><li>• The data width and frequency are fixed at 16 bits and 44.1 KHz respectively. A large amount of wave data is required.</li><li>• Monopolizes the CD drive through constant CD access. This method monopolizes the CD drive.</li><li>• The tempo cannot be changed in real-time.</li></ul>

## 3.2 Sequence Play

### 3.2.1 Map Change

A game contains many scenes, and normally a different area map is used for each scene. Therefore, always specify an area map before playing a sequence. Select the desired area map from the available sound area maps, and execute map change. After map change ends, transfer the sound data according to the map. This completes the preparation for sequence play.

Execute map change at least once after activating the sound driver, and then each time the area map changes. Follow steps 1 through 6 below to execute a map change.

#### Map Change

Sequence	Operation	Description
1	Stop all sounds.	Execute the <b>Sound Initial (10h)</b> command. Use parameters to specify stop and initialization of all sounds.
2	Change the area map.	Execute the <b>Map Change (08h)</b> command. Use a parameter to specify the number of the area map to be changed.
3	Transfer the sound data.	Transfer the sound data to the area starting from SH address 25A0B000. If the sound data is divided into several files, transfer the data in each file according to the area map.
4	Report transfer completion.	For all sound data that was transferred, change the transfer completed bits in the sound area map CRNT work area (from SH address 25A00500h) to 1.
5	Set the DSP.	If necessary, execute the <b>Effect Change (83h)</b> command.
6	Set the mixer.	Execute the <b>Mixer Change (87h)</b> command. Set the mixer even if the DSP is not being used.

### 3.2.2 Sequence Play

After map change is completed, the sequence can be played at any time. Execute the sound control commands (such as sequence start and sequence stop) to play a sequence. See Chapter 4, *Commands and Status Information* for details on issuing commands.

**Note 1:** The system may run uncontrolled if the area map is changed while sounds are being produced or if the sound data being used is replaced. Even if sequence play is stopped and sound is not being produced, sound processing may occur internally. Be sure to execute step 1 above before changing the area map or the sound data.

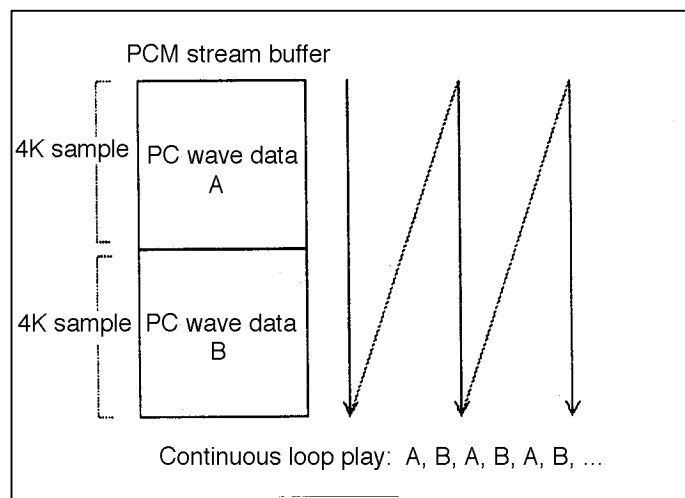
**Note 2:** The DSP work RAM is used in DSP internal processing. When the DPS is operating, it constantly rewrites the contents of this area. Caution is necessary for area map changes that change the size and address of the DSP work RAM. If the area map is changed during DSP operation, transferred sound data in the original DSP work RAM is destroyed.

### 3.3 PCM Stream Play

#### 3.3.1 How Stream Play Works

To play data, the PCM stream play method plays long PCM wave data through loop play. In loop play, after one segment of the PCM wave data is played, the next wave data segment is written to the sound memory. This process is repeated until the entire wave data is played. This method can be used to play long PCM wave data, such as speech, long special effects, and recorded background music that normally will not fit into sound memory.

After the current wave data segment is played, the system writes the next wave data segment. The following figure shows that after wave data A is played, wave data B begins to play while the system writes the next wave data to data A. By repeating this process, the system can play wave data of any length without interruption.



#### 3.3.2 PCM Stream Play Procedure

Execute PCM stream play according to the procedure described below. For details on issuing the sound control commands, see Chapter 4, *Commands and Status Information*. Execute the following procedure for both the left and right channels.

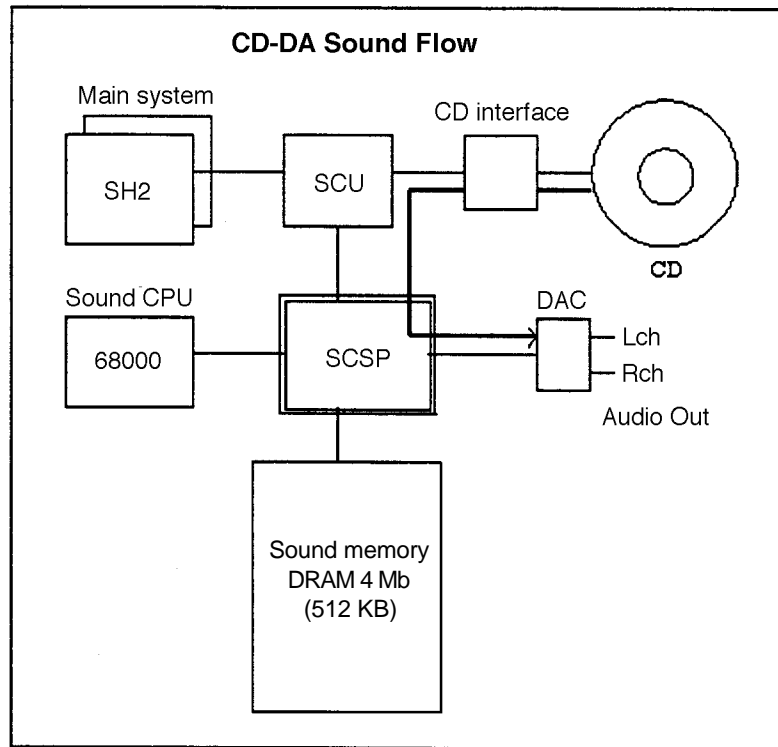
Sequence	Operation
1	Transfer the first wave data to the wave data A and B areas.
2	Execute the <b>PCM start (85h)</b> command of the sound control commands.
3	After data A replay ends, transfer the next wave data to the wave data A area.
4	After data B replay ends, transfer the next wave data to the wave data B area.
5	Repeat the operations for 3 and 4 above.
6	Execute the sound control command <b>PCM stop (86h)</b> to end play

**Note:** The buffer size, the transfer method, and the transfer timing are determined independently by the main system. Select the optimum procedure for the scene without being bound to the above procedure.

### 3.4 CD-DA Play

#### 3.4.1 How CD-DA Play Works

The CD-DA play method uses hardware to output audio data recorded on a CD audio track. The recorded data is played unchanged according to the same principle used in data recorders. Audio data read from the CD is transferred automatically to the SCSP. Audio output can then be started by setting the CD-DA level on the SCSP.



#### 3.4.2 CD-DA Play

To execute CD-DA play, read the audio data from the CD drive and set audio output from the SCSP.

Call the following two functions from the CDC system library to read audio data. For details on the CDC library, see the *CDC Library User's Manual*.

**[CDC library functions]**

1. CDC\_CdInit: Initializes the CD drive.
2. CDC-CdPlay: Plays the CD.

To set audio output from the SCSP, execute sound control commands **CD-DA Level (80h)** and **CD-DA Pan (81h)**. See Chapter 4, *Commands and Status Information* for details on issuing these commands.

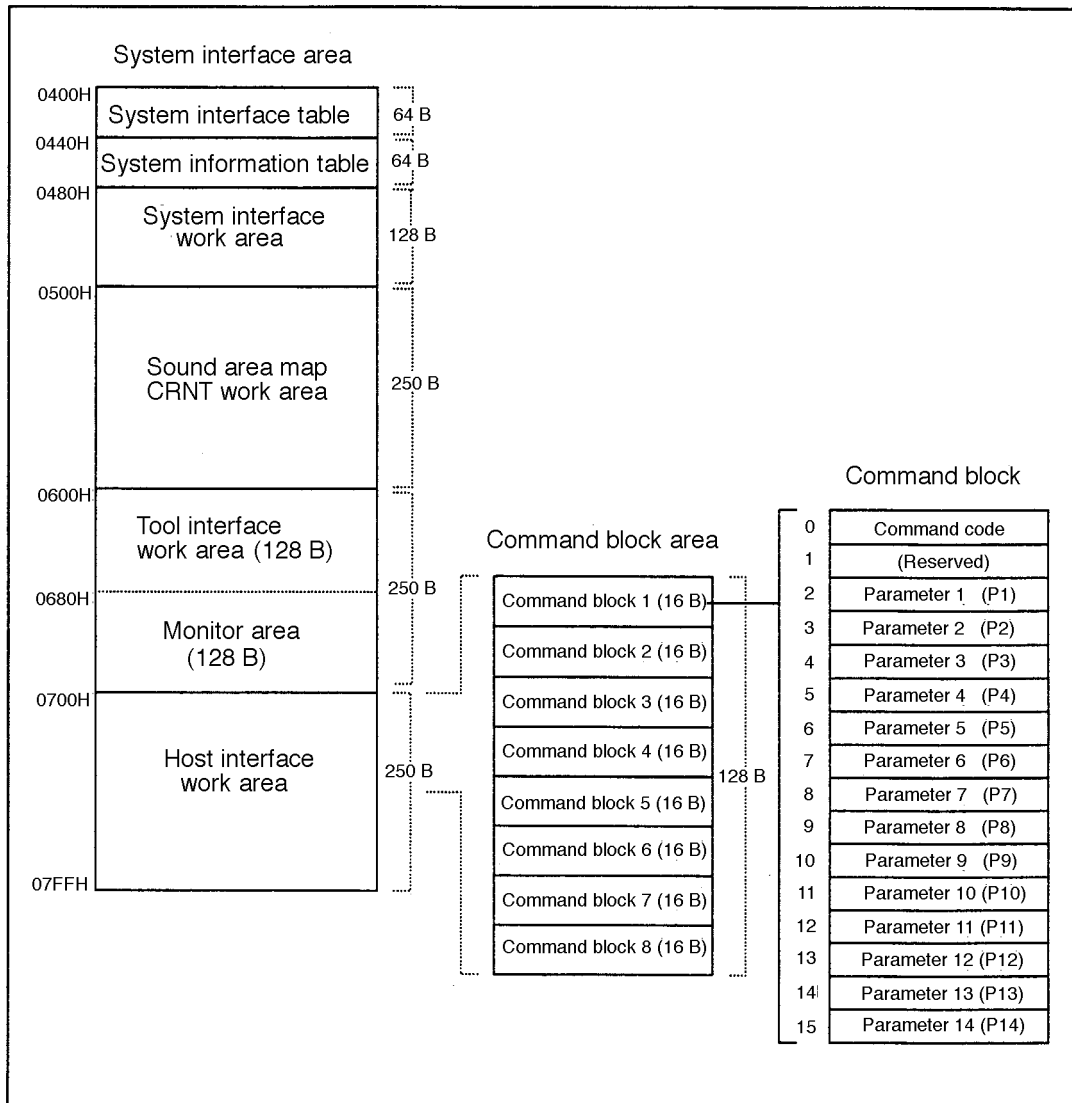


## Chapter 4 Commands and Status Information

### 4.1 Command Area

Control commands are issued for the command block area shown below. Write the command code and the necessary parameters (P1 to P14) to a 16-byte command block. The command block area contains eight command blocks, which can be used to execute up to eight commands concurrently.

For information on the write procedure, see section 6.1, *Executing Commands*.



## 4.2 Details on Command

The sound control commands are categorized into four types according to the type of sound to be played. The related play type for each command is described in parentheses (SPC) in the command name column.

- S: A sequence play command or a command related to sequence play  
P: A PCM stream play command or a command related to PCM stream play  
C: A CD-DA play command or a command related to CD-DA play  
S/P/C: Command related to two or all of the above, other command

For details on each parameter, refer to section 4.2.1 beginning on page 19 while reading this table.

