

YAMAHA LSI

YM3812

FM OPERATOR TYPE-L (OPLII)

APPLICATION MANUAL

YAMAHA

YM3812 APPLICATION MANUAL

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1. OPLII

1-1 Overview

The FM Operator Type-LII (OPLII) is a new type of sound generator designed for use with Captain systems and videotext systems. This device uses the same frequency modulation (FM) system used in our Music Synthesizer Yamaha-DX7 and other similar instruments. This allows for the production of a wide variety of sounds using software control. This sound generator is also equipped with functions for the production of rhythm sounds. This does not use the FM system, but instead creates sound through combining various sound frequencies, including white noise.

The OPLII has also has a built-in low frequency oscillator for vibrato and AM effects, reducing the amount of programming required to produce special effects.

As this sound output from LSI is digital, a D/A converter such as YM3014 is necessary.

1-2 Features

- FM sound generation system for realistic sound
- Mode selection of simultaneous voicing of 9 sounds or 6 melody sounds and 5 rhythm sounds
- Built-in vibrato oscillator/amplitude modulation oscillator (AM)
- Composite sine wave speech synthesis also possible
- Input/output TTL compatible
- Si-gate CMOS-LSI
- 5V single power supply

1-3 Overview of FM System

FM refers to Frequency Modulation and is a system using combinations of the higher harmonics created by modulation. This allows for the generation of waveforms containing high harmonics and non-harmonic sounds using circuitry which is relatively simple. The correspondence between the modulation index and spectrum distribution of higher harmonics is extremely natural allowing for production of a wide range of sounds from natural instruments to electronic sounds.

The following formula (1) expresses the four parameters relevant to FM sound generation.

$$F = A \sin(\omega c t + I \sin \omega m t) \quad (1)$$

Where A is the output amplitude, I is the modulation index, and ωm and ωc are the frequencies of the carrier and modulator respectively. Formula (1) can also be expressed in the following manner :

$$F = A [J_0(I) \sin \omega c t + J_1(I) \{ \sin(\omega c + \omega m)t - \sin(\omega c - \omega m)t \} + J_2(I) \{ \sin(\omega c + 2\omega m)t + \sin(\omega c - 2\omega m)t + \dots \}] \quad (2)$$

$J_n(I)$ is a type 1 Bessel function of the n series. As (2) indicates, the amplitude of each of the harmonics is expressed by the Bessel function of the modulation index. Formula (1) indicates that FM sound generation is highly effective for combining special music and sound effects. This is not a string-type sound source which does not provide an even distribution of higher harmonics. The feedback FM of this system is shown in (3) below :

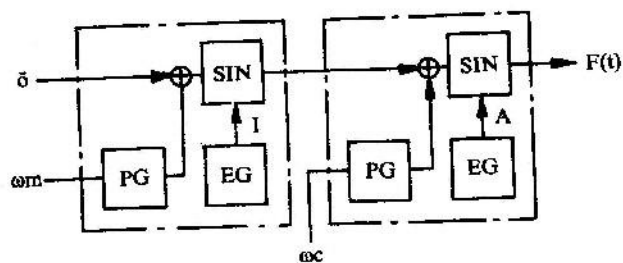
$$F = A \sin(\omega c t + \beta F) \quad (3)$$

Where β is the feedback factor. This feedback FM is also possible by string-type sound generation where the higher harmonic spectrum is a sawtooth wave.

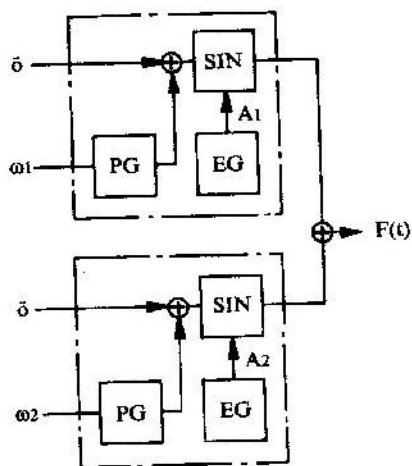
The following three function blocks are necessary for FM sound generation :

- a. Phase generator (PG) to generate ωt
- b. Envelope generator (EG) allowing for amplitude A and modulation index to be expressed as time functions
- c. Sin table (sin)

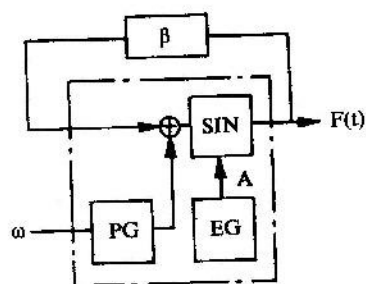
Combining these three elements into a single allows for configuration of the FM system shown in Fig. 1-1. Thus, if the concept of these units (operator cells : OP) is used, FM sound generation is a matter of setting the frequency and EG parameters within the units, and then combining the data between units.



$$a. F(t) = A \sin(\omega_c t + I \sin \omega_m t)$$



$$b. F(t) = A_1 \sin \omega_1 t + A_2 \sin \omega_2 t$$



$$c. F(t) = A \sin(\omega t + \beta F(t))$$

Fig. 1-1 FM sound generation using unit cells

2. Overview of Functions

2-1 Main Functions

The OPILL is equipped with a total of three voicing modes : nine sound simultaneous voicing mode, 6 melody/5 rhythm sound voicing mode, and composite sine wave speech synthesis mode. Each of these modes can be selected by software.

a) **9 sound simultaneous voicing mode :**

This mode allows for simultaneous voicing of nine FM sounds having different voices. Both the rhythm bit (R) and speech synthesis bit (CSM) must be set to "0".

b) **6 melody/5 rhythm sound voicing mode :**

When the OPLII is set to this mode, the number of melody sounds which be simultaneously voiced is reduced by three to six, but five rhythm sounds are added (bass drum, snare drum, tom tom, top cymbal, and high hat cymbal). The bass drum is created using FM sound generation, the tom tom by sine waves, and the other three rhythm instruments are simulated by composite frequencies. This mode is effective when connected to Captain or similar systems using text.

c) **Speech synthesis mode :**

Speech synthesis using the OPLII is by the composite sine wave speech synthesis method. Voices are simulated using 3 to 6 sine waves and pitch.

In addition to the above voicing modes, the OPLII is also equipped with a built-in vibrato oscillator and amplitude modulation oscillator. These effects can be used to create a sound which closely simulates the sound of natural instruments. Inclusion of these functions allows for a reduction in required programming. The OPLII also has both a long and short timer allowing for use as reference signals for scanning of key switches and tempo clock. The short timer can be used as a pitch generator for composite sine wave speech synthesis.

2-2 Pin Layout

Vcc	1	24	ϕM
\overline{IRQ}	2	23	ϕSY
\overline{IC}	3	22	NC
A0	4	21	MO
\overline{WR}	5	20	SH
\overline{RD}	6	19	NC
\overline{CS}	7	18	D7
NC	8	17	D6
NC	9	16	D5
D0	10	15	D4
D1	11	14	D3
GND	12	13	D2

* NC : No Connection

TOP VIEW(24pin DIP, 24pin SOP)

2-3 Description of Pin Functions

- a) ϕM
Master clock for OPLII
- b) $\phi SY \cdot SH$
Clock needed for converting digital output of OPLII into analog signals (ϕSY) and synchronization signal (SH). These signals allow for direct connection to the YM3014 D/A converter.
- c) $D_0 \sim D_7$
8 bit bidirectional data bus.
- d) $\overline{CS} \cdot \overline{RD} \cdot \overline{WR} \cdot A_0$
Control data bus comprised of $D_0 \sim D_7$.

