GENESIS

SOUND SOFTWARE MANUAL

INDEX

This manual explains memory mapping and way of accessing especially. FM sound generation and PSG are explained in another manual.

I. Z80 MAPPING

A. Z80 Map

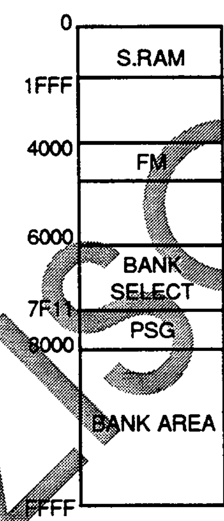
We show the memory at right.
I/O is contained in memory map.

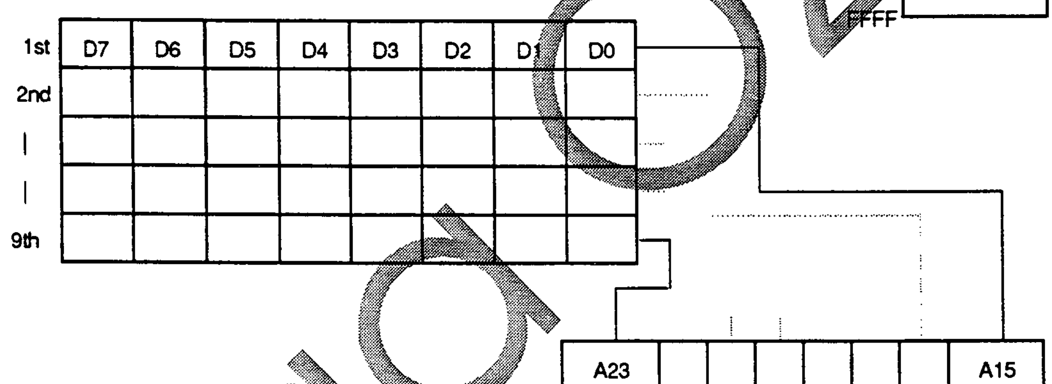
1. Program Area

Program, data and scratch are in 0 to 1FFFH, in S-RAM.

2. BANK

From 8000H-FFFFH is window of 68K memory. Z-80 can access all of 68K memory by BANK switching. BANK select data create 68K address from A15 to A23. You must write these 9 bits one at a time into 6000H serially, byte units, 7F11 using the LSB.





3. I/O

4000H FM1 register select (Channel 1-3)
4001H FM1 DATA

4002H FM2 register select (Channel 4-6)
4003H FM2 DATA

PSG address is in 7F11H.

B. Interrupt

Z-80 gets the only VIDEO vertical interrupt. This interrupt is generated 16ms period and 64ms length.

II. 68K CONTROL OF Z-80

A. Z80 Start-Up

Z-80 Operation Sequence:

1. BUS REQ ON
2. BUS RESET OFF
3. 68K copies program into Z-80 S-RAM
4. BUS RESET ON
5. BUS REQ OFF
6. BUS RESET OFF

BUS REQUEST

- BUS REQ ON
  DATA 100H (WORD) → $A11100
- BUS REQ OFF
  DATA 0H (WORD) → $A11100

RESET Z-80

- RESET ON
  DATA 0H (WORD) → $A11200
- RESET OFF
  DATA 100H (WORD) → $A11200

  This period requires 26ms.
  Also FM sound source is cleared at the same time.

CONFIRMATION OF BUS STATUS

This information is in $A11100, bit 0.

0 - Z-80 is using
1 - 68K can access

B. Z80 Handshake

If you access the HANDSHAKE area (A00000 - A07FFF) you must use BUS REQ. 68K has to access the Z-80 S-RAM by byte.

III. FM SOUND CONTROL

A. 68K Accesses the FM Source

68K needs BUS REQ when accessing the FM source, because this memory is controlled by Z-80.

B. Z80 Accesses the FM Source

Z-80 normally controls the FM (4000H - 4003H).

IV. PSG CONTROL

PSG accepts access of 68K and Z-80 any time, but you have to coordinate 68K and Z-80 accesses.

PSG is in $C00011 from 68K and in 7F11H from Z-80.

OVERVIEW

The Yamaha 2612 Frequency Modulation (FM) sound synthesis IC resembles the Yamaha 2151 (used in SEGA's coin-operated machines) and the chips used in Yamaha's synthesizers.