Command Name	Command	Parameters
<b>Sequence Start</b> (S-)	01h	[Sequence play start] P1: Sound control number (0-7) P2: Sequence data bank number (0-15) P3: Song number in sequence data (0-127) P4: Priority level (0-31)
<b>Sequence Stop</b> (S-)	02h	[Sequence play stop] P1: Sound control number (0-7)
<b>Sequence Pause</b> (S-)	03h	[Sequence play pause] P1: Sound control number (0-7)
<b>Sequence Continue</b> (S-)	04h	[Sequence play continue] P1: Sound control number (0-7)
<b>Sequence Volume</b> (S-)	05h	[Sequence volume and fade settings] P1: Sound control number (0-7) P2: Sequence volume (0-127) P3: Fade rate (0-255)
<b>Tempo Change</b> (S-)	07h	[Sequence play tempo change] P1: Sound control number (0-7) P2: Dummy (not used) P3-P4: Tempo value (+32767 --> -32768)
<b>Map Change</b> (S-)	08h	[Area map switching] P1: Area map number (0-255)
<b>MIDI Direct Control</b> (S-)	09h	[MIDI direct output] P1: MIDI command word (00h-FFh) P2: MIDI channel word (00h-FFh) P3: MIDI data 1 (00h-7Fh) P4: MIDI data 2 (00h-7Fh) • For details, see "MIDI Direct Output" in Chapter 6, <i>Supplemental Information</i> .

Command Name	Command	Parameters
<b>Volume Analyze Start</b> (-C)	0Ah	[Input level analysis start for digital audio in] No parameters
<b>Volume Analyze Stop</b> (-C)	0Bh	[Input level analysis stop for digital audio in] No parameters
<b>DSP Stop</b> (SPC)	0Ch	[DSP initialization] No parameters <ul style="list-style-type: none"> <li>This command is supported for compatibility with the host and will be deleted in the future. Use the <b>Sound Initial (10h)</b> command instead.</li> </ul>
<b>Sound All Off</b> (SPC)	0Dh	[All sound stop] No parameters <ul style="list-style-type: none"> <li>This command is supported for compatibility with the host and will be deleted in the future. Use the <b>Sound Initial (10h)</b> command instead.</li> </ul>
<b>Sequence Pan</b> (S-)	0Eh	[Sequence Pan setting] P1: Sound control number (0-7) P2: Bit 7 0: Pan control OFF 1: Pan control ON Bits 6-0 MIDI Pan data (00h-7Fh) <ul style="list-style-type: none"> <li>For data values, see "Details on MIDI Pan Data."</li> </ul>
Reserved	0Fh	-
<b>Sound Initial</b> (SPC)	10h	[Sound initialization] P1: Stop all sequence play (0-1) P2: Stop all PCM stream play (0-1) P3: Stop CD-DA play (0-1) P4: Initialize DSP (0-1) P5: Initialize mixer (0-1) <ul style="list-style-type: none"> <li>From P1 to P5, all execution instructions are per 01h. with 00h, no action is taken.</li> </ul>
<b>3D Control</b> (SPC)	11h	[3D sound control] P1: Channel number (0-1) P2: Distance (0-127) P3: Horizontal position (0-127) P4: Vertical position (0-127)
<b>Qsound Control</b> (SPC)	12h	[Qsound control] P1: Channel number (0-7) P2: Pan position (0-30)

Command Name	Command	Parameters
<b>CD-DA Level</b> (-C)	80h	[CD-DA output level setting] P1: CD-DA level left, 8 levels (00h-E0h) P2: CD-DA level right, 8 levels (00h-E0h) Level values: 00h (off), 20h, 40h, 60h, 80h, A0h, C0h, E0h (max)
<b>CD-DA Pan</b> (-C)	81h	[CD-DA output Pan setting] P1: CD-DA Pan left, 32 levels (0-31) P2: CD-DA Pan right, 32 levels (0-31)
<b>Total Volume</b> (SPC)	82h	[Total volume setting] P1: Total volume, 16 levels (0-15)
<b>Effect Change</b> (SPC)	83h	[Total effect switching] P1: Effect bank number (0-15)
<b>PCM Start</b> (-P)	85h	[PCM stream play start] P1: Bit 7 0: Mono 1: Stereo Bits 6-5 Not used Bit 4 0: 16-bit PCM 1: 8-bit PCM Bits 3-0 Play-start PCM stream play number (0-7) P2: Bits 7-5 Direct sound output level, 8 levels (0-7) Bits 4-0 Direct sound output Pan, 32 levels (0-31) P3-P4 PCM stream buffer start address (0000h-FFFFh) P5-P6 PCM stream buffer size (0000h-FFFFh) P7-P8 Pitch word (0000h-7FFFh) P9: Bits 7-3 Effect input channel (0-15) [P9=Rch] Bits 2-0 Effect input level, 8 levels (0-7) P10: Bits 7-3 Effect input channel (0-15) [P10=Lch] Bits 2-0 Effect input level, 8 levels (0-7)
<b>PCM Stop</b> (-P)	86h	[PCM stream play stop] P1: Play-stop PCM stream play number (0-7)
<b>Mixer Change</b> (SPC)	87h	[Mixer switching] P1: Tone data bank number (0-15) P2: Mixer number in tone data (0-127)

Command Name	Command	Parameter
<b>Mixer Parameter Change</b> (SPC)	88h	[Mixer parameter change] P1: Effect output channel (0-17) P2: Bits 7-5      Effect output level, 8 levels (0-7) Bits 4-0      Effect output Pan, 32 levels (0-31)
<b>Hard Check</b> (-)	89h	[Hard check] P1: Check items    00h: DRAM 4 Mb read/write 01h: DRAM 8Mb read/write 02h: SCSP MIDI 03h: Sound source output (L/R) 04h: Sound source output (L) 05h: Sound source output (R)
<b>PCM Parameter Change</b> (-P-)	8Ah	[Parameter change for PCM stream play] P1: PCM stream play number (0-7) P2: Bits 7-5      Direct sound output level, 8 levels (0-7) Bits 4-0      Direct sound output Pan, 32 levels (0-31) P3-P4: Pitch word (0000h-FFFFh) P5: Bits 7-3      Effect input channel (0-15) [P5=Rch] Bits 2-0      Effect input level, 8 levels (0-7) P6: Bits 7-3      Effect input channel (0-15) [P6=Lch] Bits 2-0      Effect input level, 8 levels (0-7)

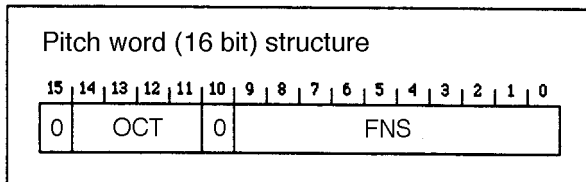
#### 4.2.1 Parameters

- **Sound Control Number (0-7)**  
Specifies a sound control number from 0 to 7 at sequence start. This parameter is used because sequence play allows up to eight concurrent sequence plays. The specified sound control number is used later to stop, pause, or restart sequence play.
- **Sequence Data Bank Number (0-15)**  
Specifies the position of the sequence bank (sequence data storage area) in the current area map if the area map contains more than one sequence bank. Since an area map can hold up to 16 sequence banks, 0 to 15 values are allowed. If the area map contains only one sequence bank, the parameter value will always be 0.
- **Song Number in Sequence Data (0-127)**  
Specifies the sequence number of the song data within the sequence data area. This parameter is used because one sequence data area can store data for multiple songs. Since one sequence data area can store up to 128 songs, 0-127 values are allowed. If the sequence data contains only one song, the parameter value will always be 0.
- **Priority Level (0-31)**  
Specifies the priority level when a sequence is played. There are 32 priority levels (0 to 31), where level 0 has the highest priority. When the 32 sound slots become full, the sounds are played according to reverse arrival priority. This priority sequence is also used to control which sounds are erased when the system plays the later arriving sounds.
- **Sequence Volume (0-127)**  
Specifies the sequence volume with a value from 0 to 127, where value 127 is the highest volume (fundamental tone). The fade rate can be set to determine how long it takes to reach the specified volume. After the fade, the final volume is kept as the current sequence volume. If fade is not necessary, set the fade rate (P3) to 0, and set only the sequence volume.
- **Fade Rate (0-255)**  
Specifies the required time for the current sequence volume to reach the specified sequence volume with a rate from 0 to 255, where 255 is the longest. The change delay becomes longer as the value increases. (If 0 is specified, fade is not performed, and only the sequence volume is set.) The fade direction is from the current sequence volume to the specified sequence volume. The fade-in and fade-out areas are determined by whether the specified sequence volume is larger or smaller than the current sequence volume.
- **Tempo Value (+32767 --> - 32768)**  
Specifies the relative tempo volume in relation to the standard tempo value (0000h). The tempo is doubled for every positive 1000h (4,096) and is halved for every negative 1000h. In other words, the interval can be controlled with a precision of 4,096 levels until the tempo is doubled (or halved). Since 0 is the standard tempo, the original tempo can be restored by specifying 0000h.

- **Area Map Number (0-255)**  
Indicates which area map from the start address within the sound area map should be specified.
- **Pan Control ON/OFF (Bit 7: 0/1)**  
Specifies whether or not sequence Pan control is to be executed. If the specified value is 1, sequence Pan control is executed. If the specified value is 0, control is not executed.
- **3D Channel Number (0-1)**  
Specifies the channel number that controls Yamaha 3D sound. A maximum of two channels can be used; specify 0 or 1.
- **3D Distance (0-127)**  
Specifies the distance from the listening point to the virtual sound source for Yamaha 3D sound. The value 0 indicates a distance of 0. The distance increases as the value becomes larger.
- **3D Horizontal Position (0-127)**  
Specifies the horizontal position of the virtual sound source for Yamaha 3D sound as viewed from the listening point. A value of 0 = straight ahead; value 32 = 90 degrees to the right; value 64 = directly behind; and value 96 = 90 degrees to the left.
- **3D Vertical Position (0-127)**  
Specifies the vertical position of the virtual sound source for Yamaha 3D sound as viewed from the listening point. A value of 0 = directly above; value 32 = directly ahead; value 64 = directly below; and value 96 = directly behind.
- **Q Channel Number (0-7)**  
Specifies the channel number that controls Q sound. Up to eight channels can be used, and values between 0 and 7 can be specified.
- **Q Pan Position (0-30)**  
Specifies the Pan position for Q sound. A value of 0 = 90 degrees to the left; value 15 = center; and value 30 = 90 degrees to the right. There are 15 levels for both left and right.
- **CD-DA Level (00h-E0h)**  
Sets the CD-DA output level. There are eight output volume levels, with 00h as the lowest (=off) and E0h as the highest.  

Off <-----> Max
00h, 20h, 40h, 60h, 80h, A0h, C0h, E0h
- **CD-DA Pan (0-31)**  
Sets the CD-DA output Pan. For more information, see section 7.2, *Details on SCSP Pan (32 Levels)*.
- **Total Volume (0-15)**  
Sets the SCSP total volumes (overall volume for all play operations). There are 16 volume levels, with 0 as the lowest (=off) and 15 as the highest.

- **Effect Bank Number (0-15)**  
Specifies the position of the DSP bank (DSP microprogram storage area) in the current area map if the area map contains more than one DSP bank. Since one area map can hold up to 16 DSP banks, 0 to 15 values are allowed. If the area map contains only one DSP bank, the parameter value will always be 0.
- **PCM Stream Play Number (0-17)**  
Specifies the PCM stream play number with a value from 0 to 7 since the PCM stream play can play up to eight streams concurrently. For stereo output, note that one play number plays two streams: Lch and Rch. The maximum total number of streams for mono and stereo is eight streams. The command is ignored if there are more than eight streams.
- **Direct Sound Output Level (0-7)**  
Sets the direct sound output level. There are eight levels with 0 as the lowest volume (=off).
- **Direct Sound Output Pan (0-31)**  
Sets the direct sound output Pan. For more information, see Section 7.2, *Details on SCSP Pan (32 Levels)*.
- **PCM Stream Buffer Start Address (0000h-FFFFh)**  
Specifies the start address of the PCM stream buffer. This parameter specifies the upper 16 bits of the 20-bit address.
- **PCM Stream Buffer Size (0000h-FFFFh)**  
Specifies the size of the PCM stream buffer. The value is the sample count for each channel. The same value is used for both mono and stereo operations.
- **Pitch Word (0000h-7FFFh)**  
Specifies the SCSP pitch register value (16 bits).