\overline{CS}	\overline{RD}	\overline{WR}	A_0	
0	1	0	0	Write address of register to OPLII
0	1	0	1	Write contents of register to OPLII
0	0	1	0	Status of OPLII is read.
0	0	1	1	Data of data bus $D_0 \sim D_7$ not assured
1	x	x	x	Set data bus $D_0 \sim D_7$ to high impedance

- e) \overline{IRQ}
Interrupt signal sent from either of two timers. When the time programmed for the timer elapses, driven to low level. Interrupts can be masked by program.
- f) \overline{IC}
Clears the contents of all registers of OPLII, and initializes OPLII.
- g) MO
Outputs music or audio signal converted into 13 bit serial signal. D/A converter (YM3014 or equivalent) is need for converting this digital output signal into its analog equivalent.
- h) GND
Ground pin
- i) V_{cc}
+ 5V power supply pin

2-4 Bus Control

Data bus control for reading and writing addresses and data of the OPLII is performed by the \overline{CS} , \overline{WR} , \overline{RD} , and A0 signals. The following four modes can be set according to the state of this four signals.

Table 2-1 Mode Selection

	\overline{CS}	\overline{RD}	\overline{WR}	A0	
1	1	x	x	x	Inactive mode
2	0	1	0	0	Address write mode
3	0	1	0	1	Data write mode
4	0	0	1	0	Status write mode
5	0	0	1	1	Inhibit

a) **Inactive mode :**

When level of \overline{CS} is "1", the data bus $D_0 \sim D_7$ has high impedance.

b) **Address write mode :**

When an address is to be written, the control signals are set to the address write mode and the address data is set in the data bus. The address of the designated internal register is set and data can be written. It should be noted that after address data has been written, 12 cycles of the master clock (ϕM) must elapse before music data is written.

c) **Data write mode :**

When the control signals are set to the data write mode, the data of $D_0 \sim D_7$ (data bus) is written to the register having the designated address. A wait period is also required after a data write, in the same manner as an address write. In this case, the wait period is 84 cycles of the master clock (ϕM) before the next data or address is written.

d) **Status read mode :**

Setting the control signals to this mode allows for output of the status information stored in the status register of OPLII.

e) **Inhibit :**

The data of the bus is meaningless when the control signals are set to this mode. Control of this data is not possible.

The following precautions must be taken regarding the address and data write modes. When an address or data is written to an internal register of the OPLII, the following wait period is required before the next operation is performed. The period varies between the address write and data write modes. The CPU generates the wait period shown in Table 2-2 for the OPLII. Data integrity cannot be assured if this wait period is ignored.

Table 2-2 Wait period

Mode	Wait period
Address write mode	12 cycles
Data write mode	84 cycles

Note: The indicated number of cycles for the wait period is the number of master clock cycles.

2-5 Channels and Slots

The OPLII is capable of voicing 9 FM sounds (9 channels). Each of these sounds has two operator cells. There is, however, actually one operator cell for the system, and thus the signal must pass through this operator cell a total of 18 times for 9 FM sounds. The order (slot number) which channel signals pass this operator cell corresponds to the register numbers. Voicing control of the various sounds is possible through control of the registers corresponding to the slots.

The F-number data for each channel controls two slots.

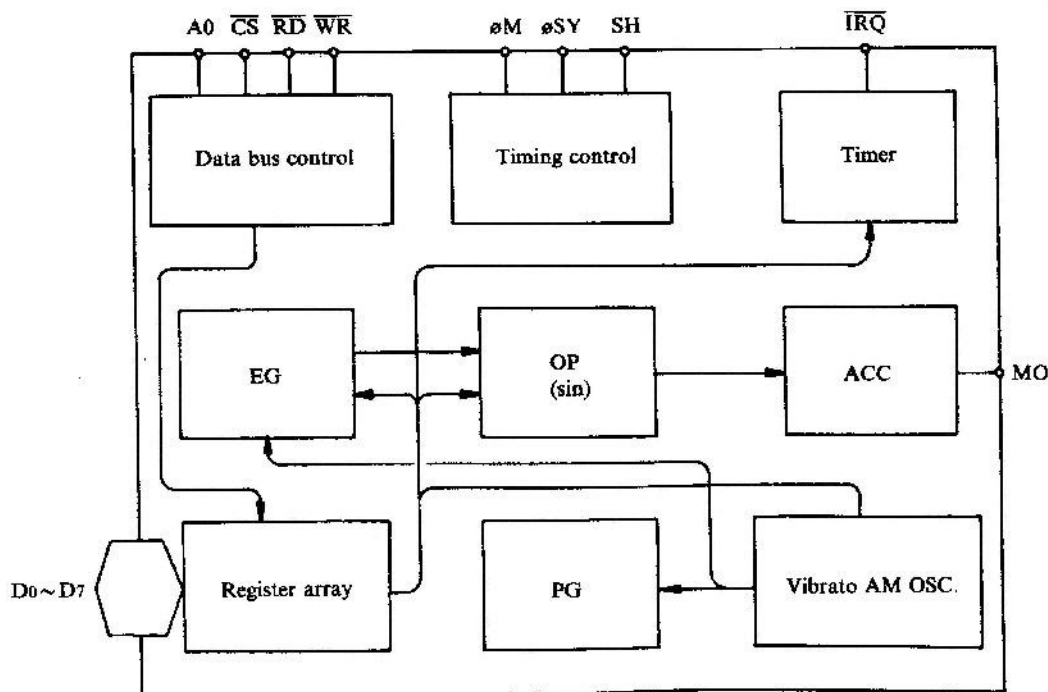
The relationship between these two slots (first slot and second slot) in the FM modulation mode is such that the first slot is always the modulating wave, and the second slot is the carrier wave. This first slot can also be set to the FM feedback mode. Refer to (2-1-9) for settings in this mode.

The relationship between channels and slots is shown in Table 2-3.

Table 2-3 Channels and Slots

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	Slot number
1	2	3	1	2	3	4	5	6	4	5	6	7	8	9	7	8	9	Channel number
1			2			1			2			1			2			Slot number for each channel
20	21	22	23	24	25	28	29	2A	2B	2C	2D	30	31	32	33	34	35	Relationship between data for each slot and registers (ex. \$20~\$35)
C0	C1	C2	C0	C1	C2	C3	C4	C5	C3	C4	C5	C6	C7	C8	C6	C7	C8	Relationship between data for each channel and registers (ex. \$C0~\$C8)

2-6 Block Diagram



2-7 Address Map

ADDRESS	D7	D6	D5	D4	D3	D2	D1	D0
01				TEST				
02	TIMER-1							
03	TIMER-2							
04	RST	MASK T1 T2					ST2 ST1	
08	CSMSEL							
20	AM	VIB	EG-TYP	KSR	MULTI			
35								
40								
	KSL		TL					
55	AR				DR			
60								
75	SL				RR			
80								
95	F-NUMBER (L)							
A0								
A8			KON	BLOCK		F-NUM (H)		
B0								
B8	DEP AM	DEP VIB	R	BD	SD	TOM	TC	HH
BD					FB			C
C0								
C8								WS
E0								
F5								

COMMENT

TEST DATA OF LSI	When the value is "0", D5 is compatible with YM3526.
DATA OF TIMER-1	
DATA OF TIMER-2	
IRQ-RESET/CONTROL OF TIMER-1,2	
CSM SPEECH SYNTHESIS MODE/NOTE SELECT	
AM-MOD/VIBRATO/EG-TYPE	
KEY-SCALE RATE/MULTIPLE	
KEY-SCALE LEVEL/TOTAL LEVEL	
ATTACK RATE/DECAY RATE	
SUSTAIN LEVEL/RELEASE RATE	
KEY-ON/BLOCK/F-NUMBER	
DEPTH(AM/VIB)/RHYTHM(BD, SD, TOM, TC, HH)	
FEEDBACK/CONNECTION	
WAVE SELECT	

Status registers

D7	D6	D5	D4	D3	D2	D1	D0
IRQ	FLG T1	FLG T2					

3. Description of Operation

All functions of the OPLII are controlled by data written from the microprocessor to the register array. The shape of the envelope for music, modulation factor, frequency, voicing, mode, and other parameters are determined according to the data which is written to the registers. Data can be combined to generate the sound of a piano, violin, or other instrument. There is an extremely large number of combinations with a high degree of complexity. This chapter deals solely with the function of the various registers. Refer to the chapter on creating music for details on the various possible combinations.

