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Saturn SCSP User's Manual

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Introduction

Definitions

Terms used in this manual are defined below.

SCSP (Saturn Custom Sound Processor)

A multi-function game sound generator LSI that combines a PCM sound generator and sound DSP.

Data

A bit indicates 0 or 1 and is the smallest unit. 8 bits combined together is called a byte, and 16 bits (or 2 bytes) is called a word. When upper and lower grade is divided into 4 bits respectively, they are called nibbles.

PAN-POT

Determines the direction or location from which the sound seems to come from in localization.

Direct Data

Sound that does not pass through the DSP, or even if it did, no effects were applied. Also called "dry data".

Effect Data

The sound generated, as a result of sending it through DSP with an effect created in the sound generator. Also called "wet data".

MIDI Standard (Musical Instrument Digital Interface)

Conversion standard for sound interval and sound length used to communicate between electronic instruments and computers.

Pulse Code Modulation Sound Generator

Sound is converted to PCM data and stored in memory. Refers to the sound source in a method by which sound data is read from memory during performance to generate the sound output. PCM divides the sound (wave form) along the time axis and converts each of the wave high values to digital data. The data that results from this operation is called PCM data.

FM (Frequency Modulation)

Indicates frequency modulation. In the context of a sound generator, it indicates the FM sound generator.

Linear

Shows a straight line. This means that the relation of the input and output are proportional. Therefore, if the wave form data is linear, a high wave value proportional to the size of the data is produced. In other words, the quantized step is in equal intervals.

LFO (Low Frequency Oscillator)

Indicates a functional frequency generator that generates wave forms with frequencies that are below the range that is audible to the human ear. The output wave form of this LFO is used for amplitude and modulation of frequency.

DSP (Digital Signal Processor)

This is mainly for calculation (multiplication and addition) and contains a circuit for high-speed calculations. The DSP within SCSP is specially customized for audio editing functions and is used when directing sound effects such as echo, reverb, chorus and filter.

Sample Count

Indicates the data count of the wave form data.

dB (Decibel)

A unit for expressing the relative intensity of sounds by using a logarithm to show that unit. The equation would be:

$$\text{Sound Ratio [dB]} = 20 \cdot \text{LOG}_{10}N \quad N = \frac{\text{Sound Volume being compared}}{\text{Standard Sound Volume}}$$

Specifically, if the comparative volume is twice the standard volume (wave form amplitude), the sound comparator would be 6 [dB]. Once would be 0 [dB], and three times would be 10 [dB] On the other hand, if the comparative volume is smaller than the standard, the value [dB] would be negative (minus value). For example, if it was 1/2 of the standard, it would be -6 [dB].

Modulator

Indicates the modulator slot when multiple slots are connected. In the diagram below, SLOT0 and SLOT1 are modulators.

Carrier

Indicates the slot which is modulated when multiple slots are connected. In the diagram below, SLOT2 is the carrier.

Prescaler

This function allows required time to be set when the counter increases one increment. Time selection and settings are done for each timer.

Symbols

The following symbols are used in this manual.

Binary

Displayed with a “B” at the end. For example, 100_B. However, in the case of 1bit, it may be shortened to just B.

Hexadecimal

Displayed with an “H” at the end. For example, 00_H, FF_H.

Units

1KByte indicates 1,024 Bytes; 1Mbit indicates 1,048,576bits.

MSB, LSB

In byte and word configuration, left is the high bit (MSB, Most Significant Bit), while right is the low bit (LSB, Least Significant Bit).

Undefined bits

In the sound generator block register or the DSP block register, bits that are not defined are shown with “-”. When writing data to undefined bits, please write a “0_B”.

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Chapter 1

Sound System Configuration

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1.1 System Configuration

SCSP is a multi-function game sound generator LSI that consists of a PCM sound generator and a sound DSP. Figure 1.1 shows the sound system and peripheral configuration. The sound processing CPU that makes up the Saturn sound block is the MC68EC000. This CPU controls the various sound blocks.

The SCSP creates and processes sound mixes. It contains a 32 slot sound generator and sound effect DSP, digital mixer and timer, and an interrupt controller.

The sound memory is connected directly to the SCSP. It has a capacity of 4Mbit (512KByte) and is used to store the sound programs and data sound wave form data, etc. This memory is accessed by the sound CPU, SCSP and the main CPU (SH-2, SCU). Besides the memory, the SCSP is connected to the main CPU, sound CPU and the D/A converter. The sound system can function independently of the main system.

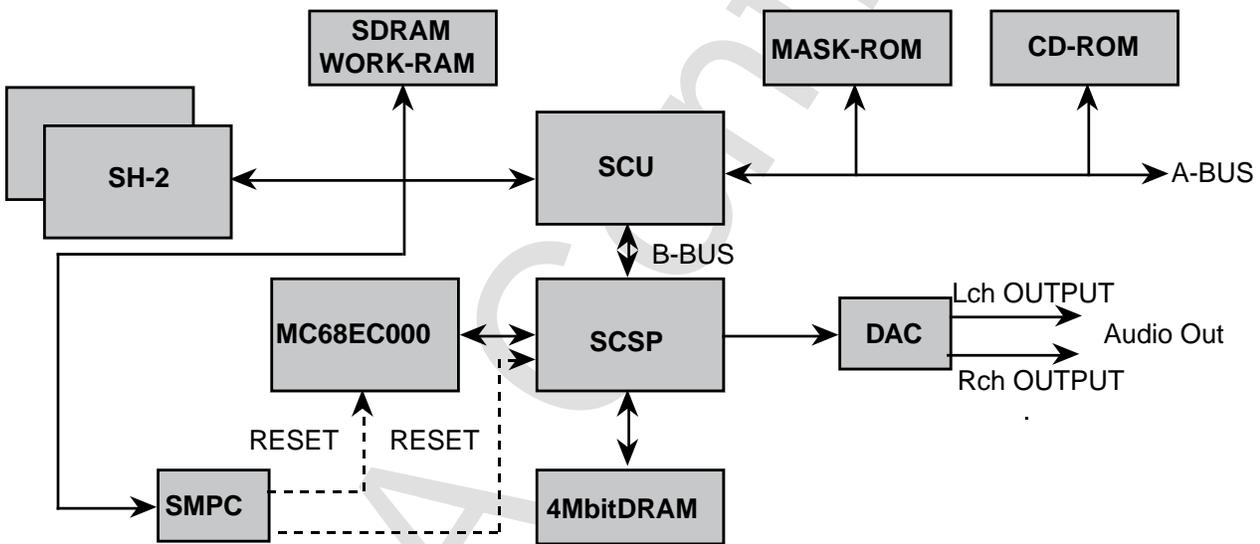


Figure 1.1 Sound System and Peripherals



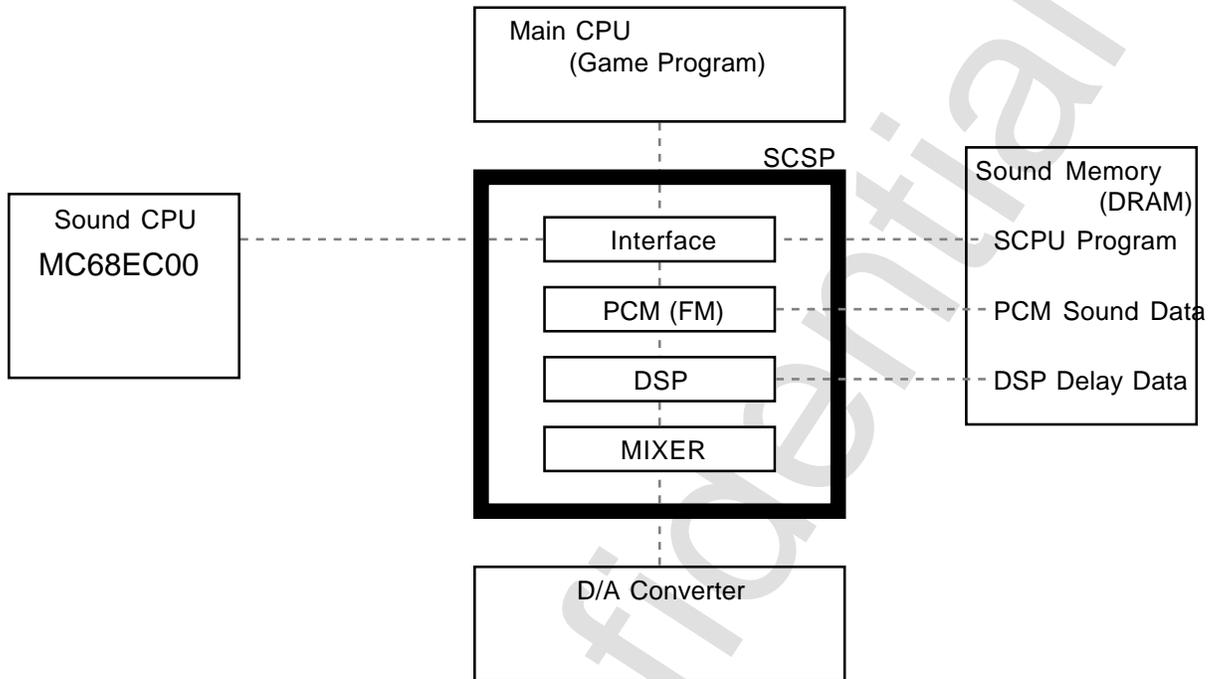


Figure 1.2 Sound System Configuration

Relation to the Main CPU

The SCSP has a main CPU interface incorporated to allow communication between the sound system and the main system (including the main CPU). As shown in Figure 1.2, the main system can access the memory and registers controlled by the SCSP through the main CPU interface.

Relation to the Sound CPU

The SCSP also incorporates a dedicated sound processing CPU interface (for the MC68EC00) to allow independent operation of the sound system against the main system. From the SCSP's point of view, the sound CPU is like a controller.

Sound Memory Map

Saturn has 4Mbit of internal sound memory. The sound memory can be accessed by the sound CPU, the SCSP, and the main CPU (SH-2, SCU). Similarly, the SCSP control register that controls the SCSP sound source (block) and DSP control that are used in the generation and processing of sound can also be accessed.

Figure 1.3 shows a sound memory map. Be aware that the memory map is referenced from the sound block side, making the addresses different when accessing from the main CPU. (See the SCU user's manual for more details.)

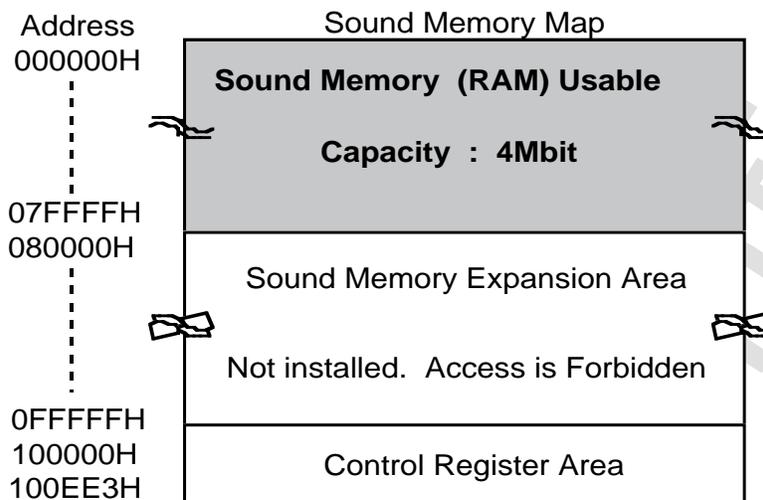


Figure 1.3 Memory Map of the Sound Memory Register

CPU programs and data, wave form data, and DSP work area (DSP delay area) are stored in the sound memory.

Relation to the D/A Converter

The D/A converter converts and outputs the digital sound signal created by the SCSP to analog sound signal.

Positioning of the Sound System

The sound system (sound CPU, SCSP), based on the sound memory, can operate independently from the main system (main CPU, video system, etc.) During this time, the sound system operates on a RAM base and so it must get the programs and data required for the sound CPU from the main system. Furthermore, since synchronization between the image and sound is required, an interface for two-way communication is also required.

To facilitate this communication the SCSP has an interface for the main system included to enable communication with the main system.

The sound CPU cannot independently read data from or write data to the main system. Information exchange between the main system and the sound system is accomplished by the main system reading data from or writing data to the sound memory (RAM).



Starting Up the Sound System

After the sound system power is turned on, it is reset by the SMPC. In this state, neither the sound CPU or SCSP will operate.

RESET

The sound system will operate after the SCSP reset has been released by the SMPC. The SCSP will initialize the internal registers, etc. for about 30 μ sec after the reset has been released. For this reason, access is not allowed during the 30 μ sec.

After the internal registers, etc., are initialized, the SCSP can be accessed by the main CPU (SH-2, SCU). Once in this state, the sound program can be downloaded. Before downloading anything, make sure to set MEM4MB bit to 1 and DAC18B bit to 0 within the sound CPU (address) 10400H address. Once done, accessing the memory and downloading files can be begun.

Table 1.1 Sound CPU Address 100400H, 100401H

(100400 H)								(100401 H)								
b15	b14	b13	b12	b11	b10	b9	b8	b7	b6	b5	b4	b3	b2	b1	b0	
-							M4	D8	VER [3 : 0]				MVOL [3 : 0]			
0	0	0	0	0	0	1	0	D3	D2	D1	D0	D3	D2	D1	D0	

M4: MEM4MB D8: DAC18B

The sound CPU reads the reset vector from the 8 bits of the sound CPU address (000000H~000007H), so always transfer the CPU reset vector to this area.

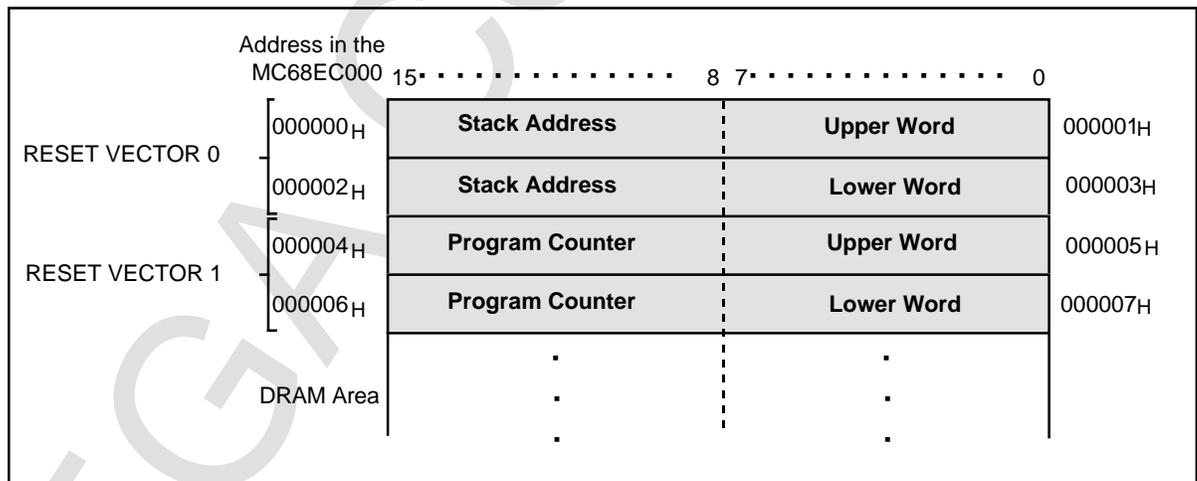


Figure 1.4 Sound Memory Range (MC68EC000, SCSP)

After the reset is released, the sound CPU reads the reset vector and jumps to the PC value address.

After download has finished, the sound CPU is reset by the SMPC enabling the sound CPU to operate.

Sound System Communication

Communication with the Main System

Communication between the main system and the sound system takes place through the SCU. The sound CPU cannot access the main system through the SCSP. Access is limited to access from the main system only.

Interrupt Signal

The main system uses the SCSP interrupt register when it sends an interrupt to the sound CPU (explained in detail later on). This is executed by writing a "1B" to bit 5 of the interrupt register (SCIPD).

All interrupts that can be applied to the sound CPU can be used as interrupts for the main CPU.

Please refer to the register explanations for sending interrupts from the sound system to the main system.

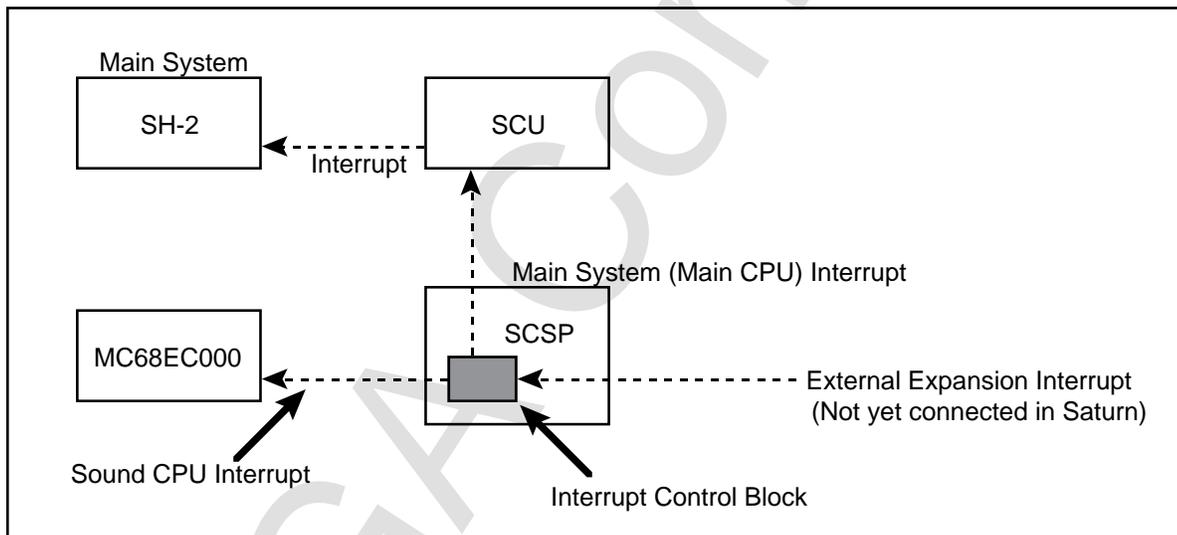


Figure 1.5 Interrupt Relations

Interrupts to the sound CPU use an auto vector method. Vector data is shown in Table 1.2.

Interrupt levels for the sound CPU can be set at different levels for each interrupt factor. (See register explanations for settings.)



Table 1.2 Sound CPU Interrupt Vector Table

Vector No.	MC68EC000 Address	Interrupt Vector Contents
0	000000 H	Reset vector initial SSP value
	000004 H	Reset vector initial PC value
2	000008 H	Bus error
3	00000C H	Address error
7	000010 H	Invalid command
5	000014 H	Calculation by 0 (Zero)
6	000018 H	CHK command
7	00001C H	TRAPV command
8	000020 H	Privilege violation
9	000024 H	Trace
10	000028 H	Line 1010 emulator
11	00002C H	Line 1111 emulator
12	000030 H	Undefined (Reserved)
13	000034 H	Undefined (Reserved)
14	000038 H	Undefined (Reserved)
15	00003C H	Uninitialized interrupt vector
16~23	000040 H~00005F H	Undefined (Reserved)
24	000060 H	Spurious interrupt
25	000064 H	Auto vector level 1 interrupt
26	000068 H	Auto vector level 2 interrupt
27	00006C H	Auto vector level 3 interrupt
28	000070 H	Auto vector level 4 interrupt
29	000074 H	Auto vector level 5 interrupt
30	000078 H	Auto vector level 6 interrupt
31	00007C H	Auto vector level 7 interrupt
32~47	000080 H~0000BF H	TRAP command vector
48~63	0000C0 H~0000FF H	Undefined (Reserved)
64~255	000100 H~0003FF H	User interrupt vector

 Normal use vectors.

 Vectors open to users.

Note:

The sound development tools supplied by this company use the auto vector level 7 interrupt, so users will not be able to use that level.

Interrupt Signals to the Main and the SCU DMA

The interrupt signal to the main CPU can also be used as triggers for the DMA transfer start that have a SCU. (Refer to the SCU manual for details.)

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Chapter 2

SCSP Overview

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2.1 LSI Overview

Since SCSP was developed with multi-media in mind, the audio features have better functions and higher quality than before. The calculation block is equipped with synthesizers that have performance comparable to those currently on the market enabling a wide variety of sounds to be produced. Also, with future expansion in mind, various interfaces are equipped. The DSP block can also produce composite sound stages, and perform special effects such as play back of various sound stages and special 3D sound positioning.

The characteristics of LSI are described below.

- Sampling Frequency
 - Sound generator block re-sampling frequency (set playback rate) 44.1KHz
 - Wave form data sampling frequency 0 (DC) ~ 44.1KHz
- Characteristics
 - 32 slots for FM or PCM use built-in.
 - One per slot (completely independent) incorporates the total of 32 LFOs.
 - Incorporation of 32 EGs of 4 segment .
 - Interface for a built-in CPUs: The main CPU and the sound CPU
 - Built-in MIDI interface.
 - Built-in 128 step DSP
 - Built-in input mixer that selects DSP input
 - Built-in output mixer to mix sound generating output and DSP outputs.
 - Incorporation of a new FM calculation method.



2.2 LSI Specifications

Table 2.1 shows the sound CPU (MC68EC000); tables 2.2 and 2.3 show SCSP LSI detailed specifications.

Table 2.1 Sound CPU Specifications

SOUND CPU MC68EC000-12 12.5MHz version (Operating frequency 11.2896MHz)		
Function	No. Included	Special Comments
CPU commands		Command system that is completely compatible with the MC68000 CPU.
Interrupt signal		level 1~7 interrupt by auto vector .
Hardware configuration		MC68000 CPU with the MC6800 (8 bit) interface removed

Table 2.2 SCSP LSI Specifications (1)

SOUND LSI SCSP (SATURN CUSTOM SOUND PROCESSOR) (Operating frequency 22.5792MHz)		
Function	No. Included	Special Comments
Sound Source		
Sound mixing method		PCM sound mixing and FM sound mixing methods (1) noise source (can be used for mixing)
Sound source block re-sampling frequency		44.1KHz fixed, primary interpolation
Wave form data format		16 and 8 bit linear 2'S complement method
Sound processor slots	32	Equal to 1 operator (1slot) in PCM and FM sound sources.
Maximum number of simultaneous sounds	1~32	When all are 4 operator (slot) type FM sounds: 8 sounds when all are PCM sounds, a total of 32 sounds. (The number of slots per FM sound can be freely set. It is also possible to mix FM and PCM sounds. As long as the total number of slots does not exceed 32, then any combination is possible.
Wave form loop function		Select from normal, reverse and alternative loops for each slot.
EG	32	4 segment AMPLITUDE-EG (Each slot has 1 base included)
LFO	32	Each slot has one installed (for amplitude/frequency modulation)
Types of LFO output wave form	4	Four types: sawtooth, rectangular, triangular, and white noise.
Sound source oscillation frequency		-8~+7 octaves with 1024 steps between octaves (nonlinear) can be set
Effect DSP block		
DSP processor speed		128 steps/Fs (multiple parallel processing type DSP Fs=44.1KHz)
Program RAM	128W	128 words x 60 bits
Coefficient data RAM	64W	64 words x 13 bits
Temporary (universal) RAM	128W	128 words x 24 bits
Multiplier accuracy		24 words x 13 bits = 26 bit output
Adder accuracy		26 words +26 bits = 26 bit output
DSP internal bus width		24 bit data bus

- 1 FM sound mixing is not limited to the 4 operator (slot) types used in the past, but 2~32 operators can be freely used for FM connections. You may also freely set the feedback.

Table 2.3 SCSP LSI Specifications (2)

Function	No. Included	Special Comments
Digital mixer block		
Output level adjustment steps		8 steps (for each slot sound, DSP output sound)
Panpot level adjustment steps		31 steps (center 1 step, left and right, 15 steps each)
DSP effect send channel count	16	Able to mix and store multiple slot output in each channel
DSP effect return channel count	16	Able to set level and panpot for each channel.
Master volume set function		Stereo capable
CPU interface		
Main CPU interface	1	SCU B-BUS interface
Sound CPU interface	1	MC68EC000 interface
Other functions, interfaces		
Timer	3	8 bit timers with prescaler
DMAC	1	DMA controller for transfer between SCSP~DRAM
Interrupt controller	1	Interrupt controller for the main and sound CPUs
MIDI interface		IN: 1 OUT: 1
Digital audio interface	1	Input stereo interface: 1
Sound memory interface	1	DRAM interface
DAC output interface	1	16 bit/ 18 bit stereo DAC interface
External interrupt signal interface	3	Not connected/used in the Saturn
DSP program library		
Various types of effect programs		REVERB (HALL, ROOM, VOCAL, PLATE, ETC.) EARLY REFLECTION ECHO / DELAY (STEREO, MONORAL) PITCH SHIFTER (SINGLE, DOUBLE, TRIPLE) CHORUS, FLANGER SYMPHONIC, SURROUND VOICE CANCEL, AUTOPAN PHASER, DISTORTION FILTER PARAMETRIC EQUALIZER



2.3 SCSP Chip Block Diagram

Figure 2.1 shows the SCSP chip block diagram.

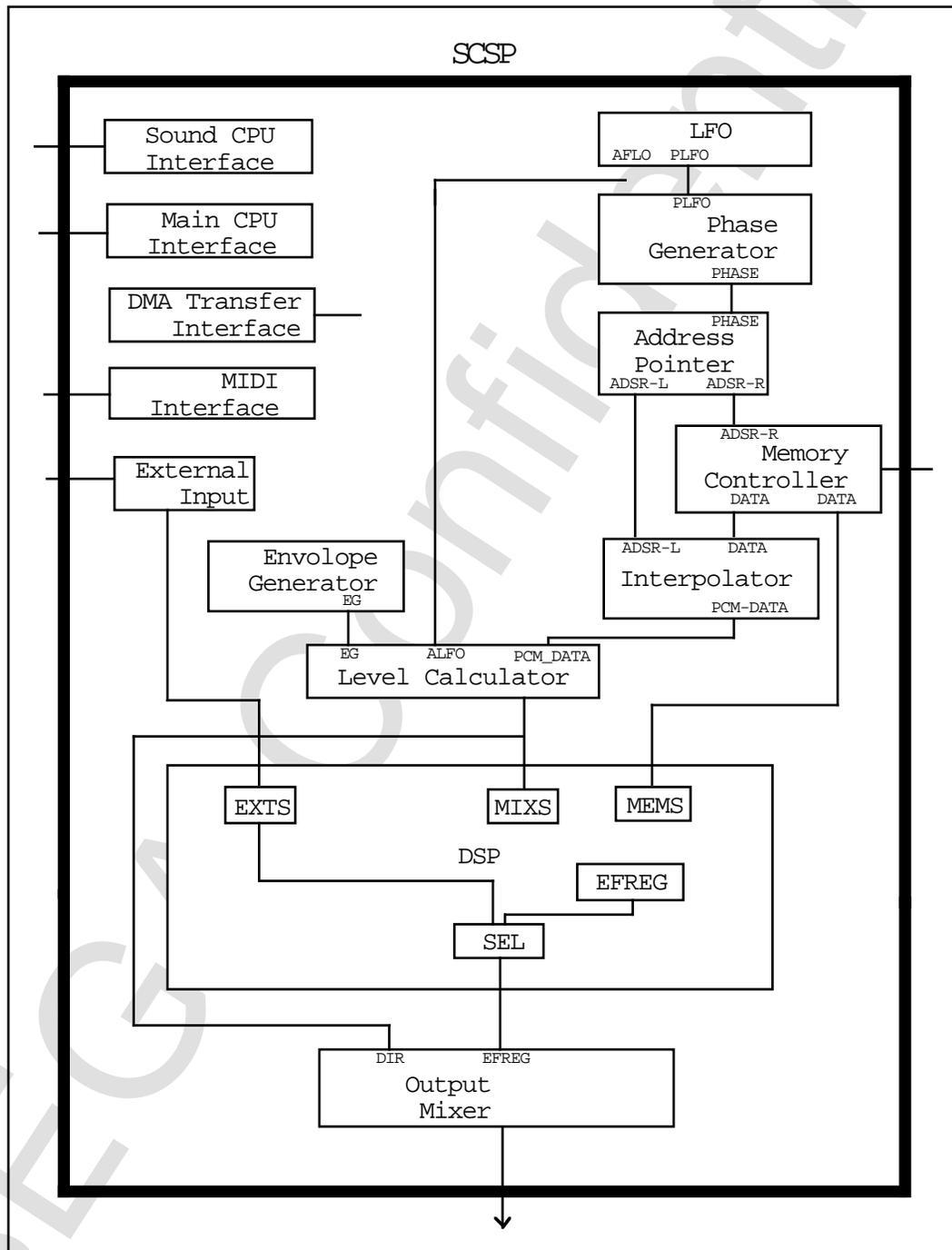


Figure 2.1 SCSP Chip Block Diagram

Items shown in SCSP Chip Block Diagram Figure 2.1 are explained below.

Sound CPU Interface

Interface that connects to the MC68EC000.

Main CPU Interface

Interface that connects the system controller (SCU).

DMA Transfer Interface

This is the SCSP built-in DMA controller. It enables data transfer between the SCSP and the sound memory.

MIDI Interface

This serial interface complies with MIDI specifications; however, some external circuits will be needed to make it compatible with a MIDI connector.

External Input (External Digital Audio Input Interface)

Interface that inputs a digital audio signal from an external instrument (external device).

L F O (Low Frequency Oscillator)

Indicates a function generator that generates wave forms of a frequency that is below the range that the human ear can hear, and is used as a wave form for various modulation.

Phase Generator

Block that calculates and outputs frequency data based on the sound generation frequency setting of the PCM data.

Address Pointer

Creates the wave form address value based on the phase information, etc., from the phase generator.



Memory Controller

Block that outputs the address created by the address pointer to the memory and reads data, etc.; it controls the sound memory.

Interpolator

Block that interpolates the wave form data.

Envelope Generator (EG)

Functional calculation block that controls the sound output level with time.

Level Calculator Block

Block that calculates the wave form output level based on the level coefficient created by EG, TL (Total Level), and ALFO.

DSP

Audio DSP that receives and holds the SCSP special sound effects.

Output Mixer

Final block that contains functions that calculate various positions and level adjustments needed to compile each sound output in stereo.

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Chapter 3

SCSP Functions

Chapter 3 Contents

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3.1 Interface

The SCSP has two internal CPU (main CPU and sound CPU) interfaces. The main CPU has higher priority than the sound CPU, so the processing speed of the sound CPU depends on the operation of the main CPU.