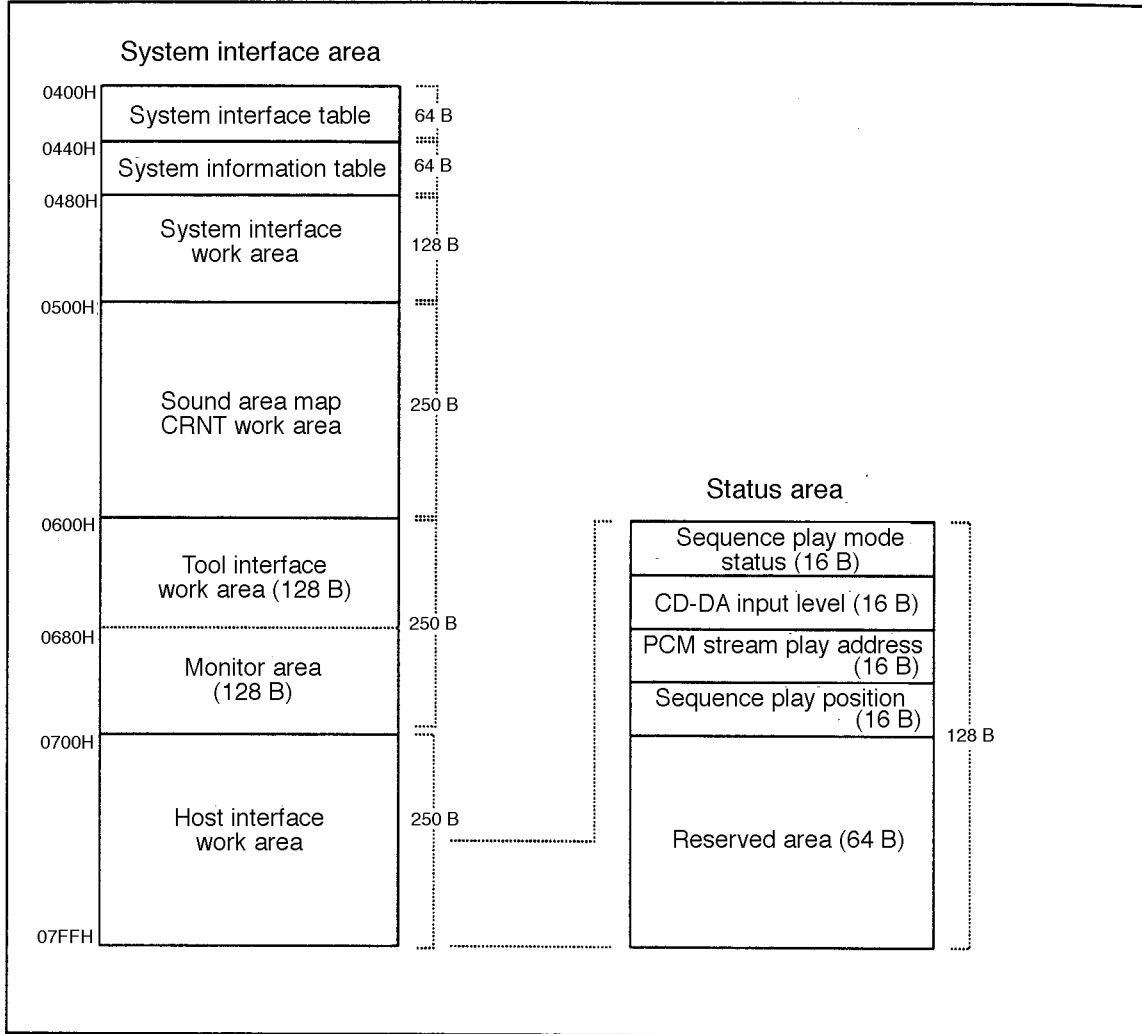
- OCT: Specifies the octave as a two's complement. The allowed values are -8 octave to +7 octave.
- FNS: Specifies the interval from the standard pitch (0) to one octave above (1023) with 1,024 levels. Bits 15 and 10 are always 0.
- Note:** For an explanation of the pitch calculation procedure, see *Calculating the Pitch* in section 6.4, *PCM Stream Play*.



- **Effect Input Channel (0-15)**  
Specifies the DSP effect in each channel. The allowed values are 0 to 15 because there are 16 DSP effect input channels. For details, see section 6.7, *DSP and Mixers*.
- **Effect Input Level, 8 levels (0-7)**  
Specifies the direct sound level to be sent to the effect input channel. The level is specified with a value from 0 to 7. There are 8 levels with 0 as the lowest level (=off).
- **Tone Data Bank Number (0-15)**  
Specifies the position of the tone bank (tone data storage area) in the current area map if the area map contains more than one tone bank. Values can be specified between 0 and 15 since one area map can hold up to 16 tone banks. If the area map only has one tone bank, the parameter value will always be 0.
- **Mixer Number in Tone Data (0-127)**  
Specifies the sequence number of the mixer data within one tone data area. This parameter is used because one tone data can store data for multiple mixers. Values can be specified between 0 and 127 since one tone data area can store up to 128 mixers. If the tone data contains only one mixer, the parameter value will always be 0.
- **Effect Output Channel (0-15, 16-17)**  
Specifies the DSP effect out channel. Since there are 16 DSP effect output channels, 0 to 15 values are allowed. When CD-DA is selected, either 16 (Lch) or 17 (Rch) is specified. For details, see section 6.7, *DSP and Mixers*.
- **Effect Output Level (0-7)**  
Specifies the effect level to be output from the effect output channel. The level is specified with a value from 0 to 7. There are eight levels with 0 as the lowest level (=off)
- **Effect Output Pan (0-31)**  
Specifies the effect Pan that is output from the effect output channel. The Pan is specified with a value from 0 to 31. For details, see section 7.2, *Details on SCSP Pan (32 Levels)*.

### 4.3 Status Area

When a sound control command is executed, the current execution state and all pertinent information are sent to the main system as the return status. The return status is stored in the status area of the system area and can be referenced when necessary.



#### 4.3.1 Sequence Play Mode and Status

+0	Sequence play mode (0-4)
+1	Sequence play status (0-FFh)

During sequence play, the current status is constantly written to the mode and status areas. To find out the current sequence play status, reference these areas.

The mode status area corresponds to sound control numbers 0 to 7 and contains a total of 16 bytes for 8 sequences of 2 bytes each. For details on the area addresses, see the section that describes the host interface work RAM found in the system area.

##### [Sequence play mode]

00h: Initial status  
 01h: Playing  
 02h: Fading  
 03h: Pause during play  
 04h: Pause during fade

##### [Sequence play status]

00h: Normal  
 80h: Outside resolution range  
 81h: No tempo data  
 82h: No event data  
 83h: Outside tempo data range  
 FFh: Data error during data decompression

#### 4.3.2 CD-DA Input Level

The CD-DA input level area stores the signal level of the digital audio input signal. When the **Volume Analyze Start (0Ah)** control command for Digital Audio In is executed, the system begins searching for the input signal and sets this area. The values are from 0 to 7FFFh, with 0 indicating the absence of an input signal and 7FFFh indicating the maximum input signal level. Because the input signal level is a momentary value, it changes as the input signal level changes.

Normally, only the total level of Lch and Rch is stored. However, a special DSP program (**DSP-3 Band Ana.EXB**) can be used to detect the input signal at three levels: high-tone band, mid-tone band, and low-tone band. This program only allows detection of the input signal level. Even if the output volume is exceeded by the **CD-DA Level (80h)** control command, the signal set in this area does not change.

**Note:** Level detection places a load on the sound driver. If level detection is not necessary, execute the **Volume Analyze Stop (0Bh)** control command for Digital Audio In to stop the process.

#### 4.3.3 PCM Stream Play Address

+0	PCM stream play position (1-15)
+1	Unused

The PCM stream play address area stores the position of the data currently being played. This position is the sample count from the beginning of the PCM stream play buffer and is indicated with a value from 0 to 15. The system can monitor 4K sample units (4 Kb for 8-bit play, 8 Kb for 16-bit play). If the value increments by 1, the play position advances by 4K samples. In stereo play, Rch and Lch have the same play address.

The PCM stream play position area corresponds to stream play numbers 0 to 7 and contains a total of 16 bytes for 8 streams of 2 bytes each. For details on the area addresses, see the section that describes the host interface work RAM found in the system area.

#### 4.3.4 Sequence Play Position

+0	Sequence play position (H)
+1	Sequence play position (L)

The sequence play position area stores the sequence play position during sequence play. Values from 0 to FFFh are stored sequentially in this area. The current value indicates the elapsed time since play was started and provides an approximation of the current play position. The value increments by 1 every 100 ms.

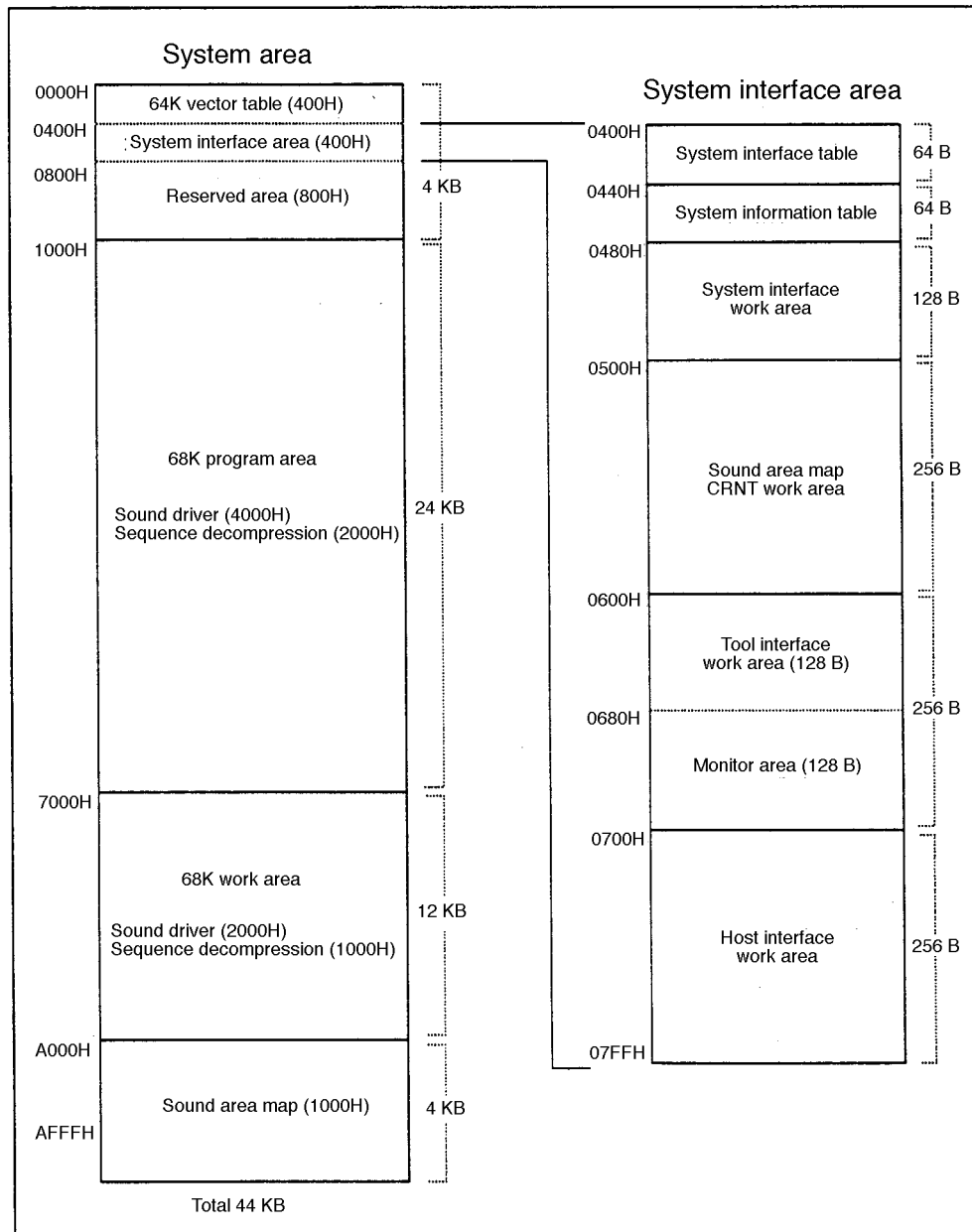
The sequence play position area corresponds to sound control numbers 0 to 7 and contains a total of 16 bytes for 8 sequences of 2 bytes each. For details on the area addresses, see the section that describes the host interface work RAM found in the system area.

