Programmer's Guide to Yamaha YMF 262/OPL3 FM Music Synthesizer

Contents

Introduction

Version 1.00 Nov-24-1994

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Disclaimer

Introduction

The chip I am going to describe is getting more and more common, but programming information is still scarce, so I have decided to fill in this gap. All information contained in this file is a result of my experience in Adlib programming, research (read: reverse engineering) and finally of my effort to write down everything necessary to understand and use this piece of hardware. No official sources (i.e. development kits, books about this topic, etc.) were available to me except:

> Adlib Programming Guide - by Tero Töttö, and
> The PC Games Programmers Encyclopedia V1.0

The information below is a combination of known features of Adlib (alias Yamaha YM 3812/OPL2) and my own uncountable experiments and failures, which brought out a lot of important details you have to know about the chip.

As far as I know, there are four major sound cards based on OPL3 chip:

- Sound Blaster Pro II (not Sound Blaster Pro I)
- Sound Blaster 16
- Adlib Gold
- Pro Audio Spectrum Plus/16

I currently have a Sound Blaster Pro II-compatible card only, so all the programming info I provide will be based on this card. (The other cards are quite similar, however. They are just wired at different I/O-port addresses.)

Note: I assume some knowledge of FM music programming (mainly Adlib FM synthesizer) in this manual. If you are new to this topic I recommend you try Adlib first before going higher. Anyway, OPL3 is a direct descendant of OPL2 (what a surprise), so most features of OPL2 are also present on OPL3.

Description of the Synthesizer

My card's user manual says: "[this card contains] ... a stereo music FM synthesizer with 20 channels consisting of four (4) operators each ... " I thought: "Wow -- that's together eighty operators. This must be a GOOD sound-card." I was wrong. Just another advertising lie.

So let's clear some facts. First, OPL3 has only thirty-six (36) operators which can be combined in several ways:

- 18 FM channels (36 operators), or
- 15 FM channels (30 ops) and 5 percussion instruments (6 ops), giving us 20 channels altogether, or
- up to 6 four-operator FM channels (max 24 ops), the rest again being divided into two-operator FM channels and drums.

From the table above you can see that not all channels can be used in four-operator (4-OP) mode -- only a part of the synthesizer is really capable of making 4-OP sounds -- the rest uses traditional two-operator (2-OP) mode.

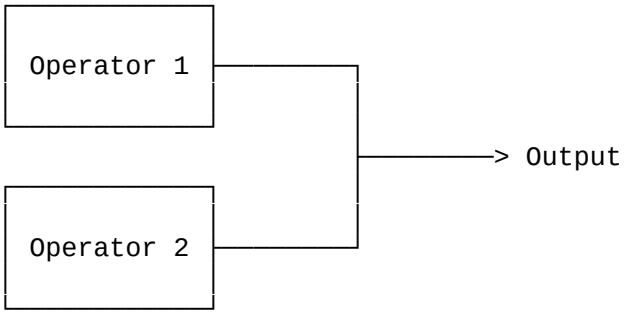
Second, the manual states this card is capable of "stereo" music. Yes, the quotes are necessary, because the stereo capabilities are very limited. You are given ability to control output going to left or right channel by turning it on and off. That's all. So the sound can flow from very left side, center and very right side. No sound panning, no special stereo effects. :-(

Well, flaming apart, back to the main topic.

The OPL3 chip is capable of making sounds in several ways:

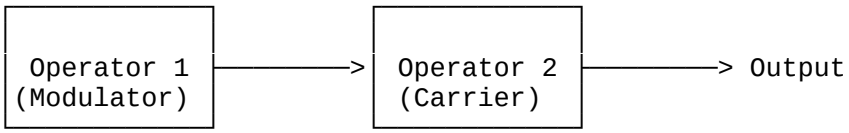
Two-operator Additive Synthesis

Output of both operators is simply added. It is the simplest way to make any sound, and it works on both OPL2 and OPL3. The diagram should make it clear.