Sound CPU Interface

The sound CPU interface is a block with specialized functions to enable it to connect to the sound CPU. By adding this interface, the sound CPU can be directly connected to the SCSP without external circuits.

Programs for the sound CPU reside in the sound memory. For this reason, all CPU programs are placed in the sound CPU address space available.

Main CPU Interface

Access between the main CPU and the interface is shown in Figure 3.1.

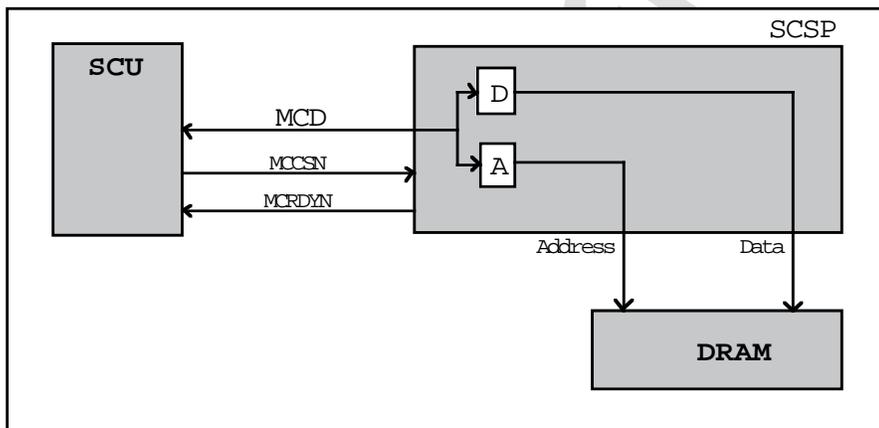


Figure 3.1 Access Overview

At the trailing edge of the select signal (MCCSN) from the main CPU, the interface is started; at the leading edge of the select signal, the interface is ended. Also, when "1" is output with respect to the ready signal (MCRDYN) to the main CPU, the select signal (MCCSN) from the main CPU and the main CPU data bus (MCD[7:0]) do not change.



The following cautions should be observed with interfacing with the main CPU.

- (1) Because the main CPU cannot access in units of 8 bits, so read and write in 16 bit units.
- (2) When there is a request to read or write, the SCSP buffer gets the address and data. Because of this, a "1" is output to the ready signal (MCRDYN) going to the main CPU causing a wait. This wait continues until the internal processing of the LSI is finished. For this reason, continuous read and writes that cause many waits to occur should be avoided except when turning on the power.
- (3) When the power is turned on, the time necessary for initialization of sound memory by continuous writing is approximately 100 msec for 4Mbit DRAM.

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3.2 Memory Access Control

When accessing sound memory from the SCSP, the following priorities are maintained.

1. PCM Data Read by PCM sound generator, accessed by DSP.
2. DRAM refresh cycle.
3. DMA transfer.
4. Access by the main CPU.
5. Access by the sound CPU.

When there is an access request with a high priority, there will be a wait state against an access request with a low priority. Also, a high priority access request will not occur while a low priority access is requested because the decision on which device to be permitted to access memory (PCM sound generator block, DSP block, main CPU, sound CPU, DMA, etc.) is made before any actual memory access is performed.

	CYCLE 0	CYCLE 1	CYCLE 2	CYCLE 3	
Memory Access Priority					
High	DSP Access	PCM Access	DSP Access	PCM Access	
	Refresh cycle	Refresh cycle	Refresh cycle	Refresh cycle	
	DMA Access	DMA Access	DMA Access	DMA Access	
	MCPU(SCU) Access	MCPU(SCU) Access	MCPU(SCU) Access	MCPU(SCU) Access	
Low	SCPU Access	SCPU Access	SCPU Access	SCPU Access	

Figure 3.2 Memory Access Priorities

The performance of the SCSP / sound CPU is determined by the distribution of the memory cycles.

There are 128 memory cycles within one sample ($1/44.1K = 22.68\mu$ sec). These 128 cycles are distributed among the various devices. The number of times the CPU accesses changes with the application, so there is no best way to access memory, but be aware of the items on the following page.



- Sound memory uses DRAM, so refresh cycles are required. In the Saturn sound system, two empty cycles are required between each sample, so the number of memory cycles that can be used by other devices is 126.
- The sound generator and DSP memory cycles have the highest priority memory cycles.
- The sound generator uses two memory cycles to produce sound from one slot. With a maximum of 32 slots, a total of 64 times of memory cycle may be used. When the EG is at the maximum attenuation state (“3FFH”), the sound generator does not access memory. Do KEY_OFF for those slots that are not producing sound.
- The DSP accesses memory a maximum of 64 times. This changes with the DSP application. Try to use the internal DSP registers as much as possible when storing data temporarily.
- The sound CPU operating speed will decrease when using the SCSP built-in DMA due to the extent of the wait state imposed on the sound CPU.

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Chapter 4

SCSP Register

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4.1 Register Map

The SCSP carries a wave form calculator unit called SLOT to realize FM sound mixing.

Figure 4.1 describes the SCSP memory map and the allocation of the individual registers that make up the memory map.

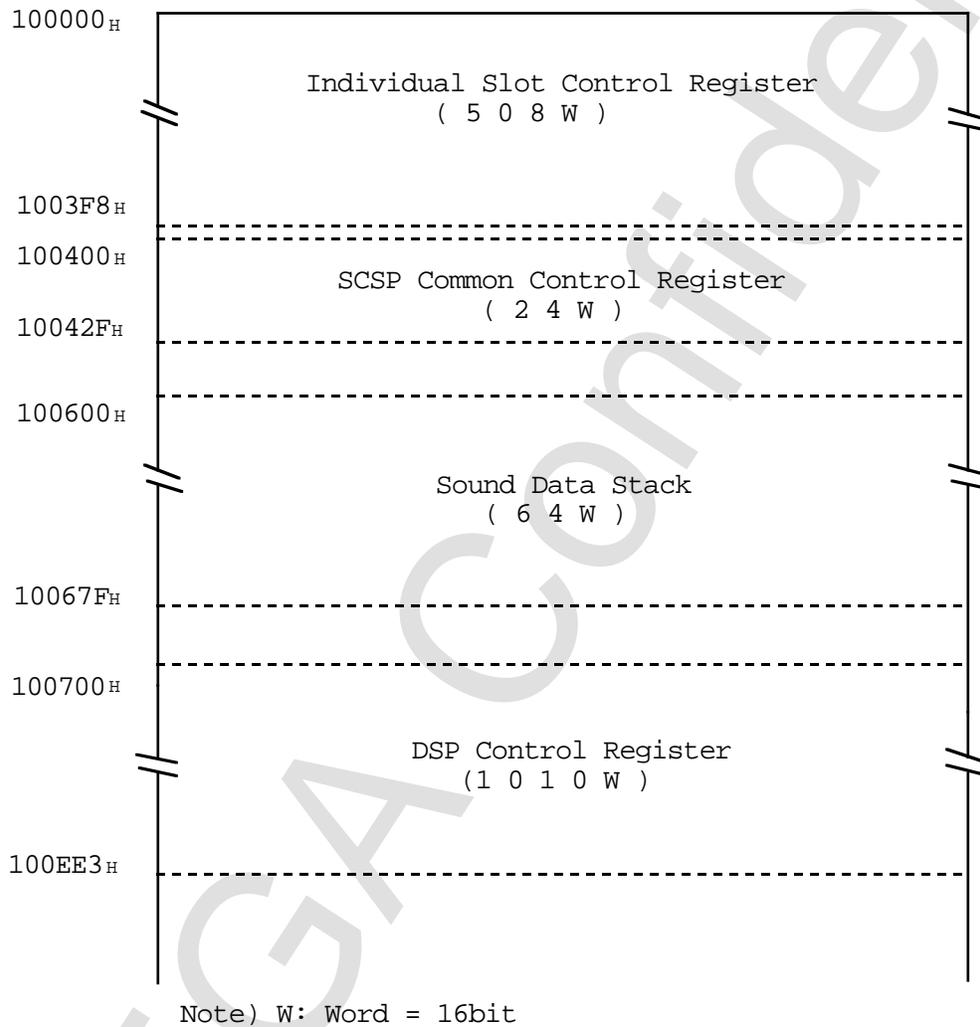


Figure 4.1 SCSP Memory Map (1906W)



The individual slot control register represents the allocation of the registers configured in each of the 32 slots (SLOT0~SLOT31).

The location of bit allocation for each register is described as [3 : 0]. For example, SA [19 : 16] represents the allocation from the 16th bit through the 19th bit of SA.

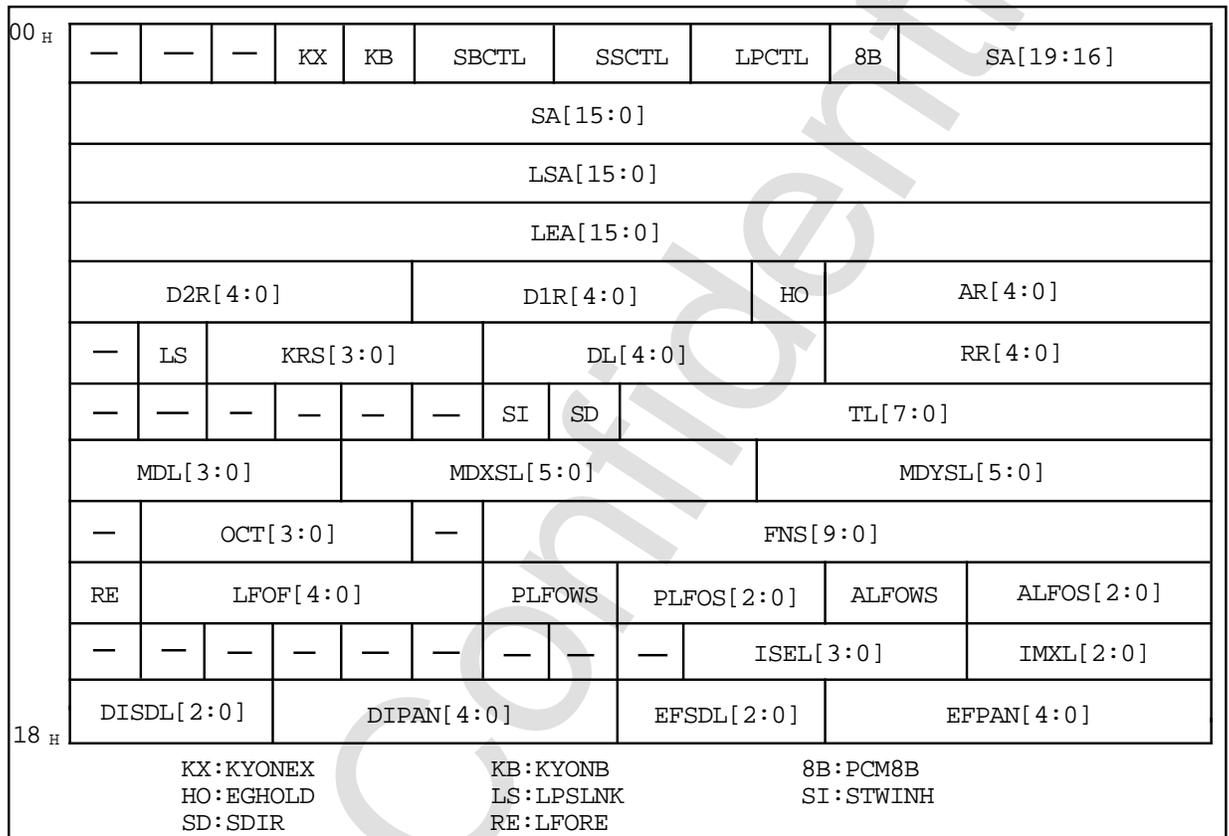


Figure 4.2 Individual Slot Control Register

Table 4.1 Individual Slot Control Register (1)

Designation	Contents
KYONEX (KX)	Execute KEY_ON
KYONB (KB)	Record KEY_ON, KEY_OFF
SBCTL	Source bit control
SSCTL	Sound source control
LPCTL	Loop control
PCM8B (8B)	Select wave form data format
SA	Start address
LSA	Loop start address

Table 4.2 Individual Slot Control Register (2)

LEA	Loop end address
D2R	Decay 2 rate
D1R	Decay 1 rate
EGHOLD (HO)	EG hold mode
AR	Attack rate
LPSLNK (LS)	Loop start link
KRS	Key rate scaling
DL	Decay level
RR	Release rate
STWINH (SI)	Stack write inhibit
SDIR (SD)	Sound direct
TL	Total level
MDL	Modulation level
MDXSL	Select modulation input X
MDYSL	Select modulation input Y
OCT	Octave
FNS	Frequency number switch
LFORE (RE)	LFO reset
LFOF	LFO frequency
PLFOWS	Select LFO frequency modulation wave form
PLFOS	LFO frequency modulation level
ALFOWS	Select LFO amplitude modulation wave form
ALFOS	LFO amplitude modulation level
ISEL	Input select
IMXL	Input mix level
DISDL	Direct data send level
DIPAN	Direct data, fixed position
EFSDL	Effect data send level
EFPAN	Effect data, fixed position



Table 4.3 Individual Slot Control Register Address Map

Slot	Address	Slot	Address
0	10000 _H ~ 100017 _H	16	100200 _H ~ 100217 _H
1	100020 _H ~ 100037 _H	17	100220 _H ~ 100237 _H
2	100040 _H ~ 100057 _H	18	100240 _H ~ 100257 _H
3	100060 _H ~ 100077 _H	19	100260 _H ~ 100277 _H
4	100080 _H ~ 100097 _H	20	100280 _H ~ 100297 _H
5	1000A0 _H ~ 1000B7 _H	21	1002A0 _H ~ 1002B7 _H
6	1000C0 _H ~ 1000D7 _H	22	1002C0 _H ~ 1002D7 _H
7	1000E0 _H ~ 1000F7 _H	23	1002E0 _H ~ 1002F7 _H
8	100100 _H ~ 100117 _H	24	100300 _H ~ 100317 _H
9	100120 _H ~ 100137 _H	25	100320 _H ~ 100337 _H
10	100140 _H ~ 100157 _H	26	100340 _H ~ 100357 _H
11	100160 _H ~ 100177 _H	27	100360 _H ~ 100377 _H
12	100180 _H ~ 100197 _H	27	100380 _H ~ 100397 _H
13	1001A0 _H ~ 1001B7 _H	29	1003A0 _H ~ 1003B7 _H
14	1001C0 _H ~ 1001D7 _H	30	1003C0 _H ~ 1003D7 _H
15	1001E0 _H ~ 1001F7 _H	31	1003E0 _H ~ 1003F7 _H

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The SCSP control register allocates the interrupt register and timer register, etc., that are commonly used.

100400 H	—	—	—	—	—	—	M4	DB	—	—	—	—	MVOL[3:0]	
	—	—	—	—	—	—	—	RBL	RBP[19:13]					
	—	—	—	OF	OE	IO	IF	IE	MIBUF[7:0]					
	—	—	—	—	—	—	—	—	MOBUF[7:0]					
100409 H	MSLC[4:0]					CA[3:0]			—	—	—	—	—	—
10040A H	//													
10040F H	—	—	—	—	—	—	—	—	—	—	—	—	—	—
100410 H	DMEA[15:1]												—	
	DMEA[19:16]					DRGA[11:1]							—	
	—	GA	DI	EX	DTLG[11:1]									—
	—	—	—	—	—	TACTL[2:0]			TIMA[7:0]					
	—	—	—	—	—	TBCTL[2:0]			TIMB[7:0]					
	—	—	—	—	—	TCCTL[2:0]			TIMC[7:0]					
	—	—	—	—	—	SCIEB[10:0]								
	—	—	—	—	—	SCIPD[10:0]								
	—	—	—	—	—	SCIRE[10:0]								
	—	—	—	—	—	—	—	—	SCILV0[7:0]					
	—	—	—	—	—	—	—	—	SCILV1[7:0]					
	—	—	—	—	—	—	—	—	SCILV2[7:0]					
	—	—	—	—	—	MCIEB[10:0]								
	—	—	—	—	—	MCIPD[10:0]								
10042F H	—	—	—	—	—	MCIRE[10:0]								
100430 H														

M4:MEM4MB
OE:MOEMP
IE:MIEMP
EX:DEXE

DB:DAC18B
IO:MIOVF
GA:DGATE

OF:MOFULL
IF:MIFULL
DI:DDIR

Figure 4.3 SCSP Common Control Register



Table 4.4 SCSP Common Control Register

Designation	Contents
MEM4MB (M4)	Memory size designation
DAC18B (DB)	Use the 18 bit D/A converter on the digital output
VER	Version number
MVOL	Master volume
RBL	Ring buffer length
RBP	Ring buffer lead address
MOFULL (OF)	Output FIFO is full
MOEMP (OE)	Output FIFO is empty
MIOVF (IO)	Input FIFO overflow
MIFULL (IF)	Input FIFO full
MIEMP (IE)	Input FIFO empty
MIBUF	MIDI input data buffer
MOBUF	MIDI output data buffer
MSLC	Monitor Slot
CA	Call Address
DMEA	DMA transfer start memory address
DRGA	DMA transfer start register address
DGATE (GA)	DMA transfer gate 0 clear
DDIR (DI)	DMA transfer direction
DEXE (EX)	DMA transfer start
DTLG	DMA transfer data count
TACTL	Timer A pre-scaler control
TIMA	Timer A count data
TBCTL	Timer B pre-scaler control
TIMB	Timer B count data
TCCTL	Timer C pre-scaler control
TIMC	Timer C count data
SCIEB	Allow sound CPU interrupt
SCIPD	Request sound CPU interrupt
SCIRE	Reset sound CPU interrupt
SCILV0	Sound CPU interrupt level bit0
SCILV1	Sound CPU interrupt level bit1
SCILV2	Sound CPU interrupt level bit2
MCIEB	Allow main CPU interrupt
MCIPD	Request main CPU interrupt
MCIRE	Reset main CPU interrupt

The sound data stack is where sound data is stored. Sound data is in two generation configuration (GENERATION A and B). Each generation can control up to 32 sound data.

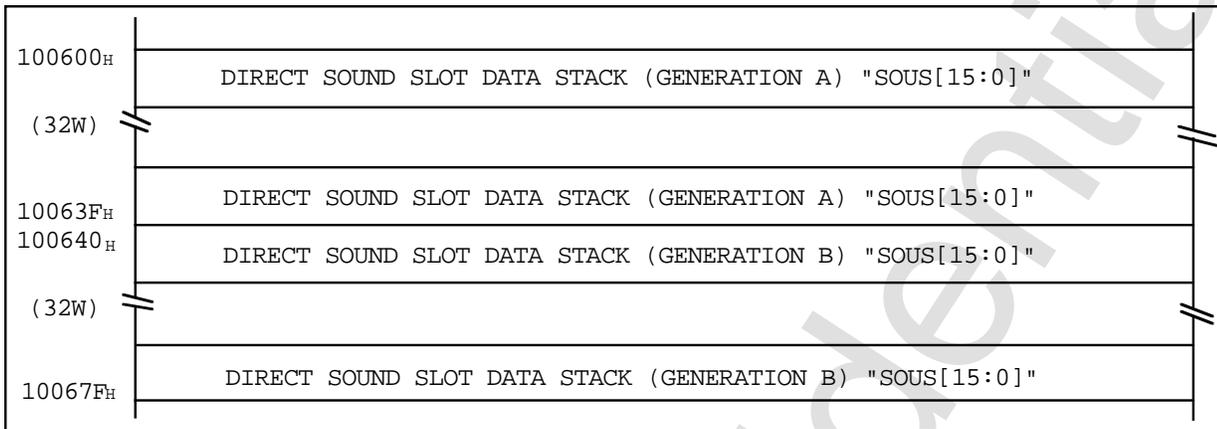


Figure 4.4 Sound Data Stack

Table 4.5 Sound Data Stack

Designation	Contents
SOUS	Sound stack



On the DSP control register, the interface block area of DSP is defined which is built within SCSP.

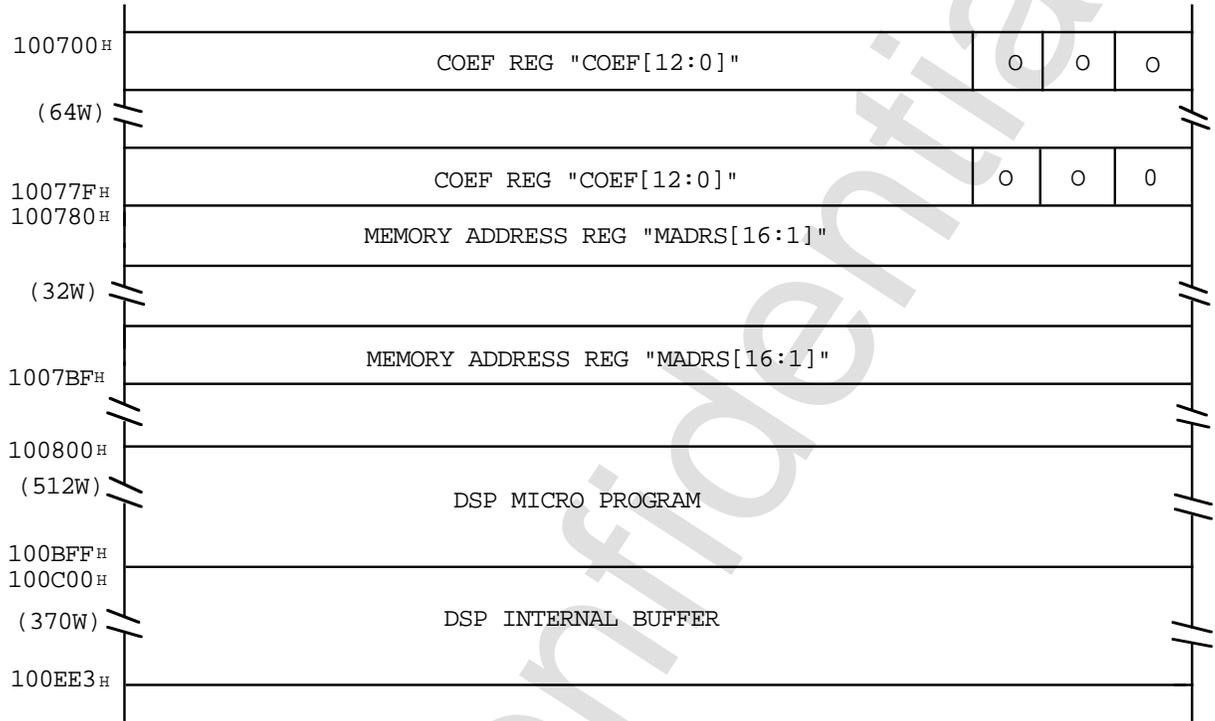


Figure 4.5 DSP Control Register

Table 4.6 DSP Control Register

Designation	Contents
COEF	DSP coefficient buffer
MADRS	Memory address register

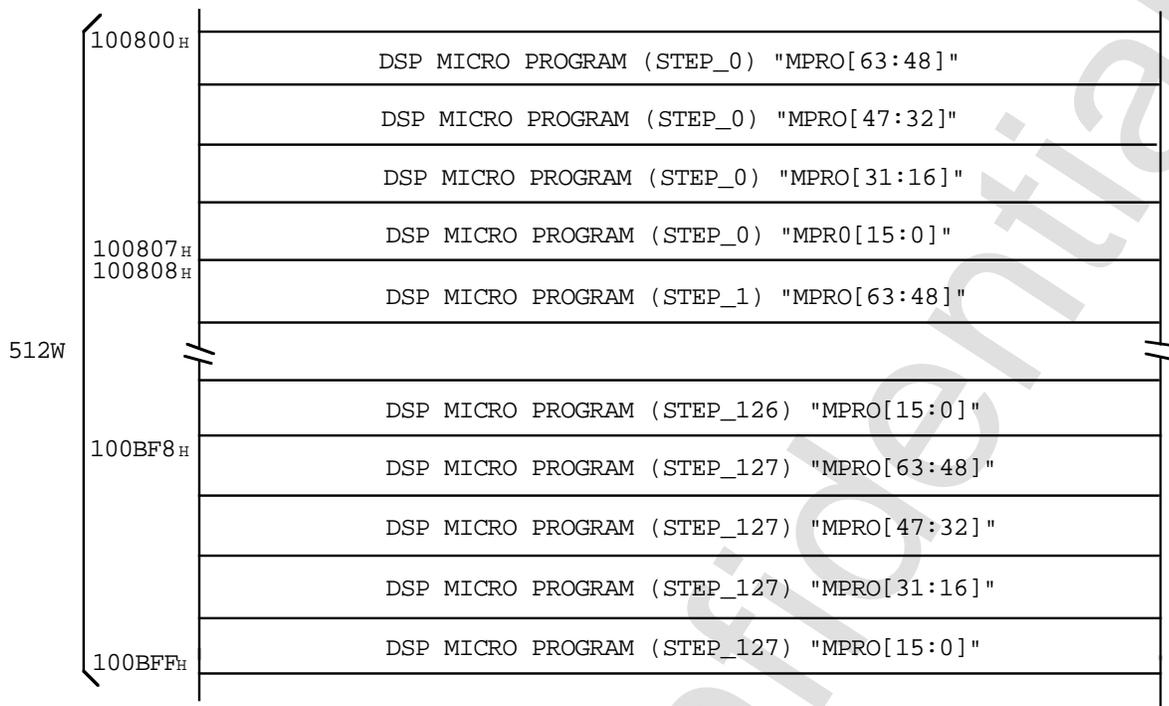
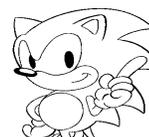


Figure 4.6 DSP Micro Program Map

Table 4.7 DSP Micro Program

Designation	Contents
MPRO	Micro program register



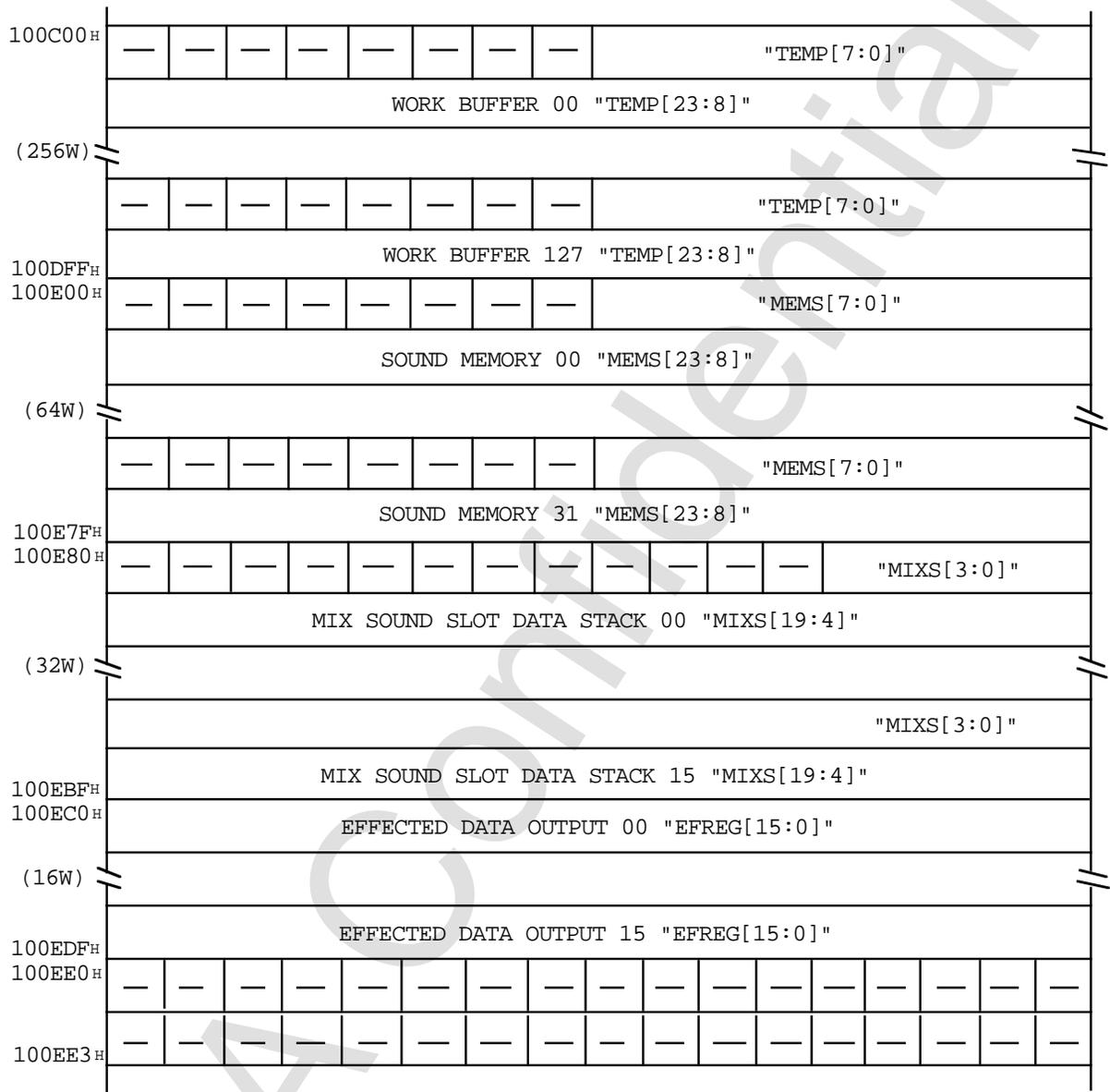


Figure 4.7 DSP Internal Buffer Map

Table 4.8 DSP Internal Buffer

Designation	Contents
TEMP	DSP temporary (universal) buffer
MEMS	Memory data stack
MIXS	Mix stack
EFREG	Effect register

4.2 Sound Source Register

The sound source (block) register is made up of the allocations shown in Table 4.9.

Table 4.9 Sound Source (Block) Register Allocation

Register Name	Bit Name
Loop Control Register	KYONEX, KYONB, SBCTL, SSCTL, SA, LSA, LEA, PCM8B, LPCTL
EG Register	EGHOLD, AR, D1R, D2R, RR, DL, KRS, LPSLNK
FM Modulation Control Register	SOUS, MDL, MDXSL, MDYSL, STWINH
Sound Volume Register	TL, SDIR
PITCH Register	OCT, FNS
LFO Register	LFORE, LFOF, ALFOWS, ALFOS, PLFOWS, PLFOS
MIXER Register	IMXL, ISEL, DISDL, DIPAN, EFSDL, EFPAN, MVOL, DAC18B
Slot Status Register	MSLC, CA
Sound Memory Config. Register	MEM4MB
MIDI Register	MIBUF, MIOVF, MIFULL, MIEMP, MOFULL, MOEMP, MOBUF
Timer Register	TACTL, TIMA, TBCTL, TIMB, TCCTL, TIMC
Interrupt Control Register	SCIPD, SCIEB, SCIRE, SCILVO, SCILV1, SCILV2, MCIPD, MCIEB, MCIRE
DMA Transfer Register	DGATE, DDIR, DEXE, DMEA, DRGA, DTLG



Below is the definition of the bit in terms of the sound generator block register, based on the classifications in Table 4.9. The symbols following the register names have the following meanings: (R) read only, (W) write only, (R/W) both read and write are possible.