**Note:** The sequence play position represents the absolute time that has elapsed. If the tempo is changed, the same value will indicate a different position. For accurate synchronization of the sequence being played, store synchronization messages in the sequence data.

## Chapter 5 System Area

### 5.1 Details on Interface Area

The system area is a fixed area used to operate the sound driver provided by SEGA. As shown in the following figure, the system area consists of five areas. The system interface area controls the entire system. (Refer to the next section for more information on the system interface area.)



5.1.1 System Interface Table (400H-43FH: 64B)

Address	Size	Data Contents	Description
0400h	4 B	Pointer to system information table	Start address of system information table (440h)
0404h	4 B	Pointer to host interface work area	Start address of host interface work area (700h)
0408h	4 B	Pointer to sound area map CRNT work area	Start address of sound area map CRNT work area (500h)
040Ch	4 B	Pointer to tool interface work area	Start address of tool interface work area (600h)
0410h	1 B	DSP program load flag	Bit 7 = 0: No data 1: DSP program load completed
0411h	1 B	Reserved area	-
0412h	4 B	Pointer to system interface work area	Start address of system interface work area (480h)
0416h	2 B	Error status	Area for storing error status of sound driver
0418h	2 B	Hardware check status	Area for storing return status of hardware check
041Ah	2 B	Command history offset address	Write pointer for sound control command history (800h-FFFh)
041Ch	4 B	Reserved area	
0420h	4 B	Error status bit map	Bit map for sound driver error status
0424h	4 B	Pointer to system interface work area	Start address of system interface work address (412h copy: for SH)
0428h	24 B	Reserved area	-

5.1.2 System Information Table (440H-47FH: 64B)

Offset	Size	Data Contents	Description
+00h	4 B	68K program start address	Start address of 68K program area (1000h)
+04h	4 B	68K program size	Size of 68K program area (6000h)
+08h	4 B	Sound area map start address	Start address of sound area map area (A000h)
+0Ch	4 B	Sound area map size	Size of sound area map area (1000h)
+10h	4 B	68 K work area start address	Start address of 68K work area (7000h)
+14h	4 B	68K work area size	Size of 68K work area (3000h)
+18h	40 B	Reserved area	-

5.1.3 System Interface Work Area (480H-4FFH: 128B)

Offset	Size	Data Contents	Description
+00h	1 B	Interrupt control word (Host ---> Sound)	Bit 7: IPT enable 0: Disables PCM play address update interrupt 1: Enables PCM play address update interrupt
+01h	1 B	Interrupt type (1-127) (Sound ---> Host)	1: PCM play address update interrupt 2-127: Not used
+02h	1 B	Corresponding stream play number (Sound ---> Host)	Corresponding stream play number when PCM play address update interrupt is executed • Bits 0-7 correspond to PCM play numbers 0-7 (bit=1: Updated)
+03h	1 B	Sound mode control word (Host ---> Sound)	Bit 7: Sound mode 0: Stereo output 1: Mono output
+04h	1 B	3D sound control word (Host ---> Sound)	Bit 7: 3D 1-ch mode 0: Not used 1: Used Bit 6: 3D 2-ch mode 0: Not used 1: Used Bit 5: Q sound 4-ch mode 0: Not used 1: Used Bit 4: Q sound 8-ch mode 0: Not used 1: Used
+05h	1 B	Q sound COEF pointer	Q sound COEF pointer
+06h	1 B	3D sound COEF pointer	3D sound COEF pointer
+07h	1 B	3D sound MADRS pointer	3D sound MADRS pointer
+08h	32 B	3D sound parameter	3D sound parameter storage area
+28h	48 B	3D sound program work	3D sound program work area (30h)
+58h	8 B	Reserved area	-
+60h	1 B	Command sequence timing flag (Host ---> Sound)	Bit 7: Timing flag 0: Idle 1: Command set completed
+61h	1 B	Command sequence control word (Host ---> Sound)	Bit 7: Command mode 0: Handshake off (V1.00-1.29) 1: Handshake on (V1.30- )
+62h	14 B	Reserved area	-
+70h	16 B	Titan work area	Reserved area for Titan compatibility (not used by SATURN)

5.1.4 Sound Area MAP CRNT Work Area (500H-5FFH: 256B)

Offset	Size	Data Contents	Description
+00h	8 B	Map information 1	<p>Map information</p> <p>8 bytes</p> <p>           E 0: Data present            1: No data            Data ID 0: Tone bank data            1: Sequence data            2: DSP program            3: DSP work RAM            ID number 0-15: Identification number            L 0: Data not transferred            1: Transfer completed         </p>
+F8h	8 B	Map information 32	

5.1.5 Tool Interface Work Area (600H-6FFH: 256B)

Offset	Size	Data Contents	Description
+00h	4 B	Reserved area	-
+04h	4 B	Wave edit start address	Start address of RAM area for wave editing
+08h	4 B	Wave edit total size	Total size of RAM area for wave editing
+0Ch	2 B	Reserved area	-
+0Eh	4 B	Tone editor start address	Start address of RAM editor for tone editor
+12h	4 B	Tone editor total size	Total size of RAM area for tone editor
+16h	2 B	Reserved area	-
+18h	4 B	TrgtMem_DSPprogAddress	Dedicated DSP linker area
+1Ch	4 B	TrgtMem_DSPprogSize	Dedicated DSP linker area
+20h	32 B	TrgtMem_Filename	Dedicated DSP linker area
+40h	4 B	TrgtMem_DSPRAMSize	Dedicated DSP linker area
+44h	2 B	TrgtMem_RBL	Dedicated DSP linker area
+46h	4 B	TrgtMem_ModElementAddress	Dedicated DSP linker area
+4Ah	4 B	TrgtMem_ModElementSize	Dedicated DSP linker area
+4Eh	1 B	TrgtMem_NumberOfElements	Dedicated DSP linker area
+4Fh	49 B	Reserved area	-
+80h	4 B	Voice 1 monitor	Voice # / Note / Velocity
:	:	:	:
+FCh	4 B	Voice 32 monitor	Voice # / Note / Velocity



**5.1.6 Host Interface Work Area (700H-7FFH: 256B)**

- Command block area (700h-77Fh: Main system ---> sound driver)

Offset	Size	Data Contents	Description
+00h	16 B	Command block 1	Command block 1 from host to sound driver
+10h	16 B	Command block 2	Command block 2 from host to sound driver
+20h	16 B	Command block 3	Command block 3 from host to sound driver
+30h	16 B	Command block 4	Command block 4 from host to sound driver
+40h	16 B	Command block 5	Command block 5 from host to sound driver
+50h	16 B	Command block 6	Command block 6 from host to sound driver
+60h	16 B	Command block 7	Command block 7 from host to sound driver
+70h	16 B	Command block 8	Command block 8 from host to sound driver

- Status area (780h-7FFh: Sound driver ---> main system)

Offset	Size	Data Contents	Description
+80h	2 B	Sequence play 0 mode/status	Sequence play mode/status for sound control no. 0
+82h	2 B	Sequence play 1 mode/status	Sequence play mode/status for sound control no. 1
+84h	2 B	Sequence play 2 mode/status	Sequence play mode/status for sound control no. 2
+86h	2 B	Sequence play 3 mode/status	Sequence play mode/status for sound control no. 3
+88h	2 B	Sequence play 4 mode/status	Sequence play mode/status for sound control no. 4
+8Ah	2 B	Sequence play 5 mode/status	Sequence play mode/status for sound control no. 5
+8Ch	2 B	Sequence play 6 mode/status	Sequence play mode/status for sound control no. 6
+8Eh	2 B	Sequence play 7 mode/status	Sequence play mode/status for sound control no. 7
+90h	2 B	Input level Lch	Digital audio input level Lch (0000h-7FFFh)
+92h	2 B	Input level Rch	Digital audio input level Rch (0000h-7FFFh)
+94h	2 B	H-vol L	High-range input level Lch (0000h-7FFFh)
+96h	2 B	H-vol R	High-range input level Rch (0000h-7FFFh)
+98h	2 B	M-vol L	Middle-range input level Lch (0000h-7FFFh)
+9Ah	2 B	M-vol R	Middle-range input level Rch (0000h-7FFFh)
+9Ch	2 B	L-vol L	Low-range input level Lch (0000h-7FFFh)
+9Eh	2 B	L-vol R	Low-range input level Rch (0000h-7FFFh)
+A0h	2 B	PCM stream play 0 address	PCM stream play position (0-15) of stream play no. 0
+A2h	2 B	PCM stream play 1 address	PCM stream play position (0-15) of stream play no. 1
+A4h	2 B	PCM stream play 2 address	PCM stream play position (0-15) of stream play no. 2
+A6h	2 B	PCM stream play 3 address	PCM stream play position (0-15) of stream play no. 3
+A8h	2 B	PCM stream play 4 address	PCM stream play position (0-15) of stream play no. 4
+AAh	2 B	PCM stream play 5 address	PCM stream play position (0-15) of stream play no. 5
+ACh	2 B	PCM stream play 6 address	PCM stream play position (0-15) of stream play no. 6
+AEh	2 B	PCM stream play 7 address	PCM stream play position (0-15) of stream play no. 7
+B0h	2 B	Sequence play 0 play position	Sequence play position (0000h-FFFFh) of sound control no. 0
+B2h	2 B	Sequence play 1 play position	Sequence play position (0000h-FFFFh) of sound control no. 1
+B4h	2 B	Sequence play 2 play position	Sequence play position (0000h-FFFFh) of sound control no. 2
+B6h	2 B	Sequence play 3 play position	Sequence play position (0000h-FFFFh) of sound control no. 3
+B8h	2 B	Sequence play 4 play position	Sequence play position (0000h-FFFFh) of sound control no. 4
+BAh	2 B	Sequence play 5 play position	Sequence play position (0000h-FFFFh) of sound control no. 5
+BCh	2 B	Sequence play 6 play position	Sequence play position (0000h-FFFFh) of sound control no. 6
+BEh	2 B	Sequence play 7 play position	Sequence play position (0000h-FFFFh) of sound control no. 7
+C0h	64 B	Reserved area	-

## Chapter 6 Supplemental Information

### 6.1 Executing Commands

Commands can be executed by using two methods: timing flag handshake (for Sound Driver version 1.30 and later), and command code handshake (for Sound Driver versions 1.00 to 1.29.)

Although either method can be selected, the default is command code handshake set for upward compatibility. When timing flag handshake is selected, set the "command sequence control word (4E1h)" in the system interface work area to the timing flag handshake side. For setting instructions, see section 5.1.3, *System Interface Work Area*.

#### 6.1.1 Timing Flag Handshake

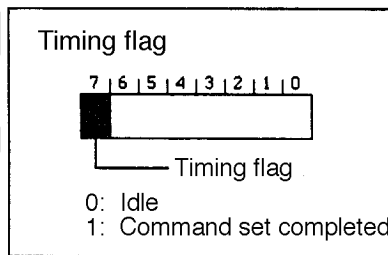
This method uses a timing flag to ensure that commands written by the main system are executed in the correct sequence. To execute commands by using a timing flag, follow the procedure below. The "timing flag (4E0h)" area is located in the system interface work area. For details, see section 5.1.3, *System Interface Work Area*.

##### Main System Processing (Issuing Commands)

1. If the timing flag is 0, the main system writes the commands to be executed into the command blocks. (Up to eight commands can be written. Spaces are permitted between the command blocks.)
2. The main system changes the timing flag to 1.

##### Sound Driver Processing (Accepting and Executing Commands)

1. If the timing flag is 1, the sound driver starts command execution sequentially from command block 1. (The command blocks are executed in sequence from block 1 to block 8. If the command code for a block is 00h, the sound driver skips that block and proceeds to the next block.)
2. After execution of a command ends, the sound driver returns the command code in a command block to 00h.
3. After processing command blocks 1 to 8, the sound driver returns the timing flag to 0.



### 6.1.2 Command Code Handshake

This method executes commands by checking whether the command code is 00h. In this method, the main system checks that the command code is 00h and then executes commands while preserving the block 1 to block 8 sequence. Similarly, the sound driver checks the sequence from block 1 to block 8 and, if a command code is written, the commands are processed again.

When command code processing ends, the sound driver clears the command code to 00h. If the command code has not been cleared to 00h, it means that the sound driver is still processing commands or is waiting to start processing. Conversely, if the command code has been cleared to 00h, the sound driver has completed all command processing.