3-1 Registers

The registers comprise an area of 777 bits as shown in the address map of 2-7. The addresses in this diagram are the subaddresses allocated to the various registers in the OPLII. Music data is written to the internal registers through these subaddresses. Thus, data is stored in the OPLII by first sending the subaddress data which will hold this data, and then sending the music data itself. When the same subaddress is to be accessed a number of times, the subaddress data need only be sent in the beginning. Music data can then be sent, without address data, to update the data.

The initial setting for all registers is "0" (initial clear = "0").

3-1-1 Test : Address (\$01)

The only use of this address is for testing of the LSI device by Yamaha.

The bits are normally all "0". The D5 bit, however, has a special meaning. The output waveform can be controlled by setting this bit to 1. (Refer to 3-1-2)

3-1-2 Timer

There are two timers: Timer 1 which has a resolution of 80μs and Timer 2 which has a resolution of 320μs. Starting, stopping, and flag control of both timers is possible. When a timer flag is set, the $\overline{\text{IRQ}}$ pin is driven to low level, and the microprocessor is notified of a timer interrupt.

i) Timer-1 : Address (\$02)

Timer 1 is an 8 bit presettable counter. If an overflow occurs, the flag is set, and the preset value is loaded. In addition to normal timer functions, Timer 1 is also used for control of composite speech synthesis. When an overflow occurs in this mode, all slots are set to Key-ON (voicing), and then immediately to Key-OFF. This operation allows for composite speech synthesis.

\$02

D7	D6	D5	D4	D3	D2	D1	D0
----	----	----	----	----	----	----	----

$$Tov(ms) = (256 - N_1) * 0.08 \quad @ \phi M = 3.6MHz$$

$$N_1 = D_7 * 2^7 + D_6 * 2^6 + \dots + D_1 * 2 + D_0$$

ii) **Timer-2 : Address (\$03)**

Timer 2 is an 8 bit presetable counter like Timer 1. The difference between the two timers is that the resolution of Timer 1 is 80μs, and the resolution of Timer 2 is 320μs.

\$03

D7	D6	D5	D4	D3	D2	D1	D0
----	----	----	----	----	----	----	----

$$Tov(\omega s) = (256 - N_2) * 0.32 \quad @ \phi M = 3.6MHz$$

$$N_2 = D_7 * 2^7 + D_6 * 2^6 + \dots + D_1 * 2 + D_0$$

iii) **Timer Control : Address (\$04)**

This register is used for start, stop, and flag control of Timers 1 and 2. These operations are controlled by the bit state of D0, D1, D5, D6, and D7.

\$04

D7	D6	D5	D4	D3	D2	D1	D0
IRQ RESET	MASK T1	MASK T2	/			ST2	ST1

D0 (ST1) : Controls starting and stopping of Timer 1.

When this bit is "1", the preset value is loaded into Timer 1, and counting started.
When this bit is "0", Timer 1 does not operate.

D1 (ST2) : Performs the same operation as D0 (ST1) for Timer 2.

D5 (MASK T2) : When this bit is "1", the flag for Timer 2 is masked (always "0"), and has no effect on operation of Timer 2.

D6 (MASK T1) : This bit masks the flag for Timer 1.

D7 (IRQ RESET) : Resets the flags for Timers 1 and 2.

When the D7 bit is set to "1", the data of D0~D6 is ignored, and the D7 bit is automatically reset to "0".

3-1-3 CSM Mode/Keyboard Split : Address (\$08)

This register sets the mode to the music mode or speech synthesis mode, and determines the keyboard split for keyboard scaling of rate.

\$08

D7	D6	D5	D4	D3	D2	D1	D0
CSM	NOTE SEL	/					

D₆ (NOTE SEL) : This bit controls the split point of the keyboard. When "0", the keyboard split is the second bit from the most significant bit (MSB) of the F-Number. The MSB of the F- Number is controlled when "1". This is illustrated below.

D₆ = "0"

0	1	2	3	4	5	6	7	Octave
0	1	2	3	4	5	6	7	Block data
1	1	1	1	1	1	1	1	F-Num*MSB
0	1	0	1	0	1	0	1	F-Num*2nd
0	1	2	3	4	5	6	7	Keyboard split number

D₆ = "1"

0	1	2	3	4	5	6	7	Octave
0	1	2	3	4	5	6	7	Block Data
0	1	0	1	0	1	0	1	F-Num*MSB
*	*	*	*	*	*	*	*	F-Num*2nd
0	1	2	3	4	5	6	7	Keyboard split number

* DON'T CARE

D₇ (CSM) : The composite sine wave speech synthesis mode is selected when "1". All channels must be in the Key-OFF state when this mode is selected.

3-1-4 AM/VIB/EG-TYP/KSR/Multiple : Address (\$20 ~ 35)

This register controls the multiple for the conversion of the frequency data given by the envelope shape and F-Number into carrier and modulating wave frequencies corresponding to the frequency components of music.

\$20 ~ \$35

D7	D6	D5	D4	D3	D2	D1	D0
AM	VIB	EG-typ	KSR	MULTI			
				2 ³	2 ²	2 ¹	2 ⁰

D₀~D₃ (MULTIPLE) : The frequencies of the carrier and modulating waves are controlled by the multiples shown in Table 3-1.

< Example >

Frequency of F-Number ωf
 Multiple for carrier wave 1
 Multiple for modulating wave 7

$$F(t) = E \sin(\omega f t + I \sin(7 \omega f t))$$

Table 3-1 Multiples

MUL	Multiple	MUL	Multiple	MUL	Multiple	MUL	Multiple
0	1/2	4	4	8	8	C	12
1	1	5	5	9	9	D	12
2	2	6	6	A	10	E	15
3	3	7	7	B	10	F	15

D₄ (KSR) : This bit gives the key scaling for the rate.

The leading and trailing edges (attack and decay) of the sound of natural instruments tends to increase as the interval becomes higher. Key scaling of the rate allows for a simulation of this phenomenon. Table 3-2 shows the value of the offset which is added to the speed for each the intervals. Thus, the actual rate for ADSR (attack, decay, sustain, and release) becomes the rate set for each with the indicated offset added.

$$\text{RATE} = 4 \cdot R + R_{ks}$$

- R is the value set for ADSR
- R_{ks} is the key scaling offset value
- The RATE=0 when R=0

Table 3-2 Key Scaling of Rate

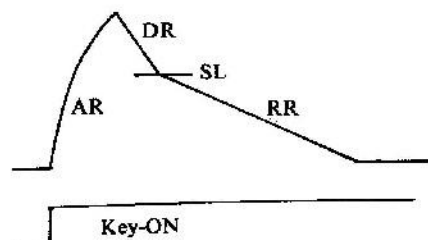
D ₄	N	R _{ks}	N	R _{ks}	N	R _{ks}	N	R _{ks}
0	0	0	4	1	8	2	12	3
	1	0	5	1	9	2	13	3
	2	0	6	1	10	2	14	3
	3	0	7	1	11	2	15	3
1	0	0	4	4	8	8	12	12
	1	1	5	5	9	9	13	13
	2	2	6	6	10	10	14	14
	3	3	7	7	11	11	15	15

* N is the key scaling number

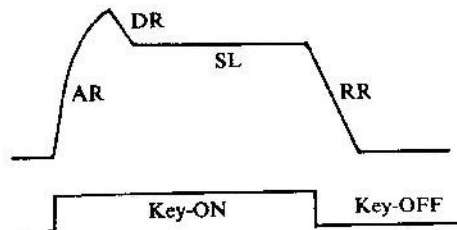
D5 (EG-TYP) : Switches between connecting sound and diminishing sound.