Its capabilities include:

- 6 channels of FM sound
- An 8-bit Digitized Audio channel (as replacement for one of the FM channels)
- Stereo output capability
- One LFO (low frequency oscillator) to distort the FM sounds
- 2 timers, for use by software.

To define these terms more carefully, an FM channel is capable of expressing, with a high degree of realism, a single note in almost any instrument's voice. Chords are generally created by using multiple FM channels.

The standard FM channels each have a single overall frequency and data for how to turn this frequency into the complex final waveform (the voice). This conversion process uses four dedicated channel components called "operators," each possessing a frequency (a variant of the overall frequency), an envelope, and the capability to modulate its input using the frequency and envelope. The operator frequencies are offsets of integral multiples of the overall frequency.

There are two sets of three FM channels, named channels 1 to 3 and 4 to 6, respectively. Channels 3 and 6, the last in each set, have the capability to use a totally separate frequency for each operator rather than offsets of integral multiples. This works well (we believe) for percussion instruments, which have harmonics at odd multiples such as 1.4 o 1.7 of the fundamental.

The 8-bit Digitized Audio Channel (DAC) exists as a replacement of FM channel 6, meaning that turning on the DAC turns off FM channel 6. Unfortunately, all timing must be done by software — meaning that unless the software has been very cleverly constructed, it is impossible to use any of the FM channels at the same time as the DAC.

Stereo output capability means that any of the sounds, FM or DAC, may be directed to the left, the right, or both outputs. The stereo is output only through the headphone jack.

The LFO, or Low Frequency Oscillator, allows for amplitude and/or frequency distortions of the FM sounds. Each channel elects the degree to which it will be distorted by the LFO, if at all. This could be used, for example, in a guitar solo.

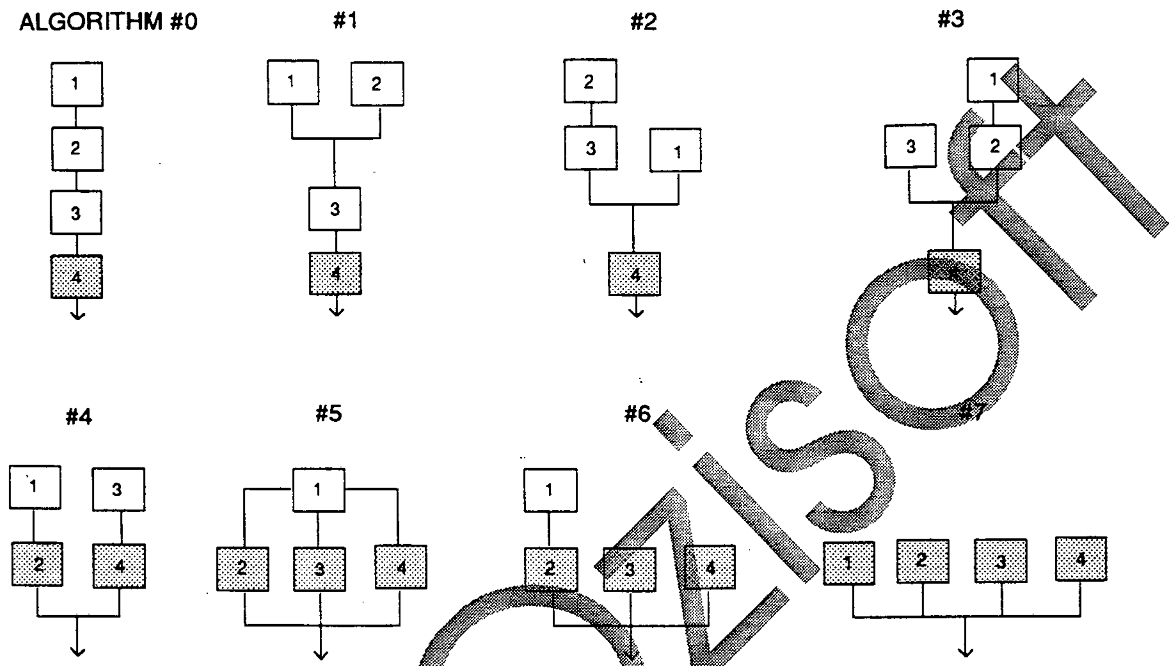
Finally, the system has two software timers which may be used as an alternative to the Z80 VBLANK interrupt. Unfortunately, these two timers do not cause interrupts — they must be read by the software to determine if they have finished counting.