Two-operator Frequency Modulation (FM) Synthesis

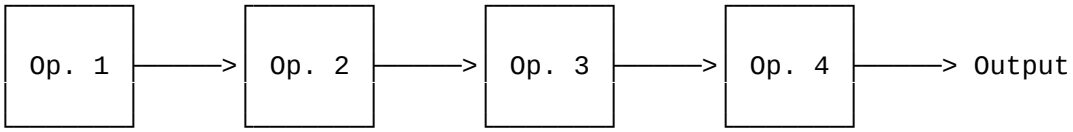
Output from the first operator (Modulator) is sent to the input of the second one (Carrier) and is used to modulate (alter) frequency of the second operator. Only the second operator produces sound. Most of interesting sounds are made this way. This also works on OPL2. Hope the picture helps.



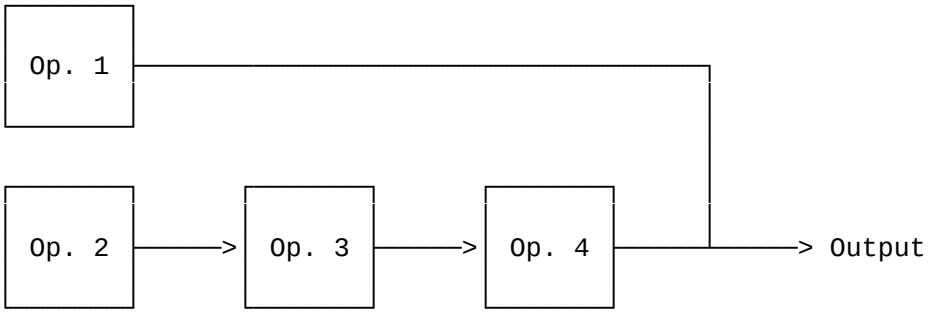
Four-operator "Mess" Modulation Synthesis

All of OPL3's 4-OP configurations are combinations of the above two modes of synthesis. OPL3 combines these two modes in four ways. I have no words to describe these four ways. Only the pictures can show their principle.

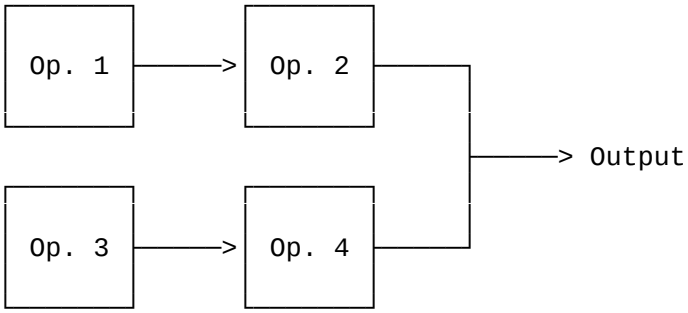
a) FM-FM Mode



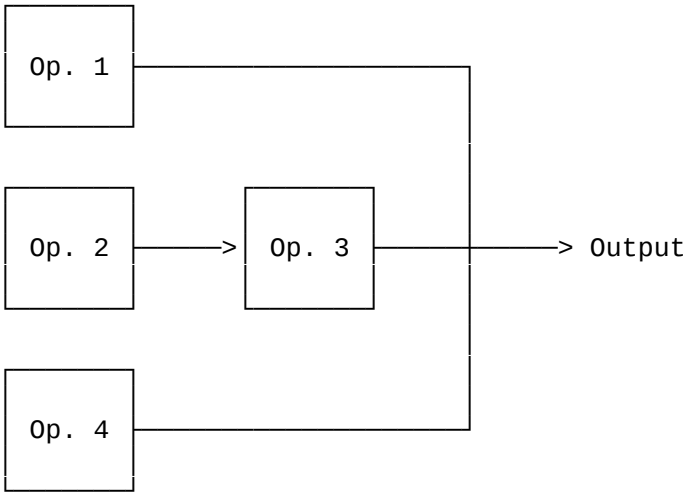
b) AM-FM Mode



c) FM-AM Mode



d) AM-AM Mode



Nice, aren't they?