A diminishing sound is selected when D5 = "0", and a connecting sound is selected when "1". The difference between these two voicing modes lies in the use of the release rate. This is illustrated in Fig. 3-1.

D5 = "0" Diminishing sound



D5 = "1" Continuing sound



AR = ATTACK RATE DR = DECAY RATE
SL = SUSTAIN LEVEL RR = RELEASE RATE

Fig. 3-1 Two Types of Voicing Modes

D6 (VIB) : ON/OFF switch for vibrato. Setting this bit to "1" causes a vibrato effect to be applied to the slot. The frequency for this is 6.4Hz (@ $\phi M = 3.6\text{MHz}$), and the depth of the vibrato is determined by VIB-DEPTH of the BD register.

D7 (AM) : ON/OFF switch for AM modulation. Setting this bit to "1" causes AM modulation to be applied to the slot. The frequency for this is 3.7Hz (@ $\phi M = 3.6\text{MHz}$), and the depth of the vibrato is determined by AM-DEPTH of the BD register.

3-1-5 KSL/Total Level : Address (\$40 ~ \$55)

The total level is used to control the modulation factor (voice) and volume for the output of the envelope generator. Level key scaling (KSL) is similar to rate key scaling. This allows for simulation of the tendency for the output level of natural instruments to decrease as the interval increases.

\$40 ~ \$55

D7	D6	D5	D4	D3	D2	D1	D0
KSL		Total Level					

D0 ~ D5 (Total Level) : The minimum resolution for attenuation is 0.75dB, and the volume can be attenuated by a maximum of 47.25dB.

Table 3-3 Total Level

	D5	D4	D3	D2	D1	D0
Degree of attenuation	24	12	6	3	1.5	0.75
	dB	dB	dB	dB	dB	dB

D6~D7 (KSL) : Bits for control of level key scaling.

The key scale mode attenuates the level as the interval increases. The degree of attenuation can be set as 0 (no attenuation), 1.5dB/octave, 3dB/octave, or 6dB/octave.

Table 3-4

D7	D6	Degree of Attenuation
0	0	0
0	1	3 dB/OCT
1	0	1.5dB/OCT
1	1	6 dB/OCT

Table 3-5 Attenuation for each F-Number when 3dB/octave

F-Num	0	1	2	3	4	5	6	7
OCT	8	9	10	11	12	13	14	15
0	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
1	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
2	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
3	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
4	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
5	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
6	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
7	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000

Unit : dB

Note: • F-Number is the value of most significant four bits
 • 1.5dB is 1/2 of above
 • 6dB is twice above

3-1-6 Attack/Decay Rate : Address (\$60 ~ \$75)

The attack rate sets the rising time for the sound. The decay rate is the diminishing time after the attack. The time settings for each of these rates are shown in Table 3-6.

\$60 ~ \$75

D7	D6	D5	D4	D3	D2	D1	D0
AR				DR			
2 ³	2 ²	2 ¹	2 ⁰	2 ³	2 ²	2 ¹	2 ⁰

3-1-7 Sustain Level/Release Rate : Address (\$80 ~ \$95)

For continuing sounds, the sustain level gives the point of change where the attenuated sound in the decay mode changes to a sound having a constant level. For diminishing sounds, the sustain level gives the point where the decay mode changes to the release mode.

For continuing sounds, the release rate defines the rate at which the sound disappears after Key-OFF. For diminishing sounds, the sustain level indicates the attenuation prior to reaching the sustain level (point of change) while the release rate indicates the attenuation after the sustain level is reached.

\$80 ~ \$95

D7	D6	D5	D4	D3	D2	D1	D0
SL				DR			
24	12	6	3	2 ³	2 ²	2 ¹	2 ⁰
dB	dB	dB	dB				

* 93dB when bits D4~D7(SL) are all "1".

* The attenuation time for the release rate is the same as that shown in table for decay rate.

Table 3-6 Attack and Decay Times for Various Rates

(The rates indicated below are those after key scaling. The rate value is obtained by taking the most significant four bits (RM) and subtracting the least significant two bits (RL) (RM-RL). RATE = RM*4 + RL)

*** EG ATTACK TIME ***	*** EG DECAY TIME ***	*** EG ATTACK TIME ***	*** EG DECAY TIME ***
RATE	RATE	RATE	RATE
ms	ms	ms	ms
(e 10% - 90%)	(e 10% - 90%)	(e 0dB - 36dB)	(e 0dB - 36dB)
15 3	0.00	15 3	2.40
15 2	0.00	15 2	2.40
15 1	0.00	15 1	2.40
15 0	0.00	15 0	2.40
14 3	0.11	14 3	2.74
14 2	0.11	14 2	3.20
14 1	0.14	14 1	3.84
14 0	0.19	14 0	4.80
13 3	0.22	13 3	5.48
13 2	0.26	13 2	6.40
13 1	0.31	13 1	7.68
13 0	0.37	13 0	9.60
12 3	0.43	12 3	10.96
12 2	0.49	12 2	12.80
12 1	0.61	12 1	15.36
12 0	0.73	12 0	19.20
11 3	0.85	11 3	21.92
11 2	0.97	11 2	25.56
11 1	1.13	11 1	30.68
11 0	1.45	11 0	38.36
10 3	1.70	10 3	43.84
10 2	1.94	10 2	51.12
10 1	2.26	10 1	61.36
10 0	2.90	10 0	76.72
9 3	3.39	9 3	87.68
9 2	3.87	9 2	102.24
9 1	4.31	9 1	122.72
9 0	5.79	9 0	153.44
8 3	6.78	8 3	175.36
8 2	7.74	8 2	204.48
8 1	9.02	8 1	245.44
8 0	11.58	8 0	306.88
7 3	13.57	7 3	350.72
7 2	15.49	7 2	409.96
7 1	18.05	7 1	490.88
7 0	23.17	7 0	613.76
6 3	27.14	6 3	701.44
6 2	30.98	6 2	817.92
6 1	36.10	6 1	981.76
6 0	46.34	6 0	1227.52
5 3	54.27	5 3	1402.88
5 2	61.95	5 2	1635.84
5 1	72.19	5 1	1963.52
5 0	92.67	5 0	2455.04
4 3	108.54	4 3	2805.76
4 2	123.90	4 2	3271.68
4 1	144.38	4 1	3927.04
4 0	185.34	4 0	4910.08
3 3	217.09	3 3	5611.52
3 2	247.81	3 2	6543.36
3 1	288.77	3 1	7854.08
3 0	370.69	3 0	9820.16
2 3	434.18	2 3	11223.04
2 2	495.62	2 2	13086.72
2 1	577.54	2 1	15708.16
2 0	741.38	2 0	19540.32
1 3	868.35	1 3	22446.08
1 2	991.23	1 2	26173.44
1 1	1155.07	1 1	31416.32
1 0	1482.75	1 0	39280.64

Note: There is no change in the envelope when the rate is "0".

3-1-8 Block/F-Number : Address (\$A0~\$B8)

Data for determining the interval and scale. The F-Number is determined by both the \$A* and \$B* registers.

\$A0~\$A8	D7	D6	D5	D4	D3	D2	D1	D0
	F-Number							
	2 ⁷	2 ⁶	2 ⁵	2 ⁴	2 ³	2 ²	2 ¹	2 ⁰

\$B0~\$B8	D7	D6	D5	D4	D3	D2	D1	D0
			Key-ON	BLOCK			F-Num	
				2 ²	2 ¹	2 ⁰	2 ⁹	2 ⁸

D0~D7 (\$A*), D0~D1 (\$B*) (F-Number): The F-Number is indicated by the total of ten bits formed from the 8 bits of the \$A* register and lower two bits of the \$B* register. The data of the F-Number gives the scale. This value is determined by the procedure outlined below.