A LITTLE BIT ABOUT OPERATORS

There are four dedicated operators assigned to every channel, with the following properties:

- An operator has an input, a frequency and an envelope (with which to modify the input), and an output.
- The operators have two types: those whose outputs feed into another operator, and those that are summed to form the final waveform. The latter are called "slots."
- The slots may be independently enabled, although Sega's software always enables or disables them all simultaneously.
- Operator one may feed back into itself, resulting in a more complex waveform.

These operators may be arranged in eight different configurations, called "algorithms." Following is a diagram of the algorithms.



SLOTS ARE INDICATED BY SHADING

Algorithm 0 - distortion guitar, "high hat chopper" (?) bass
Algorithm 1 - harp, PSG (Programmable Sound Generator) sound
Algorithm 2 - Bass, electric guitar, brass, piano, woods
Algorithm 3 - strings, folk guitar, chimes
Algorithm 4 - flute, bells, chorus, bass drum, snare drum, tom-tom
Algorithm 5 - brass, organ
Algorithm 6 - xylophone, tom-tom, organ, vibraphone, snare drum, base drum
Algorithm 7 - pipe organ

REGISTER OVERVIEW

The system is controlled by means of a large number of registers. General system registers are:

- timer values and status, software use
- LFO enable and frequency, to distort the FM channels
- DAC enable and amplitude
- output enables for each of the six FM channels
- number of frequencies to be used in FM channels 3 and 6.

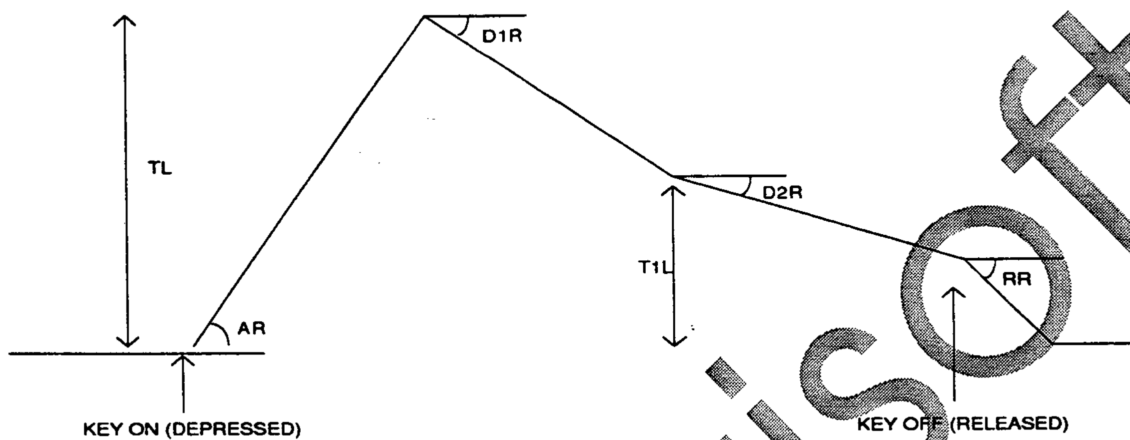
Usually, an FM channel has only one overall frequency, but if so elected, FM channels 3 and 6 use four separate frequencies, one for each operator.

The remainder of the registers apply to a single FM channel, or to an operator in that channel. Registers that refer to the channel as a whole are:

- frequency number (in the standard case)
- algorithm number
- extent of self-feedback in operator 1
- output type, to L, R, or both speakers. This can only be heard if headphones are used.
- the extent to which the channel is distorted by the LFO.

Registers that refer to each operator make up the remainder. The four operators' connections are determined by the algorithm used, but the envelope is always specified individually for each operator. In the case of FM channels 3 and 6, the frequency may be specified individually for each operator.

ENVELOPE SPECIFICATION



The sound starts when the key is depressed, a process called "key on." The sound has an attack, a strong primary decay, followed by a slow secondary decay. The sound continues this secondary decay until the key is released, a process called "key off." The sound then begins a rapid final decay, representing, for example, a piano note, after the key has been released and the damper has come down on the strings.

The envelope is represented by the above amplitudes and angles, and a few supplementary registers. Used in the above diagram are:

TL — Total level, the highest amplitude of waveform.

AR — Attack rate, the angle of initial amplitude increase. This can be made very steep if desired. The problem with slow attack rates is that if the notes are short, the release (called "key off") occurs before the note has reached a reasonable level.