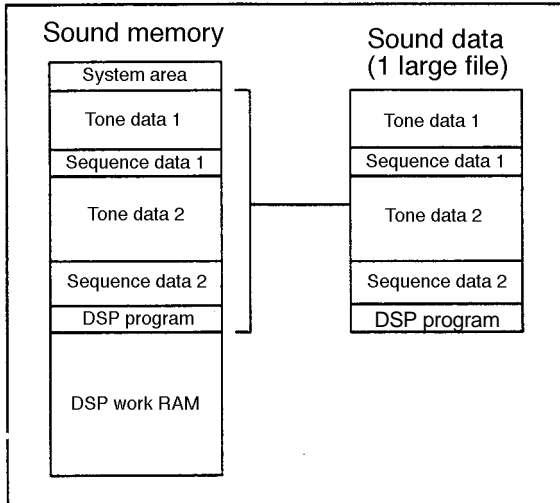
#### Notes on Using the Command Code Handshake Method

- When issuing a command, first write all command parameters, and then write the command code. When the sound driver detects a command code, it immediately retrieves the command parameters. If the command code is written first, the code may be processed with unexpected command parameters.
- The sound driver checks the command blocks in sequence from block 1 to block 8. If a command is set to any empty block that is found, the commands may not be executed in the same sequence written by the main system. If the execution sequence is switched, a system malfunction may occur. Therefore, to execute the commands in the desired intended execution sequence, set the commands so that the execution sequence will not switch.
- In this procedure, the main system and the sound driver operate asynchronously, and the command sequence may be switched even when the above notes are followed. Use only one command block to prevent sequence switching in situations when the execution sequence must be maintained, then check to ensure that the previous command was processed before issuing the next command.

## 6.2 Handling Sound Data

The sound data for all 512 KB of sound memory can either be stored in one file or in separate files. See below for the advantages and disadvantages of each method. Select the best method for the particular game or scene.

- Storing all sound data into one file



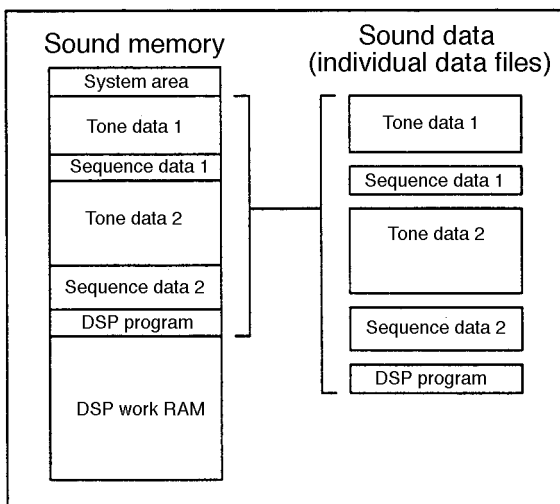
### Advantages:

The sound data need to be transferred only once, and data management is easy. SEGA recommends this method for most situations.

### Disadvantages:

The file size is large, and the data volume becomes large. Although some space may be wasted, this should not be a problem considering the CD capacity.

- Storing sound data into separate files



### Advantages:

Data can be stored without wasting space.

### Disadvantages:

There are many files to manage. The large number of files may lead to problems such as transfer errors, incorrect versions, and incorrect mapping.

For the reasons described above, SEGA recommends that the sound data normally be stored in one large file. If data replacement is necessary, store the replacement data in a separate data file.

## 6.3 CD-DA Play

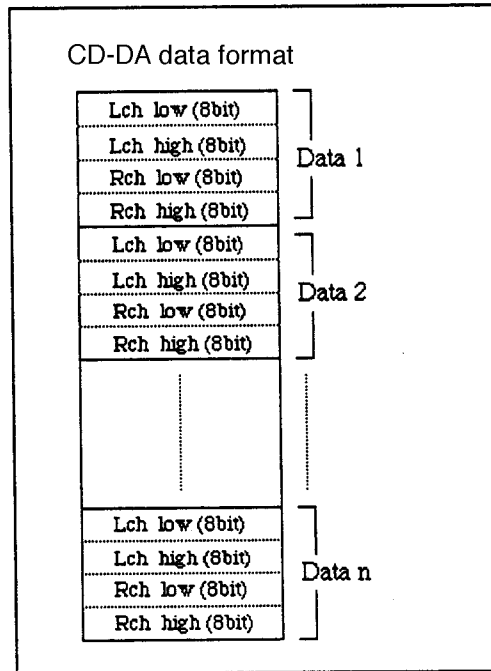
### 6.3.1 Characteristics of CD-DA Play

Audio data is transferred from the CD interface directly to the SCSP and output as audio sound. Therefore, the main system only has to execute a play request to the CD interface and does not need to read or transfer data to the sound memory. CD-DA play places the smallest load on the main system and the sound system, but monopolizes the CD drive while sounds are being played. In other words, the CD drive can only be accessed to output sounds. In situations that allow CD drive monopolization, this method produces the best quality sound available in this sound system.

The frequency and data width are fixed at 44.1 KHz and 16 bits, which are the same as the CD standards. Also, the pitch and tempo cannot be changed during play. Furthermore, implementing this high quality sound requires an extremely large amount of audio data. Just as in normal music CDs, one CD will only hold from 10 to 20 melodies. Because the CD will also contain the game program and the image data, the CD actually will hold only about five to six melodies.

### 6.3.2 Creating CD-DA Data

CD-DA data is the recorded PCM audio wave data, and is saved as a data file. Although the amount of data is large, CD-DA is the simplest audio data to use because manipulation and detailed processing are not required. CD-DA data can be created with the sound tool wave editor provided by SEGA, or any other commercial wave editing tool, as long as it outputs 16-bit stereo PCM data at 44.1 KHz. Melodies and effects can be digitally recorded, or analog audio sounds can be converted to digital. For mono play, the same data is recorded to both Lch and Rch.



**Note 1:** Some tools output PCM wave data in which the 16-bit low/high relationship is reverse. Generally, Intel family CPUs feature a low/high format, while the Motorola family CPUs feature a high/low format. If the CPU outputs data in the Motorola format, convert the data to the Intel format. If necessary, prepare a tool that facilitates inversion of the low/high format.

**Note 2:** Mixer settings are necessary to use a DSP (for effects) with the CD-DA. When using a DSP, prepare the necessary mixer data, and execute the **Mixer Change (87h)** command. Even if there is no sequence play, tone data is necessary for the mixer settings.

## 6.4 PCM Stream Play

### 6.4.1 Handling the Data Area

Execution of PCM stream play requires a data area (PCM stream buffer) for playing the PCM wave data. The PCM stream buffer can be allocated to any location in the sound memory. Normally, an area that is not being used by tone or melody data is used. However, a tone bank that is not used during PCM stream play can also be used.

The current play position can be known within a 4K sample unit. This means that PCM stream play can be executed as long as at least two 4K sample buffers are prepared. The total size of the PCM stream buffers can be set to hold anywhere up to 64K samples per channel.

### 6.4.2 Buffer Write Timing

When PCM stream play starts, the current wave play position is written to the host interface work area. The play position is a number from 0 to 15. The value is 0 at the start of play, and increments by 1 each time play advances by 4K samples. When play reaches the end of the buffer, the play position value returns to 0 and play is repeated from the beginning of the buffer.

Buffer Allocation Examples	Play Position Values
If two 4K sample buffers are allocated	0, 1, 0, 1, 0, 1, ...
If three 4K sample buffers are allocated	0, 1, 2, 0, 1, 2, 0, 1, 2, ...
If sixteen 4K sample buffers are assigned	0, 1, 2, 3, ..., 12, 13, 14, 15, 0, 1, 2, 3, ...

Set up the main system so that it monitors the play position when sending the next wave data. Because the PCM stream buffers can hold up to 64K samples, anywhere from 2 to 16 buffers can be allocated.

An interrupt can be executed to the main system whenever the play position is updated. Use this feature when necessary. The system interface work area contains control information for this interrupt. For information on the settings, see section 5.1.3, *System Interface Work Area*.

### 6.4.3 Transfer Load

Because the PCM play slot is fixed by the hardware to 44.1 KHz, a sample is played every 22.68  $\mu$ sec after play starts. In other words, wave data for 44,100 samples must be supplied continuously for each channel per second. This number of sample converts to 44,100 bytes for 8-bit PCM and 88,200 bytes for 16-bit PCM.

The transfer time, which includes data read from the CD up to data transfer to the sound memory, becomes part of the main system load. The transfer time depends on the main system hardware and can be calculated from the CD revolution wait and seek times, the SH (or DMA) transfer clock, and the number of transfer bytes.

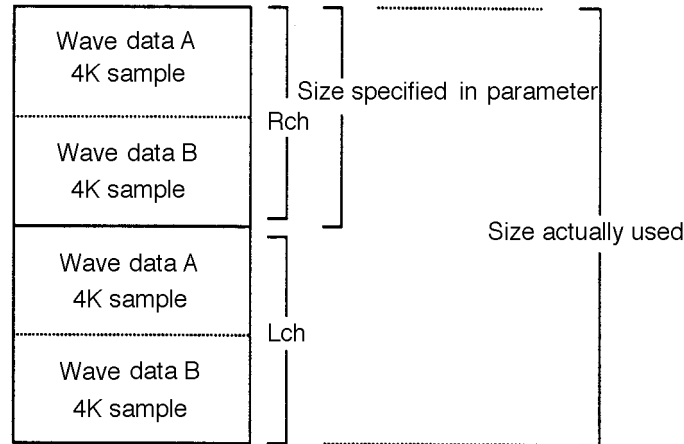
However, the number of bytes described above is the data size at 44.1 KHz. If the frequency can be halved to 22 KHz, the data volume can also be halved.

Consequently, the transfer load is also halved. Conversely, in stereo play or concurrent multi-channel play, the load increases correspondingly.



#### 6.4.4 Stereo Play

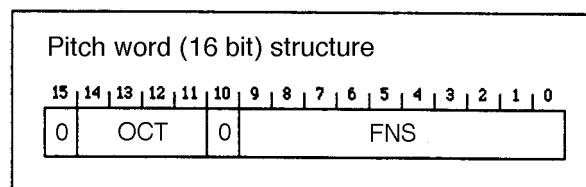
PCM stream play can be executed in stereo. For stereo play, twice the memory used in mono play is necessary. The parameter size specified in the command parameter at the start of play is the sample count for one channel. For stereo play, allocate an area that is twice the specified parameter size.



#### 6.4.5 Playing Wave Data at Frequencies Other than 44.1 KHz

The SCSP plays wave data always at a frequency of 44.1 KHz. The sound system supports wave data play at other frequencies by using a pitch parameter. For example, if wave data is played unaltered at 22 KHz, the play pitch is doubled. By halving the play pitch, the wave data can be played at the original pitch.

The pitch range can be up to  $\pm 8$  octaves (OCT). In addition, one octave can be divided into 1,024 levels (FNS). With a range of this precision, the sound system can handle any frequency. If the sampling frequency is known, calculate the pitch (OCT and FNS values) beforehand according to the pitch calculation procedure described in the next section. During PCM stream play, this pitch is assigned to a parameter so that the sampling frequency can be reproduced unaltered.



### Calculating the Pitch

Use the equation shown below to calculate the pitch. To calculate FNS, select the OCT value from the following table and substitute the sampling frequency value for Fs. For example, if the sampling frequency is 44.1 KHz, Fs is 44.1; if the sampling frequency 16 KHz, Fs is 16, etc.

Calculation Equation

$$FNS = \frac{2^{-OCT} \times 1024 \times Fs}{44.1} - 1024$$

[Empty box for calculation]

OCT Values

Sampling frequency	5.5125 KHz→	11.025 KHz→	22.05 KHz→	44.1 KHz→	88.2 KHz→
OCT value	-3	-2	-1	0	1

- For example, if wave data sampled at 16 KHz is played, the values for OCT and FNS are as shown below. The table tells us that the value for OCT is -2. From the equation, FNS is calculated as 462 (1CEh).

$$FNS = -1024 = -1024 = 462 (1CEh)$$

Consequently, the value for parameter P7-P8 Pitch Word for PCM stream play becomes 71CEh.

### 6.4.6 Notes

#### Continuous Transfer

When transferring wave data, allow the sound CPU to operate by avoiding long, continuous transfers. Note that the sound CPU operation is disabled during sound memory access by the main system, which has a higher priority. If continuous transfer is necessary, divide the transfer into segments no longer than 1 millisecond each. The sound CPU is then able to operate during the breaks when no data is being transferred.