D2~D4 (BLOCK): Gives information on the octave.

D5 (Key-ON): Bit corresponding to ON/OFF of the keyboard. When this bit is "1", the channel is ON and voiced. Key-OFF when "0".

* F-Number/Block

With the OPLII, the required frequency can be obtained by giving the phase increment corresponding to this frequency. This increment is determined by the F-Number, Block, and Multiple information.

First, the increment of the desired frequency is obtained. This is given by the formula shown below.

$$\Delta P = f_{mus} * 2^{19} / f_{sam} \qquad f_{sam} = fM / 72 \qquad \text{————— ①}$$

f_{mus} : desired frequency
 f_{sam} : sampling frequency (50kHz)
 fM : frequency of input clock (3.6MHz)

The above allows for the phase increment to be obtained. As many of bits would be required to express this value, only the data for a single octave is given, and this increment is shifted for each octave (2 ×, 4 ×). This allows for expression of the increment in the following manner.

$$\Delta P = 2^B * F' * MUL$$

②

B : octave information
F' : increment limited to a single octave
MUL : Multiple data

Formulas ① and ② permit the increment (F') to be expressed in 10 bits. The F-Number and Block can be expressed as follows.

$$F = (f_{mus} * 2^{19} / f_{sam}) / 2^{b-1} \quad @MUL = 1$$

F : F-Number data
b : Block data

Table 3-7-1 F-Number (1)

Scale	Frequency (4oct)	F-Number	SB*		SA*							
			D1	D0	D7	D6	D5	D4	D3	D2	D1	D0
C#	277.2	363	0	1	0	1	1	0	1	0	1	1
D	293.7	385	0	1	1	0	0	0	0	0	0	1
D#	311.1	408	0	1	1	0	0	1	1	0	0	0
E	329.6	432	0	1	1	0	1	1	0	0	0	0
F	349.2	458	0	1	1	1	0	0	1	0	1	0
F#	370.0	485	0	1	1	1	1	0	0	1	0	1
G	392.0	514	1	0	0	0	0	0	0	0	1	0
G#	415.3	544	1	0	0	0	1	0	0	0	0	0
A	440.0	577	1	0	0	1	0	0	0	0	0	1
A#	466.2	611	1	0	0	1	1	0	0	0	1	1
B	493.9	647	1	0	1	0	0	0	0	1	1	1
C	523.3	686	1	0	1	0	1	0	1	1	1	0

Table 3-7-2 F-Number (2)

Scale	Frequency (4 ~ 5oct)	F-Number	SB*		SA*							
			D1	D0	D7	D6	D5	D4	D3	D2	D1	D0
G	392.0	514	1	0	0	0	0	0	0	0	1	0
G#	415.3	544	1	0	0	0	1	0	0	0	0	0
A	440.0	577	1	0	0	1	0	0	0	0	0	1
A#	466.2	611	1	0	0	1	1	0	0	0	1	1
B	493.9	647	1	0	1	0	0	0	0	1	1	1
C	523.3	686	1	0	1	0	1	0	1	1	1	0
C#	554.4	727	1	0	1	1	0	1	0	1	1	1
D	587.3	770	1	1	0	0	0	0	0	0	1	0
D#	622.2	816	1	1	0	0	1	1	0	0	0	0
E	659.3	864	1	1	0	1	1	0	0	0	0	0
F	698.5	916	1	1	1	0	0	1	0	1	0	0
F#	740.0	970	1	1	1	1	0	0	1	0	1	0

3-1-9 Feedback/Connection : Address (\$C0 ~ \$C8)

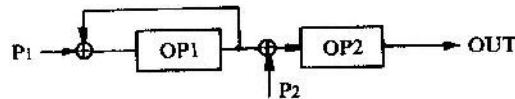
This register determines the modulation factor for self-feedback and the type of FM modulation.

\$C0 ~ \$C8

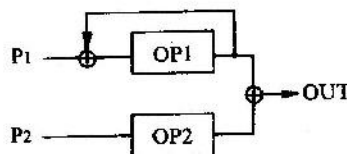
D7	D6	D5	D4	D3	D2	D1	D0
				Feed back			CONNECTION
				2 ³	2 ²	2 ⁰	

D0 (CONNECTION) : Connection controls the manner in which two slots are connected. The FM modulation mode is selected when the data is "0". The two slots are connected in parallel for sine wave output in the composite mode when the data is "1".

< "0" >



< "1" >



D1 ~ D3 (FEEDBACK) : Gives the modulation factor for feedback FM modulation of the first slot.

Table 3-8 Modulation Factor

	0	1	2	3	4	5	6	7
Modulation Factor	0	$\pi/16$	$\pi/8$	$\pi/4$	$\pi/2$	π	2π	4π

3-1-10 AM VIB-Depth/Rhythm : Address (\$BD)

This allows for control of AM and vibrato (VIB) depth, selection of rhythm mode, and ON/OFF control of various rhythm instruments.

\$BD	D7	D6	D5	D4	D3	D2	D1	D0
AM-DEPTH								
VIB-DEPTH								
RHYTHM								
				BD	SD	TOM	TOP-CY	HH

D0~D5 (RHYTHM) : The OPLII is set to the rhythm sound mode when D5="1". Channels 7~9 (refer to page 9) are the channels for rhythm sounds. Thus, music (melody section) is limited to 6 sounds. D0~D4 allow for ON/OFF control of the various rhythm instruments. This means that the \$B6, \$B7, and \$B8 Key-ON registers must always be "0". Slots 13~18 correspond to the rhythm sounds shown in Table 3-9. Data such as rate must be input as a value appropriate to each rhythm sound.

Table 3-9 Rhythm Slot

Instrument	Slot
BD	13.16
SD	17
TOM	15
TOP · CYM	18
HH	14

D6 (VIB-DEPTH) : The vibrato depth is 14 cent when D6="1", and 7 cent when "0".

D7 (AM-DEPTH) : The depth for AM is 4.8dB when D7="1", and 1dB when D7="0".

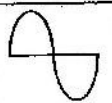

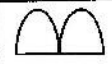

3-1-11 Wave Select

When bit D5 of address \$φ1 is "0", the OPLII is fully compatible with YM3526 (OPL); there are no differences between the two devices. If a sine wave is input in this mode, the output will be a sine wave like the input. When bit D5 of address \$φ1 is "1", the input sine wave will be output as the distorted wave shown in Table 3-10.

\$E0~\$F5

D7	D6	D5	D4	D3	D2	D1	D0
						WAVE SELECT	

Table 3-10 Wave Select

D1	D0	Waveform
0	0	
0	1	
1	0	
1	1	

3-2 Phase Generator (PG)

The phase generator is a circuit which obtains a phase value corresponding to the required frequency through accumulation of the increments for each unit of time. This increment is created from the frequency information (F-Number, Block, and Multiple) sent from the registers. This circuit is also equipped with a vibrato generator allowing for creation of a vibrato effect through combining the output of this generator and frequency information.

3-3 Envelope Generator (EG)

The envelope generator controls the rates for attack, decay, and release, sustain level, total level, etc. These parameters give the changes in voice and level which occur over time. The dynamic range of the envelope generator is 96dB (resolution of 0.1875). Indication for the envelope generator is logarithmic or in terms of degree of attenuation. The basic envelope shape is shown in Fig. 3-2. The special characteristics of this shape is that the change in level during the attack is exponential while it is linear during the other sections of the envelope. The crossover from attack to decay occurs at the 0dB point, and the decay changes to sustain when the sustain level is reached. Release begins when at the Key-OFF point. The effects of total level, level key scaling, and amplitude modulation are added to the envelope generator for changing the shape of the envelope.