DIR — The angle of initial amplitude decrease.

T1L — The amplitude at which the slower amplitude decrease starts.

D2R — The angle of secondary amplitude decrease. This will continue indefinitely unless "key off" occurs.

RR — The final angle of amplitude decrease, after "key off."

Additional registers are:

RS — Rate scaling, the degree to which envelopes become shorter as frequencies become higher. For example, high notes on a piano fade much more quickly than low notes.

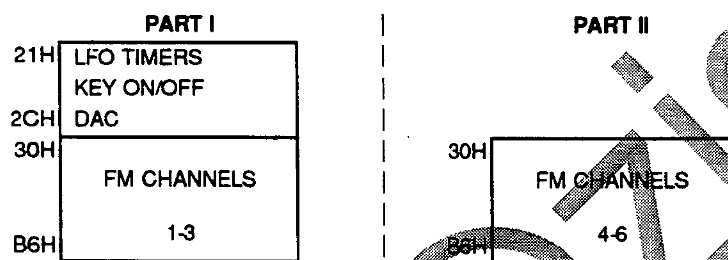
AM — Amplitude Modulation enable, whether or not this operator will allow itself to be modified by the LFO. Changing the amplitude of the slots changes the loudness of the note; changing the amplitude of the other operators changes its flavor.

SSG-EG — A proprietary register whose usage is unknown. It should be set to zero.

The FM-2612 may be accessed from either the 68000 or the Z-80. In both cases, however, the bus is only 8 bits wide.

The FM-2612 is accessed through memory locations 4000H - 4003H in the Z80 case, or A04000H - A04003H in the 68000 case. These will be referred to as 4000 to 4003.

The internal registers of the FM-2612 are divided as follows:



To write to Part I, write the 8-bit address to 4000 and the data to 4001. To write to Part II, write the 8-bit address to 4002 and the data to 4003.

Caution: Before writing, read from any address to determine if the YM-2612 I/O is still busy from the last write. Delay until Bit 7 returns to 0.

Caution: In the case of registers that are "ganged together" to form a longer number — for example the 10-bit Timer A value or the 14-bit frequencies — write the high register first.

READ DATA: Reading from any of the four locations.

D7						D0	
BUSY	X	X	X	X	X	OVERFLOW A	OVERFLOW B

BUSY — 1 if busy, 0 if ready for new data.
OVERFLOW — 1 if the timer has counted up and overflowed. See Register 27H.

PART I MEMORY MAP

22H	X	X	X	X	LFO EN	LFO FREQ		
24H	TIMER A							
25H	X	X	X	X	X	X	TIMER A	
26H	TIMER B							
	CH3	RESET		ENABLE		LOAD		
27H	MODE	B	A	B	A	B	A	
28H	OPERATOR			X	CHANNEL			
2AH	DAC							
2BH	DAC EN	X	X	X	X	X	X	X

30H+	X	DT1		MUL	
40H+	X	TL			
50H+	RS		X	AR	
60H+	AM	X	X	D1R	
70H+	X	X	X	D2R	
80H+	D1L				RR
90H+	X	X	X	X	SSG-EG

30H	CH1, OP1
31H	CH2, OP1
32H	CH3, OP1
34H	CH1, OP2
34H	CH2, OP2
36H	CH3, OP2
38H	CH1, OP3
39H	CH2, OP3
3AH	CH3, OP3
3CH	CH1, OP4
3DH	CH2, OP4
3EH	CH3, OP4

Each of 30H-90H has twelve entries, three channels x four operators.

Channels 1-3 become channels 4-6 in Part II.

PART I MEMORY MAP (cont.)

A0H+	FREQ. NUM				
A4H+	X	X	BLOCK	FREQ. NUM	
A8H+	CH 3 SUPPLEMENTARY FREQ. NUM				
ACH+	X	X	CH 3 SUPP BLOCK		CH3 SUPP FREQ NUM
B0H+	X	X	FEEDBACK		ALGORITHM
B4H+	L	R	AMS	X	FMS

Each of the above has three entries. All follow the pattern

A0H	CH1
A1H	CH2
A2H	CH3

with the exception that A8H and ACH follow the pattern

A8H	CH3, OP2
A9H	CH3, OP3
AAH	CH3, OP4

"PART II" is a duplication of 30H-B4H, where channels 1-3 are replaced by 4-6.