#### DSP Usage

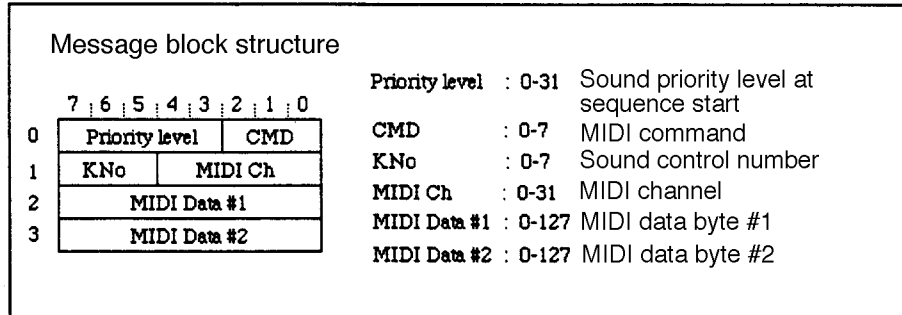
Mixer settings are necessary when using a DSP (for effects) in PCM stream play. To use a DSP, prepare the necessary mixer data and execute the mixer setting command. Tone data is necessary for the mixer settings even if there is no sequence play.

**Note:** A sample is the smallest unit of PCM wave data, and the size of a sample depends on the data width. If the data width is 8 bits, the sample size is one byte; if the data width is 16 bits, the sample size is two bytes. A 4K sample therefore indicates 4,096 bytes for 8-bit PCM, and 8,192 bytes for 16-bit PCM.

## 6.5 MIDI Direct Output

The sound driver is a sound source that uses MIDI messages to generate sounds. Sounds can be generated with the sound driver by sending MIDI messages directly without creating sequence data. The **MIDI Direct Control** command is the interface for MIDI direct output and has parameters that specify the message block to be sent to the sound driver.

The message block is a 4-byte data block that has the sound start time parameter added to the MIDI message. The following figure shows the structure of the message block.



Correspondence between CMD value and actual MIDI event:

CMD value	Corresponding MIDI event
0	(80h-8Fh) Note Off Event
1	(90h-9Fh) Note On Event
2	(A0h-AFh) After Touch
3	(B0h-BFh) Control Change
4	(C0h-CFh) Program Change
5	(D0h-DFh) Channel Pressure
6	(E0h-EFh) Pitch Wheel
7	(F0h-FFh) System Message

### 6.5.1 Usage Procedure

Execute the **MIDI Direct Control (09h)** command with the 4-byte message block as parameters P1, P2, P3, and P4. The following are the advantages and disadvantages of doing so. Consult with the sound creator before selecting the method to be used.

#### Advantages

- Real-time control is permitted when sequence data cannot be deployed in advance.
- Data creation tasks from sequence creation to compression are unnecessary.
- The load is reduced because the data compression and decompression processes are omitted.

#### Disadvantages

Although this method is versatile, in certain cases, a sophisticated program must be written based on an understanding of a large amount of information. (Using this method requires an understanding of MIDI channels and the corresponding tones, as well as an understanding of the relationship between the DSP program and mixer specifications. Consultation with the sound creator is necessary.)

## 6.6 Fade-In and Fade-Out

### 6.6.1 Fade-In Procedure

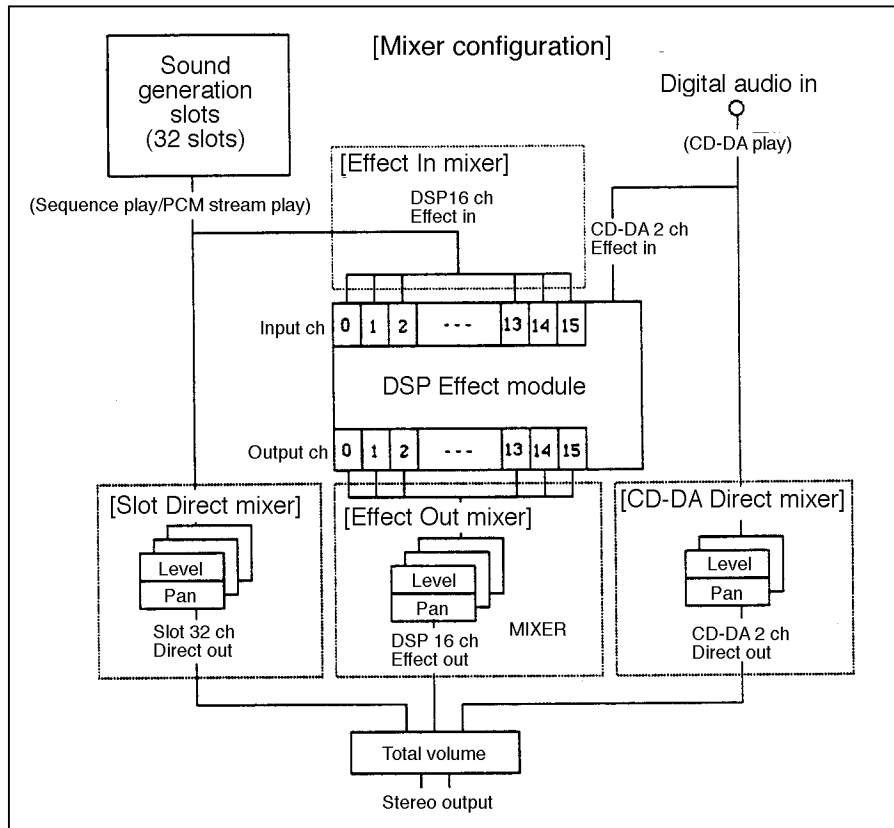
Specify a fade rate in the **Sequence Volume** command, and execute the command before issuing the **Sequence Start** command. (If a fade rate is not specified, only the current sequence volume changes.) Immediately after the **Sequence Start** command is executed, the volume fades in from the current sequence volume until the specified sequence volume is reached. The fade-in curve can be controlled by using the **Sequence Volume** command more than once.

### 6.6.2 Fade-Out Procedure

Specify "Volume=0" in the **Sequence Volume** command and execute the command. The volume fades out until "volume=0" is reached, according to the fade rate. The fade-out curve can be controlled by using the **Sequence Volume** command more than once.

## 6.7 DSP and Mixers

The relationship between the DSP and the mixers is very important when controlling sound. This section explains the relationship between the DSP and the mixers.



- Slot Direct mixer: Direct (without passing through the DSP) output level and Pan from the sound slot
- CD-DA Direct mixer: Direct (without passing through the DSP) output level and Pan from the CD-DA
- Effect In mixer: Input level to the DSP
- Effect Out mixer: Effect (through the DSP) output level and Pan from the DSP

### 6.7.1 Introduction to Mixers

Mixers are hardware units that balance the volume and position (left-right distribution) of the SCSP sound source. The sound source itself has two types: a sound slot and a digital audio input. However, because each type can be passed through the DSP, the mixers can be divided into the following four mixer blocks:

- Slot Direct mixer: Direct (without passing through the DSP) output level and Pan from the sound slot
- CD-DA Direct mixer: Direct (without passing through the DSP) output level and Pan from the CD-DA
- Effect In mixer: Input level to the DSP
- Effect Out mixer: Effect (through the DSP) output level and Pan from the DSP

Of these mixers, the Slot Direct mixer and the Effect In mixer are already set in the tone data and do not require special control. To control the CD-DA Direct mixer, use the special **CD-DA Level (80h)** and **CD-DA Pan (81h)** commands that have been prepared.

In this manual, "mixer" refers to the Effect Out mixer and the 16 channel effect level and Pan settings that are output by the DSP.

### 6.7.2 DSP and Mixers

The mixer is the exit setting for sounds output by the DSP. Even if effect processing operates normally and effects are properly incorporated, the effect sounds will not be output if the mixer is not set correctly. The mixer is the volume and position setting data for the 16 DSP output channels. The sound creator creates this data as part of the sound data.

All sounds for sequence play, PCM stream play, and CD-DA play enter the DSP. Controlling the mixer requires a correct understanding of which sounds enter and exit which channels in each scene. When switching effects, check the corresponding mixer and mixer specifications beforehand.

### 6.7.3 Eliminating Noise During Effect Change

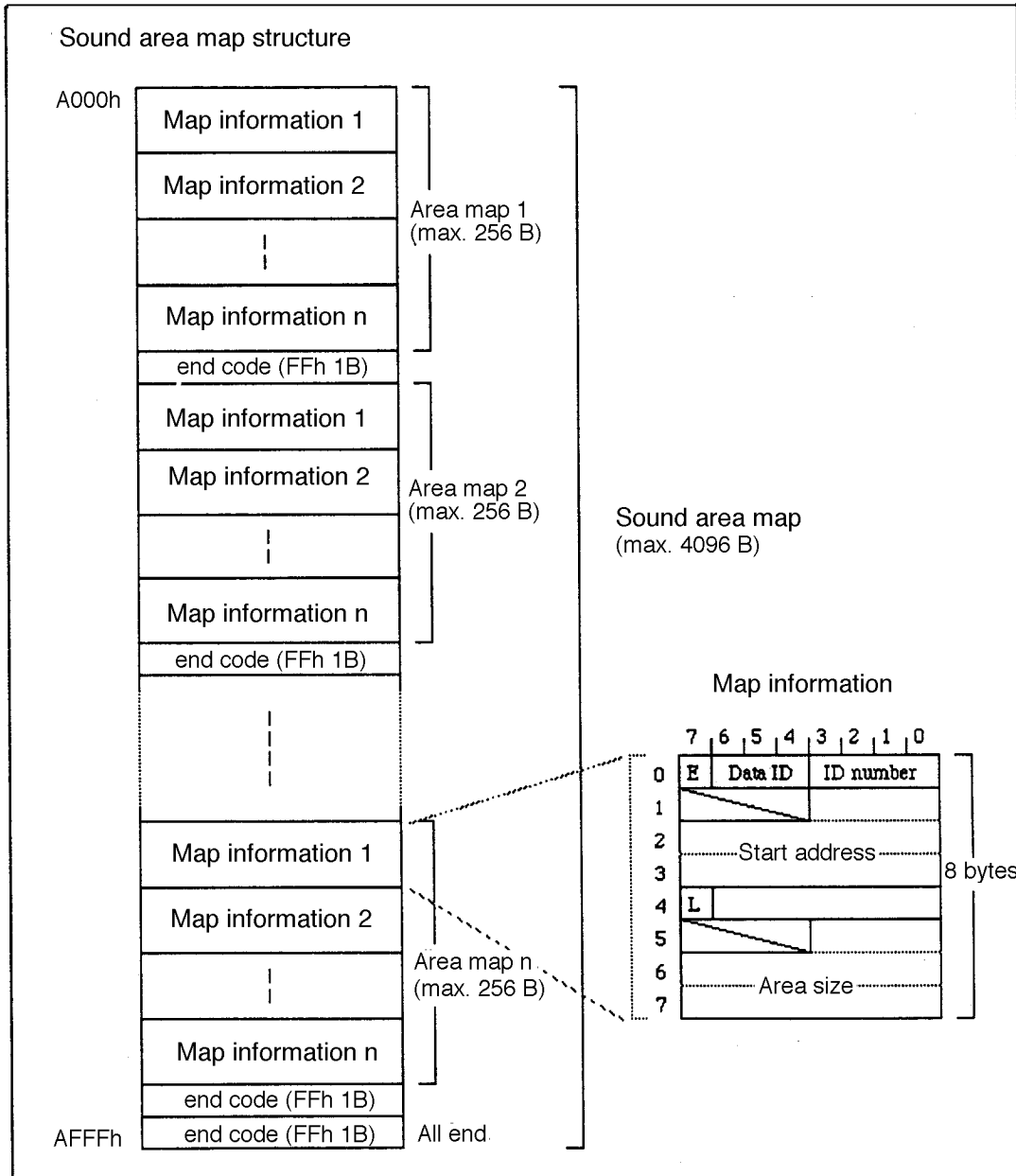
When an effect is switched, certain effect types may generate a large amount of noise. In such cases, use the mixer controls to prevent noise generation. Instructions are as follows:

1. Prepare a mixer that does not output any effects (mixer in which the output level for all 16 channels is 0).
2. Use the **Mixer Change (87h)** command to switch the mixer. After the mixer is switched, no effects are output.
3. Execute the **Effect Change (83h)** command to switch the DSP program.
4. Wait about one second. Other processing can be performed during this time. Although noise is actually generated, the noise is not output externally because the mixer is closed. Note that the time until the noise disappears varies slightly depending on the sound being produced at the time and the environment, including the DSP program and the size of the delay RAM. Make any necessary adjustments.
5. Use the **Mixer Change (87h)** command to return to the mixer that outputs the previous effects.