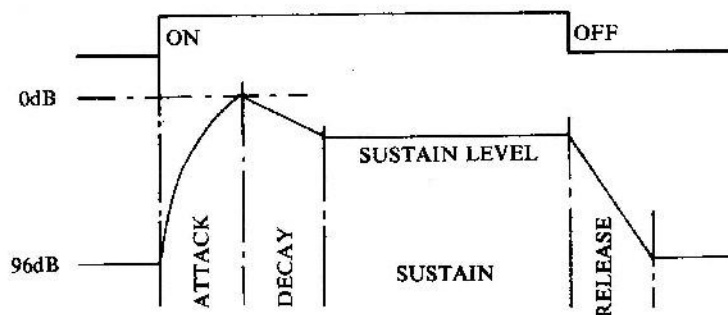


Fig. 3-2 Envelope Waveform

3-4 Operator (OP) and Accumulator (ACC)

3-4 Operator (OP) and Accumulator (ACC)

The operator is a circuit for FM calculation. The operator calculates the SIN value based upon the phase output from the phase generator, and this is combined with the output of the envelope generator. If the result is the modulating wave, it is sent back to the input of the operator. If music, the output is sent to the accumulator. The data for feedback and connection controls this transfer.

The accumulator accumulates the operator output for the various channels. The results of this calculation are converted to offset binary data consisting of a 10 bit mantissa section (including sign) and 3 bit exponent section. This data is output from the LSB as shown in Fig. 3-3.

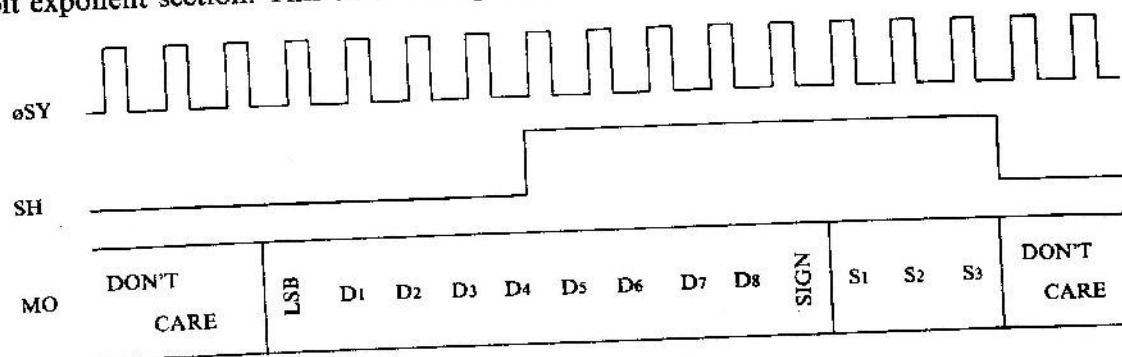


Fig. 3-3 Output Timing

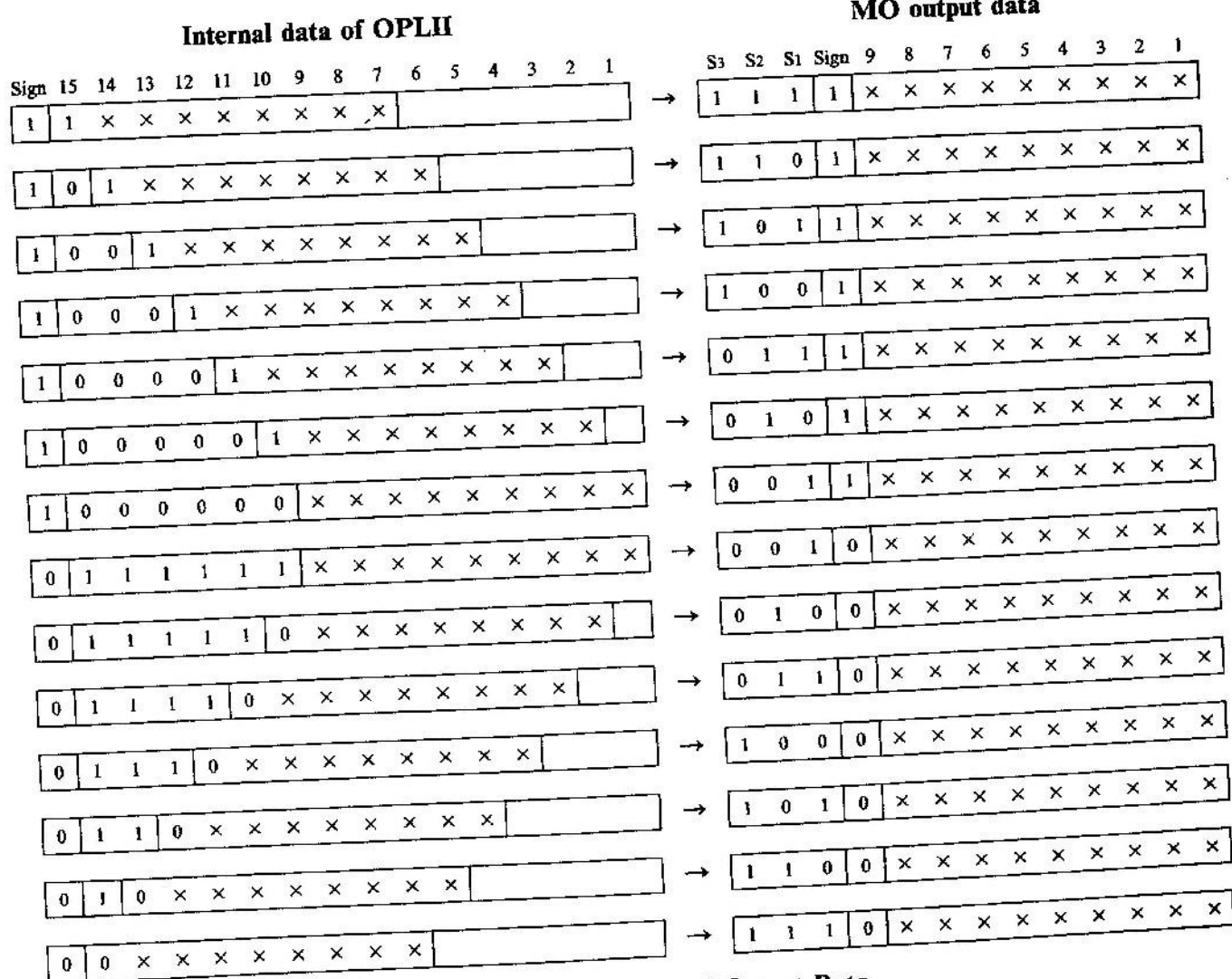


Fig. 3-4 Internal Data and Output Data

3-5 Status Information and Interrupt Signals

The two timers of OPLII are capable of setting flags according to a set period. These flags can be read as status signals or interrupt signals. Thus, the load on the CPU can be reduced by using the timers as tempo counters or for generating keyboard scanning signals. The interrupt signals have an open drain allowing for linking to other chips.

Status signal	D7	D6	D5	D4	D3	D2	D1	D0
	IRQ	Timer 1	Timer 2					

D5 (Timer 2 FLAG) : Flag signal set by Timer 2. Set to "1" when Timer 2 reaches the set time. This data remains until reset.

D6 (Timer 1 FLAG) : Same operation as D5 for Timer 1.

D7 (IRQ) : Set to "1" when either D5 or D6 is "1".

5. Creation of Music

This chapter deals with the data which can be input to the registers of the OPLII for creating piano or brass instrument music.

5-1 Concept of Sound Creation

The basic concept behind FM sound generation is to first fully understand the characteristics of the desired instrument. For example, the envelope of piano sounds is such that there is a sharp attack when the keys are pressed, and then the sound gradually disappears as the key is held down. There are also a large number of harmonic overtones during the attack with this number decreasing over time until a nearly constant harmonic configuration is attained. After such characteristics are understood, the means of attaining this sound through FM sound generation can be considered. The output amplitude can be determined from the envelope characteristics with the harmonic structure determining the modulation exponent. As this structure of harmonic overtones is related to the frequency of the operator, the frequency ratios can also be determined to a certain degree. In this manner, the characteristics of the desired sound roughly determine the FM parameters. Then, the various parameters are adjusted while listening to the sound, until the desired voice is obtained.