The Registers:

22H	X	X	X	X	LFO EN	LFO FREQ

LFO EN — 1 is enabled, 0 disabled.

LFO FREQ

	0	1	2	3	4	5	6	7
Hz	3.98	5.56	6.02	6.37	6.88	9.63	48.1	72.2

The LFO (Low Frequency Oscillator) is used to distort the FM sounds' amplitude and phase. It is <u>triply</u> enabled, as there is:

a. a global enable in Register 22H
b. a sensitivity enable on a channel by channel basis, in Registers B4H-B6H
c. an amplitude enable on an operator by operator basis in Registers 60H-6EH.

If the LFO is desired, enable it by Register 22H. Next, select which channels will be affected by the LFO, to what degree, and whether their amplitude or frequency is affected, by setting Registers B4H-B6H. Finally, if a channel's amplitude is affected, make sure that it is only the "slots" that are affected by setting Registers 60H-6EH.

24H	TIMER A MSBs

25H	X	X	X	X	X	X	TIMER A LSBs

Registers 24H and 25H are ganged together to form 10-bit TIMER A, with Register 25H containing the least significant bits. They should be set in the order 24H, 25H. The timer lasts

18* (1024 - TIMER) microseconds

TIMER A = all 1's → 18μs = 0.108 ms
TIMER A = all 0's → 18,400μs = 18.4 ms

26H	TIMER B

8-bit TIMER B lasts

288 * (256 - TIMER B) microseconds

TIMER B = all 1's → 0.288 ms
TIMER B = all 0's → 73.44 ms

27H	CH3 MODE	RESET		ENABLE		LOAD	
		B	A	B	A	B	A

Register 27H controls the software timers and the Channel 3 (and 6) mode, two entirely separate items.

CH3 MODE	D7	D6	
NORMAL	0	0	Channel 3 is the same as the others.
SPECIAL	0	1	Channel 3 has four separate frequencies.
ILLEGAL	1	X	

A normal channel's operators use offsets of integral multiples of a single frequency. In SPECIAL mode, each operator has an entirely separate frequency. Channel 3 operator 1's frequency is in Registers A2 and A6. Operators 2 and 4 are in Registers A8 and AC, A9 and AD, and AA and AE, respectively.

No one at Sega has used the timer feature, but the Japanese manual says:

LOAD 1 starts the timer, 0 stops it.

ENABLE 1 causes timer overflow to set the read register flag. 0 means the timer keeps cycling without setting the flag.

RESET writing a 1 clears the read register flag, writing a 0 has no effect.

28H	OPERATOR	X	CHANNEL

This register is used for "key on" and "key off." "Key on" is the depression of the synthesizer key. "Key off" is its release. The sequence of operations is: set parameters, key on, wait, key off. When key off occurs, the FM channel stops its slow decline and starts the rapid decline specified by "RR", the release rate.

In a single write to Register 28H, one sets the status of all operators for a single channel. Sega always sets them to the same value, on (1) or off (0). Using a special channel 3, I believe it is possible to have each operator be a separate note, so there is possible justification for turning them on and off separately.

OPERATOR				X	CHANNEL
4	3	2	1		

D2	D1	D0	
0	0	0	Channel 1
0	0	1	2
0	1	0	3
1	0	0	Channel 4
1	0	1	5
1	1	0	6

2AH	DAC DATA

Register 2AH contains 8 bit DAC data.

2BH	DAC EN	X	X	X	X	X	X	X

If the DAC enable is 1, the DAC data is output as a replacement for channel 6. The only channel 6 register that affects the DAC is the stereo output portion of Register B4H.

Registers 30H-90H are all single-operator registers. Please see page 8 for how the twelve channel-operator combinations are arranged.

30H+	X	DT1	MUL

Both DT1 (Detune) and MUL (multiple) relate the operator's frequency to the overall frequency.

MUL ranges from 0 to 15_{10}, and multiplies the overall frequency, with the exception that 0 results in multiplication by 1/2. That is, MUL = 0 to 15 gives x 1/2, x 1, x 2, ... x 15.

DT1 gives small variations from the overall frequency x MUL. The MSB of DT1 is a primitive sign bit, and the two LSBs are magnitude bits. See the next page for a diagram.