However, the effect component is not output for about one second while the effect is being switched. In step 2 when the mixer is switched, the abrupt volume change may generate a faint noise, which cannot be avoided. To eliminate the noise completely, use the **Total Volume (82h)** command to reduce the volume gradually by levels. An abrupt volume change can cause noise. Note that using the **Total Volume** command changes all sounds.

## 6.8 Sound Area Map

The sound area map links and combines all area maps in a game into one map and outputs it with the sound emulator (one of the sound development tools.) A one-byte end code (FFh) is inserted at the end of each area map, and an all-end code (FFh) is inserted at the end of the sound area map. The size of the sound area map varies, depending on the number of area maps. The maximum size a sound area map can be is 4,096 bytes.



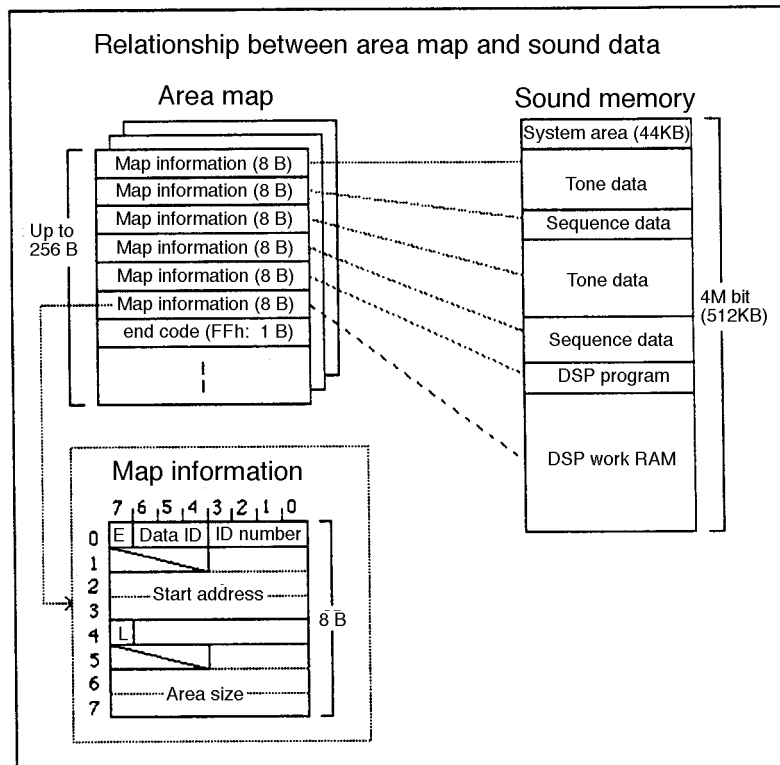


### 6.8.1 Introduction to Area Maps

A game has several scenes, and frequently the music changes when a scene changes. When the music changes, the sound data allocation (how the 512-KB sound memory is used) also changes. An area map represents how data is arranged for a particular scene. The three types of allocated data are tone data, sequence data, and the DSP program. Therefore, the area map displays where and how much of the sound memory each of the three types use. When the DSP is used, a DSP work RAM becomes necessary. In this case, the memory map also specifies the location and size of the DSP work RAM.

Normally, sequence play is enabled if an area map contains map information for both sound data and sequence data. However, to build a system with high memory and development efficiency, divide frequently used melodies and sound effects and allocate them to separate locations.

If the sound does change when the game scene changes, the number of required area maps would be equal to the number of scenes. However, area map, as used here, refers to the area map viewed in terms of sound; therefore, one area map does not necessarily correspond to one game scene. In cases when the same sound is used after the game scene changes, the area map does not have to be changed. In addition, the area map does not have to be changed if only part of the sound data is replaced.



### 6.8.2 Area Map Structure

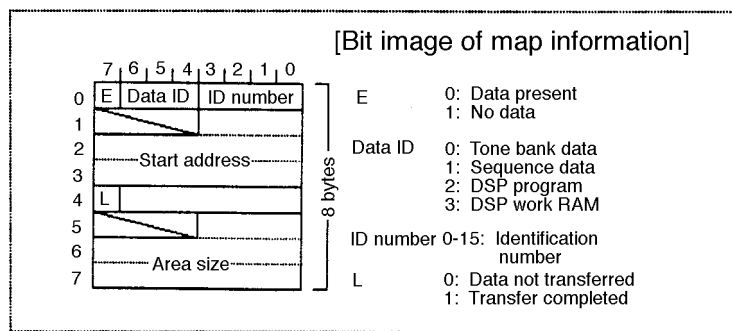
An area map links and combines several map information units into one unit. The size of area maps vary.

An area map can have up to 32 map information units, and the size of the area map is determined by the number of units.

Because the size of an area map varies, a one-byte end code is inserted at the end of the area map.

### 6.8.3 Details on Map Information

A map information unit is an 8-byte data block that describes where sound data is mapped in memory. The following figure shows the structure of a map information unit.



#### Map Information Data

Data	Meaning	Description
E	Data end bit	End bit of the map information. The data is one-byte long, and the actual value is FFh.
Data ID	Data type	Type of data to be stored in the defined area.
ID number	Identification number in data type	Identification number if there are several data items of the same type.
Start address	Start address	First address of the defined area.
L	Transfer completed bit	Flag indicating that data was transferred to this area.
Area size	Area size	Size of the defined area.

## 6.9 3D Sound

The SATURN sound driver supports two types of 3D sound: Q sound and Yamaha 3D sound. Use sound control commands **3D Control (11h)** and **Q Sound Control (12h)** to specify any desired position. The virtual sound source moves to the specified position.

For both Q sound and Yamaha 3D sound, the movement of the virtual sound source is random in most cases. Thus, the position movement controls are all performed by the main system. The sound driver only moves the sound image to the specified position. However, in cases involving simple linear movement or that require more intelligent control by the sound driver, the necessary processing can be performed for each project.

The sound driver provides four DSP programs as Yamaha DSP extension modules: Q sound 4 channel, Q sound 8 channel, Yamaha 3D sound 1 channel, and Yamaha 3D sound 2 channel. When using 3D sound, download the necessary DSP program with the **Effect Change (83h)** command.

### 6.9.1 Q Sound

Q sound is able to Pan sound over wide angles of up to 180 degrees horizontally. The position type is only Pan; however, up to eight channels can be controlled simultaneously.

- Channels (0-7): Either four channels (0-3) or eight channels (0-7) can be specified.
- Pan position (0-30): Position 15 is the center. Sound can be panned up to 15 levels left and right.

### 6.9.2 YAMAHA 3D Sound

Yamaha 3D sound allows the virtual sound source to be moved freely to positions spanning 360 degrees horizontal, 360 degrees vertical, and 128 distance levels. Although only up to two channels can be used, the channels can be specified to any combination of up, down, left, and right.

- Channels (0-1): Either one channel (0) or two channels (1) can be specified.
- Distance (0-127): The value 0 indicates a distance of 0. The distance increases as the value increases.
- Horizontal position (0-127): The 360-degree horizontal position is specified with 128 levels. The value 0 indicates directly ahead; 32, 90 degrees to the right; 64, directly behind; and 96, 90 degrees to the left.
- Vertical position (0-127): The 360-degree vertical position is specified with 128 levels. The value 0 indicates directly above; 32, directly ahead; 64, directly below; and 96, directly behind.

**Note:** 3D sound is presented as a Yamaha DSP extension module (DSP program), and cannot be used until the DSP extension modules are provided. At this time (2/28/95), 3D sound is still not supported.

## 6.10 Sound Initial Command

For parameters P1 to P5 of the **Sound Initial (10h)** command, the corresponding process is executed when the parameter value is 01h. For all other values, the parameter is ignored. This command is used mainly for setting conditions immediately before the **Map Change** command is executed.

### Details on P1 to P5

- When 01h is specified for P1, the **Sound Initial** command stops all sequences that are currently generating sounds. This specification does not affect other sounds (PCM stream play or CD-DA play).
- When 01h is specified for P2, the **Sound Initial** command stops all PCM stream plays that are currently generating sounds. This specification does not affect other sounds (sequence play or CD-DA play).
- When 01h is specified for P3, the **Sound Initial** command turns off the output for the CD-DA input components **EXTS0** and **EXTS1** by setting the direct components **EFSDL** and **EFPAN** to 00H. Therefore when the CD-DA input is manipulated with effects and the results are output, the volume is not turned off. This specification does not affect other sounds (sequence play or PCM stream play).

Sound address      100217H.b <--- 00H  
                          100237H.b <--- 00H

- When 01h is specified for P4, the **Sound Initial** command initializes the DSP unit in the sound source. Actually, the data shown below is set to the DSP registers, the output of the effect component is turned off, and access (read/write) to the D-RAM by the DSP is prohibited. In addition, the mixer is initialized but then returned. This specification does not affect other sounds (direct components of sequence play, CD-DA play, and PCM stream play).

**IMXL** <--- 00H  
**EFSDL, EFSPAN** <--- 00H  
**MPRO** <--- 00000000H  
**COEF** <--- 00H  
**TEMP** <--- 000000H

- When 01h is specified for P5, the **Sound Initial** command writes **EFSDL=EFSPAN=0** for all **EFREG** units (the mixer units), and disables output of the effect components. This specification does not affect other sounds (direct components of sequence play, CD-DA play, and PCM stream play).

## 6.11 Hardware Check Command

Details on the check items performed by the **Hard Check (89h)** command are described below. During check execution, a value from 0 to 5 is specified for command parameter P1. The parameter value determines the check to be performed. For all checks, one of the following values is stored in the system interface table (address 418h: word) as the check result.

- 7FFFH: Memory check error
- 8000H: Memory check OK

P1 Value	Check Description
00h	The command executes read/write check of the sound D-RAM (4 Mb). Note that during this check, the DSP for which internal memory access should be prohibited is initialized. The check writes, reads, and compares low/high for all bits and stores the results into the system interface table.
01h	The command executes read/write check of the sound D-RAM (8 Mb). The other features are as described in the previous item.
02h	The command executes an operation check of the SCSP built-in MIDI. When executing this check, short the input-output terminals for the external MIDI. The check results are as described above.
03h	The command outputs a square wave from both L and R sides. Note that during this check, SCSP slot 025B0000H is forcibly used. Also the sound is turned off automatically after a few seconds. When this command is executed, map data, melody data, and tone data are not necessary.
04h	The command outputs a square wave from the L side only. The other features are as described above.
05h	The command outputs a square wave from the R side only. The other features are as described above.

## Chapter 7 Supplement

### 7.1 Details on Error Status Bit Map

Bit Position	Error Description
bit 31	68K CPU address error
bit 30	68K CPU bass error
bit 29	68K CPU exception processing
bit 13	RAM area for target DSP was insufficient during effect change.
bit 12	Invalid value was set for RAM of target DSP during effect change.
bit 11	RAM for target DSP was not found in map during effect change.
bit 10	Target DSP program was not downloaded during effect change.
bit 9	Target DSP program was not found in map during effect change.
bit 8	Target sequence bank was not downloaded during sequence start.
bit 7	Target sequence song number was not found in sequence bank during sequence start.
bit 6	Target sequence bank was not found in map during sequence start.
bit 5	Target program number was not found in tone bank during MIDI program change.
bit 4	Tone bank for target control number and MIDI channel was not set during MIDI program change.
bit 3	Tone bank for target control number and MIDI channel was not set during MIDI control change (mixer change).
bit 2	Target tone bank was not found in map during mixer change or tone bank change. Or, the tone bank was not downloaded.
bit 1	Target map was not found in area starting from A000h during map change.
bit 0	Target mixer was not found in tone bank during mixer change. Or, the tone bank was not set.

**Note:** Bits 24 to 31 are not cleared automatically. When necessary, clear these bits from the host side. Bits 0 to 23 are automatically cleared during normal operation.