5-2 Basics of Sound Creation

FM sound generation uses effects obtained by using a modulator to modulate the carrier. Thus, the pitch, tone, and level of the music can be determined by skillful manipulation of the basic FM parameters (carrier output level, modulator output level, feedback level of modulator, frequency of carrier, and frequency of modulator). The relationship between these parameters and the parameters of the OPLII is shown in Table 5-1.

- i) FM connection (CONNECTION = "0")
All of the FM characteristics shown in Table 5-1 can be expressed. As operator 1 is equipped for self-feedback, combination with operator 2 allows for a two stage FM connection for high harmonic output.
- ii) Parallel connection (CONNECTION = "1")
The sum of the two operators is obtained with operator 2 always generating a sine wave. Thus, harmonically shifting the frequency of the two operators allows for effects such as the coupler effect of a pipe organ. In addition, operator 1 is equipped with the same feedback capabilities as described above for the output of higher harmonics.

5-3 Example of Sound Creation

Table 5-1 Basics of Sound Creation

Item	Applicable parameters	MIN—(change in sound)—MAX
Output level of carrier	TOTAL LEVEL (Data for A, D, S, and R) Key Scale data	Min. level —————> Max. level
Output level of modulator		Round sound —————> Bright sound
Feedback level of modulator	FB	Normal tone —————> Sharp tone (Noise)
Frequency of carrier	MULTI (BLOCK/F-Number)	Low pitch —————> High pitch
Frequency of modulator		Near harmonics —> Far harmonics

i) Electric Piano

- (1) Selection of connection
The connection is set to "0". Almost all voices can be obtained from this connection. This allows for both accenting on the attack of operator 1 and for rich high harmonics.
- (2) Determining the frequency for operators
The MULTIPLE for both operators are set to "1" in order to obtain higher harmonic frequencies which are integer multiples.
- (3) Output level of operators
The output of the modulator is altered to adjust the tone. When determining the level for operator 1, the bass is first set so that the sound is like a piano with a rich range of higher harmonic frequencies, and then the change in the treble is adjusted by level scaling for operator 1. Level scaling for the treble is required until the output is almost a sine wave.
- (4) EG setting
The next step is to determine the envelope for level and timbre. First, operator is set for an envelope which provides a sharp attack but a relatively prolonged sound. Operator 1 which forms the modulator is set so that there are a great number of harmonic overtones during the attack followed by a constant timbre with no changes. Key scaling is also used for operator 2 to provide level adjustment. Rate scaling of the treble should be used for a sharp sound.
- (5) Readjustment of data
The procedure for sound creation is nearly complete. The timbre can be altered to a certain degree by the envelope generator settings, etc. Here, the output level of the operator and the feedback level are readjusted to touch up the sound. For example, if you feel that the metallic echo is too strong, the level of operator 1 can be reduced.
- (6) Adding effects
Finally, a tremolo effect is added using the LFO to create a sound which is like an electric piano. This can be done using the amplitude modulation (AM) function of the OPLI, or by programming the total level to be updated at a period of 2~6Hz (use of a triangular wave is possible).

ii) Trumpet

- (1) Selection of connection
The connection for brass instruments is also "0". The bold brass sound of a trumpet can be created by adjusting the feedback level of operator 1.
- (2) Operator output
The total level for operator 1 (modulator) is set to a moderate value in the range of \$10 to \$28. The feedback level should be set to the maximum level of 7 for bright reverberation.
- (3) Frequency of operators
Basically, both operators should be set to a multiple of 1.
- (4) EG
There should a slow attack for both operators. For brass sounds, the attack of the modulator comes completely after the attack of the carrier. This allows for the characteristics attack of brass instruments to be captured.
- (5) Key scaling
Clarity will be lacking in the treble due to the slow attack of the envelope. Slight rate scaling is needed to prevent an unnatural sound when playing fast passages.

(6) LFO

No matter how well a brass instrument is played, the pitch will quiver during long tones. A vibrato effect should be added to express this.

5-4 Creation of Rhythm Sounds

Rhythm sounds are created using channels 7, 8, and 9. These three channels (6 slots) can be used to create a total of five sounds. The bass drum (BD), however, uses two slots for generation of FM sounds. Thus, a bass drum sound can be created using the same basic procedure as described in (a)~(c). This explanation will deal only with the remaining four sounds: high hat, top cymbal, tom tom, and snare drum.

The OPLII is equipped with a noise generator which allows for combining a number of frequencies and a white noise generator for creating rhythm instruments. This noise generator uses 8 and 9 channel frequency information (Block/F-Number/Multi), the proper phase output for various rhythm instruments is output by combining this signal with white noise. The resultant signal is sent to the operators. Experience has shown that the best ratio for the two set frequencies is 3 : 1 ($f7ch = 3 * f8ch$). So far, the phase data for the various instruments has been obtained. Finally, envelope information is combined with this output. As the envelope is set for one slot which corresponds to a single rhythm instrument, values which express the characteristics of the instrument are set in the parameter registers, in the same manner as melody instruments. (Refer to 3-1-10).

The above allows for the generation of various rhythm sounds.

6. Electrical Characteristics

1. Absolute Maximum Ratings

	Rating	Units
Pin voltage	-0.3 ~ 7.0	V
Operating ambient temperature	0 ~ 70	°C
Storage temperature	-50 ~ 125	°C

2. Recommended Operating Conditions

Item	Symbol	Minimum	Typical	Maximum	Unit
Power voltage	V _{CC}	4.5	5	5.5	V
	GND	0	0	0	V

3. DC Characteristics

Item		Symbol	Conditions	Minimum	Typical	Maximum	Unit
Input high level voltage	All input	V _{IH}		2.0			V
Input low level voltage	All input	V _{IL}				0.8	V
Input leak current	$\overline{\text{aM}}, \overline{\text{WR}}, \overline{\text{RD}}, \text{Ao}$	I _{LI}	V _I = 0 ~ 5V	-10		10	μA
Output leak current	D0 ~ D7, $\overline{\text{IRQ}}$	I _{LO}	V _I = 0 ~ 5V	-10		10	μA
Output high level voltage	Output expect $\overline{\text{IRQ}}$	VOH1	IOH1 = 0.4mA	2.4			V
		VOH2	IOH2 = 40μA	3.3			V
		VOL	IOL = 2.0mA			0.4	V
Output low level voltage	All output	VOL				0.4	V
Pullup resistance	$\overline{\text{IC}}, \overline{\text{CS}}$	R _U		80		400	kΩ
Input capacity	All input	C _I				10	pF
Output capacity	All output	C _O				10	pF
Power voltage		ICC				30	mA

4. AC Characteristics

Item		Symbol	Conditions	Minimum	Typical	Maximum	Unit
Input clock frequency	aM	fc	Fig. A-1	2.0	3.58	4.0	MHz
Input clock duty cycle	aM			40	50	60	%
Input clock rise time	aM	t _{CR}	Fig. A-1				ns
Input clock fall time	aM	t _{CF}	Fig. A-1				ns
Address setup time	Ao	t _{AS}	Fig. A-2, Fig. A-3	10			ns
Address hold time	Ao	t _{AH}	Fig. A-2, Fig. A-3	20			ns
Chip select write width	$\overline{\text{CS}}$	t _{CSW}	Fig. A-2	100			ns
Chip select read width	$\overline{\text{CS}}$	t _{CSR}	Fig. A-3	200			ns
Write pulse write width	$\overline{\text{WR}}$	t _{WW}	Fig. A-2	100			ns
Write data setup time	D0 ~ D7	t _{WDS}	Fig. A-2	20			ns
Write data hold time	D0 ~ D7	t _{WDH}	Fig. A-2	30			ns
Read pulse width	$\overline{\text{RD}}$	t _{RW}	Fig. A-3				ns
Read data access time	D0 ~ D7	t _{ACC}	Fig. A-3			200	ns
Read data hold time	D0 ~ D0	t _{RDH}	Fig. A-3	10			ns
Output rise time	aSY	t _{OR1}	Fig. A-4			100	ns
	MO-SH	t _{OR2}	Fig. A-5			150	ns
Output fall time	aSY	t _{OF1}	Fig. A-4			100	ns
	MO-SH	t _{OF2}	Fig. A-5			150	ns
Reset pulse width	$\overline{\text{IC}}$	t _{ICW}	Fig. A-6	80/fc			s