When reading data from the write only register (bit), that value becomes "0B". When writing data to the read only register (bit), always write "0B".

Loop Control Register

KYONEX (W) ;KeY_ON EXecution

A "1B" written here will execute KEY_ON, OFF for all of the slots.

KYONB (R/W) ;KeY_ON Bit

Registers KEY_ON,OFF. (If you wish to KEY_ON simultaneously, the "KYONB" of the slot you want to turn ON must be set to "1B".)

Table 4.10 KYONB Function

bit	Function
0	Registers KEY_OFF
1	Registers KEY_ON

"KYONEX" and "KYONB" exist in each slot. The sequence of KEY_ON and KEY_OFF is shown in Figure 4.8.

There is no need to write a "0B" in "KYONEX" after writing a "1B". Also a "1B" in "KYONEX" is used for all slots, so you don't need to set to "1B" per a specific slot.

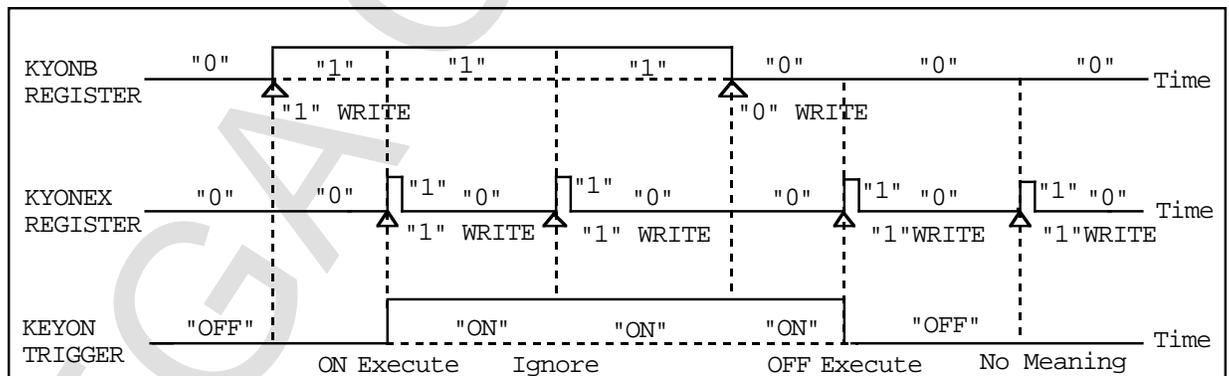


Figure 4.8 KEY_ON and KEY_OFF Sequence

SBCTL[1:0] (R/W) ;Source Bit Control

Specifies bit reversal operation of the sound input data. The reversal function becomes valid for bits with "1B" written to them.

Table 4.11 SBCTL Function

Bit	Function
SBCTL0	Reverse selection for bits other than the source wave form data sign bit.
SBCTL1	Reverse selection for the source wave form data sign bit.

SSCTL[1:0] (R/W) ;Sound Source Control

Designates the data to be used as sound input data. When using the wave form data in sound memory to produce a sound, write "0B". When this register is "1B", the slot with respect to which setting has been applied (the LFO accompanied with each slot) will output noise.

The block diagram of noise generated when "SSCTL"="1B" and the relation to LFO is shown with ALFO added in Figure 4.9

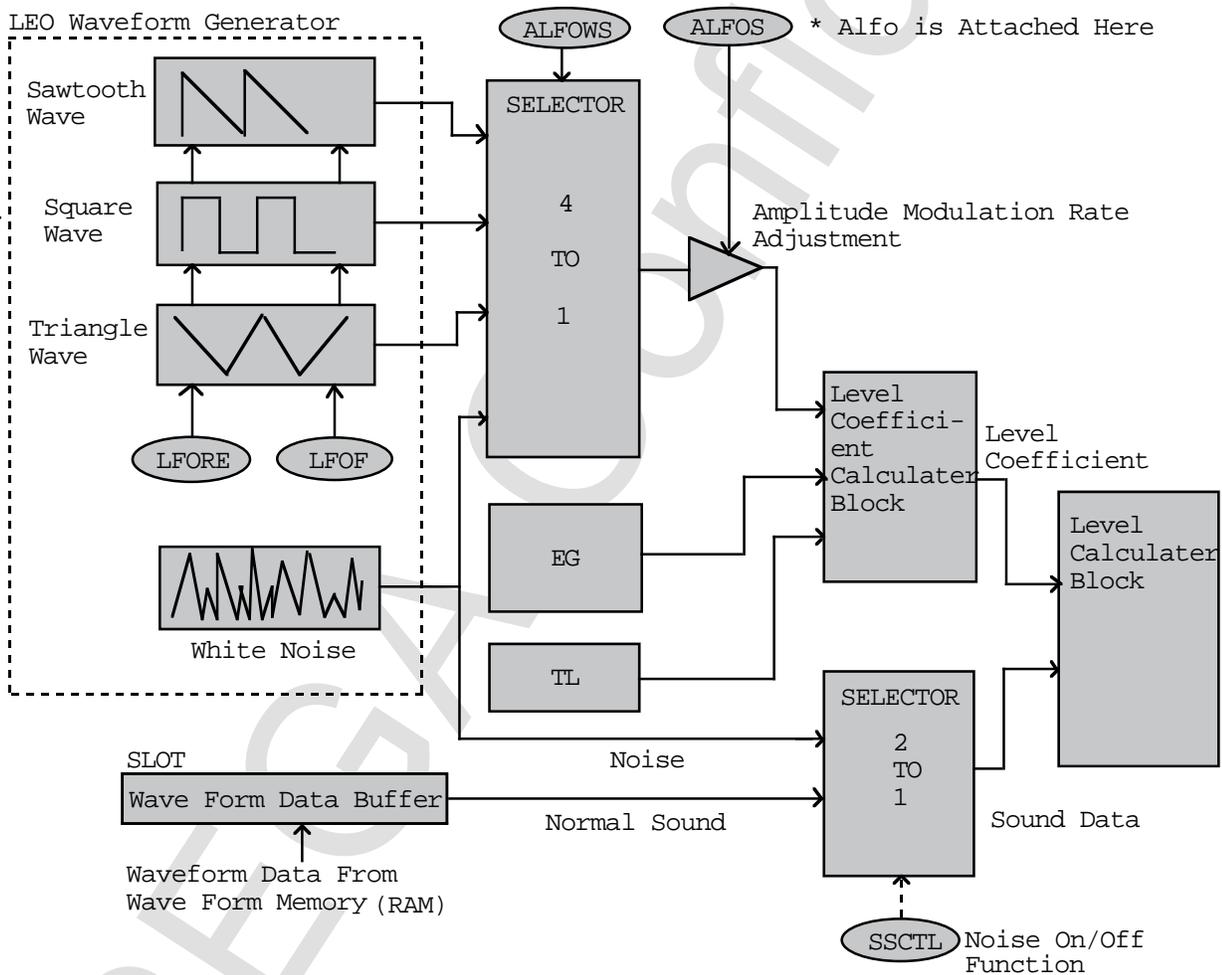


Figure 4.9 Relation of the Noise Generation Block Diagram and LFO



Each slot utilizes the LFO noise oscillator output to output noise. Here, the LFO parameters in Figure 4.9 (“LFORE”, “LFOF”, “ALFOWS”, “ALFOS”, “PLFOWS”, “PLFOS”) will not affect noise as voice data.

Also, when using LFO in low-frequency modulation with the LFO wave form selection at noise (ALFOWS=“3H”, or PLFOWS=“3H”) as in Figure 4.9, reset with the “LFORE” will not function. The frequency also cannot be changed. This means that when selecting noise, the LFO can select wave form but cannot change in terms of other parameters.

Table 4.12 SSCTL Internal Configuration

SSCTL	Data Used
0	External DRAM data
1	Internally generated data (noise)
2	Internally generated data (ALL “0”)
3	Cannot be used

SA[19:0] (R/W) ; Start Address

Specifies the waveform data start address in byte address when using memory waveform data to generate sound. However, if the waveform data is “16 bit PCM” (“PCM8B”=“0”), always set the register’s lsb (SA0) to “0B”.

LSA[15:0] (R/W) ; Loop Start Address

Represents the sound data loop start address in sample count from the “SA”.

LEA[15:0] (R/W) ; Loop End Address

Represents the sound data loop end address in sample count from the “SA”.

PCM8B (R/W) ; PCM 8Bit

Designates the format of the wave form data.

Table 4.13 Types of Sound Data

PCM8B	Sound Data
0	16Bit PCM data 2’S complement
1	8Bit PCM data 2’S complement

LPCTL[1:0] (R/W) LooP ConTroL
 Sets the loop format.

Table 4.14 Types of Loops

LPCTL	Loop Format
0	Loop OFF
1	Normal loop
2	Reverse loop
3	Alternative loop

Loop processing or sound memory access will end with either of the following two conditions:

- (1) After release, when the attenuation volume reaches maximum, or
- (2) When the loop is OFF, and the read point reaches the loop end point.

When using normal or reverse loops, set the data corresponding to "SA+LSA" (loop start address) and the "SA+LEA" (loop end address) to the same value. Using the same method with the alternative loop can make the pitch the same as the normal or reverse loops.

Figure 4.10 shows specific examples of loop types.

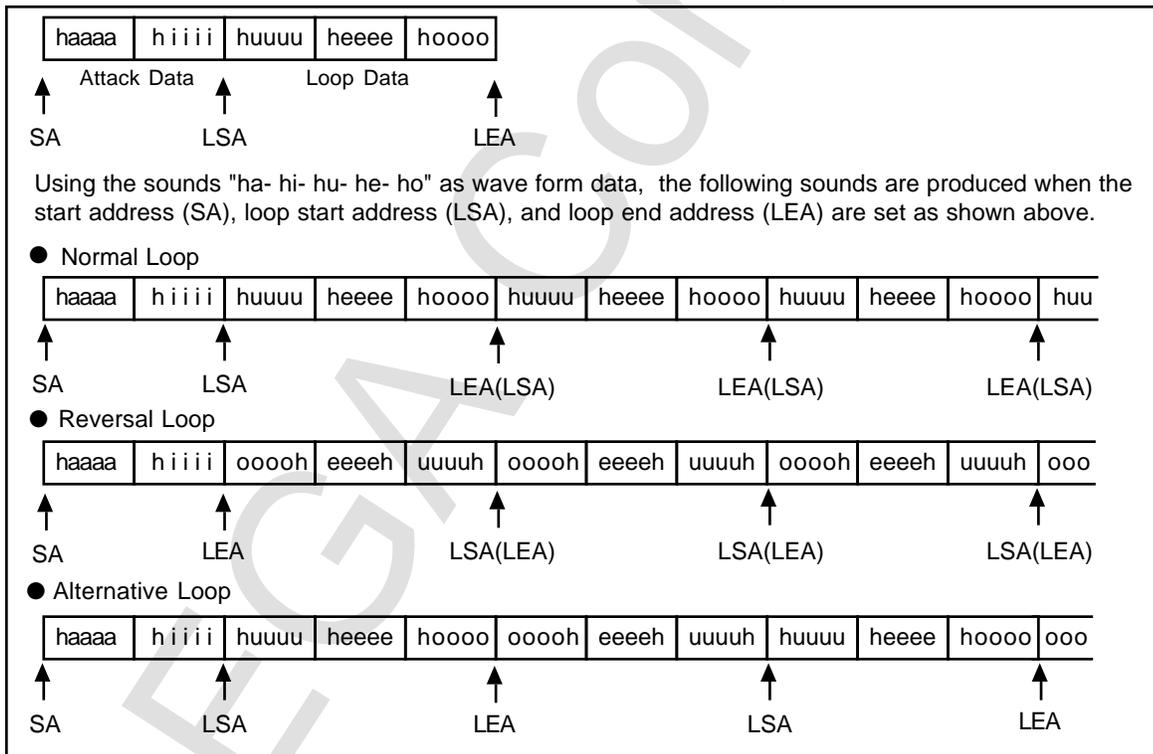


Figure 4.10 Loop Types



The wave form for each of the normal loop, reversal loop and the alternative loop can be represented as in figure 4.11. Note that the uuuu, eeee, oooo wave forms are reversed from huuh, heeh, hooh wave forms.

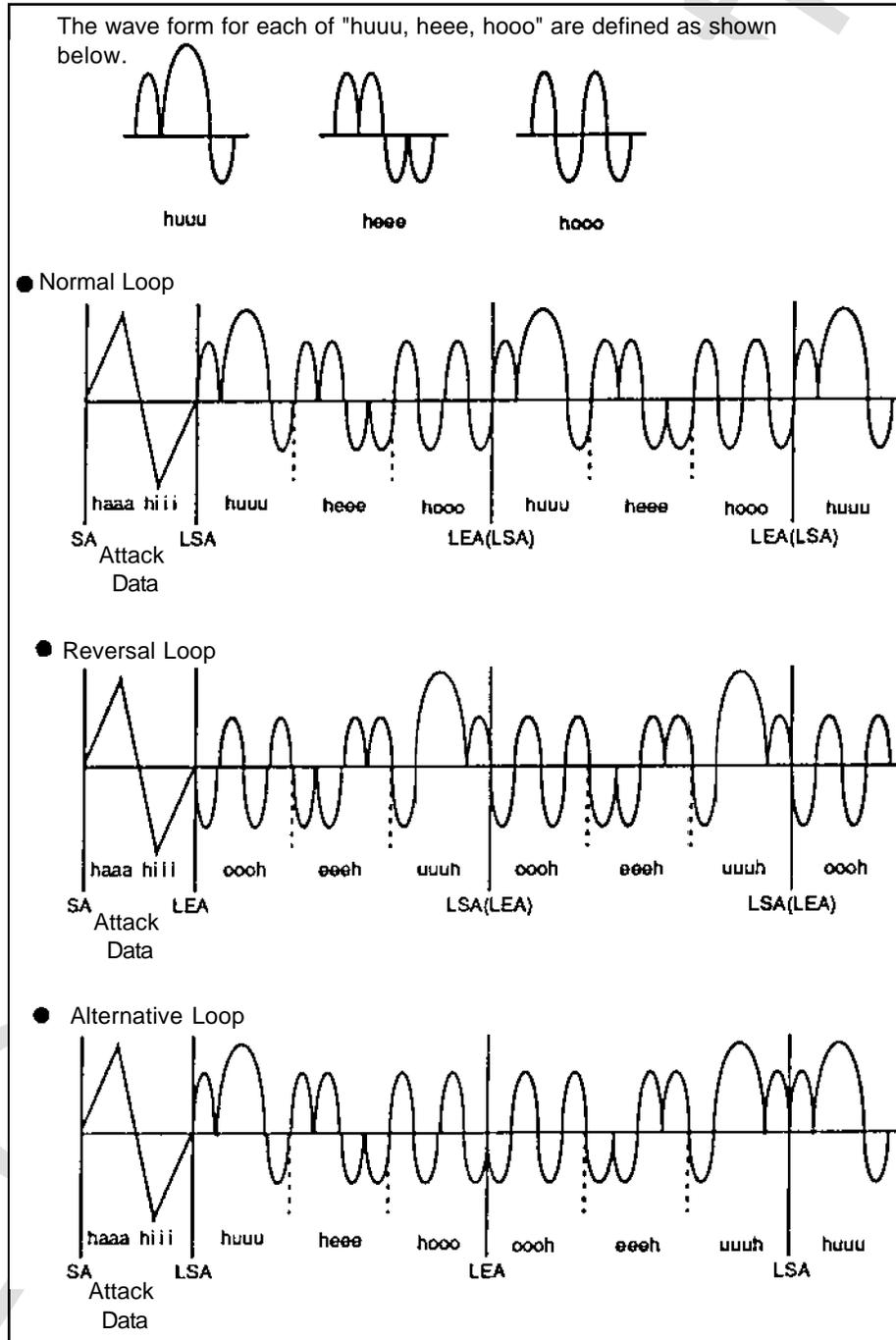


Figure 4.11 Loop Wave Forms

EG Register

EG represents the change occurred by sound attenuation time, and has the following four states.

- Attack State (Attack segment)
Indicates the start of the sound (start up).
- Decay 1 state (Decay 1 segment)
Indicates attenuation from the maximum volume.
- Decay 2 state (Decay 2 segment)
Indicates attenuation even lower than decay 1.
However, if DR2 is set to "0", the sound is maintained rather than attenuated.
- Release state (Release segment)
Indicates the attenuation until the sound disappears after KEY_OFF.

However, in all cases of sound generation state, EG does not necessarily go through all four states. Depending on the timing of KEY_OFF various envelope curves are drawn. An example is shown below.

- (a) When KEY_OFF is executed during attack state transition (Figure 4.12).

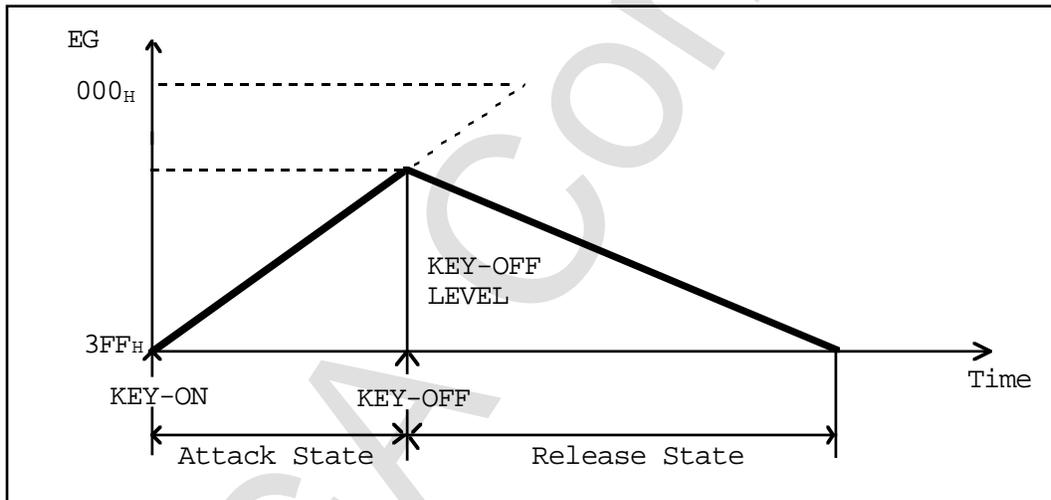


Figure 4.12 KEY_OFF During Attack State Transition

When KEY_OFF is executed, from that level (KEY_OFF LEVEL) the sound is attenuated following the release rate ("RR") setting. Therefore, in this case, the envelope curve skips the decay 1 and decay 2 states. In this case, EG value does not reach "000H", rather, with KEY_OFF as a turning point, it increases to "3FFH".



(b) When KEY_OFF is executed during decay 1 segment transition.
(Figure 4.13).

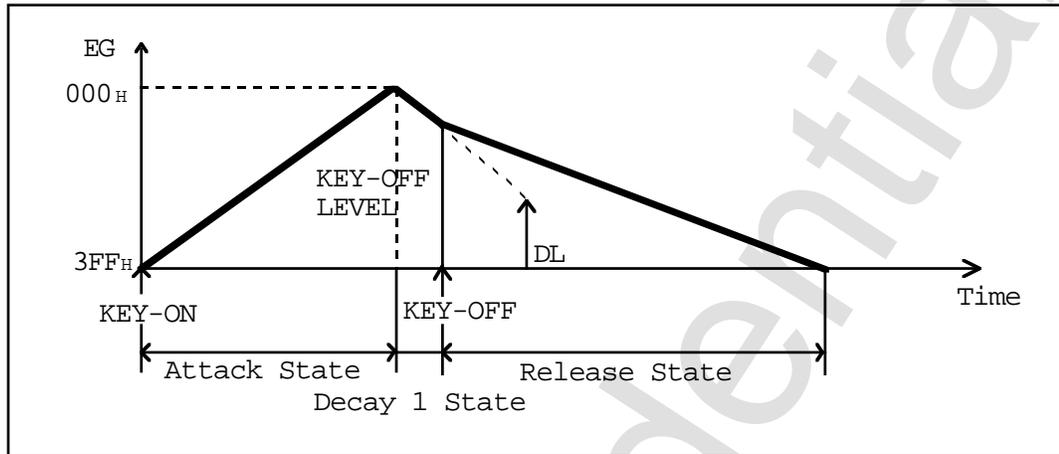


Figure 4.13 KEY_OFF During Decay State Transition

When the decay 1 state starts, it attenuates towards the DL (decay level) following the D1R (decay 1 rate) set value.

If KEY_OFF is executed during the above operation, the sound starts to attenuate from the level KEY_OFF was executed (KEY_OFF LEVEL) according to the value set in RR (release rate).

AR[4:0] (R/W) ; Attack Rate

Designates the change volume of EG in the attack state. When "AR"="00H", the change volume (level attenuation volume) is minimum (0). When "AR"="1FH", the change volume (level attenuation volume) is maximum (MAX).

EGHOLD (R/W) EG HOLD mode

Designates whether to maintain or change the attack value. As shown in Figure 4.12, when this bit is "1B" the attack value is held at "000H". Also when this bit is "0B", it changes according to the value designated by the AR register. In the hold mode, the time that EG retains "000H" (time until it reaches segment 2) is determined by the "AR" value.

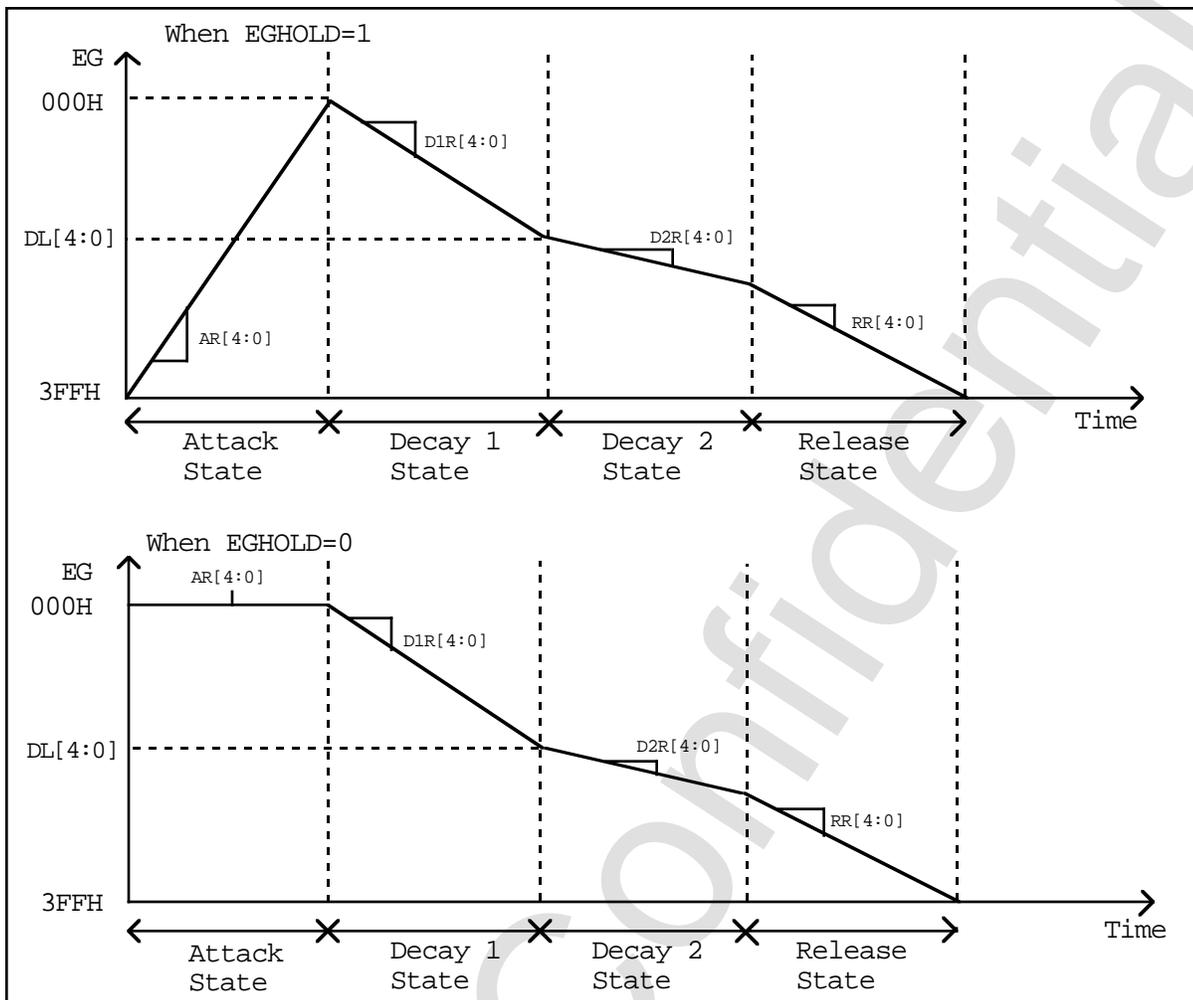


Figure 4.14 Change in the Attenuation Volume

D1R[4:0] (R/W) ; Decay-1 Rate

Designates the EG change volume in the decay 1 state. When "D1R"="00H", the change volume (level attenuation volume) is minimum (0). When "D1R"="1FH", the change volume (level attenuation volume) is maximum (MAX).

D2R[4:0] (R/W) ; Decay-2 Rate

Designates the EG change volume in the decay 2 state. When "D2R"="00H", the change volume (level attenuation volume) is minimum (0). When "D2R"="1FH", the change volume (level attenuation volume) is maximum (MAX).



RR[4:0] (R/W) ; Release Rate

Designates the change volume of the EG in the release state. When "RR"="00H", the change volume (level attenuation volume) is minimum (0). When "RR"="1FH", the change volume (level attenuation volume) is maximum (MAX).

DL{4:0] (R/W) ; Decay Level

Designates the upper 5 bit of the attenuation level (EG) that moves from decay 1 state to decay 2 state. When the upper decay 1 state attenuation level upper 5 bits match the DL value, the state moves to decay state 2. When "DL"="00H", the level is maximum (MAX). When "DL"="1FH", the level is minimum (MIN).

KRS[3:0] (R/W) Key Rate Scaling

Designates the level of EG key rate scaling condition. 00H indicates minimum scaling; 0EH designates maximum scaling. When set to 0FH, it designates that scaling OFF.

LPSLNK (R/W) ; LooP Start LiNK

The function of the "LPSLNK" (loop start link) is to synchronize the transition to decay 1 state from the loop start and EG attack states.

When "LPSLNK" ="0" there is no relation between the EG state transition and the loop start point position. When "LPSLNK" ="1", the following two changes can be seen.

- 1) When, in terms of time, EG reaches "000H" in the attack state faster than when the wave form read address reaches the loop start point ("SA"+"LSA") (Figure 4.15). In this case, EG will reach the MAX level first (P.1). However, since moving to the next segment is not possible (decay 1 state) until the wave form read address reaches the loop start point, EG is maintained at MAX level. Next, when the wave form read address reaches the loop start point (P.2) EG will move to the decay 1 state.

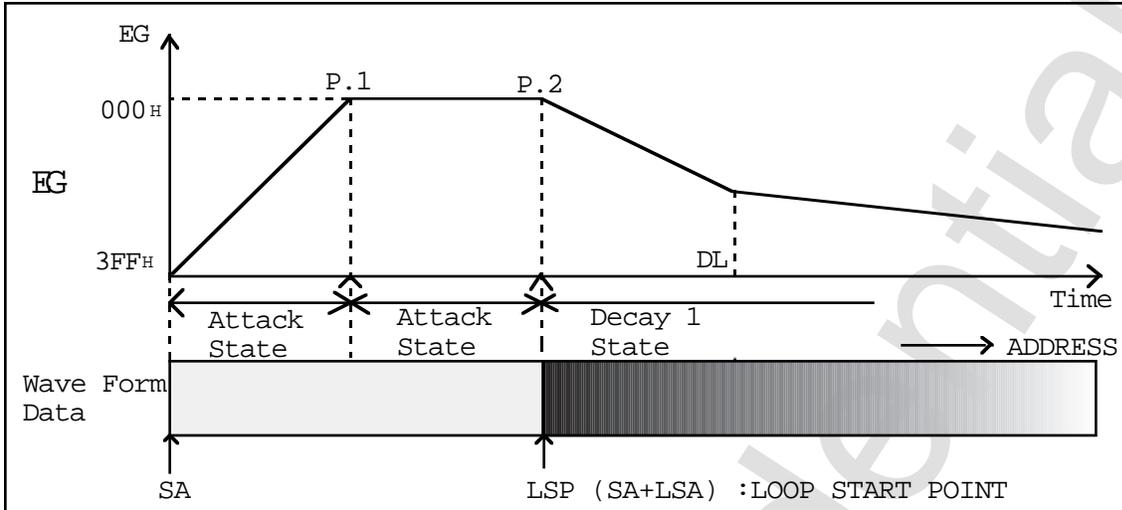


Figure 4.15 Transition (1) from the Attack State to Decay 1

2) In terms of time, EG reaches "000H" in the attack state slower than when the wave form read address reaches the loop start point ("SA"+"LSA"). This pattern has an additional two patterns.