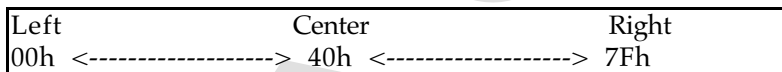
## 7.2 Details on SCSP Pan (32 Levels)

Pan	L (dB)	R (dB)	Position	Pan	L (dB)	R (dB)	Position
00h	-00.0	-00.0	C	10h	-00.0	-00.0	C
01h	-03.0	-00.0	R1	11h	-00.0	-03.0	L1
02h	-06.0	-00.0	R2	12h	-00.0	-06.0	L2
03h	-09.0	-00.0	R3	13h	-00.0	-09.0	L3
04h	-12.0	-00.0	R4	14h	-00.0	-12.0	L4
05h	-15.0	-00.0	R5	15h	-00.0	-15.0	L5
06h	-18.0	-00.0	R6	16h	-00.0	-18.0	L6
07h	-21.0	-00.0	R7	17h	-00.0	-21.0	L7
08h	-24.0	-00.0	R8	18h	-00.0	-24.0	L8
09h	-27.0	-00.0	R9	19h	-00.0	-27.0	L9
0Ah	-30.0	-00.0	R10	1Ah	-00.0	-30.0	L10
0Bh	-33.0	-00.0	R11	1Bh	-00.0	-33.0	L11
0Ch	-36.0	-00.0	R12	1Ch	-00.0	-36.0	L12
0Dh	-39.0	-00.0	R13	1Dh	-00.0	-39.0	L13
0Eh	-42.0	-00.0	R14	1Eh	-00.0	-42.0	L14
0Fh	∞	-00.0	R15	1Fh	-00.0	∞	L15

## 7.3 Details on MIDI Pan Data (Correspondence between SCSP and Pan)

MIDI Pan	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F
00h-0Fh	1Fh	1Fh	1Fh	1Fh	1Eh	1Eh	1Eh	1Eh	1Dh	1Dh	1Dh	1Dh	1Ch	1Ch	1Ch	1Ch
10h-1Fh	1Bh	1Bh	1Bh	1Bh	1Ah	1Ah	1Ah	1Ah	19h	19h	19h	19h	18h	18h	18h	18h
20h-2Fh	17h	17h	17h	17h	16h	16h	16h	16h	15h	15h	15h	15h	14h	14h	14h	14h
30h-3Fh	13h	13h	13h	13h	12h	12h	12h	12h	11h	11h	11h	11h	10h	10h	10h	10h
40h-4Fh	00h	00h	00h	00h	01h	01h	01h	01h	02h	02h	02h	02h	03h	03h	03h	03h
50h-5Fh	04h	04h	04h	04h	05h	05h	05h	05h	06h	06h	06h	06h	07h	07h	07h	07h
60h-6Fh	08h	08h	08h	08h	09h	09h	09h	09h	0Ah	0Ah	0Ah	0Ah	0Bh	0Bh	0Bh	0Bh
70h-7Fh	0Ch	0Ch	0Ch	0Ch	0Dh	0Dh	0Dh	0Dh	0Eh	0Eh	0Eh	0Eh	0Fh	0Fh	0Fh	0Fh

- The values in the table are Pan data values (00h-1Fh: 32 levels) for the SCSP.
- The MIDI Pan data consists of 128 levels. When MIDI Pan data is converted to SCSP Pan data, the last 2 bits are ignored.



## 7.4 Check Points

Symptom	Cause
No sounds are output.	<ul style="list-style-type: none"> <li>Is the sound driver operating?            Execute any <b>Sound Control</b> command if uncertain. If the executed command code is cleared to 00h, the driver is operating.            Alternatively, check sound memory address 8827h (byte) (SH address 25A08827h). If the sound driver is operating, the address value is incremented every two msec.</li> </ul>
	<ul style="list-style-type: none"> <li>Is sound data transfer completed?            For sequence play to take place, the tone data and the sequence data must be transferred successfully to the sound memory.</li> </ul>
	<ul style="list-style-type: none"> <li>Is the transfer completed bit set?            After sound data is transferred, the L bit (transfer completed) must be set in the sound area map CRNT work area. Otherwise, sounds will not be generated.</li> </ul>
	<ul style="list-style-type: none"> <li>Do the area map and the transferred data match?            Sound will be not generated if the transferred sound data does not match contents of the area map. Check the consistency, transfer address, and size of the area map and the transferred data.</li> </ul>
	<ul style="list-style-type: none"> <li>Is the sound driver version correct?            If the driver used to create the sounds and the driver incorporated in the game are different, sounds may not be generated correctly. If two different driver versions area used, the sequence compression format may be different. Check that the version is correct.</li> </ul>
	<ul style="list-style-type: none"> <li>Is the sound driver data file destroyed?            If the sound driver (<b>SDDRVS.TSK</b>) is transferred through a network UNIX server and the transmitting source is a Macintosh file with text attribute, the return codes may be converted to data. Before transferring the sound driver, check whether the attribute of the Macintosh file is text.</li> </ul>



Symptom	Cause
Effects are not applied.	<ul style="list-style-type: none"> <li>Has the DSP program been transferred? Effects can be applied only after the DSP program is successfully transferred to the sound memory.</li> </ul>
	<ul style="list-style-type: none"> <li>Is the transfer completed bit set? After sound data is transferred, the L bit (transfer completed) must be set in the sound area map CRNT work area. Otherwise, sounds will not be generated.</li> </ul>
	<ul style="list-style-type: none"> <li>Was effect change executed? Effects can be applied only if the DSP program in the sound memory is actually downloaded to the DSP. Execute the <b>Effect Change</b> command that corresponds to the effects used.</li> </ul>
	<ul style="list-style-type: none"> <li>Was mixer change executed? Sound effects can be output only if the correct mixer is set. Execute the <b>Mixer Change</b> command that corresponds to the effects used.</li> </ul>
	<ul style="list-style-type: none"> <li>Is sound sent to the DSP via tone data? Each tone in the effect data has a setting that indicates the effect to which the sound is to be connected (sent). If this setting is incorrect, effects will not be applied. Contact the sound creator and check the settings.</li> </ul>
No sounds are output after the sound driver was upgraded to a higher version.	<ul style="list-style-type: none"> <li>Was a bank change inserted in the sequence data? The sequence data must contain bank and program changes. For tool compatibility, sound will be generated on the development tool even if the data does not contain a bank change. When played on actual units, the sequence data must contain a bank change.</li> </ul>
	<ul style="list-style-type: none"> <li>Was a different sequence compression format used? Older versions (1.1x) of the sound driver use a different sequence compression format than the new versions (1.2x). Check that the correct version is being used.</li> </ul>
	<ul style="list-style-type: none"> <li>Does the program contain data size-dependent processing? When the version of the sound driver is upgraded, the driver size changes. Check whether the driver size is being processed by a constant for the previous version.</li> </ul>

Symptom	Cause
When map change is executed: <ul style="list-style-type: none"> <li>• no sounds are output.</li> <li>• operation becomes abnormal.</li> </ul>	<ul style="list-style-type: none"> <li>• Were all sound generation processes stopped before the map change? The sound driver executes internal processing even if no sound is being output. Before executing map change, execute the <b>Sound Initial (10h)</b> command to stop all sound generation processes.</li> </ul>
	<ul style="list-style-type: none"> <li>• Was the DSP work RAM address changed? The data in the DSP work RAM is always being overwritten by DSP internal processing. If the map is switched without stopping the DSP, the data that was transferred to the space corresponding to the DSP work RAM in the previous map is destroyed.</li> </ul>
	<ul style="list-style-type: none"> <li>• Was effect change executed? The DSP is stopped before executing map change. After map change, reexecute the effect change command, as necessary.</li> </ul>
	<ul style="list-style-type: none"> <li>• Was mixer change executed? In some cases, the mixer is initialized before map change. Always reexecute the mixer change command after map change.</li> </ul>
When data is played on an actual unit: <ul style="list-style-type: none"> <li>• no sounds are output.</li> <li>• sound balance is poor.</li> </ul>	<ul style="list-style-type: none"> <li>• Is the bank change at the beginning of the sequence data? When a standard MIDI file is created, there is a sequencer that rearranges the sequence of control changes having the same delta time. Shift the clock to ensure that the bank change is always first. If there is a program change or there is a <b>MIDI Volume</b> command before the bank change, sounds may not be output or the balance may be poor.</li> </ul>
Abnormal operation occurs.	<ul style="list-style-type: none"> <li>• Are the correct commands being executed in the correct order? With processing that has a set command sequence, check that the correct commands are being executed in the correct sequence. A history of the sound control commands is stored in the reserved area of the system area. (The latest 16 bytes x 128 commands are stored in the 2,048-byte region from 800h to FFFh. Also, see the explanation in the next section.)</li> </ul>

**PCM Stream Sounds Are Not Output**

Activate the sounder driver, and then execute the **PCM Start (85h)** command with the following parameters. A square wave sound should be output. Preparation of PCM wave data is not necessary.

	CMD	P1	P2	P3	P4	P5	P6	P7	P8	P9	P10
25A00700H <--	85h,	00h,	00h,	E0h,	00h,	01h,	00h,	FFh,	50h,	00h,	00h,

If the command outputs a sound, either the parameter settings or the PCM data is incorrect. If the command does not output a sound, check the following items.

Symptom	Cause
PCM stream sounds are not output	<ul style="list-style-type: none"> <li>Check the external line to the speaker, the volume, and other settings.</li> </ul>
	<ul style="list-style-type: none"> <li>Check the sound driver operation. Check whether the sound driver is operating. For instructions, see the items described for the symptom "no sounds are output."</li> </ul>
	<ul style="list-style-type: none"> <li>Check the sound driver version. The version is recorded in ASCII characters starting from sound memory address 1010h (SH address 25A01010h).</li> </ul>
	<ul style="list-style-type: none"> <li>Was the target PCM data correctly transferred to the sound memory? Note that command parameters P3 and P4 specify only the upper 16 bits of the 20-bit PCM play start address. Example) If P3=12h and P4=34h, the actual address is 12340h (25A12340h).</li> </ul>
	<ul style="list-style-type: none"> <li>Is the master volume of the sound source set to ∞ dB (volume=off)? Write 0Fh (-0 dB: maximum volume) to the byte starting from sound memory address 100401h (SH address 25B00401h). Note: This address cannot be read.</li> </ul>
	<ul style="list-style-type: none"> <li>Is the setting for PCM volume parameter P2 in the command correct? Note: Bits 7, 6, 5 = volume; bits 4,3,2,1,0 = Pan P2=E0h (volume = maximum, Pan = center) P2=C0h (volume = -6 dB, Pan = center) . . . P2=00h (volume = ∞ dB, Pan = center)</li> </ul>
	<ul style="list-style-type: none"> <li>Are the settings for pitch parameters P7 and P8 in the command correct? Bits 7 and 2 of P7 cannot be used (fixed at 0). If P7 and P8 are both set to 00h, PCM stream data is played at 44.1 KHz.</li> </ul>

## 7.5 Reference Data for Error Occurrences

If a sound-related problem occurs, check the following areas for possible reference data when investigating the cause.

### Error Status Bit Map

If an error occurs during sound driver operation, an error bit corresponding to the error type is set into the system interface table (4 bytes from 420h). For details on the error types, see section 7.1, *Details on Error Status Bit Map*.

The bit map contains bits that are automatically cleared when operation returns to normal and bits that are not cleared automatically after once set. When checking an error, clear the bits from the main system as necessary.

### Sound Control Command History

A history of the sound control commands is stored in the reserved area (2,048 bytes from 800h to FFFh) of the system area. Control commands (16 bytes) that the sound driver received from the main system and executed are constantly being written to this area in execution sequence.

The commands are written in ring buffer format. When the area becomes full, the commands are overwritten beginning from the oldest command. A record of the latest 128 commands always remains in the area. (When the number of processed commands exceeds 128, commands are written in sequence from the beginning of the reserved area.) In such cases, the next data write position is set in the system interface table (one word starting from 41Ah) as an offset address from 800h. The table therefore indicates the position of the data that was written last.

In the command history area (16 bytes), bytes 15 and 16 (one word area) contains the elapsed time since the previous command process. These bytes can be used to guarantee the command issuance timing. One count is approximately two milliseconds.

Reference the sound control command history area to check the following types of items:

- Whether the correct command was output from the main system if no sounds are being output.
- Whether the same command is being executed repeatedly.
- Whether commands are being executed in the correct sequence for processing in which the command sequence is set.
- Whether many commands are being executed in an extremely short time interval.

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