D6	D5	D4	MULTIPLICATIVE EFFECT
0	0	0	No change
0	0	1	x (1 + E)
0	1	0	x (1 + 2E)
0	1	1	x (1 + 3E)
1	0	0	No change
1	0	1	x (1 - E)
1	1	0	x (1 - 2E)
1	1	1	x (1 - 3E)

where E is a small number

40H+	X	TL

TL (total level) represents the envelope's highest amplitude, with 0 being the largest and 127_{10} the smallest. A change of one unit is about 0.75 dB.

To make a note softer, only change the TL of the slots (the output operators). Changing the other operators will affect the flavor of the note.

50H+	RS	X	AR

Register 50H contains RS (rate scaling) and AR (attack rate). AR is the steepness of the initial amplitude rise, shown on page 4.

RS affects AR, D1R, D2R and RR in the same way. RS is the degree to which the envelope becomes narrower as the frequency becomes high.

The frequency's top five bits (3 octave bits and 2 note bits) are called KC (key code) in the following rate formulas:

RS=0 $\Rightarrow$ Final Rate = 2 * Rate + (KC/8)
RS=1 $\Rightarrow$ Final Rate = 2 * Rate + (KC/4)
RS=2 $\Rightarrow$ Final Rate = 2 * Rate + (KC/2)
RS=3 $\Rightarrow$ Final Rate = 2 * Rate + KC**

** Always rounded down.

As rate ranges from 0-31, this means that the RS influence ranges from small (at 0-3) to very large (at 0-31).

60H+	AM	X X	D1R

D1R (first decay rate) is the initial step amplitude decay rate (see page 4). It is, like all rates, 0-31 in value and affected by RS.

AM is the amplitude modulation enable, whether or not __this__ operator will be subject to amplitude modulation by the LFO. This bit is not relevant unless both the LFO is enabled and Register B4's AMS (amplitude modulation sensitivity) is non-zero.

70H+	X X X	D2R

D2R (secondary decay rate) is the long tailoff of the sound that continues as long as the key is depressed.

90H+	X X X X	SSG-EG

This register is proprietary and should be set to zero.

The final registers relate mostly to a single channel. Each register is tripled; please see the diagram on page 9.

A0H+	FREQ NUM		
A4H+	X X	BLOCK	FREQ NUM
A8H+	CH 3 SUPP FREQ NUM		
ACH+	X X	CH 3 SUPP BLOCK	CH 3 SUPP FREQ NUM

Channel 1's frequency is in A0 and A4H.
Channel 2's frequency is in A1 and A5H.
Channel 3, if it is in normal mode (please see page 12) is in A2 and A6H.

If channel 3 is in special mode:
Operator 1's frequency is in A2 and A6H
Operator 2's frequency is in A8 and ACH
Operator 3's frequency is in A9 and ADH
Operator 4's frequency is in AA and AEH.

The frequency is a 14-bit number that should be set high byte, low byte (e.g., A4H then A0H). The highest 3 bits, called the "block," give the octave. The next 10 bits give position in the octave, and a possible 12-tone sequence is:

Low	617
	653
	692
	733
	777all in base 10
	823
	872
	924
	979
	1037
	1099
High	1164

This sequence should be used inside each octave.

B0H+	X	X	FEEDBACK	ALGORITHM

Feedback is the degree to which operator 1 feeds back into itself. In the voice library, self feedback is represented as this:

The ALGORITHM is the type of inter-operator connection used. Please see the list of the eight operators on page 3.

B4H+	L	R	AMX	X	FMS

Register B4H contains stereo output control and LFO sensitivity control.
L — Left output, 1 is on, 0 is off
R — Right output, 1 is on 0 is off
Note: The stereo may only be heard by headphones.

AMS (amplitude modulation sensitivity) and FMS (frequency modulation sensitivity) are the degree to which the channel is affected by the LFO. If the LFO is disabled, this register need not be set. Additionally, amplitude modulation is also enabled on an operator-by-operator level.

AMS	0	1	2	3
dB	0	1.4	5.9	11.8

FMS	0	1	2	3	4	5	6	7
% of a halftone	0	± 3.4	± 6.7	± 10	± 14	± 20	± 40	± 80

TEST PROGRAM

Here a tested power-on initialization and sample note in the "Grand Piano" voice (page 27).