- When the wave form read address reaches the loop start point, and the "SCL" at this point (EG level) is larger than the "DL" (decay level) (reversed when compared to the actual EG level). (Figure 4.16)

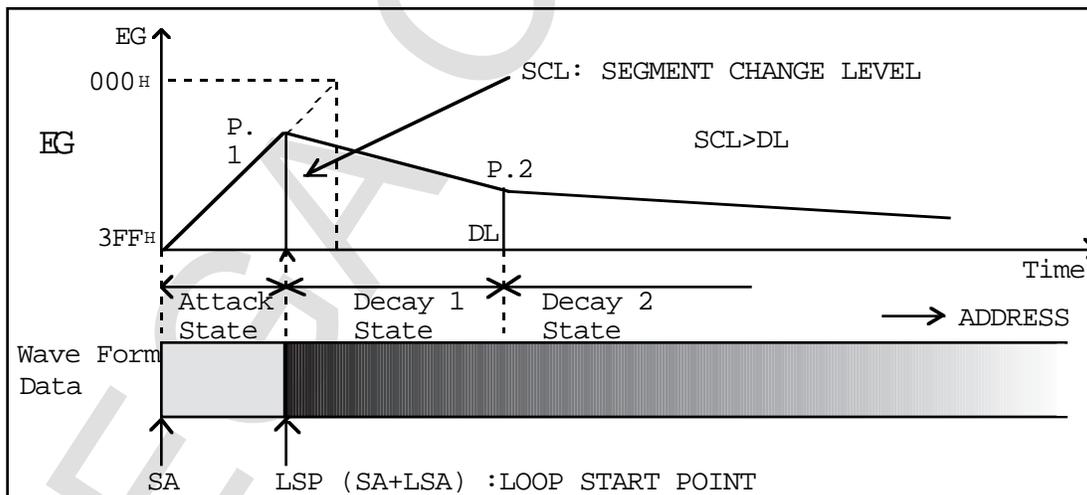


Figure 4.16 Transition (2) from the Attack State to Decay 1



After the wave form read address reaches the loop start point, the EG will shift to the decay 1 state (P.1). Next, when the EG value reaches the DL (decay level), it will shift to the decay 2 state (P.2).

- When the wave form read address reaches the loop start point, and the "SCL" at this point (EG level) is smaller than the "DL" (decay level) (reversed when compared to the actual EG level). (Figure 4.17)

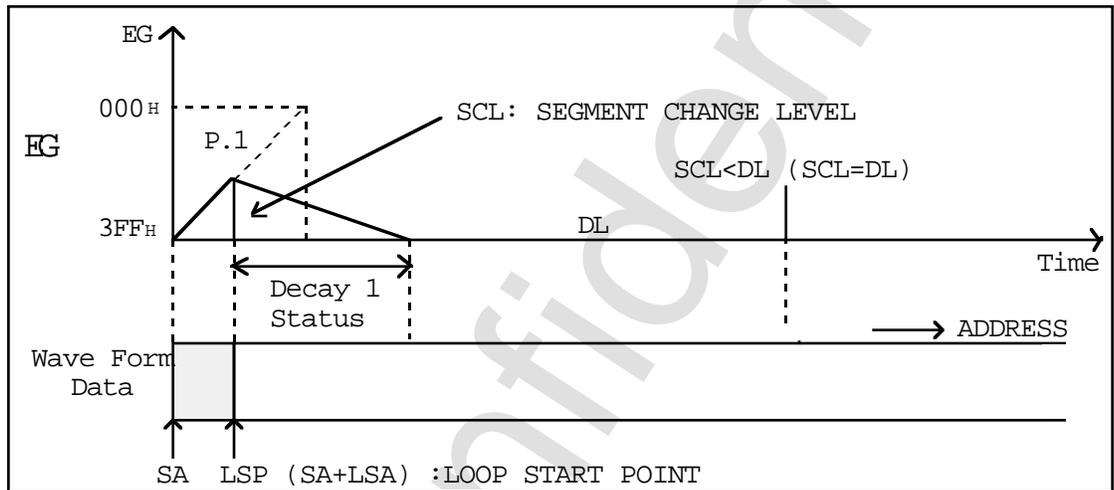


Figure 4.17 Transition (3) from the Attack State to Decay 1

After the wave form read address reaches the loop start point, the EG will shift to the decay 1 state (P.1). After this, since the EG value will no longer reach the "DL" (decay level), it will continue to retain level 0 (EG value is "3FFH") without moving to decay 2 state.

FM Modulation Control Register

SOUS[15:0] (R/W) ; SOUnd Stack

This is a data buffer for a slot output. It is a 64 word ring buffer that can hold two generations of sampling data. This is used in addition to processing between slots and modulation.

MDL[3:0] (R/W) ; MoDuLation level

Specifies the effects (modulation rate) of modulation to the modulation input source.

Table 4.15 Modulation Rate According to the Register Set Value

MDL [3:0]	0 ~ 4	5	6	7	8	9	A	B	C	D	E	F
Modulation Rate ($\pm n \pi$)	0	1/16	1/8	1/4	1/2	1	2	4	8	16	32	64

MDXSL[5:0] (R/W) ; MoDuLation input-X SeLect

Designates the source to be used as modulation input X. The modulation input X source slot is specified by two complimentary numbers showing the relative number from the current slot. Also, when this value is within the range "1CH" to 3BH" it shows the latest generation; if out of range it shows one sample past the latest generation. (See Table 4.16)

MDYSL[5:0] (R/W) ; MoDuLation input-Y SeLect

Designates the source to be used as modulation input Y. The modulator input Y source slot is indicated by two complimentary numbers showing the relative number from the current slot. Also, when this value is within the range of "1CH" to 3BH" it represents the latest generation; if out of range it represents 1 sample past the latest generation. (See Table 4.16)

STWINH (R/W) ; Stack Write INHibit

When this bit is "1B", writing the slot output to the direct data stack ("SOUS") is inhibited. Normally "0B" is fine.



Using the FM Sound Generator Method

The FM sound generator method is a method in which modulation is performed by adding different wave form generator output to the wave form generator phase input. FM voice mixing is accomplished by a wave form calculator unit called a slot. Figure 4.18 shows a block diagram of a slot.

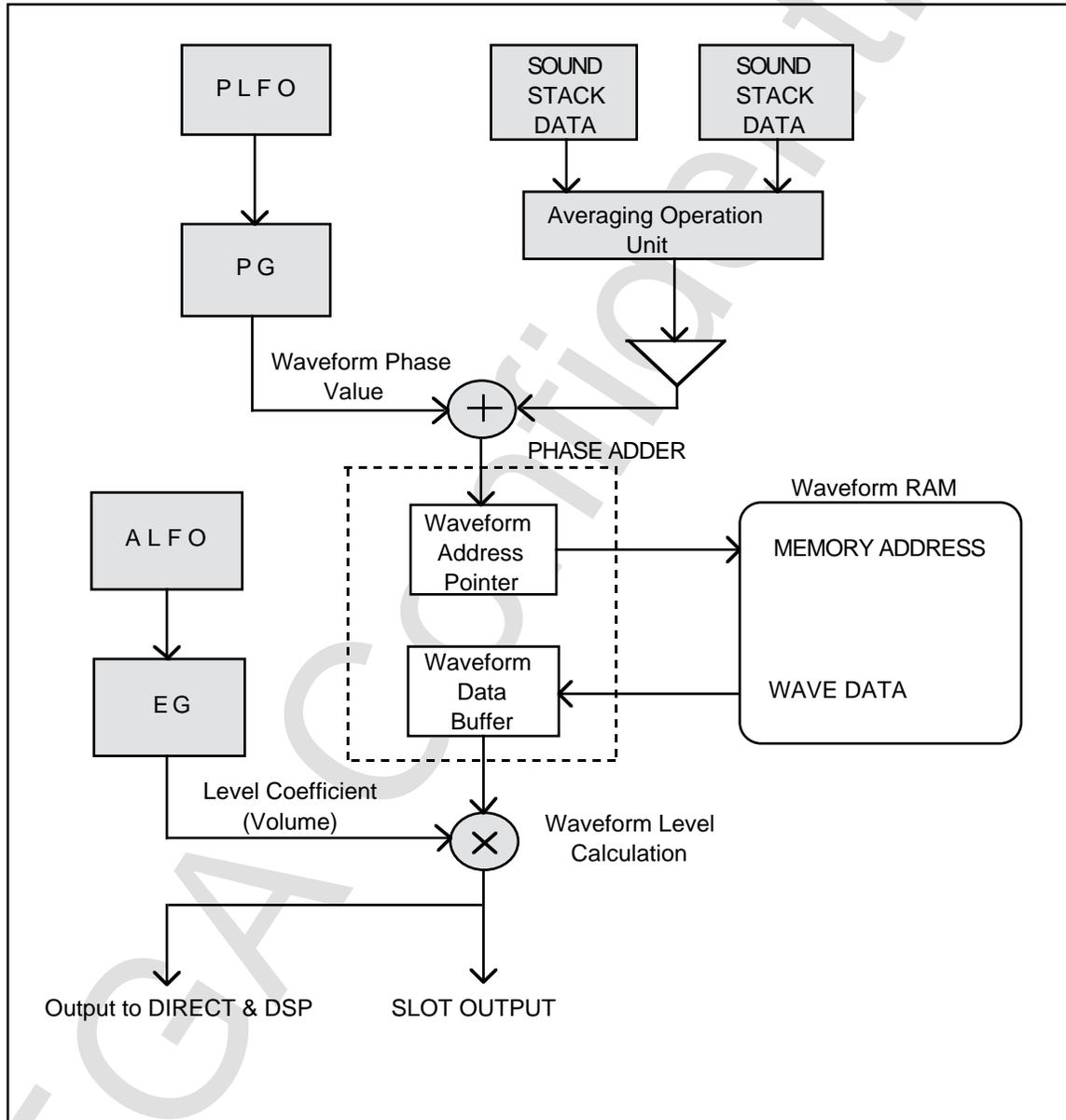


Figure 4.18 SLOT Block Diagram

To output a sound from a slot, wave form data required to run the slot is written to the wave form RAM. Sound is started by the wave form read address, loop start address, loop end address, sound pitch, sound level, setting the various EG settings, and turning KEY_ON. SCSP operates as one large cycle of 22.68[μ sec] as a whole.

The PG (Phase Generator) controls the reading rate of the waveform, which is determined by the sound generation frequency setting. As long as the sound frequency isn't changed, the PG creates a fixed value per 1Fs cycle (22.68[μ sec] = 1/44.1KHz), and it does a cumulative addition per Fs cycle and outputs. As shown in Figure 4.19, this is added to each address pointer. The resulting values are output to the address bus as wave form address and the wave form data in RAM corresponding to those wave form addresses are read.

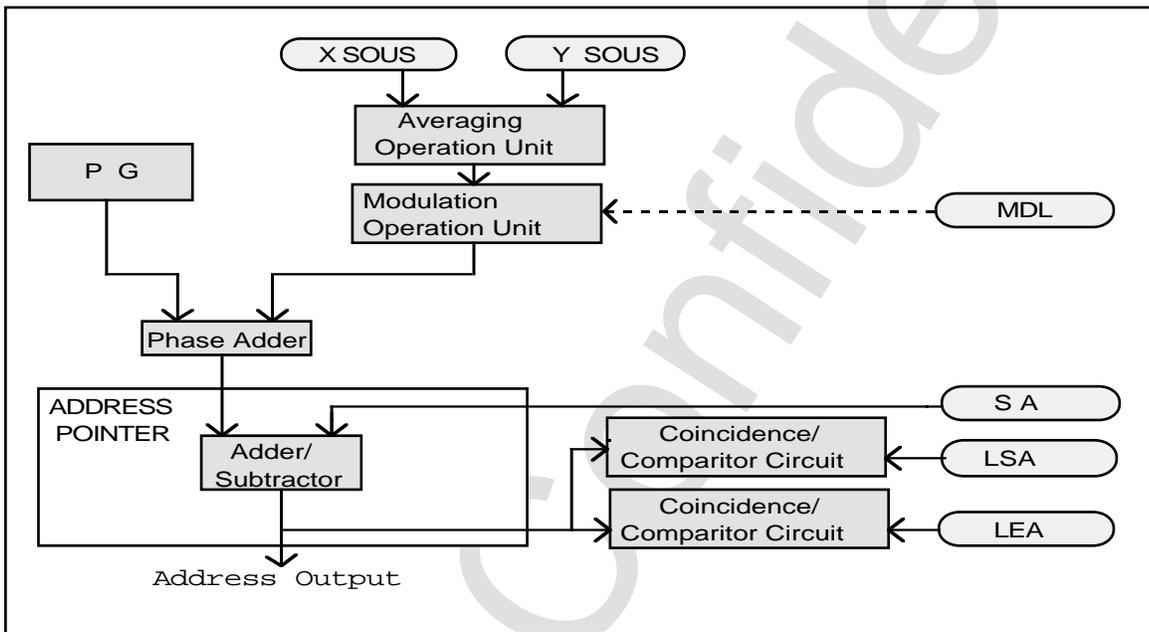


Figure 4.19 Wave Form Address Generator

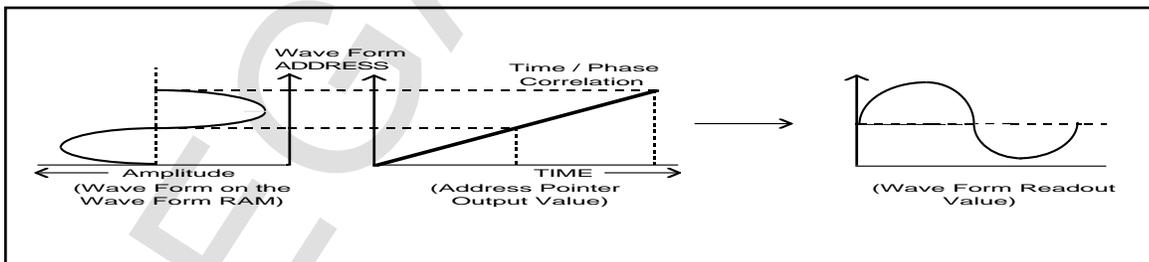


Figure 4.20 Creating Wave Form Addresses and Reading Wave Form Data



When the oscillation frequency is constant (PG internal generations are constant for all F_s cycles), the address pointer output becomes a straight line (as shown in Figure 4.20), and can be read without changing wave form on the wave form RAM.

The address pointer may appear to be a straight line, but because calculations are performed per $1F_s$ cycle, they will create a line like the stairs shown in Figure 4.21. These steps become the PG internal generator value.

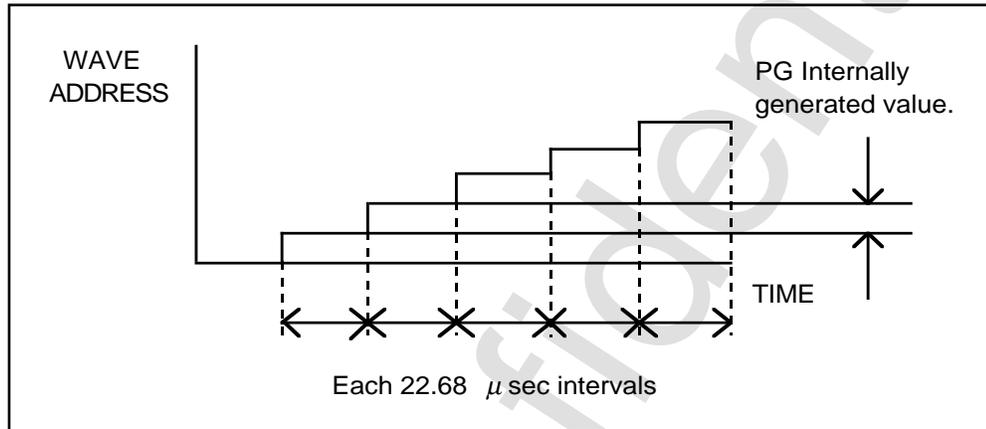


Figure 4.21 Expanded Address Pointer Output Diagram

If the frequency is increased, the PG creation value will increase, causing sharp slope for the PG output and address pointer output value (function).

On the other hand, if the frequency is decreased, the PG creation value will be reduced, causing a softer slope for the PG output value and address pointer output value (function) as shown in Figure 4.22.

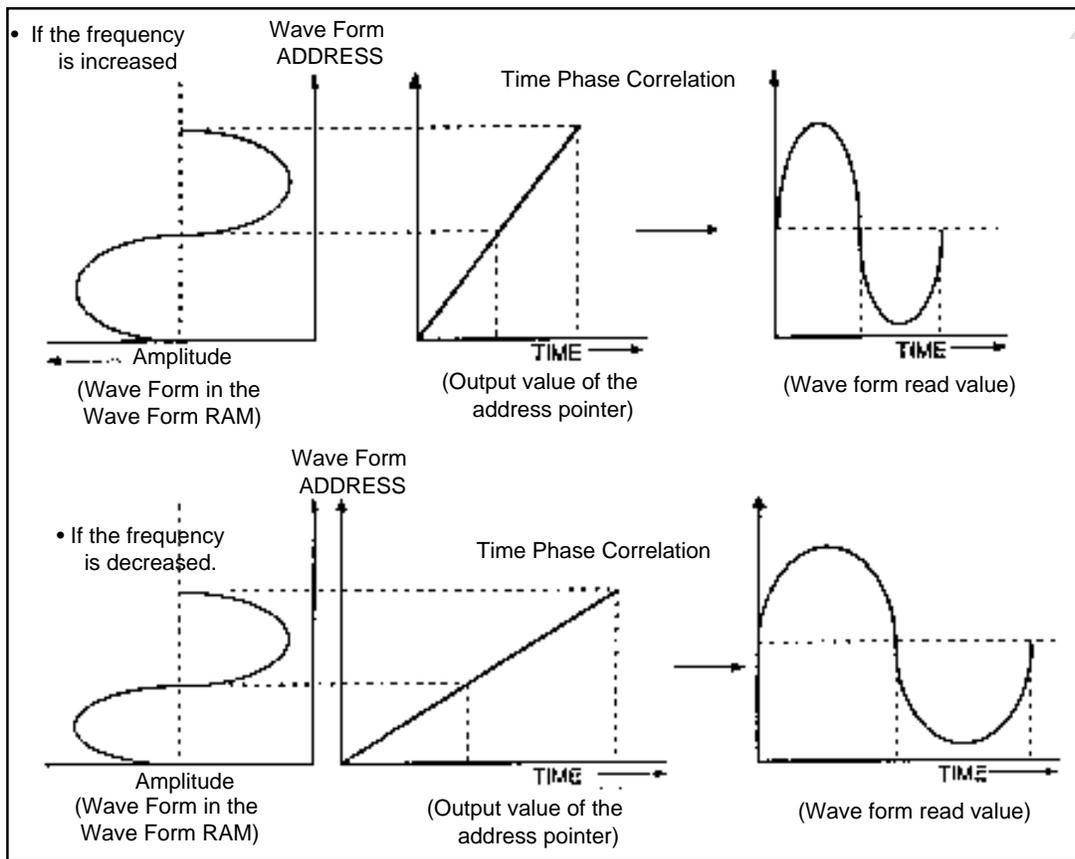


Figure 4.22 Frequency Address Pointer Output Value

When FM voice mixing is performed, short cycle wave forms are looped. Therefore, the output value (function) of the address pointer is similar to Figure 4.23 below.

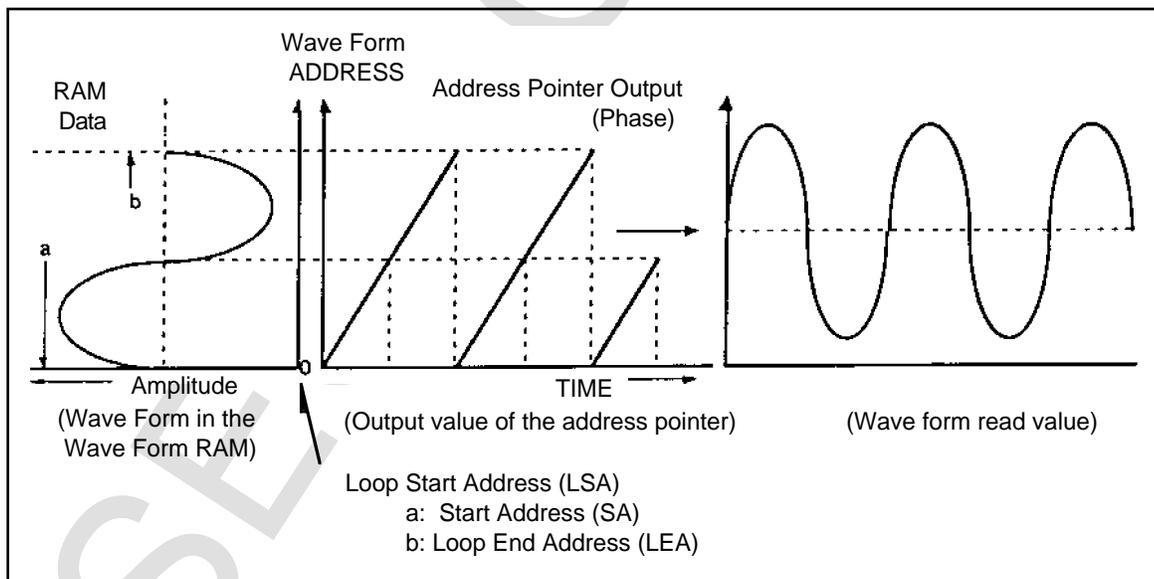
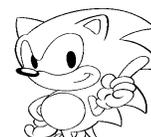


Figure 4.23 Address Pointer Output Value During FM Voice Mixing Execution(1)



The start address is set in address “a”, while the loop start address is set to “0000H”. The wave form read address and the wave form loop start address are both set to the same address. Furthermore, the “b” address is set with the loop end address, so it becomes the wave form loop end address. Results are shown in Figure 4.23.

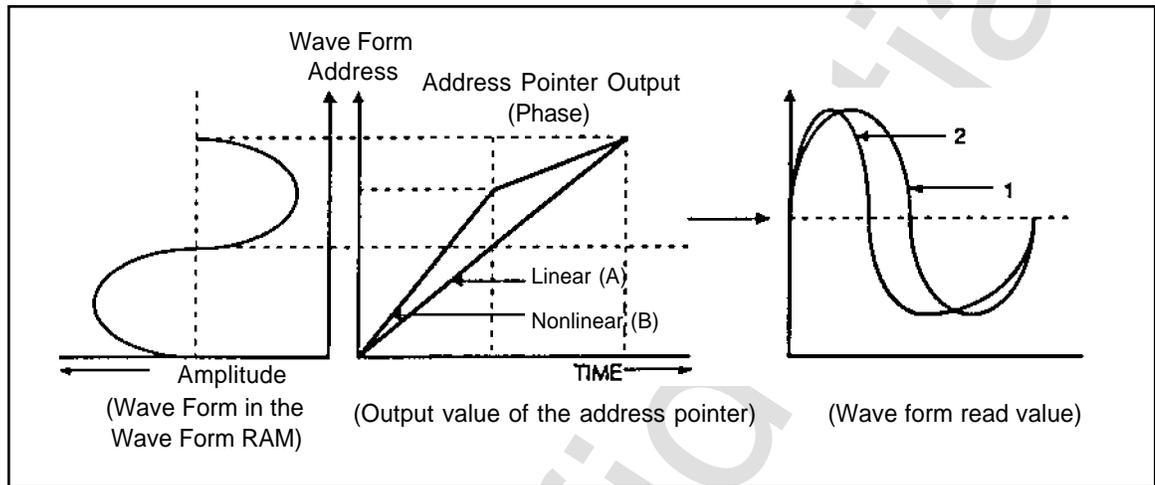


Figure 4.24 Frequency Address Pointer Output Value During FM Voice Mixing Execution (2)

If you take the address pointer values and plot them linearly (Wave form A) it would show the RAM wave form output (wave form 1). However, if it is a non-linear function (wave form B) then the wave form read method will be different and (2) wave form will be output. If the wave form changes in this manner, the tone of the sound will change.

The FM voice mixing method is applied by changing the phase value over time to distort the wave form. The FM voice mixing method uses the address pointer output value (phase value) in a non-linear form. Actually, it uses the method which adds the output values of other (in some cases, it adds its own) slots shown in Figure 4.24 where the output values from other slots (same slot depending on conditions) are added. Up until this point, normal mode has been used, but it is possible to set reverse loop and alternative loop by changing the “LPCTL” register. Loop data that can be designated in “LPCTL” register are shown in Figures 4.25, 4.26 and 4.27.

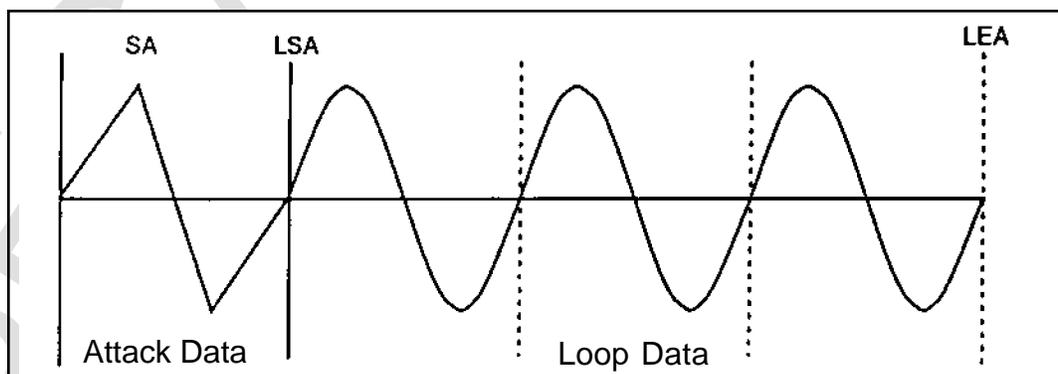


Figure 4.25 Normal Loop

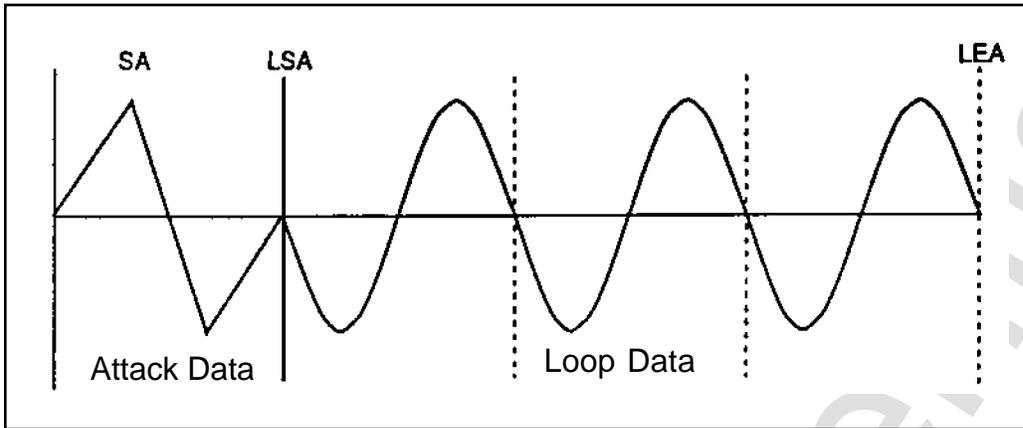


Figure 4.26 Reverse Loop

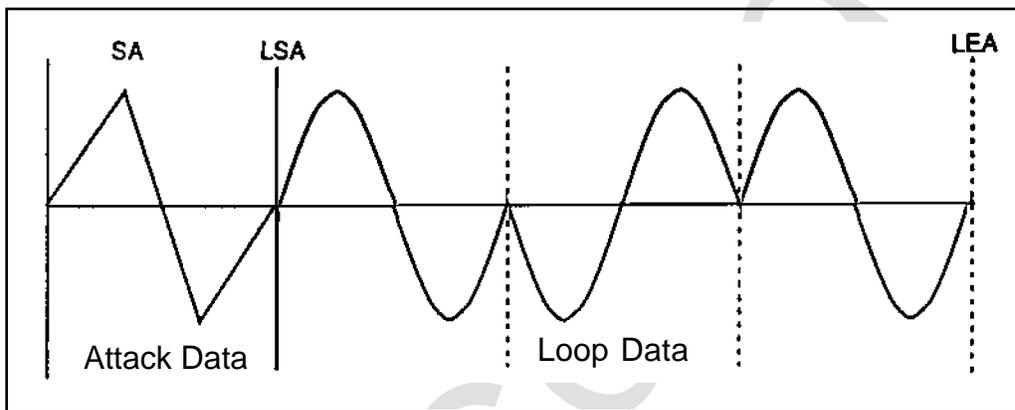


Figure 4.27 Alternative Loop

Because the premise is that the data corresponding to "LSA" and "LEA" to be the same in terms of normal loop and reverse loop, create the loop data by copying "LSA" data to "LEA".

The set value of alternative loop must always be set so that "LSA" < "LEA." (If in any of the loops "LSA" > "LEA", operation can not be guaranteed.) Also, by placing data with overlapping loop start point and loop end points the alternative pitch can be made to match the pitch in other loop modes. When all of the wave form loop modes are limited to the alternative loop, data corresponding to "LSA" and "LEA" need not be the same.



Figure 4.28 shows how the actual FM voice mixing is performed in a block diagram.

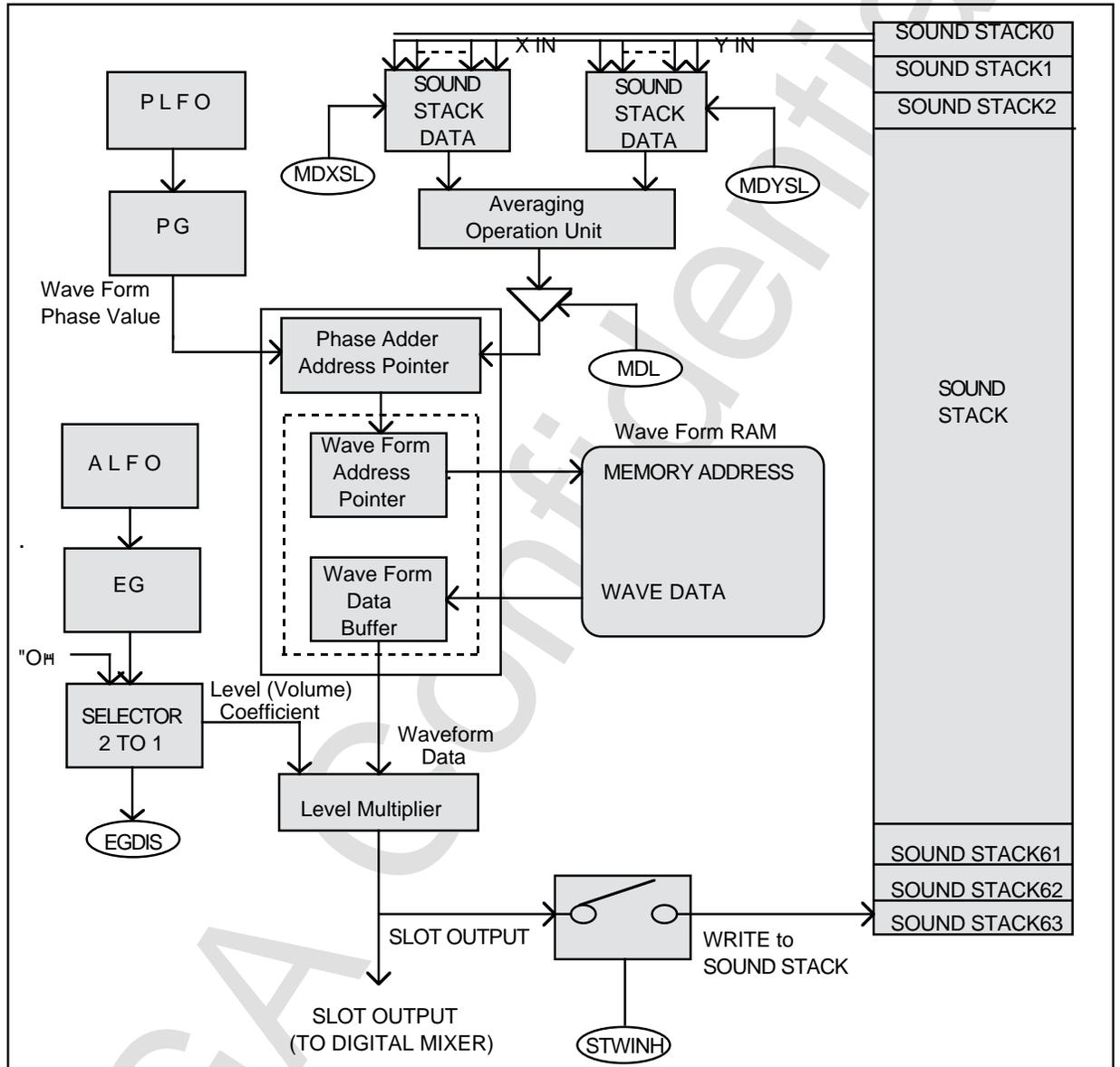


Figure 4.28 FM Sound Generator Configuration

Each block in the FM sound generator configuration diagram is explained below.