7. Timing Diagrams (Timing is based upon settings of $V_{IH} = 2.0V$ and $V_{IL} = 0.8V$)

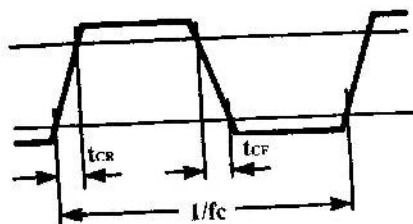
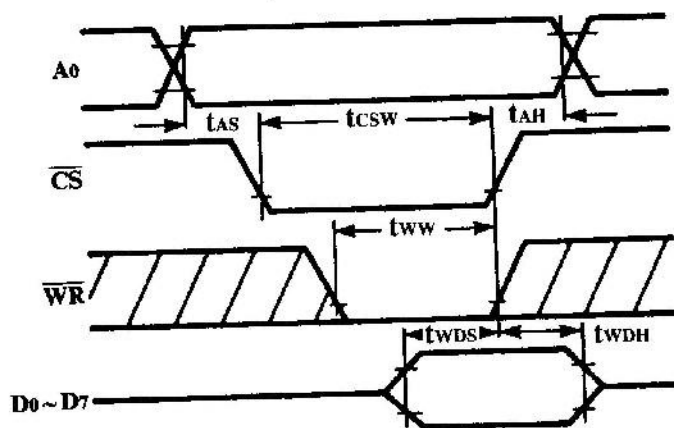
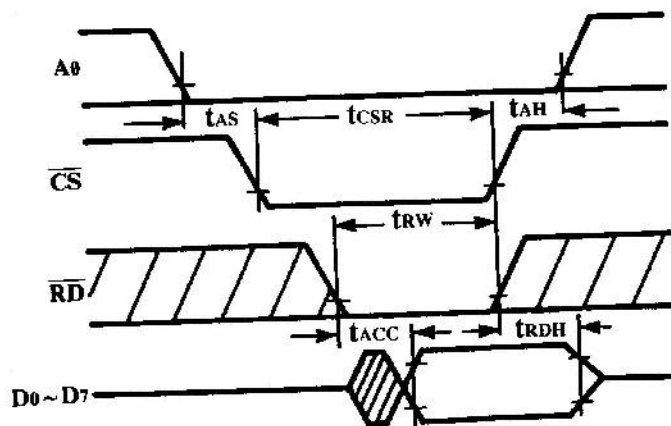


Fig. A-1 Clock Timing



Note: t_{CSW} , t_{ww} and t_{wdh} are based on either \overline{CS} or \overline{WR} being driven to high level.

Fig. A-2 Write Timing



Note: t_{ACC} is based on whichever of \overline{CS} or \overline{RD} goes to the low level last. t_{CSR} , t_{rw} and t_{rdh} are based on either \overline{CS} or \overline{RD} being driven to high level.

Fig. A-3 Read Timing

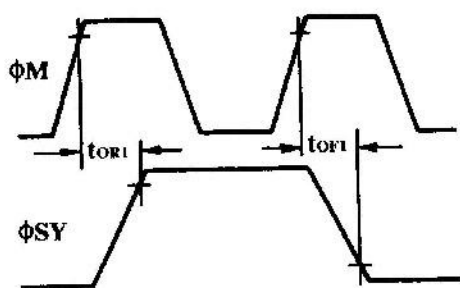


Fig. A-4 ϕM and ϕSY

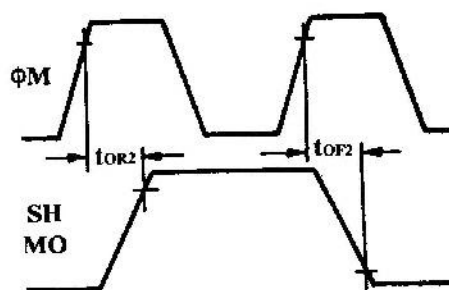


Fig. A-5 ϕM and $SH \cdot MO$

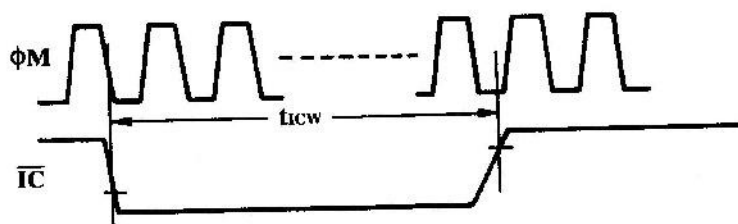
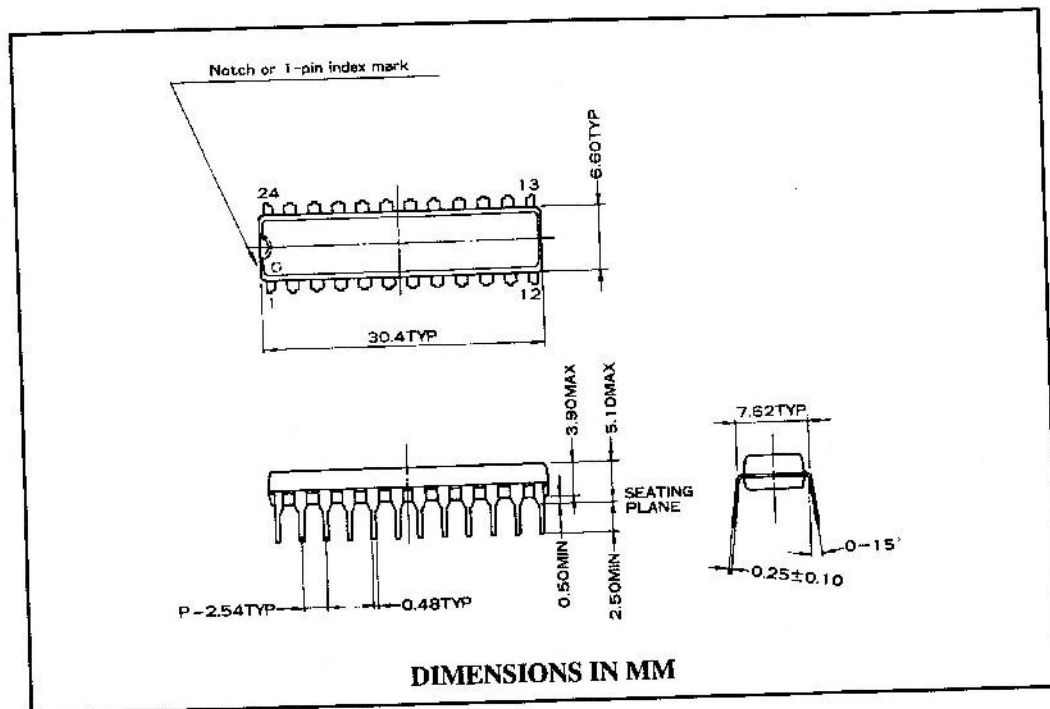


Fig. A-6 Reset Pulse

8. Package Dimensions

(1) YM3812



(2) YM3812-F