Register	Value	Comments
22H	0	LFO off
27H	0	Channel 3 mode normal
28H	0	Off
28H	1	Off
28H	2	Off
28H	4	Off
28H	5	Off
28H	6	Off
2BH	0	DAC off
30H	71H	DT1/MUL
34H	0DH	
38H	33H	
3CH	01H	
40H	23H	Total Level
44H	2DH	
48H	26H	
4CH	00H	
50H	5FH	RS/AR
54H	99H	
58H	5FH	
5CH	94H	
60H	5	AM/D1R
64H	5	
68H	5	
6CH	7	
70H	2	D2R
74H	2	
78H	2	
7CH	2	

Register	Value	Comments
80H	11H	D1L/RR
84H	11H	
88H	11H	
8CH	A6H	
90H	0	Proprietary
94H	0	
98H	0	
9CH	0	
B0H	32H	Feedback/Algorithm
B4H	C0H	Both speakers on
28H	00H	Key off
A4H	22H	Set frequency
A0H	69H	
28H	F0H	Key on
<wait>		
28H	00H	Key off

Notes:
1. Write address then data.
2. Loop until read register D7 becomes 0.
3. Follow MSB/LSB sequence.

Programmable Sound Generator (PSG)

The PSG contains four sound channels, consisting of three tone generators and a noise generator. Each of the four channels has an independent volume control (attenuator). The PSG is controlled through output port $7F.

Tone Generator Frequency

The frequency (pitch) of a tone generator is set by a 10-bit value. This value is counted down until it reaches zero, at which time the tone output toggles and the 10-bit value is reloaded into the counter. Thus, higher 10-bit numbers produce lower frequencies.

To load a new frequency value into one of the tone generators, you write a pair of bytes to I/O location $7F according to the following format:

First Byte:	1	R2	R1	R0	d3	d2	d1	d0
Second Byte:	0	0	d9	d8	d7	d6	d5	d4

The R2:R1:R0 field selects the tone channel as follows:

R2	R1	R0	Tone Channel
0	0	0	#1
0	1	0	#2
1	0	0	#3

10-bit data is: (msb) d9 d8 d7 d6 d5 d4 d3 d2 d1 d0 (lsb)

Noise Generator Control

The noise generator uses three control bits to select the "character" of the noise sound. A bit called "FB" (Feedback) produces periodic noises or "white" noise:

FB	Noise Type
0	Periodic (like low-frequency tone)
1	White (hiss)

The frequency of the noise is selected by two bits NF1:NF0 according to the following table:

NF1	NF0	Noise Generator Clock Source
0	0	Clock/2 (higher pitch, "less coarse")
0	1	Clock/4
1	0	Clock/8 (lower pitch, "more coarse")
1	1	Tone Generator #3

Note: "Clock" is fixed in frequency. It is a crystal controlled oscillator signal connected to the PSG.

When NF1:NF0 is 11, Tone Generator #3 supplies the noise clock source. This allows the noise to be "swept" in frequency. This effect might be used for a jet engine runup, for example.

To load these noise generator control bits, write the following byte to I/O port $7F:

Out ($7F):								
	1	1	1	0	0	FB	NF1	NF0

Attenuators

Four noise attenuators adjust the volume of the three tone generators and the noise channel. Four bits A3:A2:A1:A0 control the attenutation as follows:

A3	A2	A1	A0	Attenuation
0	0	0	0	0 db (maximum volume)
0	0	0	1	2 db Note: a higher attenuation results in a quieter sound
0	0	1	0	4 db
0	0	1	1	6 db
0	1	0	0	8 db
0	1	0	1	10 db
0	1	1	0	12 db
0	1	1	1	14 db
1	0	0	0	16 db
1	0	0	1	18 db
1	0	1	0	20 db
1	0	1	1	22 db
1	1	0	0	24 db
1	1	0	1	26 db
1	1	1	0	28 db
1	1	1	1	-Off-

The attenuators are set for the four channels by writing the following bytes to I/O location $7F:

Tone Generator #1:	1	0	0	1	A3	A2	A1	A0
Tone Generator #2:	1	0	1	1	A3	A2	A1	A0
Tone Generator #3:	1	1	0	1	A3	A2	A1	A0
Noise Generator:	1	1	1	1	A3	A2	A1	A0

EXAMPLE

When the Mk3 is powered on, the following code is executed:

```
        LD  HL,CLRTB        ; clear table
        LD  C,PSG_PRT       ; psg port is $7F
        LD  B,4             ; load four bytes
        OTIR
        (etc.)
.
.
CLRTB   defb $9F, $BF, $DF, $FF
```

This code turns the four sound channels off. It's a good idea to also execute this code when the PAUSE button is pressed, so that the sound does not stay on continuously for the pause interval.