SOUND STACK

Stores the output of each slot. The output values (sound stack internal data) from two slots can be input into the SCSP slot.

Averaging Operation Unit

Each slot has X and Y modulator input slots. These values must be added together to combine them into one. To avoid having the results of the addition go into overflow, the two input values are multiplied by 1/2 before adding them together. By incorporating this method, the SCSP can take two input values and convert them into one output data.

By showing input data as XD and YD and the output data as ZD, the following equation can be produced.

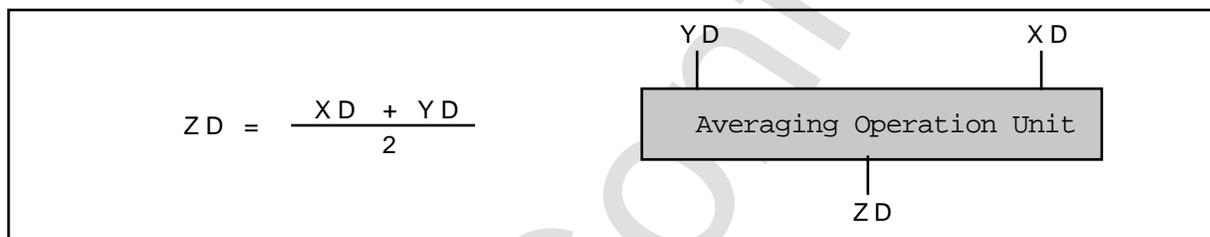


Figure 4.29 Averaging Operation Equation

Because this equation is the same as the equation for obtaining the average value, this block is called the averaging operation unit.

MDL[Modulation Level (Variation Amount)]

Used to adjust the degree of the FM effect through the external slot input.

Phase Adder

Adds (subtracts) the phase value generated by PG (phase generator), and the phase value generated by the MDL calculation after passing through the averaging arithmetic unit.



Level Multiplier

Multiplies the wave form read from the wave form memory, the ALFO, TL (Total Level), and the level coefficient generated by the EG. Also adjusts the actual wave form output level.

Wave Form RAM

RAM for sound.

Wave Form Address Pointer

This block generates the actual wave form memory address by taking the SA (start address) set for each slot and the wave form phase value output from the phase adder and adding or subtracting them. The loop control and the loop status determines whether to add or subtract.

Wave Form Data Buffer

Memory used for temporary storage of wave forms read from the wave form memory.

PG (Phase Generator)

Controls the read speed of the sound frequency wave form. (Actually, it performs skip reading.)

EG (Envelope Generator)

Generates the time-based variations of the value (envelope curve) based on each rate or level setting. The value created here is multiplied with the wave form data after it is sent to the level multiplier. This causes timing changes in the wave form output.

The connections between the slots (actually, connections between the sound stack and the slots) are important for FM voice mixing. The slot calculations in Figure 4.28 FM voice mixing diagram can be shown like Figure 4.30 below.

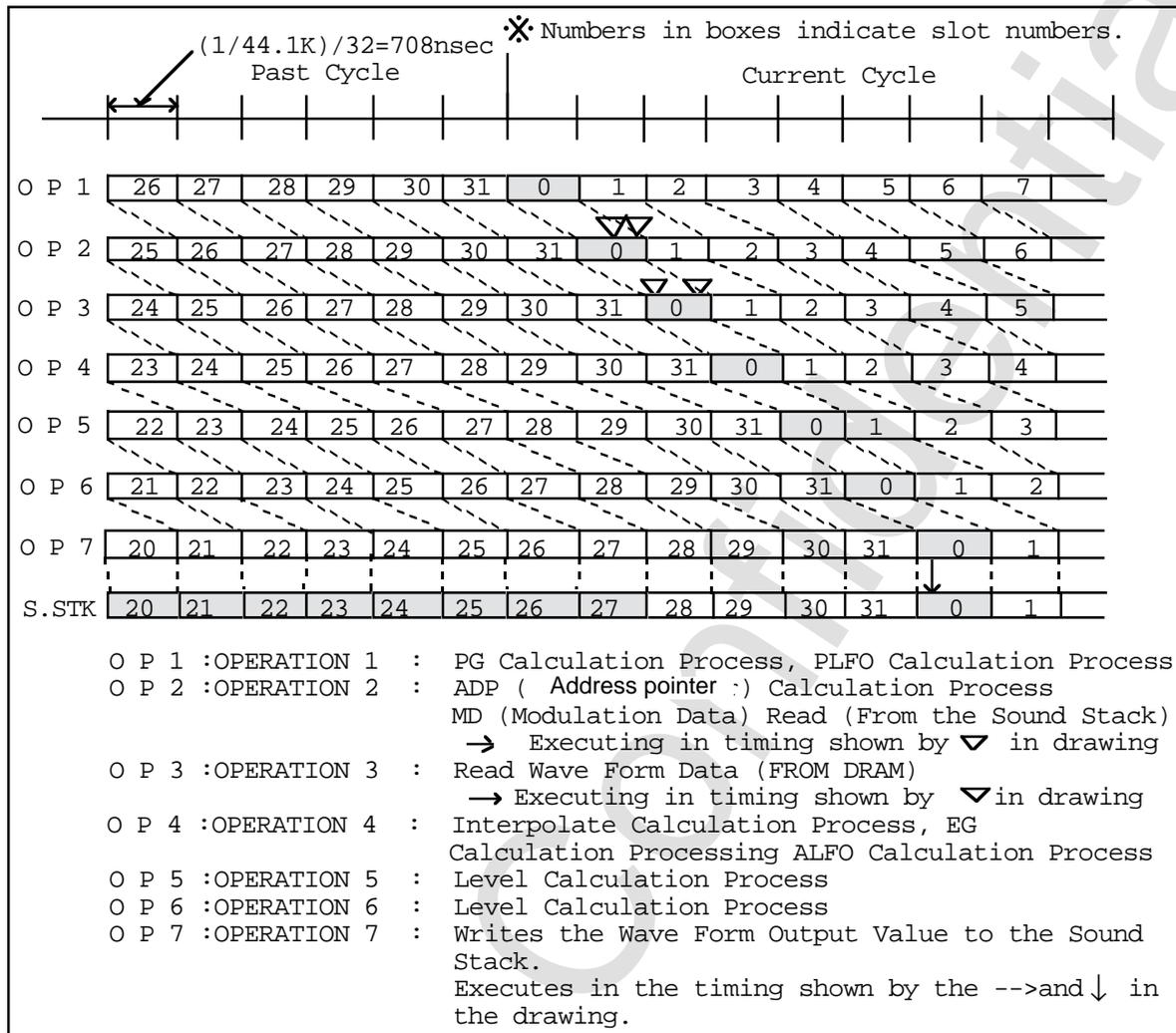


Figure 4.30 Slot Calculation and Sound Stack State

Slot calculation starts with the PG calculation process and the PLFO calculation process (OP1), with the remainder executed in the following cycle: ADP (address pointer) calculation processing and MD (modulation) data read (OP2), wave form data read (OP3), interpolate function process and EG calculation processing and ALFO calculation processing (OP4), level calculation processing (OP5) (OP6), wave form output value sound stack write (OP7). The calculation process of the level calculation takes a long time, so two cycles are required. If looking at a snapshot of the calculation process, for example, slot 3 is doing PG calculation processing and PLFO calculation processing while ADP calculation processing and MD read are done in slot 2. Meanwhile, wave form read in slot 1, interpolate calculation process, EG calculation process and ALFO calculation process in slot 0, level calculation process in slot 31 and slot 30, and sound stack writing in slot 30 are also being executed.



Because slot calculation starts at slot 0 and moves towards slot 31, the data written to the sound stacks also start at a low number and move up. However, there is a time difference between the time the slot calculation starts and the time the slot is written to the sound stack, which is equal to the calculation execution time. For example Figure 4.31 shows that it would take six cycles worth of time for slot 0 to be written to the sound stack.

As shown in Figure 4.30, when connecting a slot to another slot, the slot being connected to must already have written its data to the sound stack when it is in the OP2 state or the connection cannot be done.

For example, since up to 27 slots are written to the sound stack, up to "27" color-coded sound data sets are applicable to slots that can be connected to slot 0 when OP2 is slot 0. An equation expressing this would be as follows.

$$\{\text{Current slot number} + 32 - \text{number of the slot to be connected (sound stack)} + A\} \geq 5$$

The current slot number represents the register slot number. However, if the sound stack of the slot to connect is in the current slot, then A is "32"; for past cycles, enter "0". Here the past cycle indicates the cycle data from one sample previous.

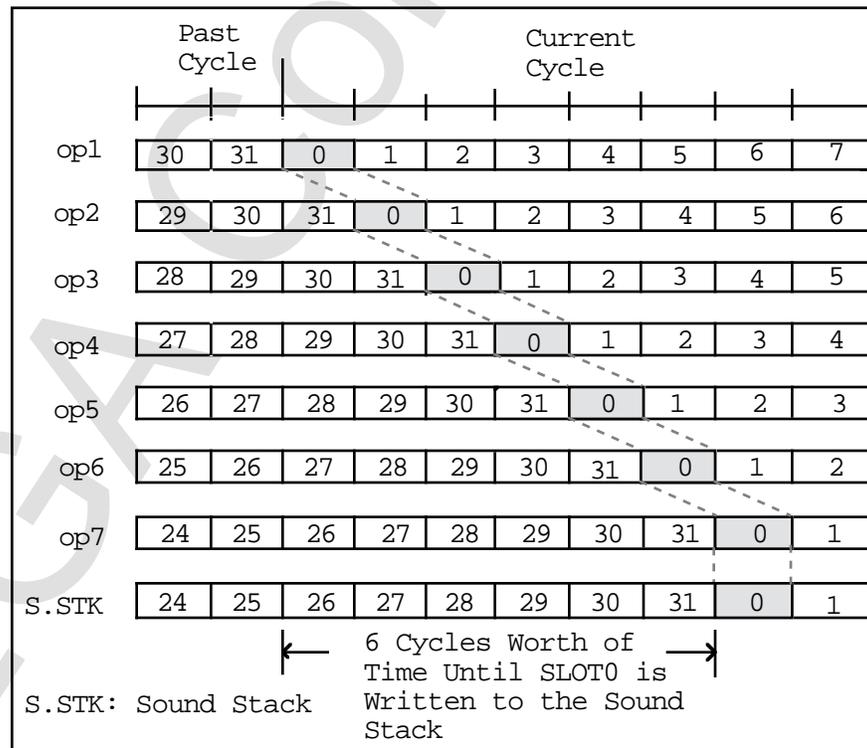


Figure 4.31 Time Lag until the Slot Is Written to the Sound Stack

At least two slots must be connected to enable voice mixing. Explained below is the calculation method used for "MDXSL" and "MDYSL" (each 6 bits).

When two slots are connected, the slot applying modulation (Modulator) is M, while the slot being modulated (carrier) is called C. The results of M-C are called J, with the resulting equation shown below.

$$M - C = J \dots \text{equation 4.1}$$

The resulting J value changes the "MDYSL" and "MDSYL" highest bit (6th in MSB) value. The conditions for this are shown below.

- Condition 1 $J \geq 28$
 - Convert the J value into hexadecimal 5 bit value.
 - If it is the newest sample then MSB (6th bit) is "0B"; if past samples, it is "1B".
- Condition 2 $J < 0$ (J is negative)
 - Calculate $32 + J$ (calculate 10 decimal)
 - Next convert the value from (1) to a hexadecimal 5 bit value.
 - If it is the newest sample then MSB (6th bit) is "0B", if past samples, then it is "1B".
- Condition 3 When the value J does not fit in neither condition 1 nor 2 above.
 - Convert the J value into hexadecimal 5 bit value.
 - If it is the newest sample then MSB (6th bit) is "1B", if past samples, then it is "0B".

Following are items to be aware of when using FM voice mixing.

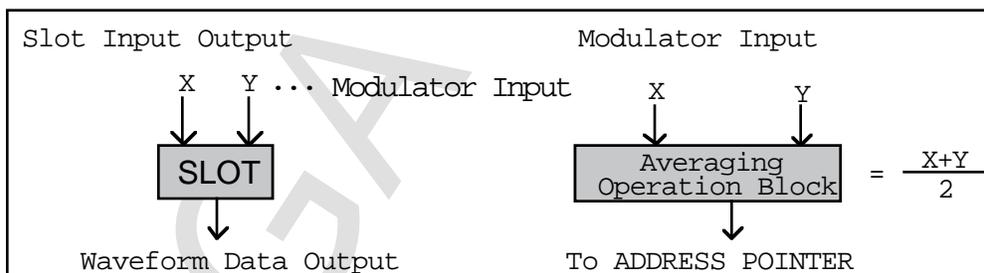


Figure 4.32 Slot Averaging Calculation



The source input into a slot is the data in the sound stack (“SOUS”). Therefore, modulating with a slot indicates that the sound stack corresponding to the modulation slot is assigned and connected to a non-modulation slot. Actual functions include selecting the sound stacks input for modulator input Y and X, using “MDXSL” and “MDYXL”; therefore, the parameters obtained using the calculation method explained at the beginning of this register is required.

Also, there are two modulator inputs to the slot, X and Y. Be aware that these may be cut in half through the averaging calculation. Therefore, if only one slot is being connected to the slot, X and Y are set with the same parameters and the same slot sound stack data should be input. If only one side is set for input, then unexpected data may get in and make the FM modulation rate could seem smaller, or other such effects.

Next, determine each parameter using the “MDXSL” and “MDYSL” calculation method with four slot configurations using the actual FM voice mixer as an example.

First, explain how to get the FM voice mixer algorithm as shown in Figure 4.33.

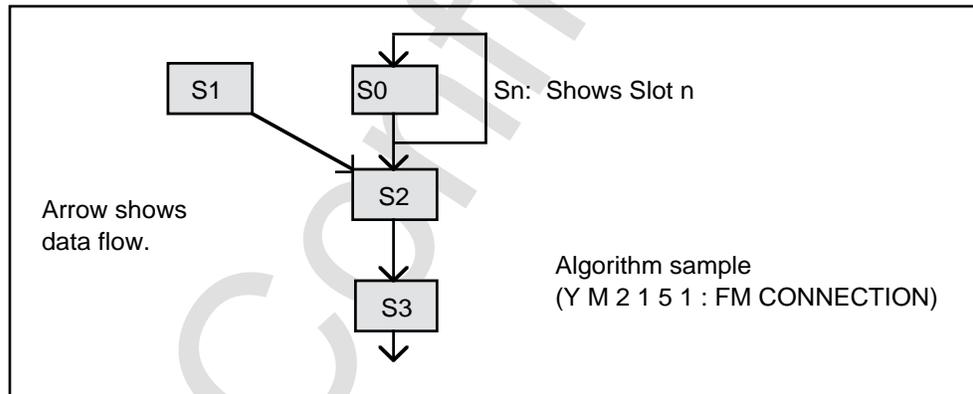


Figure 4.33 4SLOT configuration Algorithm

- Slot0
Slot0 output is used as input for slot 2, but the output of slot0 is used for the input of slot0. This is called self feedback.
- Slot1
Slot1 output is used as input for slot2.
- Slot2
Slot2 uses the output of slot1 and slot0 as its input. This output is used as input for slot3.

- Slot3
Slot3 uses slot2 output as input. Slot3 output is not connected to anything, but is used as voice output.

(1) Parameters set in SLOT0.

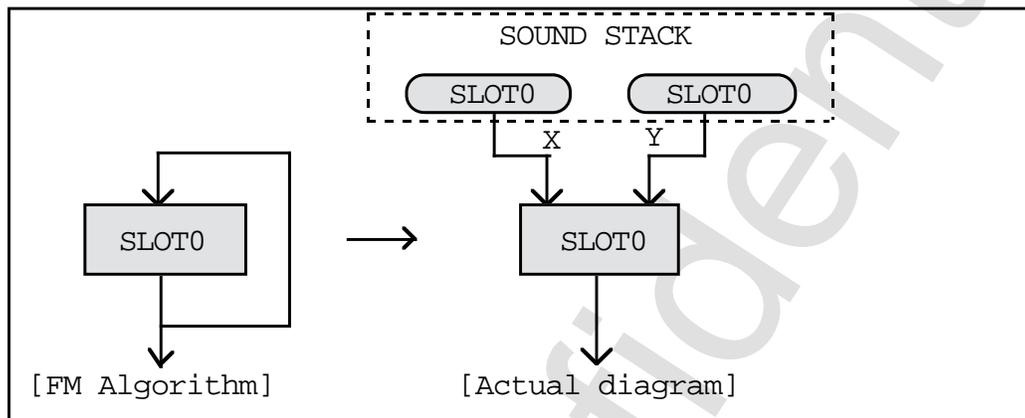


Figure 4.34 SLOT0 Algorithm

For a self feedback type configuration such as slot 0, connect the sound stack that stores the slot 0 output data to the modulation input X and Y.

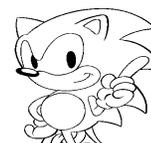
There are three different ways of input methods to input the latest and past sample to both modulation input X and Y, or latest sample to one side and past sample to another since there are latest and past sample currently calculated in sound stack.

⚠ Caution: In the case of self feedback, oscillation may occur as the same sample is input. So please input latest samples to one side and past samples to the other. The values of <MDXSL and MDYSL are both 0 for the modulator slot and the modulated slot, so entering these values into equation 4.1 produces the following results.

$$0 - 0 = 0$$

This meets the requirements of condition 3 in equation 4.1, so MSB of MDXSL and MDYSL are as follows.

Latest samples	MSB=1 (<u>1</u> 00000) _B	= 20 _H
Past samples	MSB=0 (<u>0</u> 00000) _B	= 00 _H



(2) Parameters set in SLOT2

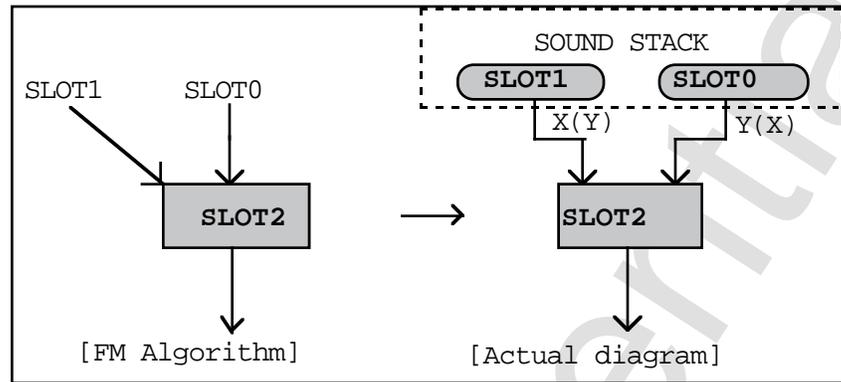


Figure 4.35 SLOT2 Algorithm

Because slot 2 is input for multiple slots, there is no need to worry about whether to use latest or past samples.

The “MDXSL” and “MDYSL” values input (slot 1 side) into slot 1 are “1” for the modulator slot and “2” for the modulated slot. Applying this to equation 4.1 results in the following.

$$1 - 2 = -1$$

This meets the requirements of condition 2 in equation 4.1, so the MSB of “MDXSL” and “MDYSL” are as follows.

$$32 + (-1) = 31 = 1FH = (011111)_B$$

Latest samples	MSB=0	(011111) _B	= 1FH
Past samples	MSB=1	(111111) _B	= 3FH

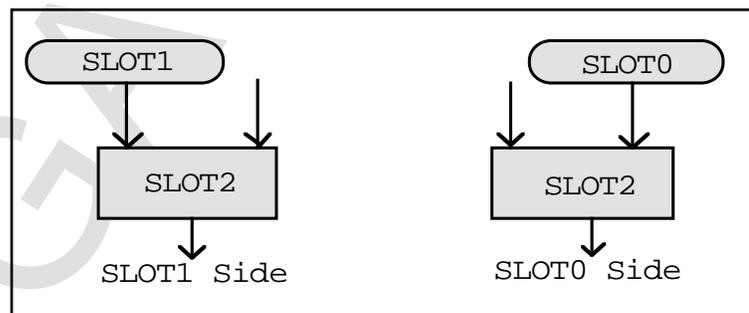


Figure 4.36 SLOT2 Algorithm (By input slot)

The “MDXSL” and “MDYSL” values input (slot 0 side) into slot 0 are “0” for the modulator slot and “2” for the modulated slot. Applying this to equation 4.1 results in the following.

$$0 - 2 = -2$$

This meets the requirements of condition 2 in equation 4.1, so the MSB of “MDXSL” and “MDYSL” are as follows.

$$32 + (-2) = 30 = 1E_H = (011111)_B$$

Latest samples MSB=0 (011110)_B = 1E_H

Past samples MSB=1 (111110)_B = 3E_H

✘ **Note:** As a rule, always input the same generation sample when inputting slots with different numbers. Past FM sound generators have used this method, so when replacing an FM sound generator tone library, use this method.

(3) Parameters Set in Slot 3

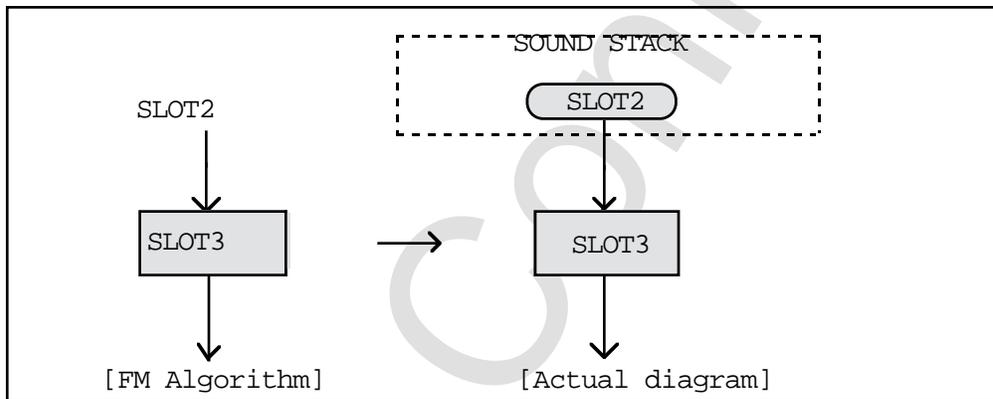


Figure 4.37 SLOT3 Algorithm

Because there is only one input for slot 3, input the output of slot 2 for the modulator input of both X and Y.

If the connected slots are different, there is no chance of oscillation, so both modulator input X and Y can have just the latest sample, or just the past samples, or the latest in one side and past in the other side. If replacing an existing FM sound generator tone library, use one sample or the other.



The “MDXSL” and “MDYSL” values input into slot 2 are “2” for the modulator slot and “3” for the modulated slot. Applying this to equation 4.1 results in the following.

$$2 - 3 = -1$$

This meets the requirements of condition 2 in equation 4.1, so the MSB of “MDXSL” and “MDYSL” are as follows.

$$\begin{aligned} 32 + (-1) &= 31 = 1FH = (011111)_B \\ \text{Latest samples} \quad \text{MSB=0} & (011111)_B = 1FH \\ \text{Past samples} \quad \text{MSB=1} & (\underline{1}11111)_B = 3FH \end{aligned}$$

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Up until now, “MDXSL” and “MDYSL” values have been achieved through calculation. However, by using the parameter table shown in Table 4.16, and from the modulator and carrier slot numbers, we can get the output of the “MDXSL” and “MDYSL” values.

Below, using the parameter table, the values for “MDXSL” and “MDYSL” can be found in Figure 4.33.

(1) Parameters set in slot0

Slot 0 is a self-feedback configuration so the carrier slot number and the modulator slot number are “00”. From this, “MDXSL” and “MDYSL” values from the parameter table are as follows.

Latest samples	20H
Past samples	00H

As explained earlier, if the same generation samples are used in the self-feedback configuration, there is a chance for oscillation, so avoid this method.

(2) Parameters set in slot2

Slot 2 uses the multiple input configuration.

First, when slot 1 is input, the carrier slot number is “02” and the modulator slot number is “01”. The MDXSL” and “MDYSL” values from the parameter table are as follows.

Latest samples	1FH
Past samples	3FH

Next, when slot 0 is an input, the carrier slot number is “02” and the modulator slot number becomes “00”. The MDXSL” and “MDYSL” values from the parameter table are as follows.

Latest samples	1EH
Past samples	3EH

(3) Parameters set in slot3

Slot 3 uses the output of slot 2, thus the carrier slot number is “03” and the modulator slot number is “02”. The “MDXSL” and “MDYSL” values from the parameter table are as follows.

Latest samples	1FH
Past samples	3FH



Table 4.16 Relation of MDXSL/MDYSL and SLOT

MDX (Y) SL		Carrier Slot Number															
Parameter Value		00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15
Latest Sample	Past Samples	Modulator Slot Number(SLOT 00 - SLOT 31)															
20H	00H	00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15
21H	01H	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16
22H	02H	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17
23H	03H	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18
24H	04H	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19
25H	05H	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20
26H	06H	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21
27H	07H	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22
28H	08H	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23
29H	09H	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
2AH	0AH	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25
2BH	0BH	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26
2CH	0CH	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27
2DH	0DH	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28
2EH	0EH	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29
2FH	0FH	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30
30H	10H	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
31H	11H	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	00
32H	12H	18	19	20	21	22	23	24	25	26	27	28	29	30	31	00	01
33H	13H	19	20	21	22	23	24	25	26	27	28	29	30	31	00	01	02
34H	14H	20	21	22	23	24	25	26	27	28	29	30	31	00	01	02	03
35H	15H	21	22	23	24	25	26	27	28	29	30	31	00	01	02	03	04
36H	16H	22	23	24	25	26	27	28	29	30	31	00	01	02	03	04	05
37H	17H	23	24	25	26	27	28	29	30	31	00	01	02	03	04	05	06
38H	18H	24	25	26	27	28	29	30	31	00	01	02	03	04	05	06	07
39H	19H	25	26	27	28	29	30	31	00	01	02	03	04	05	06	07	08
3AH	1AH	26	27	28	29	30	31	00	01	02	03	04	05	06	07	08	09
3BH	1BH	27	28	29	30	31	00	01	02	03	04	05	06	07	08	09	10
3CH	1CH	28	29	30	31	00	01	02	03	04	05	06	07	08	09	10	11
3DH	1DH	29	30	31	00	01	02	03	04	05	06	07	08	09	10	11	12
3EH	1EH	30	31	00	01	02	03	04	05	06	07	08	09	10	11	12	13
3FH	1FH	31	00	01	02	03	04	05	06	07	08	09	10	11	12	13	14

MDX (Y) SL		Carrier Slot Number															
Parameter Value		16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
Latest Sample	Past Samples	Modulator Slot Number(SLOT 00 - SLOT 31)															
16H	00H	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
17H	01H	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	00
18H	02H	18	19	20	21	22	23	24	25	26	27	28	29	30	31	00	01
19H	03H	19	20	21	22	23	24	25	26	27	28	29	30	31	00	01	02
20H	04H	20	21	22	23	24	25	26	27	28	29	30	31	00	01	02	03
21H	05H	21	22	23	24	25	26	27	28	29	30	31	00	01	02	03	04
22H	06H	22	23	24	25	26	27	28	29	30	31	00	01	02	03	04	05
23H	07H	23	24	25	26	27	28	29	30	31	00	01	02	03	04	05	06
24H	08H	24	25	26	27	28	29	30	31	00	01	02	03	04	05	06	07
25H	09H	25	26	27	28	29	30	31	00	01	02	03	04	05	06	07	08
26H	0AH	26	27	28	29	30	31	00	01	02	03	04	05	06	07	08	09
27H	0BH	27	28	29	30	31	00	01	02	03	04	05	06	07	08	09	10
28H	0CH	28	29	30	31	00	01	02	03	04	05	06	07	08	09	10	11
29H	0DH	29	30	31	00	01	02	03	04	05	06	07	08	09	10	11	12
2EH	0EH	30	31	00	01	02	03	04	05	06	07	08	09	10	11	12	13
2FH	0FH	31	00	01	02	03	04	05	06	07	08	09	10	11	12	13	14
30H	10H	00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15
31H	11H	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16
32H	12H	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17
33H	13H	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18
34H	14H	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19
35H	15H	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20
36H	16H	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21
37H	17H	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22
38H	18H	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23
39H	19H	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
3AH	1AH	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25
3BH	1BH	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26
3CH	1CH	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27
3DH	1DH	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28
3EH	1EH	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29
3FH	1FH	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30

✳ How to use the table

First find the carrier side slot number and then find the modulator slot number which is input into the carrier side. Find the number of sample that corresponds in the horizontal column.

Next, the “MDL” (Modulation Level) will be explained. “MDL” is the modulation rate (how much modulation) parameter used to set the modulation signal added to the modulation input X and Y. If this value becomes large, the frequency modulation will be quite deep; if the value becomes small, the level will be more shallow. (Figure 4.38)

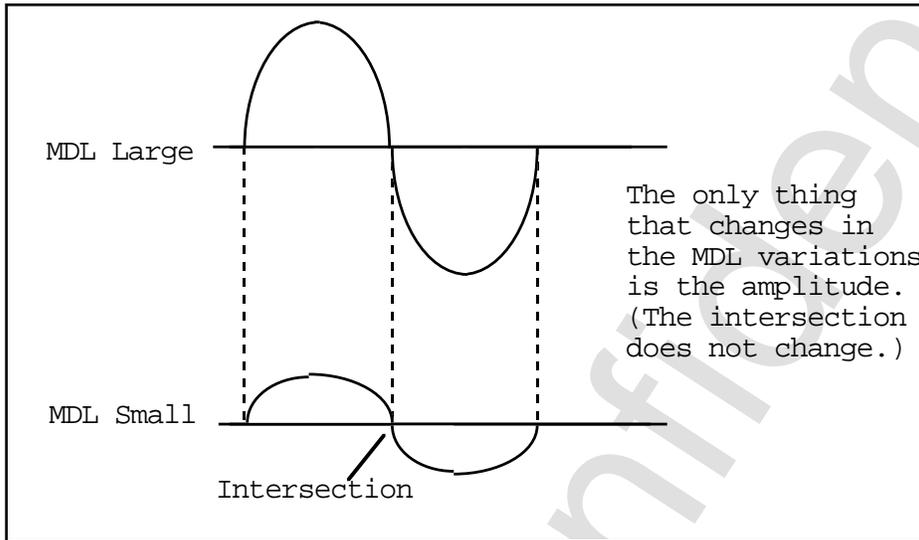


Figure 4.38 MDL Modulation Rate

Here the modulation rate of the “MDL” set value is explained. Table 4.15 shows the modulation rate becomes $\pm n \pi$ by the “MDL” setting.

Each sound slot data item input into the modulation input X and Y is averaged in the averaging calculation block. Data input into X is XD, into Y is YD, and let the output to averaging calculation block be ZD, then the following equation can be represented.

$$ZD = (XD + YD) / 2$$

This “ZD” is adjusted by the “MDL” and sent to the slot address pointer (phase adder).



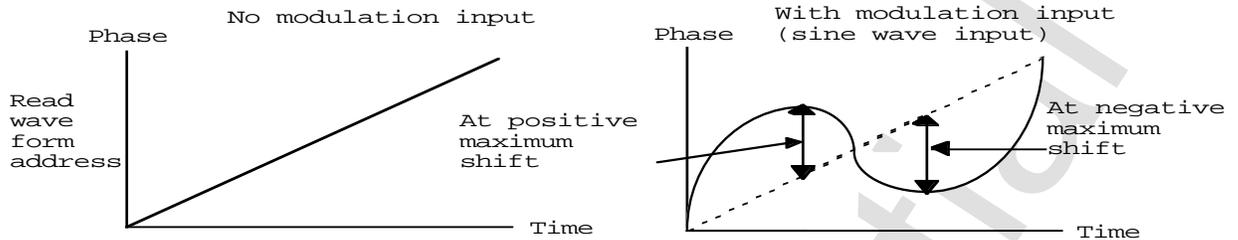


Figure 4.39 Maximum Shift of the Wave Form Read Address

When modulation wave form is not added to FM voice mixing, the time phase function is linear. However, if the modulation waveform is added, because a phase shift equivalent to the modulation waveform will be added, the linear time-phase waveform will then become non-linear. This shift becomes maximum when ZD assumes the \pm maximum value. MDL modulation rate indicates the wave form address shift (maximum shift) when the wave form data (sine wave) is at 1 cycle = 1Kword and ZD is \pm the maximum value.

Table 4.17 Address Maximum Shift Value According to the Register Setting Value

MDL [3:0]	0 ~ 4	5	6	7	8	9	A	B	C	D	E	F
Address max. shift \pm	0	32	64	128	256	512	1024	2048	4096	8192	16384	32768

Compare Table 4.16 and Table 4.17 to get the function $\pi = 512$ (when MDL=AH). In Table 4.16, there is wave form data (sine wave) consideration where 1 cycle = 1K word (1024 words), but mathematically 1 wave form cycle length is defined as 2π . Therefore, in the current FM voice mixer the sine wave used is 1 cycle = 1 Kword so the maximum shift of 512 can be expressed as π . Using π to express the shift is only valid when 1 cycle of wave form data is 1 K word. Otherwise, use the expression method by address shift shown in Table 4.17.

When using the FM voice mixer, always have three cycles of wave form data. The reason has to do with address shift. The SCSP hardware must hold up to 1Kword address shift (MDL="AH"), but if the amount exceeds 1Kword, the hardware will automatically execute clipping processing, so there is no need to have extra wave form data equal or greater than 1Kword.

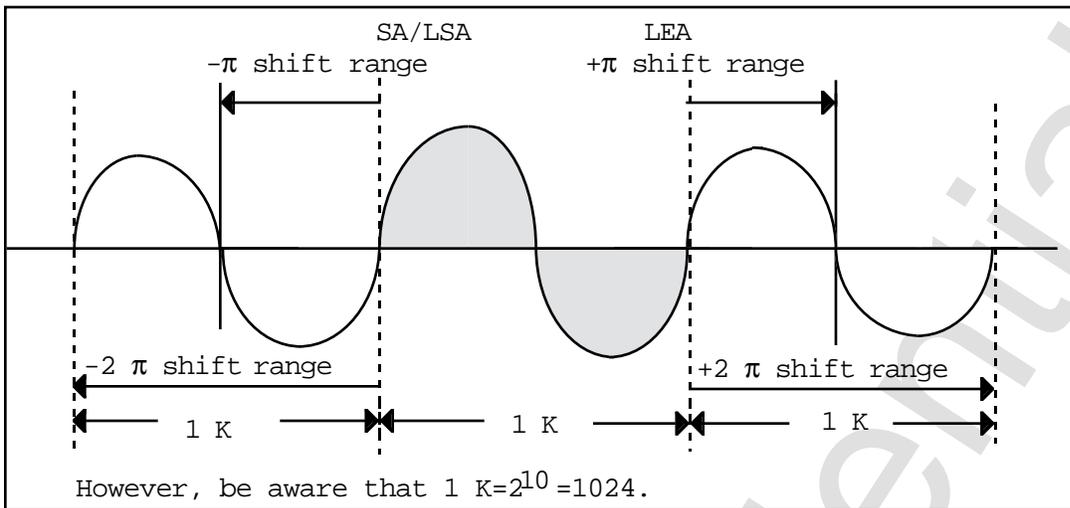


Figure 4.40 Address Shift during FM Voice Synthesis

During FM voice mixing, place only the additional amount required for address shift. As a rule, operation will run without problems if π worth of wave form data is set before "SA", and after "LEA" and is $\pm \pi$. However, using three cycles worth of data as shown in Figure 4.40 is the ideal setting method.

✘ Concerning Clipping Process

This relates to the address shift described earlier. The address shift range shifts a maximum of ± 32768 addresses, so before and after the basic wave form respectively 32 cycles (32K words) and 16 cycles (16K words) worth of wave form data is required for total memory requirements of 65K words.

For this reason, the SCSP clips (process to prevent shift from exceeding a limit) shift that exceeds 1K word and returns the shift to 0. So if there is 3K word worth of wave form data no matter how much shift there is, it is more than enough to cover requirements. Clipping process was done because the valid address bits available for shift was 10.

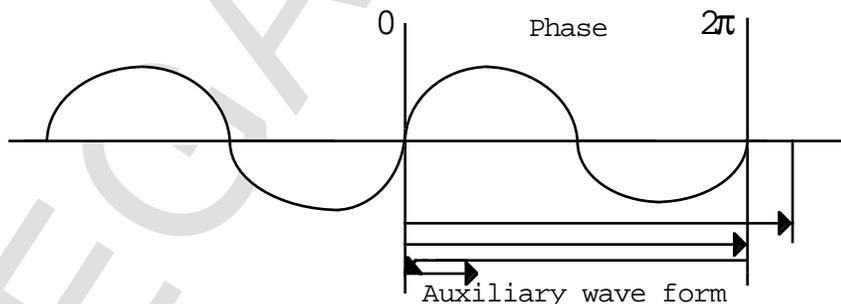


Figure 4.41 Wave Data During Clipping Process

As shown in Figure 4.41, even if the wave form exceeds the auxiliary waveform after the shift, it returns to 0 at the point where phase exceeds 2π .



The following is an explanation of the SCSP FM configuration (algorithm). The SCSP slot is configured with two inputs and one output so the number of slots that can be connected (modulation can be applied) to one slot is a maximum of two.

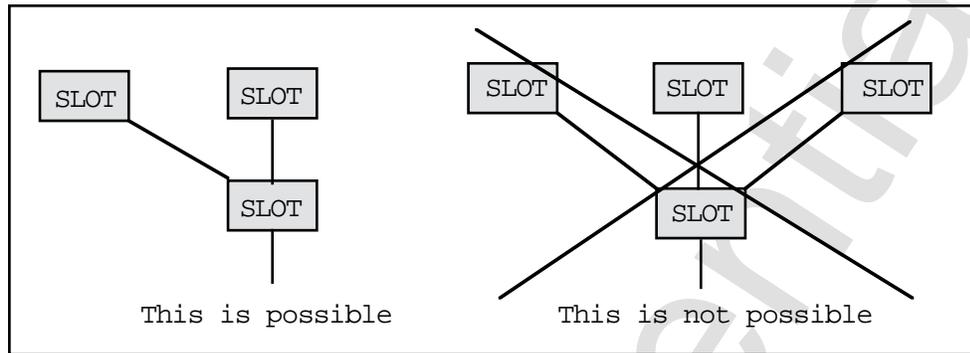


Figure 4.42 Slot Connection Count

As shown in Figure 4.42, modulation is possible if it's up to two modulators. Also if the modulation data (source) is brought up from the sound stack, FM configurations shown in Figure 4.43, Figure 4.44, Figure 4.45, and Figure 4.46 are possible.

(Figure 4.43 is a self-feedback slot with still another slot modulating the first slot; Figure 4.44 combines multi-stage slots for a multi-stage feedback; Figure 4.45 combines the multi-stage feed back and self feedback for a composite-type feedback; Figure 4.46 shows a composite modulation type slot modulated by a multi-stage feed back and further modulated by another slot.)

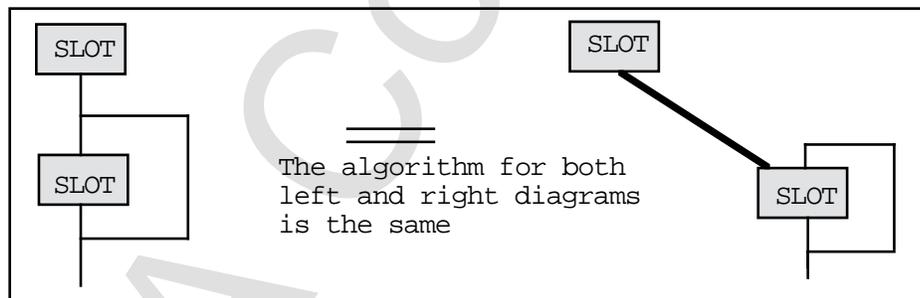


Figure 4.43 Self-Feedback Modulation

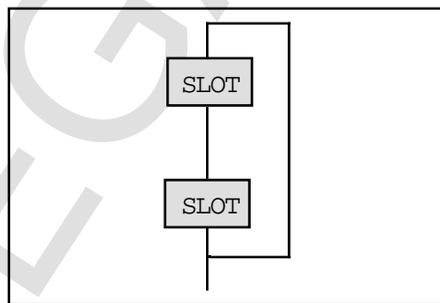


Figure 4.44 Multi-Stage Feedback

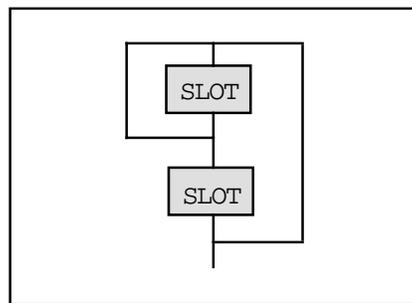


Figure 4.45 Composite Feedback

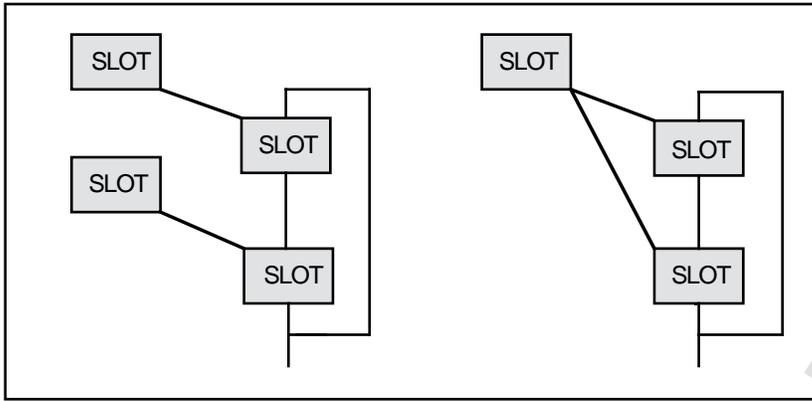


Figure 4.46 Composite Modulation

Figure 4.47 and 4.48 show specific examples of a basic FM configuration which should be used as a reference for FM voice mixing. Each slot upper side has two modulation inputs while the line on the lower side shows the slot output.

In addition, make sure that the dotted lines are also connected (refer to the forementioned figures).

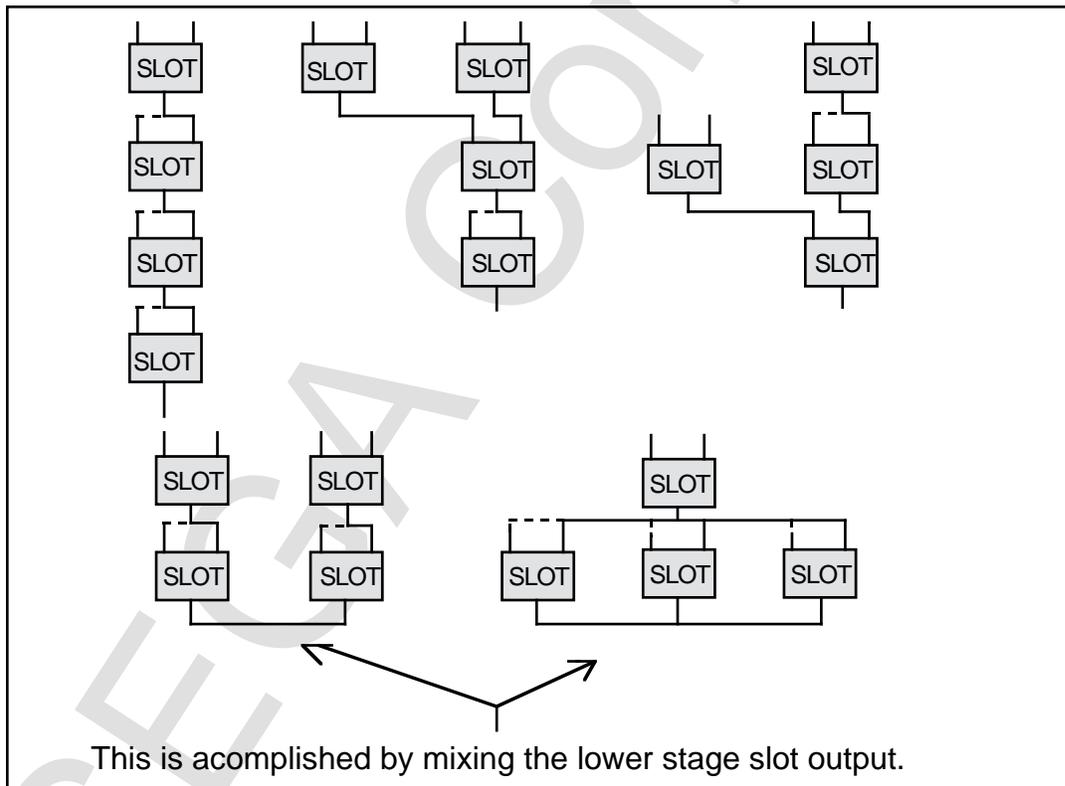


Figure 4.47 FM Configuration Algorithm Pattern 1



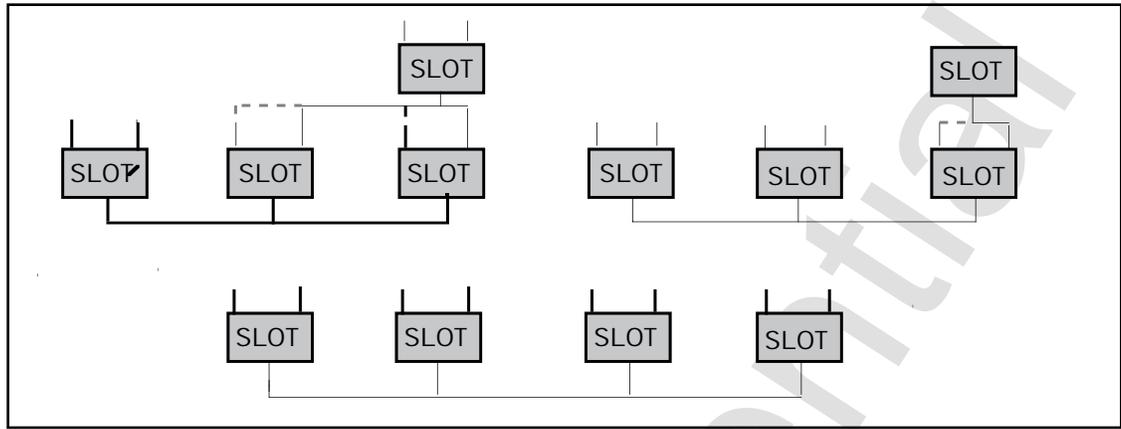


Figure 4.48 FM Configuration Algorithm Pattern 2

If the uppermost slot in FM voice mixing does not have self-feedback, there is nothing to be connected. In this situation, set "0~4" values to "MDL" value and the modulation rate to "0". This is how modulation input is set to "0".

✘ Definition of the uppermost slot.

The uppermost slot is the highest slot in each of the algorithm towers. In Figure 4.49, there are three towers: the S0 tower, the S1, S2 tower, and the S3, S4, S5, S6 tower. The uppermost slots are S0, S1, and S6.

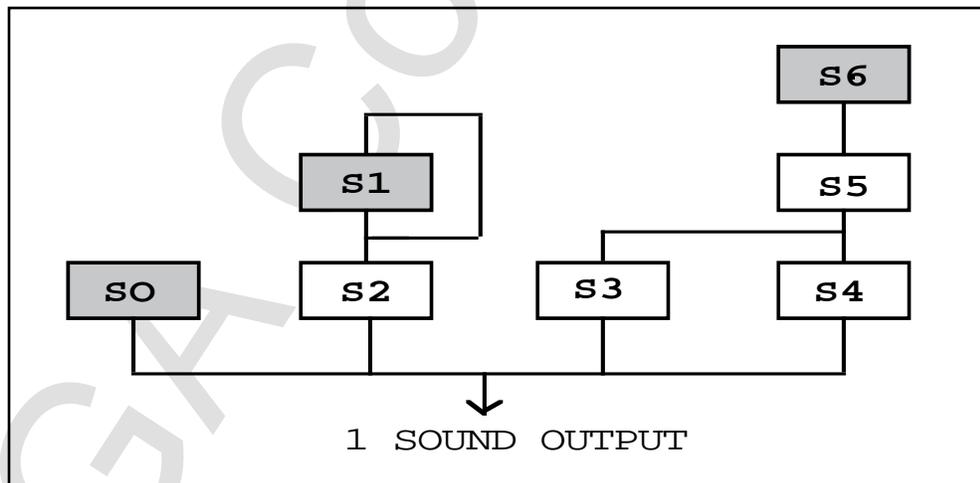


Figure 4.49 7 SLOT FM Configuration

Volume Register

TL[7:0] (R/W) ; Total Level

The total value of attenuation volume of the area where bit is set to "1", which is within total level 8 bit, becomes the actual attenuation volume [dB]. Therefore, the larger the value set in "TL", the larger the attenuation volume, making the sound volume less. On the other hand, the smaller the value set in "TL," the smaller the attenuation level, and the greater the sound volume will be . For example, "the attenuation volume with "FFH" would be as shown below.

$$- 48 - 24 - 12 - 6 - 3 - 1.5 - 0.8 - 0.4 = -95.7[\text{dB}]$$

Table 4.18 TL, Attenuation, and Waveform Amplitude

TL bit	bit 7	bit 6	bit 5	bit 4	bit 3	bit 2	bit 1	bit 0
TL Attenuation	- 48	- 24	- 12	- 6	- 3	- 1.5	- 0.8	- 0.4
Amplitude Ratio	1/256	1/16	1/4	1/2	$1/\sqrt{2}$	$1/\sqrt[4]{2}$	$1/\sqrt[8]{2}$	$1/\sqrt[16]{2}$
Real Number Ratio (times)	0.00391	0.06250	0.25000	0.50000	0.70711	0.84090	0.91700	0.95760

If only bit 4 is set to "1", then the amplitude of the output wave form after the TL calculation will be 1/2 of the amplitude of the wave form in memory, as shown in Figure 4.50.

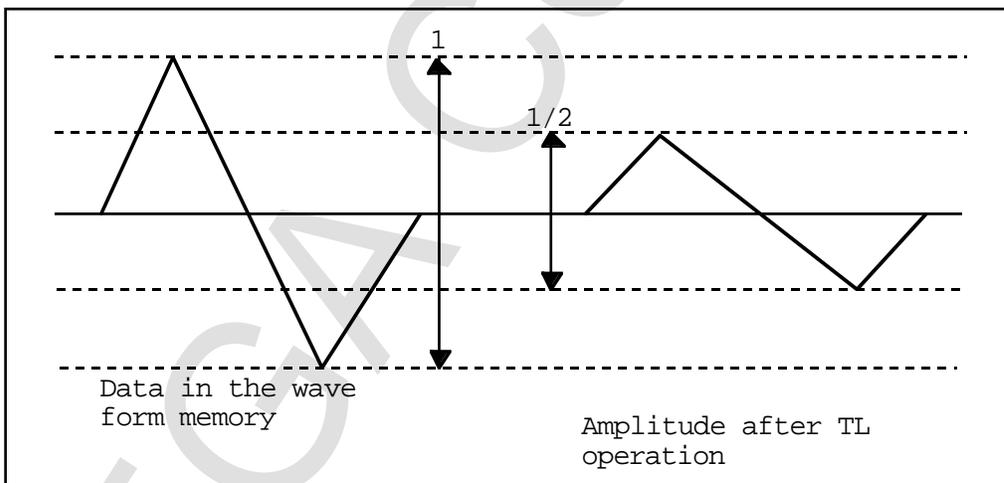


Figure 4.50 Wave Data when TL bit 4 = 1

SDIR (R/W) ; Sound DIRect

This flag is used to determine whether or not to directly output the sound data. When this bit is "1," then sound data is output without being multiplied by EG, TL, ALFO, etc., calculations.



PITCH Register

OCT[3:0] (R/W) ; OCTave

The function of the "OCT" register is to increase or decrease the sound generation frequency by octave with respect to the wave form data stored in memory.

FNS[9:0] (R/W) ; Frequency Number Switch

The function of this register is to adjust the tone between the octaves raised or lowered with "OCT." When the "FNS" and "OCT" values are both "0," the tone matches the sampling source data. Figure 4.51 shows the relation of the "FNS" and "OCT" values.

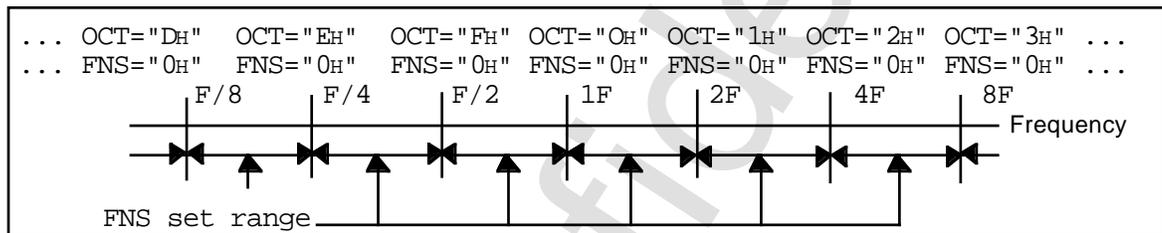


Figure 4.51 Relation of OCT and FNS

The actual pitch (n) is calculated by the equations shown below.

$$\begin{aligned}
 n(\text{Cent}) &= 1200 \times \text{LOG}_2 \left(\frac{1024 + FNS}{1024} \right) \\
 &= 1200 \times \frac{\text{LOG}_{10} \left(\frac{1024 + FNS}{1024} \right)}{\text{LOG}_{10} 2} \\
 &= 1200 \times \frac{\text{LOG}_{10} \left(\frac{1024 + FNS}{1024} \right)}{0.30103} \\
 &= 1200 \times \frac{\text{LOG}_{10}(1024 + FNS) - 10 \times 0.30103}{0.30103}
 \end{aligned}$$

The value found with this equation is expressed in units of "Cents."

One “cent” is $2^{1/1200} = 1.000577789$ times. Also, one octave is 1200 cents. When cent = n, then $2^{n/1200}$ times $\{=(1.000577789)^n\}$ against the original frequency. Therefore, the equation below is used to find the n cent high tone frequency F_n [Hz] in relation to the basic frequency tone F_o [Hz].

$$F_n \text{ [Hz]} = F_o \times 2^{n/1200}$$

Table 4.19 shows the actual frequencies corresponding to the cent count (number of cents).

Table 4.19 Actual Frequency Corresponding to Cent Count

Cent Count	Actual Frequency Value (X F_o)
0	1.000000000
100	1.059463094
200	1.122462048
300	1.189207115
400	1.259921050
500	1.334839854
600	1.414213562
700	1.498307077
800	1.587401052
900	1.681792830
1000	1.781797436
1100	1.887748625
1200	2.000000000

Next, using the sound of C4 (do) sampled at a rate of 44.1KHz as an example, the method for setting “FNS” and “OCT” will be explained.

$$FNS = 2^{10} \times (2^{P/1200} - 1)$$

By setting each parameter as shown in Table 4.20 from this equation, you can output at any frequency.



Table 4.20 FNS.OCT Parameter Table

Note Name	Note No.	PITCH [Cent]	FNS [9:0] [DEC]	FNS [9:0] [HEX]	OCT [3:0] [HEX]
B3	59	1100	909.1	38D	F
C4	60	0	0.0	0	0
C4#	61	100	60.9	03D	0
D4	61	200	125.4	07D	0
D4#	63	300	193.7	0C2	0
E4	64	400	266.2	10A	0
F4	65	500	342.9	157	0
F4#	66	600	424.2	1A8	0
G4	67	700	510.3	1FE	0
G4#	68	800	601.5	25A	0
A4	69	900	698.2	2BA	0
A4#	70	1000	800.6	321	0
B4	71	1100	909.1	38D	0
C5	72	0	0.0	0	1

LFO Register

LFO is a low-frequency oscillator that is used for modulation of sound signals. The LFO is contained in each slot and has the amplitude modulation output and frequency modulation output for that slot.

There are four types of output wave form from the LFO. These are: sawtooth wave, rectangular wave, triangle wave, and noise (white noise). All of these can be selected freely.

The LFO block diagram is shown in Figure 4.52. All waves (except the noise) can be reset and the frequency changed. Excluding the noise waveform, resetting as well as changing the frequency of sawtooth waveform, rectangular waveform, and triangular waveform are allowed. Also, the modulation rate and the LFO wave form can be set independently for both the amplitude modulation and the frequency modulation side.

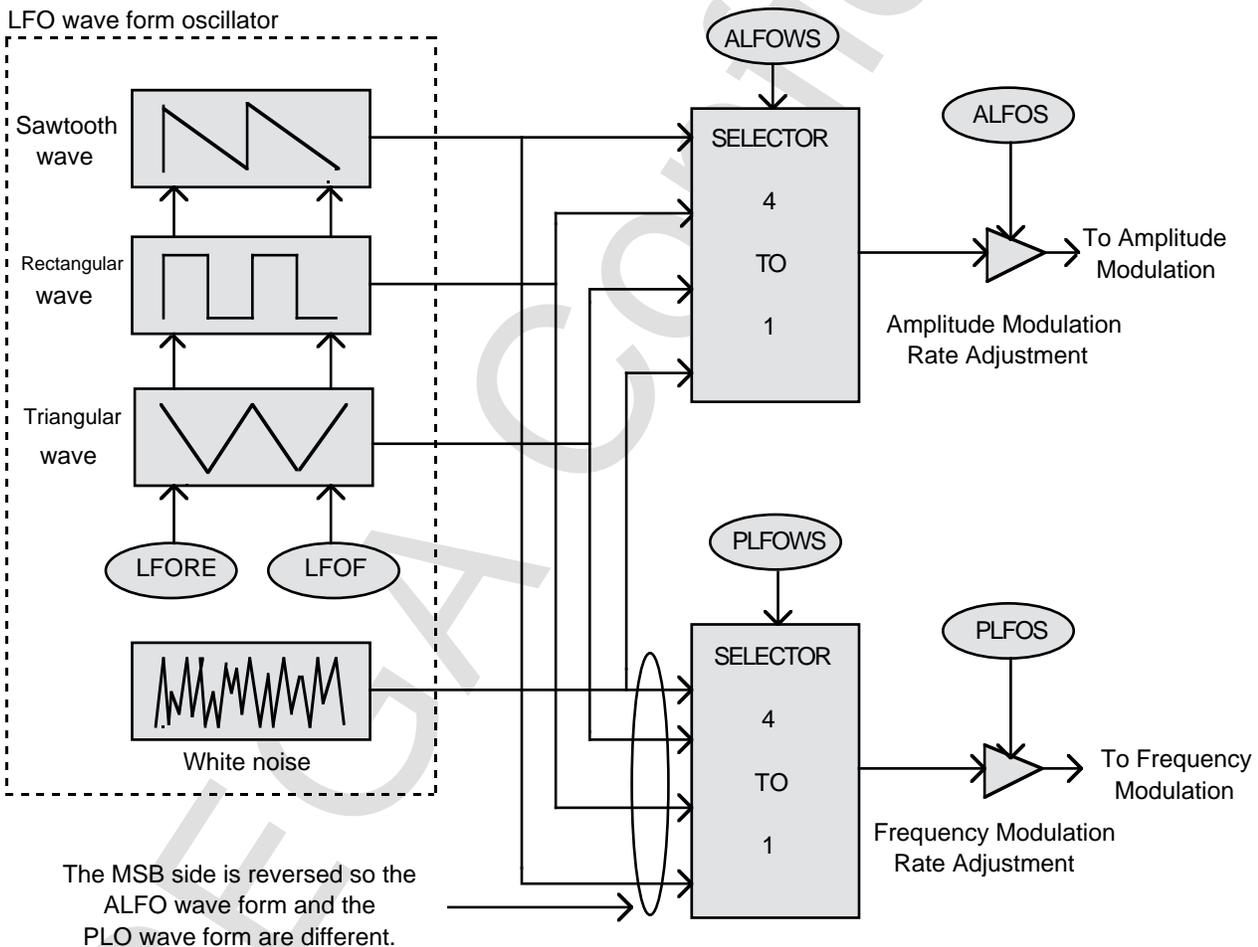


Figure 4.52 LFO Block Diagram



LFORE (R/W) ;LFO REset

Sets the LFO reset to yes or no. If this bit is set to "1", the LFO is reset. If "0" is written then operation starts.

LFOF [4:0] (R/W) ;LFO Frequency

Designates the LFO oscillation frequency.

Table 4.21 Oscillation Frequency of the Oscillator

LFOF	Oscillation Frequency (Hz)	LFOF	Oscillation Frequency (Hz)
00 _H	0.17	10 _H	2.87
01 _H	0.19	11 _H	3.31
02 _H	0.23	12 _H	3.92
03 _H	0.27	13 _H	4.79
04 _H	0.34	14 _H	6.15
05 _H	0.39	15 _H	7.18
06 _H	0.45	16 _H	8.6
07 _H	0.55	17 _H	10.8
08 _H	0.68	18 _H	14.4
09 _H	0.78	19 _H	17.2
0A _H	0.92	1A _H	21.5
0B _H	1.10	1B _H	28.7
0C _H	1.39	1C _H	43.1
0D _H	1.60	1D _H	57.4
0E _H	1.87	1E _H	86.1
0F _H	2.27	1F _H	172.3

ALFOS[2:0] (R/W) ;Amplitude-LFO Sensitivity

Sets the degree of amplitude modulation through the LFO. The tremor effect (a phenomenon that occurs when the sound volume is changed in a cyclic pattern) can be expressed through this amplitude modulation.

ALFOWS[1:0] (R/W) ; Amplitude-LFO Wave Select

Designates AM modulation wave form as shown in Table 4.22.

Table 4.22 LFO AM Modulation Wave Form by LFO

ALFOWS	AM Modulation (ALFO)		
	Volume	ALFO[7:0]	
0	-0 dB	0	
		FF	
1	-0 dB	0	
		FF	
2	-0 dB	0	
		FF	
3	-0 dB	0	***** ***** *****Noise***** *****
		FF	

PLFOWS[1:0] (R/W) ; Pitch-LFO Wave Select

Designates PM modulation wave form as shown in Table 4.23.

Table 4.23 LFO PM Modulation Wave Form

PLFOWS	AM Modulation (PLFO)		
	Pitch	PLFO[7:0]	
0	+	7F	
	0	00	
	-	80	
1	+	7F	
	0	00	
	-	80	
2	+	7F	
	0	00	
	-	80	
3	+	7F	***** ***** *****Noise***** *****
	0	00	
	-	80	

ALFO[7:0] and PLFO[7:0] show the LFO output bit width (data).



PLFOS[2:0] (R/W) ; Pitch-LFO Sensitivity

Specifies the degree of frequency modulation through the LFO. The vibrato effect (a phenomenon that occurs when the sound frequency is changed in a cyclic pattern) can be expressed through this frequency modulation (Table 4.24).

Table 4.24 Rates of Amplitude and Frequency Modulation

ALFOS	Mixing to the EG	PLFOS	Effect on Pitch
0	No effect	0	No effect
1	Displacement of 0.4 dB	1	Displacement of ± 7 cent
2	Displacement of 0.8 dB	2	Displacement of ± 13.5 cent
3	Displacement of 1.5 dB	3	Displacement of ± 27 cent
4	Displacement of 3 dB	4	Displacement of ± 55 cent
5	Displacement of 6 dB	5	Displacement of ± 112 cent
6	Displacement of 12 dB	6	Displacement of ± 230 cent
7	Displacement of 24 dB	7	Displacement of ± 494 cent

MIXER Register

The digital mixer block in the SCSP is used to adjust the level and balance of the various sound signals. It is made up of the direct mixer adjustment block, the DSP input (stage adjust) mixer block, the DSP output (stage adjust) mixer block and the last output (stage adjustment) mixer block. Figure 4.53 shows the digital mixer block diagram.

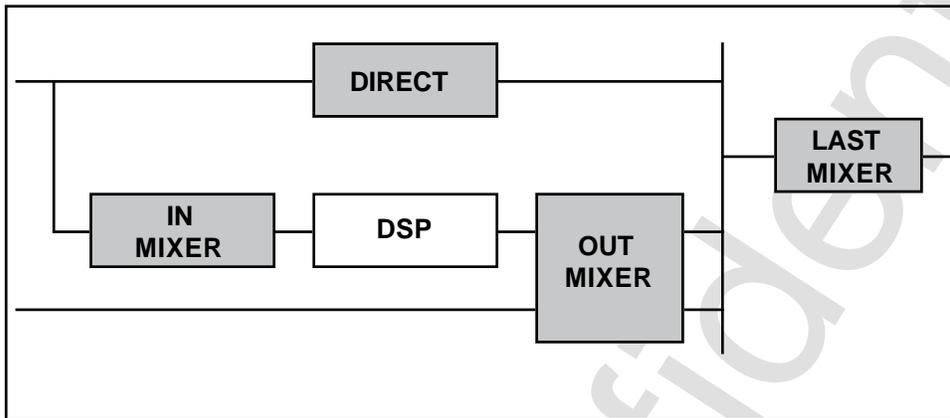


Figure 4.53 Digital mixer block diagram.

Direct Sound Adjustment Block

This mixer block controls the circuit that connects the output of each slot directly to the DAC output. It can control the output level (“DISDL”) and output balance (“DIPAN”) for each slot.

DSP Input Step Adjustment Block

Conducts mixings required to input the output of each slot to the DSP (“MIXS”). In reality, in the “ISEL” of each slot, the “MIXS” that inputs audio signal is selected and the input level is adjusted by the “IMXL”. When mixing multiple sounds with the “ISEL”, each of the sounds must be balanced.

Because the “MIXS” can mix the output from multiple slots and then input the result, the same effect like BGM reverb can be applied to many sounds. If effects at the DSP are applied, the sound data is sent to “MIXS”.

DSP Output Step Adjustment Block

The sounds that have had effects applied in the DSP or sounds that were brought in from an external digital input pass through mixing processing and are eventually compiled in stereo. The output level (“EFSDL”) and output balance (“EFPAN”) can be adjusted for sound signals that are output from the “EFREG” that correspond to DSP, and “EXTS” that receive external digital audio input. The data compiled here is mixed with the direct sound mixer.



Final Step Output Adjustment Block

Combines the direct component and the effect component of the sound and adjusts the output level to the DAC. The final output level is adjusted by "MVOL".

- * About the direct component and the effect component
When effects are applied to sound, the sound without effects applied (dry) and sounds with effects applied (wet) are mixed with the appropriate balance for each of the screens. The internal effect program separates the dry and wet data in the DSP. Because the sounds are mixed together in the end, when creating dry and wet data in the DSP, there is no need to output the direct component. This is accomplished by setting DISDL to "0" for no output. Figure 4.54 shows the circuits of the direct component and the effect component.

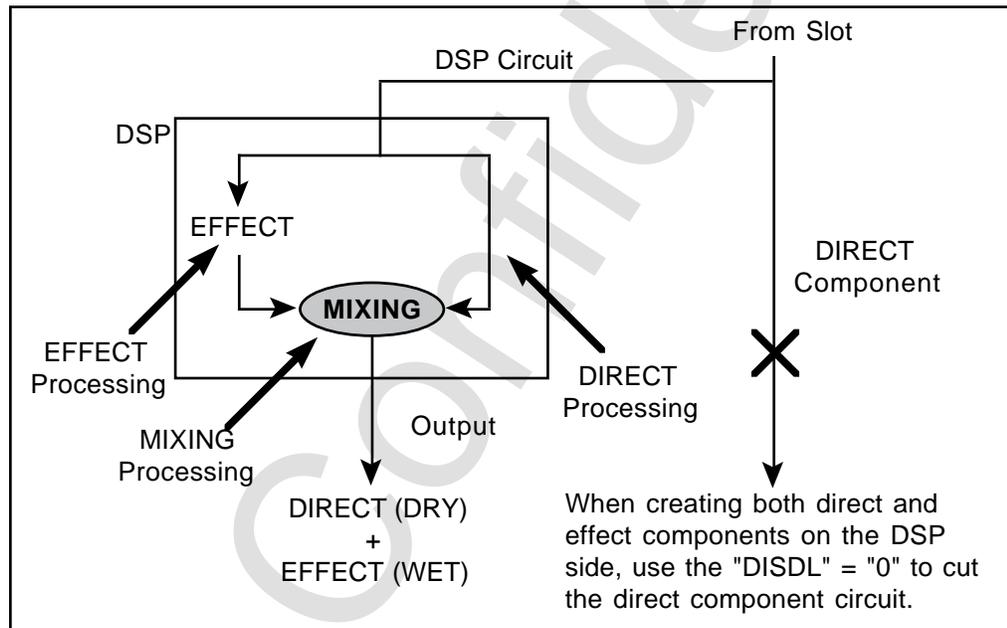


Figure 4.54 Direct and Effect Component Circuits

When rewriting the DSP program, the DSP operation becomes unstable and the sound will not be output properly from "EFREG". Set "EFSDL" to "0H" to prevent sound from being output.

- ✱ About fixed position
The SCSP digital mixer supports 31 levels of panning, but to set it in even more detail, use the DSP built into SCSP, as shown in Figure 4.55, to perform (process) panning.

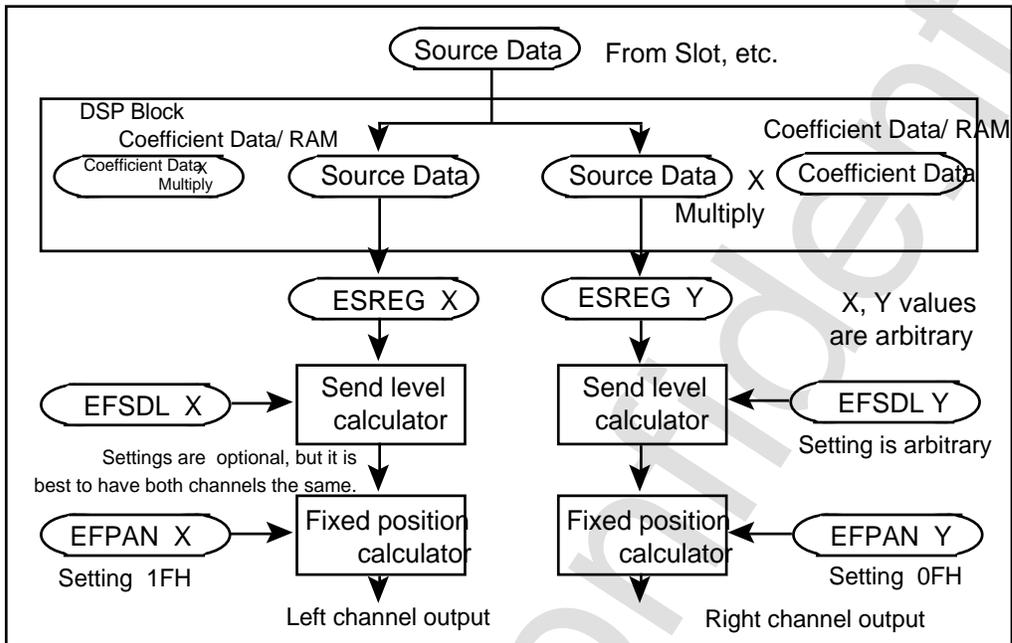
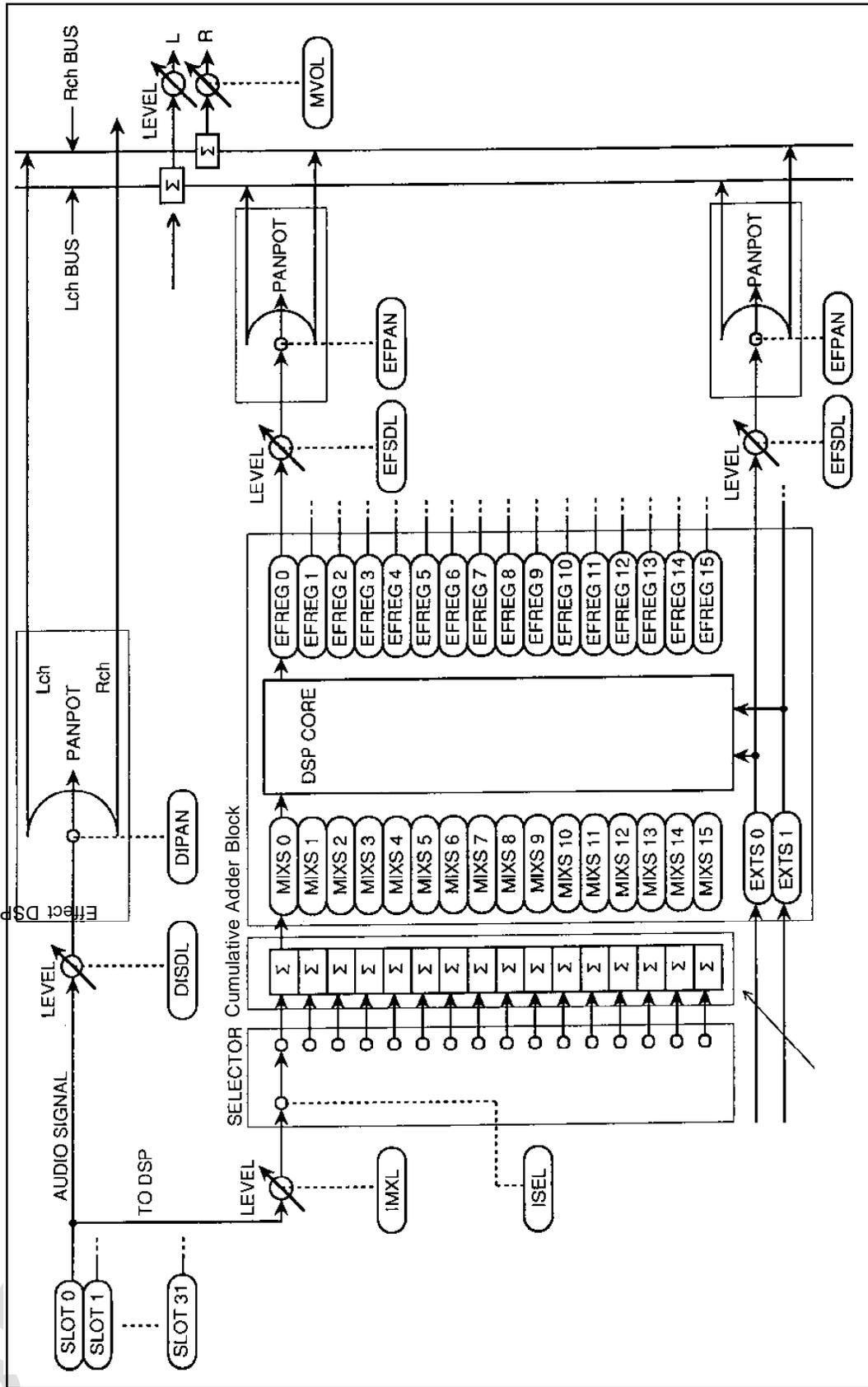


Figure 4.55 DSP Panning Calculation

Figure 4.56 shows a digital mixer block diagram.





Next, the method for setting each of the parameters based on Figure 4.56 will be explained.

The settings of “IMXL”, “ISEL”, “DISDL”, and “EPSDL” are required as described below to prevent overflow when many sounds are produced, and to prevent the overall volume from being too low when only a few sounds are produced.

(a) DSP Input Side Mixer

The parameters that are equivalent to the DSP input side mixer are the “IMXL” and “ISEL” parameters.

“MIXS” (the mix stack will be explained later) can input the output of multiple slots. This register has the ability to mix multiple data. Here the “IMXL” value must be set at the correct level to prevent “MIXS” from overflowing.

The number of available input sounds when the “IMXL” value is changed is shown in Table 4.25. For example, if “IMXL” is set to 7H against the input source and the input level of “MIXS” to “0[dB]”, then the number of available input slots would be one sound worth.

Table 4.25 Relation of the source count that can input into “IMXL” and “MIXS”

“IMXL” Value [2:0]	Level [dB]	Level Magnification	Sounds Available for Input (source count)
0H	-MAX	X0.000000	-- Sounds
1H	-36	X0.015625	64 Sounds
2H	-30	X0.031250	32 Sounds
3H	-24	X0.062500	16 Sounds
4H	-18	X0.125000	8 Sounds
5H	-12	X0.250000	4 Sounds
6H	-6	X0.500000	2 Sounds
7H	-0	X1.000000	1 Sounds

(b) “DISDL” and “EFSDL”

“DISDL” and “EFSDL” have basically the same philosophy as “IMXL,” but they must determine the sound output total count.

The sound output total count is the total sound output including the direct, effect and external input components. This indicates the cumulative total sources of L/R in Figure 4.56 that is being output. (L/R are counted as 1 together.) From this it is clear that the sum total of slots outputting sound is not equal.



(c) "DIPAN" and "EFPAN"

These registers are set at the center when setting value is either "00H" or "10H". By setting the MSB to "0B" and increasing the size of the lower 4 bits value, the fixed position will move to the right side.

By setting the MSB to "1B" and increasing the size of the lower 4 bits value, the fixed position will move to the left side.

IMXL[2:0] (R/W) ; Input MiXing L_evel

Designates the mix stack input level by slot when the sound slot output data is input into the DSP mix stack ("MIXS").

Table 4.26 Mix Stack Register Input level

IMXL	Level [dB]
0	- MAX (does not mix)
1	-36
2	-30
3	-24
4	-18
5	-12
6	-6
7	-0

ISEL[3:0] (R/W) ; Input SElect

Designates the mix stack number for each slot when the sound slot output data is input into the DSP mix stack ("MIXS").

The mix stack ("MIXS") finds the total of the input for all the slots for DSP input. The mix stack does not have a protect function for overflows, so set it so that the total of all slots does not exceed "0 [dB]."

DISDL[2:0] (R/W) ; Direct SenD Level

Designates the output level by slot when direct data is output to the D/A converter.

Table 4.27 D/A Converter Output Level

DISDL	Level [dB]
0	-∞ (not sent)
1	-36
2	-30
3	-24
4	-18
5	-12
6	-6
7	-0

DIPAN[4:0] (R/W) ; Direct PANpot

Specifies the fixed position (pan) for each slot when direct data is sent out.

Table 4.28 Fixed Position (Pan) Data by DIPAN

DIPAN	Left Fixed (dB)	Rt Fixed (dB)	DIPAN	Left Fixed (dB)	Rt Fixed (dB)
00H	-00.0	-00.0	10H	-00.0	-00.0
01H	-03.0	-00.0	11H	-00.0	-03.0
02H	-06.0	-00.0	12H	-00.0	-06.0
03H	-09.0	-00.0	13H	-00.0	-09.0
04H	-12.0	-00.0	14H	-00.0	-12.0
05H	-15.0	-00.0	15H	-00.0	-15.0
06H	-18.0	-00.0	16H	-00.0	-18.0
07H	-21.0	-00.0	17H	-00.0	-21.0
08H	-24.0	-00.0	18H	-00.0	-24.0
09H	-27.0	-00.0	19H	-00.0	-27.0
0AH	-30.0	-00.0	1AH	-00.0	-30.0
0BH	-33.0	-00.0	1BH	-00.0	-33.0
0CH	-36.0	-00.0	1CH	-00.0	-36.0
0DH	-39.0	-00.0	1DH	-00.0	-39.0
0EH	-42.0	-00.0	1EH	-00.0	-42.0
0FH	-∞	-00.0	1FH	-00.0	-∞



EFSDL[2:0] (R/W) ; Effect SenD Level

Specifies the output level for each slot when wave form data that has passed through the DSP and had effect process applied (effect data) is output to the D/A converter.

Table 4.29 Send Level to the D/A Converter

EFSDL	Level [dB]
0	$-\infty$ (does not send)
1	-36
2	-30
3	-24
4	-18
5	-12
6	-6
7	0

EFPAN[4:0] (R/W) ; Effect PANpot

Specifies by slot the fixed position of the external input waveform data and the waveform data that passed through the DSP and had the effect process applied (effect data).

Table 4.30 Fixed Position Data from EFPAN

EFPAN	Left Output (dB)	Rt Output (dB)	EFPAN	Left Output (dB)	Rt Output (dB)
00H	-00.0	-00.0	10H	-00.0	-00.0
01H	-03.0	-00.0	11H	-00.0	-03.0
02H	-06.0	-00.0	12H	-00.0	-06.0
03H	-09.0	-00.0	13H	-00.0	-09.0
04H	-12.0	-00.0	14H	-00.0	-12.0
05H	-15.0	-00.0	15H	-00.0	-15.0
06H	-18.0	-00.0	16H	-00.0	-18.0
07H	-21.0	-00.0	17H	-00.0	-21.0
08H	-24.0	-00.0	18H	-00.0	-24.0
09H	-27.0	-00.0	19H	-00.0	-27.0
0AH	-30.0	-00.0	1AH	-00.0	-30.0
0BH	-33.0	-00.0	1BH	-00.0	-33.0
0CH	-36.0	-00.0	1CH	-00.0	-36.0
0DH	-39.0	-00.0	1DH	-00.0	-39.0
0EH	-42.0	-00.0	1EH	-00.0	-42.0
0FH	$-\infty$	-00.0	1FH	-00.0	$-\infty$

“EFSDL” and EFPAN” can set individual settings for each of “EFREG” or “EXTS”. Figure 4.31 shows the register address for “EFSDL” and EFPAN” that correspond to each “EFREG” or “EXTS”.

Table 4.31 EFSDL, and EFPAN Register Addresses for Each EFREG and EXTS

Source	Address	Data								Source	Address	Data							
		7	6	5	4	3	2	1	0			7	6	5	4	3	2	1	0
EFREG	100017	EFSDL0				EFPAN0				EFREG	100137	EFSDL9				EFPAN9			
EFREG	100037	EFSDL1				EFPAN1				EFREG	100157	EFSDL10				EFPAN10			
EFREG	100057	EFSDL2				EFPAN2				EFREG	100177	EFSDL11				EFPAN11			
EFREG	100077	EFSDL3				EFPAN3				EFREG	100197	EFSDL12				EFPAN12			
EFREG	100097	EFSDL4				EFPAN4				EFREG	1001B7	EFSDL13				EFPAN13			
EFREG	1000B7	EFSDL5				EFPAN5				EFREG	1001D7	EFSDL14				EFPAN14			
EFREG	1000D7	EFSDL6				EFPAN6				EFREG	1001F7	EFSDL15				EFPAN15			
EFREG	1000F7	EFSDL7				EFPAN7				EXTS0	100217	EFSDL16				EFPAN16			
EFREG	100117	EFSDL8				EFPAN8				EXTS1	100237	EFSDL17				EFPAN17			

MVOL[3:0] (W) ; Master VOLUME

Represents the master volume output to the D/A converter. Because it is used to control the overall output level, lowering the “MVOL” for an output that has overflowed to a lower level will not remove the clipping noise. (To remove the clipping noise, correct the “DISDL” and “EFSDL” settings to remove the overflow.)

DAC18B[3:0] (W) ; DAC out 18Bit

When setting the digital output as 18bit D/A converter interface, set this bit to “1B”. When 16 bit, set “0B.” (Normally connected to an 16 bit type D/A converter so this register should be set to “0B”.)

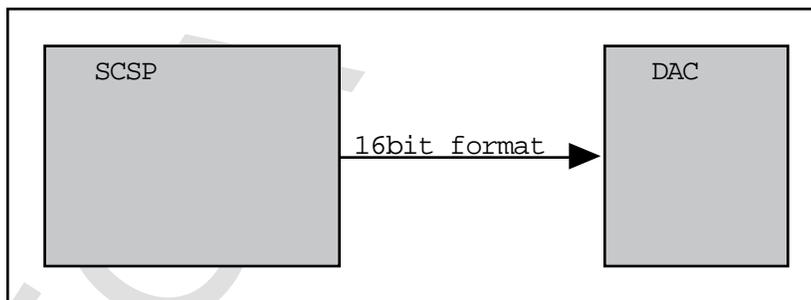


Figure 4.57 SCSP and DAC Connections



Slot Status Register

MSLC (w) ; Monitor Slot Call

Sets the number of the slot to be monitored with the "CA" register. In "0_H" it is "SLOT0", in "1F_H" it is "SLOT31"

CA (R) ; Call Address

The slot which has its number set in this register can read out the number of samples (number of samples = the number of wave form data samples) from the current output waveform data "SA" (start address) address. The value output here in the LSB (least significant bit) indicates 4K (4096) samples, so when the output value is "1_H", the slots being monitored are outputting at least more than "SA" to 4K samples of wave form addresses.

Sound Memory Configuration Register

MEM4MB (w) ; MEMory 4MBit

Designates the capacity of the memory connected to the SCSP.

Table 4.32 Memory Size

MEM4MB	Memory Capacity
0	Uses 1Mbit DRAM
1	Uses 4Mbit DRAM

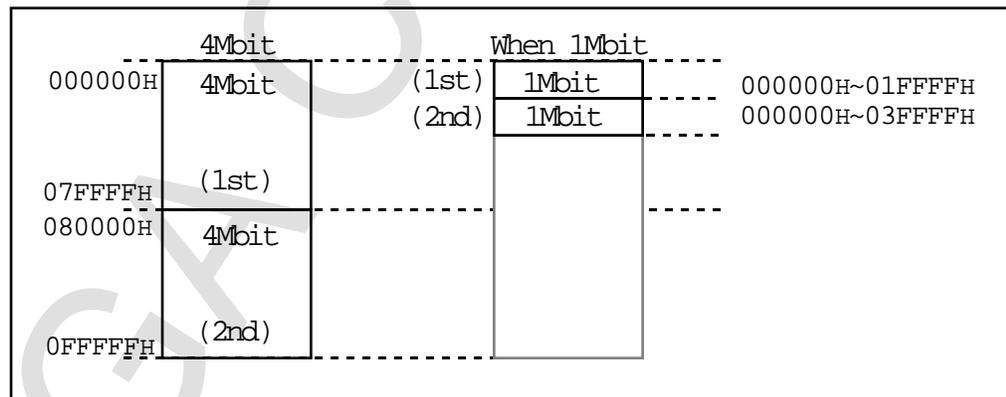


Figure 4.58 Memory Address Mapping Diagram

MIDI Register

The SCSP has a 31.25Kbps transfer rate MIDI serial interface, however, a MIDI peripheral circuit and MIDI DIN connector are not included. As a result, MIDI applications cannot be created. Figure 4.59 shows the MIDI interface block diagram.

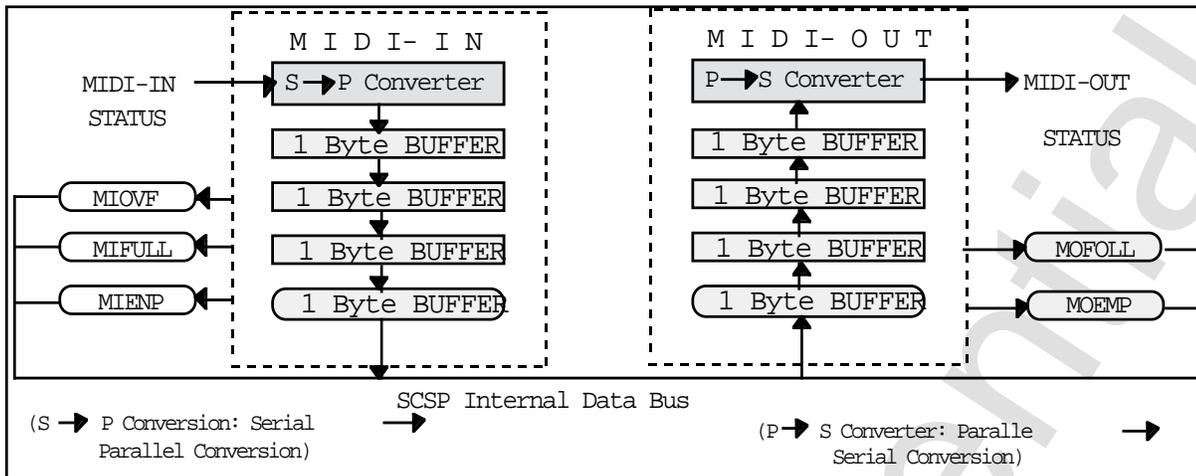


Figure 4.59 MIDI-I/F Block Diagram

MIBUF[7:0] (R) ; Midi Input BUffer

This is a MIDI input data buffer.

External data transferred into the MIDI-IN side is automatically stored in the MIDI-IN buffer "MIBUF".

MIOVF (R) ; Midi Input Over-Flow

When the MIDI-IN buffer is completely full of data and more data is transferred to the MIDI-IN, "MIOVF" changes to "1B" to indicate an overflow has taken place. When an overflow happens, the MIDI communications will not work correctly, causing a MIDI transmission error.

MIFULL (R) ; Midi Input FULL

When all 4bytes of the MIDI-IN buffer is completely full of data, the "MIFULL" will change to "1B" to indicate that the buffer is full.

MIEMP (R) ; Midi Input EMPTY

"1B" is set when the input FIFO is empty. When the internal MIDI-IN buffer is emptied through the CPU, etc., reading data, and when there is nothing in the MIDI-IN buffer, the "MIEMP" changes to "1B" to show that the buffer is empty.

Figure 4.60 is a diagram of the MIDI-IN and the interrupt generation block. Interrupts occur when the "MIEMP" is "1B" and receives external data and changes to "0B". When the MIDI-IN buffer is emptied through reading, etc., and MIEMP changes back to 1B, then the interrupt is released.



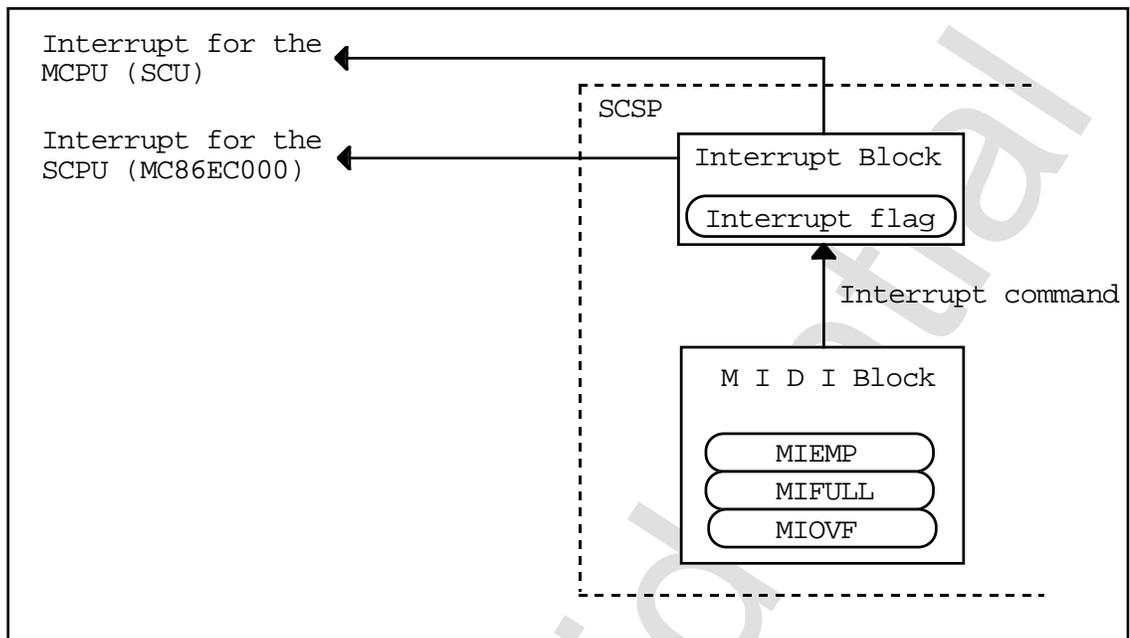


Figure 4.60 MIDI IN Block and the Interrupt Generation Block

MOFULL (R) ; Midi Output FULL

When all 4bytes of the MIDI-OUT buffer is completely full of data, the “MOFULL” will change to “1B” to indicate that the buffer is full.

MOEMP (R) ; Midi Output EMPty

When all of the data in the MIDI-OUT buffer has been sent out and no data is transferred into the MIDI buffer, “MOEMP” changes to “1B” to indicate that the buffer is empty. At the same time, an interrupt can be used to tell the CPU that the buffer is empty. Figure 4.61 shows the MIDI-OUT and interrupt generation block.

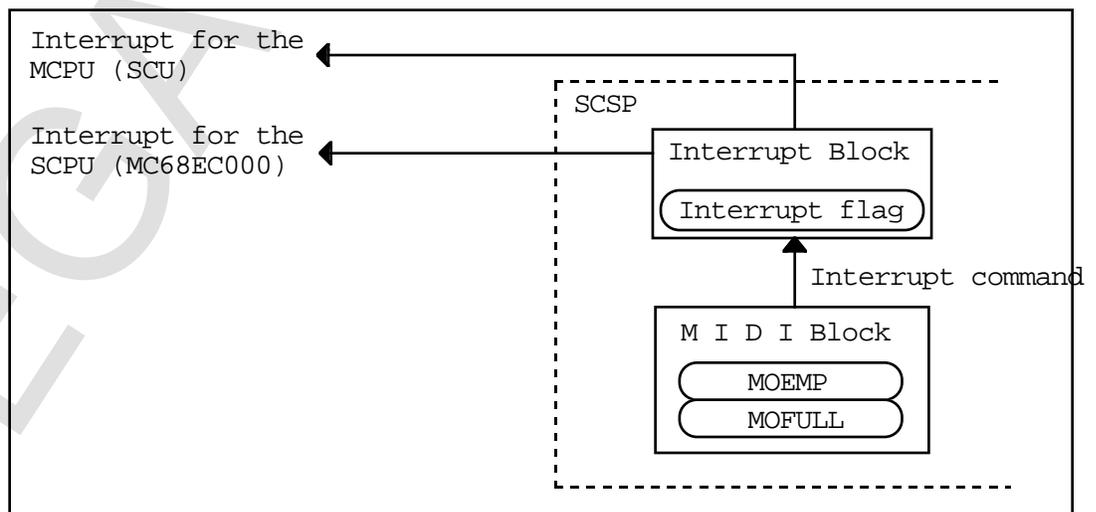


Figure 4.61 MIDI OUT Block and the Interrupt Generation Block

MOBUF[7:0] (W) ; Midi Output BUffer

This is a MIDI output data buffer. For MIDI-OUT-side transfer, write data (to be transferred) into "MOBUF". Afterwards, the data is automatically transferred.

Timer Register

The SCSP has three pre-scaler 8 bit up count timers: A, B and C. The pre-scaler execute time is set individually by "TACTL", "TBCTL" and "TCCTL."

TACTL[2:0] (W) ; Timer-A ConTrol

Designates the increment cycle for timer A.

Table 4.33 Increment Cycle for Timer A

TACTL	Increment Cycle
0	Once every sample
1	Once every 2 samples
2	Once every 4 samples
3	Once every 8 samples
4	Once every 16 samples
5	Once every 32 samples
6	Once every 64 samples
7	Once every 128 samples

TBCTL[2:0] (W) ; Timer-B ConTrol

Designates the increment cycle for timer B.

Table 4.34 Increment Cycle for Timer B

TBCTL	Increment Cycle
0	Once every sample
1	Once every 2 samples
2	Once every 4 samples
3	Once every 8 samples
4	Once every 16 samples
5	Once every 32 samples
6	Once every 64 samples
7	Once every 128 samples



TCCTL[2:0] (W) ; Timer-C Control
 Designates the increment cycle for timer C.

Table 4.35 Increment Cycle for Timer C

TCCTL	Increment Cycle
0	Once every sample
1	Once every 2 samples
2	Once every 4 samples
3	Once every 8 samples
4	Once every 16 samples
5	Once every 32 samples
6	Once every 64 samples
7	Once every 128 samples

Table 4.36 shows the count cycle in relation to the setting values in each of "TACTL", "TBCTL" and "TCCTL". Table 4.37 shows the shortest interrupt time ("TIMA"="TIMB"="TIMC"="FEH") and the longest interrupt time ("TIMA"="TIMB"="TIMC"="00H").

Table 4.36 Count Cycle in Relation to the Settings of TACTL, TBCTL and TCCTL

TACTL, TBCTL, TCCTL Setting values	Count Cycle (Compared with 1Fs=1/44.1K)	Actual Count Cycle Time [μsec]
0 H	F s	22.6757
1 H	F s/2	45.3515
2 H	F s/4	90.7029
3 H	F s/8	181.4059
4 H	F s/16	362.8118
5 H	F s/32	725.6236
6 H	F s/64	1451.2472
7 H	F s/128	2902.4943

Table 4.37 Shortest and Longest Interrupt Time

TACTL, TBCTL, TCCTL Values	Shortest Interrupt Time [μsec]	Longest Interrupt Time [msec]
0 H	22.6757	5.8050
1 H	45.3515	11.6100
2 H	90.7029	23.2200
3 H	181.4059	46.4399
4 H	362.8118	92.8798
5 H	725.6236	185.7596
6 H	1451.2472	371.5193
7 H	2902.4943	743.0385

Based on Tables 4.36 and 4.37, the following equation can be used to find the interrupt time.

$$\text{Interrupt Time} = \{255 (\text{FF}_{\text{H}}) - \text{TIMA (B, C) settings}\} \times \text{count cycle time}$$

Counting begins immediately after the settings are set to "TIMA", "TIMB" and "TIMC". When the count reaches FFH, an interrupt is sent from the timer valid for an interrupt. (When the timer is not used, please prohibit the use of interrupt.)

TIMA[7:0] (W) ; TIMer-A count data

This is timer A. This timer is an up counter. When all of the bits reach "1B" a request for an interrupt occurs.

TIMB[7:0] (W) ; TIMer-B count data

This is timer B. This timer is an up counter. When all of the bits reach "1B" a request for an interrupt occurs.

TIMC[7:0] (W) ; TIMer-C count data

This is timer C. This timer is an up counter. When all of the bits reach "1B" a request for an interrupt occurs.



Registers are explained below. There are two types of registers: those that control interrupts for the main (these have names that begins with MC~); and those that control the interrupts for the sound (these have names that begins with SC~).

SCIPD[10:0] ; Sound-Cpu Interrupt PenDing

This register monitors the interrupts for the sound CPU (interrupt flag). When an interrupt request occurs, the appropriate interrupt request flag is set to "1B", so by reading the CPU "SCIPD" register, it can be determined which interrupt just occurred. Also, no matter what the enable register ("SCIEB") is set at, all interrupt requests are monitored. The corresponding flag can be reset by the interrupt reset register ("SCIRE"). Only bit 5 can be read or written; all others are read only. Writing "1B" to bit 5 applies an interrupt to the sound CPU. However, writing "0B" is invalid.

SCIEB[10:0] (R/W) ; Sound-Cpu Interrupt EnaBle

This register enables interrupts to the sound CPU. Setting it to "1B" enables the corresponding hardware interrupt. Reading "SCIPD" will determine whether there are interrupts, regardless of the "SCIEB" setting. Setting to "0B" is invalid

SCIRE[10:0] (W) ; Sound-Cpu Interrupt REset

This is the sound CPU, interrupt request reset flag. When set to "1B", the corresponding hardware interrupt is reset. (If the "SCIRE" for the bit with an interrupt occurring is set to "1B", the "SCIPD" also will change from "1B" to "0B".)

MCIPD[10:0] (R) ; Main-Cpu Interrupt PenDing

This register monitors the interrupts for the main CPU (interrupt flag). When an interrupt request occurs, the appropriate interrupt request flag is changed to "1B," so by having the CPU read the "MCIPD" register, it can be determined which interrupt just occurred. Also, all interrupt requests are monitored regardless of what the enable register ("MCIEB") is set to. The corresponding flag can be reset by the interrupt reset register ("MCIRE"). Only bit 5 can be read or written; all others are read only. Writing "1B" to bit 5 applies an interrupt to the main CPU. However, writing "0B" is invalid.



MCIEB[10:0] (R) ; Main-Cpu Interrupt EnaBle

This register enables interrupts to the main CPU. Setting it to "1B" enables the corresponding hardware interrupt. Reading "MCIPD" determines whether there are interrupts, regardless of the "MCIEB" setting. Setting to "0B" is invalid.

MCIRE[10:0] (R) ; Main-Cpu Interrupt REset

This is the main CPU interrupt request reset flag. When set to "1", the corresponding hardware interrupt request is reset. (If the "MCIRE" for the bit with an interrupt occurring is set to "1B", the "MCIPD" also will change from "1B" to "0B".)

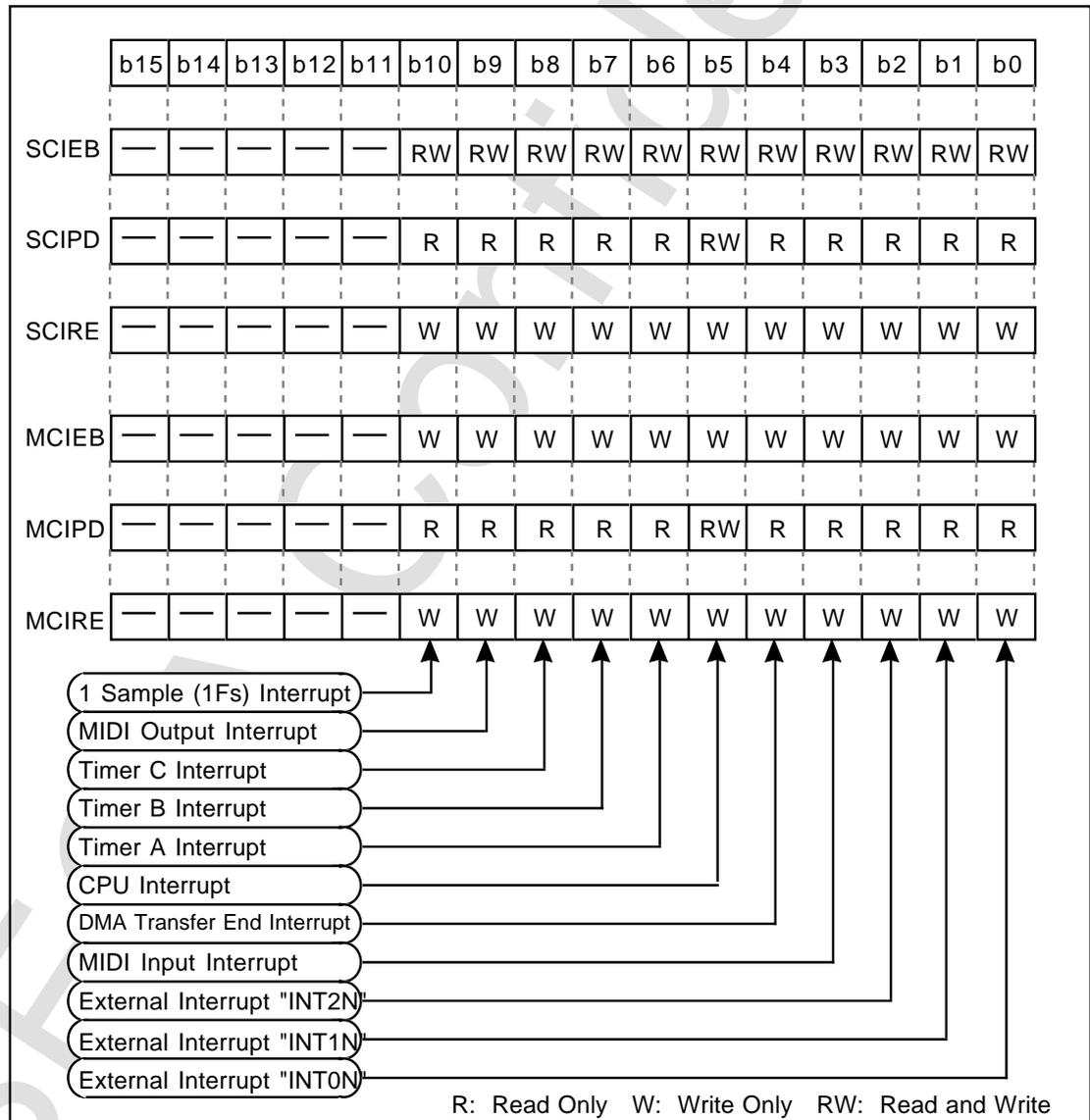


Figure 4.63 Interrupt Register Bit Accommodation

Table 4.38 Bit Factor Interrupt Registers

Bit	Reason for Interrupt
0	Applicable to the external interrupt input "INT0N" interrupt input.
1	Applicable to the external interrupt input "INT1N" interrupt input.
2	Applicable to the external interrupt input "INT2N" interrupt input.
3	Applicable to the MIDI input interrupt Interrupt occurs when the MIDI-IN side FIFO buffer memory captures data from an empty state. Automatic cancellation occurs when data is all read from the FIFO buffer and the buffer is empty.
4	Applicable to DMA transfer end interrupt Interrupt occurs when the DMA transfer using the SCSP internal DMA is finished. (When data transfer of the block set in "DLG" [length (volume)] is completely finished.)
5	Applicable to the CPU manual interrupt By writing to the CPU (main or sound), an interrupt can be applied to the sound or main CPU. Interrupt is applied when "1B" is written. ("0B" is invalid.)
6	Applicable to a timer A interrupt
7	Applicable to a timer B interrupt
8	Applicable to a timer C interrupt
9	Applicable to the MIDI output interrupt An interrupt request occurs when the MIDI-OUT side FIFO buffer memory becomes empty. The interrupt is automatically cancelled when data is written into the MIDI-OUT buffer memory (register) and it is no longer empty.
10	Applicable to one interrupt each sample (1 sample= 22.68μsec= 1/44.1K intervals).
11~15	Invalid

✳ About SCILV0, 1, 2

This is the register that sets the auto vector interrupt level for the sound CPU. Each register is partitioned in units of one bit per every interrupt factor. When setting, view Figure 4.65 vertically.

In the example of bit 7, timers B, C and MIDI output interrupts, and the interrupt level for each sample are all set at once. Level is set with a 3 bit code, but each bit is distributed among SCILV0, 1 and 2.

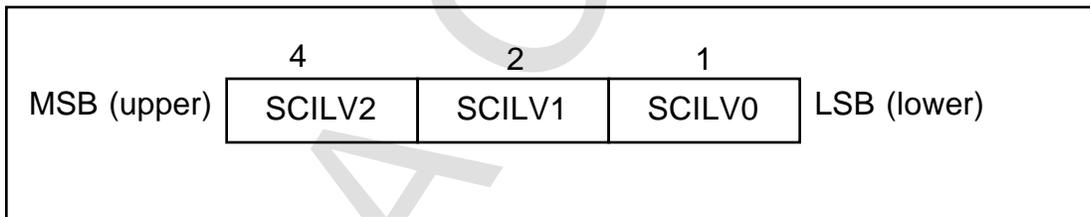


Figure 4.64 3 Bit Code and Register Accommodation

The 3 bit code used to set the level is like Figure 4.64. For example, if it is set at 101B, the interrupt level is 5. "000B" is level 0 interrupt, so an interrupt would not be applied.

However, the actual format of the register is shown in Figure 4.65. Be careful when setting the values.



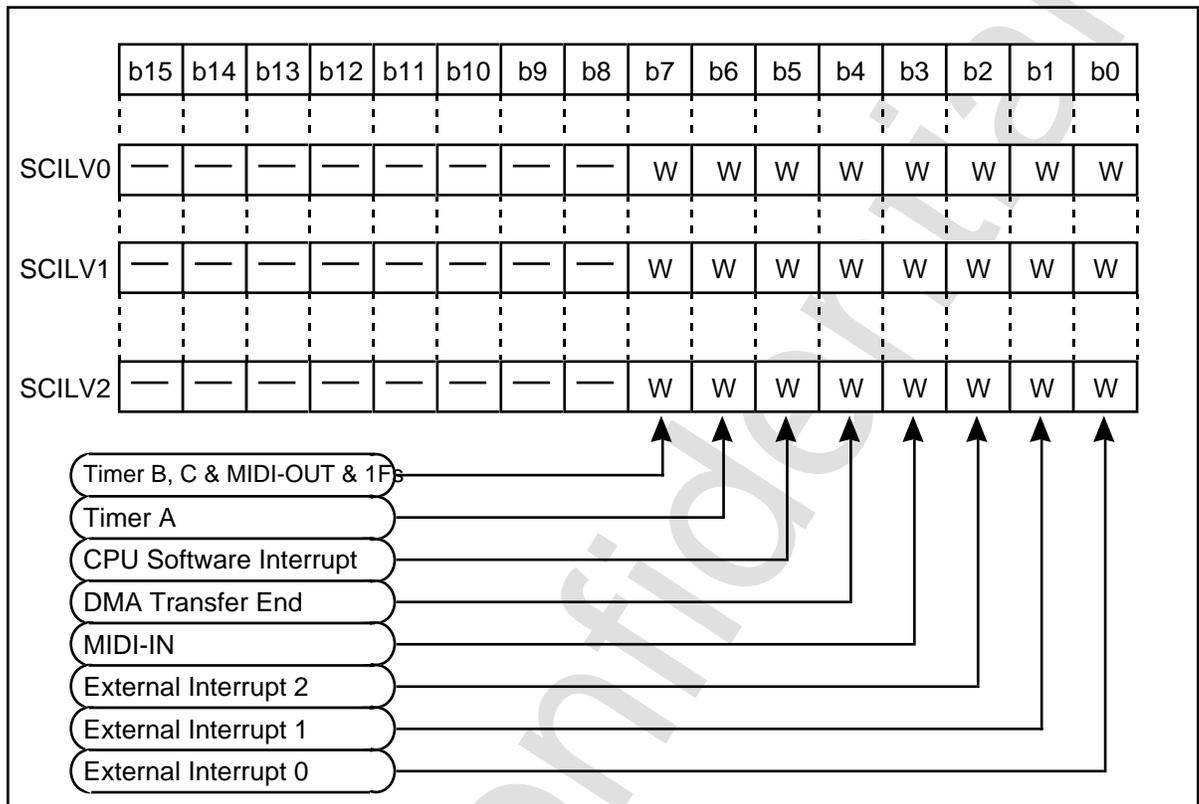


Figure 4.65 Format of the Interrupt Level Set Register

SCILV0[7:0] (W) ;Sound-Cpu Interrupt LeVel bit0

Specifies bit0 of the sound CPU interrupt level code as defined in bit application.

SCILV1[7:0] (W) ;Sound-Cpu Interrupt LeVel bit1

Specifies bit1 of the sound CPU interrupt level code as defined in bit application.

SCILV2[7:0] (W) ;Sound-Cpu Interrupt LeVel bit2

Specifies bit2 of the sound CPU interrupt level code as defined in bit application.

Examples of interrupt settings are explained below.

Conditions: Apply a level 6 interrupt to the sound CPU while timer A applies an interrupt to the main CPU.

Procedure: Currently, items without an interrupt applied set the timer after all of the interrupt settings are finished, .

- 1: Set the sound CPU interrupt level to 6
 - Level 6 in the 3 bit code is "110B".
 - The bit that controls timer A is bit 6, so set bit 6 in "SCILV0" to "0B".
(At 100425H address in byte, "00H", or at 100424H address in words "0000H".)
 - Bit 6 in "SCILV0" to "1B".
(At 100427H address in byte, "40H", or at 100426H address in words "0040H".)
 - Bit 6 in "SCILV0" to "1B".
(At 100429H address in byte, "40H", or at 100428H address in words "0040H".)

- 2: Set parameters in registers "MCIEB" and "SCIEB" so that interrupts will be applied to both the main and sound CPUs.
 - Set bit 6 in register "MCIEB" to "1B" so that an interrupt will be applied by timer A.
(At 10042BH address in byte, "40H", or at 10042AH address in words "0040H".)
 - Set bit 6 in register "SCIEB" to "1B" so that an interrupt will be applied to the sound CPU as well.
(At 100421H address in bytes, "40H", or at 10042EH address in words "0040H".)
 - Furthermore, by setting the parameters in the timer, count will begin immediately afterwards. When an overflow occurs, an interrupt will be applied.

- 3: Use the reset registers to remove the interrupts from either the sound or main CPUs.



DMA Transfer Register

The DMA inside the SCSP can only transfer between the SCSP internal control register and the sound memory. From this the maximum bytes transferable is 3812 bytes (EE4H), (allocated in the empty memory from 100000H~100EE3H). Also, the registers relating to the DMA cannot be changed with the DMA transfer. During DMA transfer address continually move in the increase direction.

- The speed of the main CPU and sound CPU may drop during DMA transfer.
- Access through DMA transfer to the control register of the DMA controller is absolutely prohibited. The operation when this transfer is performed is not guaranteed at all.
- DMA transfer is word (16bit) transfer.

DGATE (R/W) ;Dma GATE (and "0")

The DMA controller block diagram is shown in Figure 4.66.

It can freely initialize the areas in the sound memory and the SCSP internal control register to "0". When this bit is "1B", "0" clear is executed (When actually starting, "DEXE" must be executed).

Deleting DGATE transfer ("0" write) will not affect the transfer source data. In addition, data will not be lost.

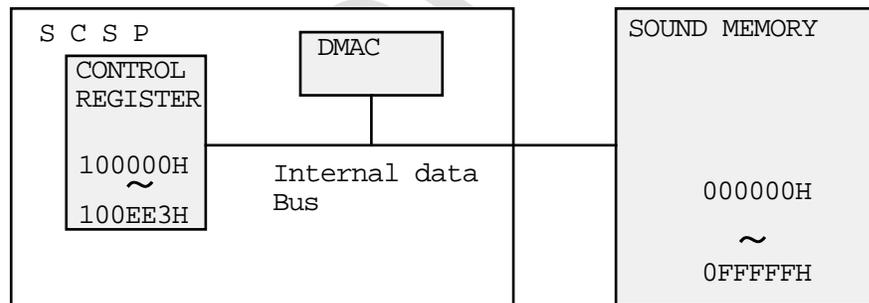


Figure 4.66 DMA Controller Block Diagram

DDIR (R/W) ; Dma (transferring) DIRection

Designates the DMA transfer direction. When this bit is "0B", data is transferred from the sound memory to the LSI block register; "1" transfers data in the opposite direction.

Table 4.39 DMA Transfer Direction

DDIR	Transfer Direction
0	From sound memory to LSI internal register
1	From LSI internal register to sound memory

DEXE (R/W) ; Dma EXEcution

Designates the start of the DMA transfer. When this bit is "1", the DMA transfer starts. Writing "0" is invalid. Also, when DMA transfer ends, this bit is automatically reset to "0".

Table 4.40 DMA Transfer

DEXE	Transfer Status
0	Invalid or transfer has ended
1	Starts DMA transfer

DMEA[19:1] (W) ; Dma MEmory start Address

Designates the sound memory address (word units) that starts the DMA transfer.

DRGA[11:1] (W) ; Dma ReGister start Address

Designates the block register byte address in the LSI (word units) that starts the DMA transfer.

DTLG[11:1] (W) ; Dma (Transferring) LenGth

Designates the word count being transferred during DMA transfer. At that time, be careful that the area of the transfer origin or destination does not exceed the area of the sound memory area or the LSI internal block register area.



4.3 DSP Memory Control Register

RBL[1:0] (W) ; Ring Buffer Length

Designates the length of the ring buffer.

Table 4.41 RBL and the Ring Buffer Length

bit	Buffer Length
0	8K word
1	16K word
2	32K word
3	64K word

RBP[19:13] (W) ; Ring Buffer (header) Pointer

Designates the first address of the ring buffer. This address designation is per 4K-word boundary.

The important base of DSP effect processing is delay processing. Delay is accomplished by storing the input data in memory and reading it out again after a certain time lag is applied. This storage area is called the ring buffer.

By combining the number of effect programs used and the effect delay timer used simultaneously, the ring buffer area settings can be set to perform the "RBL" and "RBP" registers.

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Chapter 5

SCSP Internal DSP Operation

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5.1 DSP Configuration

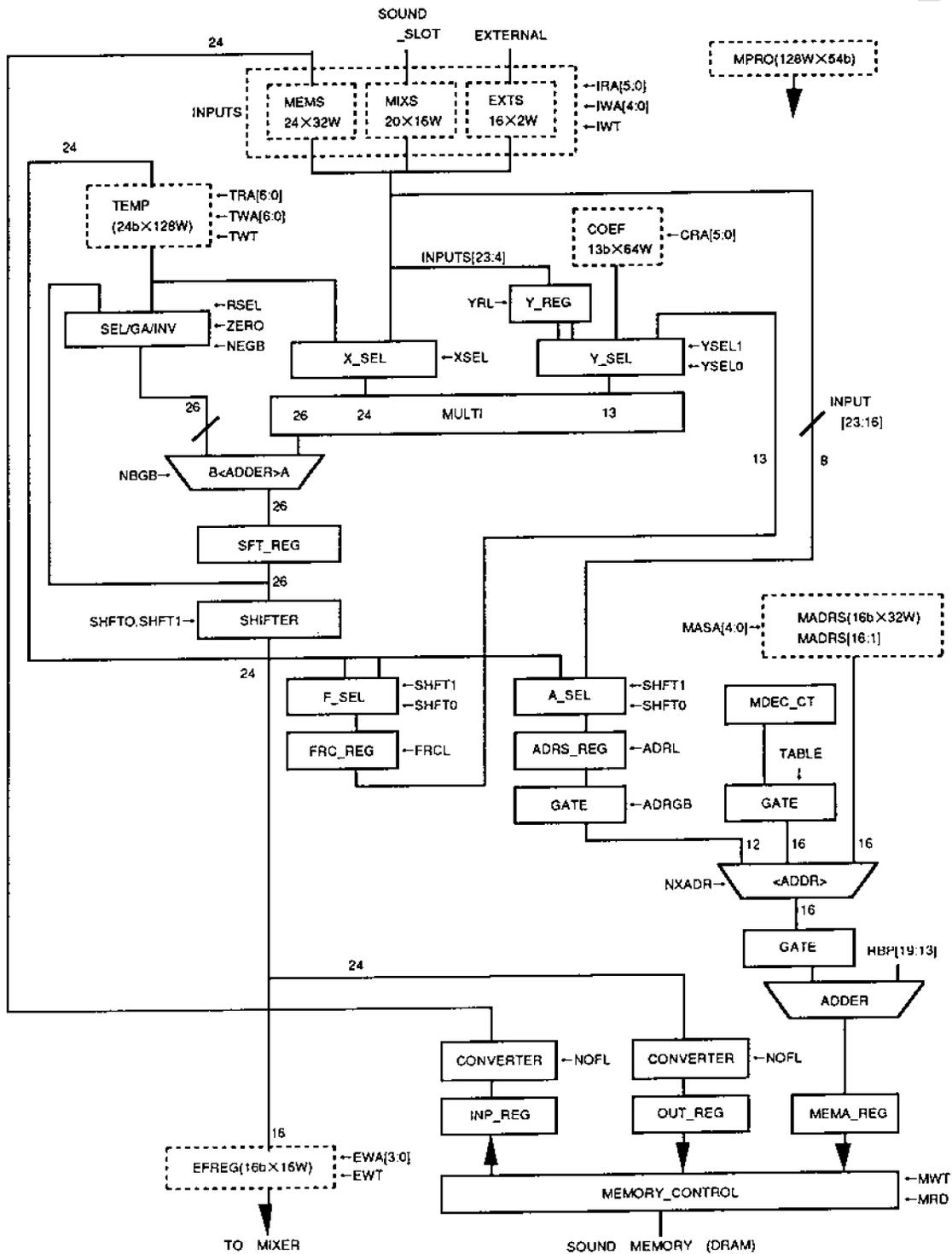


Figure 5.1 DSP Configuration Diagram



5.2 DSP Internal RAM

EXTS[15:0] ; EXTernal Stack

Represents the digital audio input data buffer. (This register cannot be accessed.)

MIXS[19:0] (R/W) ; MIX Stack

Represents the sound data buffer from the input mixer.

MEMS[23:0] (R/W) ; MEMory Stack

Represents the input data buffer from the sound memory.

TEMP[23:0] (R/W) ; TEMPorary register

Represents the DSP work buffer (128 words). The DSP work buffer is in ring buffer configuration, the pointer decreases at every sample.

COEF[12:0] (R/W) ; COEFicient register

Represents the DSP coefficient buffer.

MADRS[16:1] (R/W) ; Memory ADdReSs register

Represents the DSP address buffer.

MPRO[63:0] (R/W) ; Micro PROgram register

Represents the DSP micro program buffer.

EFREG[15:0] (R/W) ; EFfect REGister

Represents the DSP output buffer.