The only way I think this can be written is a math formula. Symbol + (plus) means additive synthesis, and * (asterisk) means frequency modulation (Op1 * Op2 means operator 1 modulates operator 2, not vice versa). Here they are:

1. FM-FM Mode: (Op1 * Op2 * Op3 * Op4) ————> Output
2. AM-FM Mode: Op1 + (Op2 * Op3 * Op4) ———> Output
3. FM-AM Mode: (Op1 * Op2) + (Op3 * Op4) ——> Output
4. AM-AM Mode: Op1 + (Op2 * Op3) + Op4 ———> Output

Percussion Mode

In this mode 6 operators are used to produce five different percussion instruments:

- Bass Drum (2 operators)
- Snare Drum (1 operator)
- Tom-Tom (1 operator)
- Cymbal (1 operator)
- Hi-Hat (1 operator)

~~Because these instruments occupy only three melodic channels, only Bass Drum, Snare Drum and Tom-Tom frequencies can be set. Cymbal and Hi-Hat frequencies are fixed.~~

Based on the behaviour of the DOSBox OPL synth, the following points hold true: ! Verify using a real chip

- Setting the block/fnum on each percussive channel affects the pitch of both instruments sharing the channel
    - The pitch on channel 7 affects both the hihat and the snare
    - The pitch on channel 8 affects both the cymbal and tomtom
    - The pitch on channel 6 (of course) affects just the bass drum
- The register 0xC0 has no effect on the percussive channels (so setting 0xC6, 0xC7 and 0xC8 will have no effect)

    - This means in OPL3 mode, the panning L/R bits are ignored and the percussive instruments are always played through both channels, as if the bits are always set.
- Changing the modulator output level (0x40) has no effect for modulator-only instruments. This means the hihat and tomtom cannot have their volume set. The other instruments (those that use the output level set in the carrier operator) can have their volume adjusted normally using the carrier output level instead.

The *AdLib Programmer's Manual* contains playback code that uses the Tom-Tom frequency for all percussion instruments. It is possible this is the source of the confusion. Here, when a note is played on the Tom-Tom channel, that note will of course cause the Top Cymbal to change as well because of the shared channel. But the code increases this pitch by seven semitones and sets that frequency for channel 7 at the same time, ensuring the Hi-Hat and Snare Drum are always set to seven semitones above the last Tom-Tom note. (See AdLib MIDI Format for details.)

This mode is identical with that of OPL2. For more details see ADLIB.DOC.

Programming the Synthesizer

OPL3 may be found at the following addresses:

Sound Blaster Pro II	220h or 240h (selectable), also 388h
Adlib Gold	388h
Pro Audio Spectrum Plus/16	? (if you have a PAS you should know it)

The base address of the synthesizer will be called "base".

The chip occupies four I/O addresses:

base+0	Primary index register (write), Status register (read)
base+1	Primary data register (write-only)
base+2	Secondary index register (write)
base+3	Secondary data register (write-only)

The index registers are used to select internal registers and data registers are used to write to them. Status register returns the state of two timers built in the chip.

OPL3 contains two sets of registers. The Primary set maps to channels 0-8 (operators 0-17) and the secondary maps to channels 9-17 (operators 18-35). The reason for this is simple: all these registers wouldn't fit into single register set.

Unlike Adlib (OPL2), OPL3 doesn't need delay between register writes. With OPL2 you had to wait 3.3 µs after index register write and another 23 µs after data register write. On the contrary OPL3 doesn't need (almost) any delay after index register write and only 0.28 µs after data register write. This means you can neglect the delays and slightly speed up your music driver. But using reasonable delays will certainly do no harm.

The data registers can't be read (they are write-only) on both OPL2 and OPL3.

Register Map

The registers are grouped in the same manner as in the OPL2 chip. Programs using both OPL2 and OPL3 chips may use the same code provided that their direct I/O interface is well written. The only thing you have to change is operator-to-register mapping, which must accomodate the fact that registers are spread between two register sets.

(The register map is nearly the same so I dared to copy it from ADLIB.DOC.)

Status Register (base+0)

D7	D6	D5	D4	D3	D2	D1	D0
IRQFlag	T1Flag	T2Flag					

Data Registers (base+1, base+3)

REG	D7	D6	D5	D4	D3	D2	D1	D0
01	Test Register		(WSEnable)	Test Register				
02	Timer 1 Count (80 µsec resolution)							
03	Timer 2 Count (320 µsec resolution)							
04*	IRQReset	T1Mask	T2Mask				T2 Start	T1 Start
04**			4-OP B-E	4-OP A-D	4-OP 9-C	4-OP 2-5	4-OP 1-4	4-OP 0-3
05**								OPL3
08	(CSW)	NOTE-SEL						
20-35	Tremolo	Vibrato	Sustain	KSR	Frequency Multiplication Factor			
40-55	Key Scale Level		Output Level					
60-75	Attack Rate				Decay Rate			
80-95	Sustain Level				Release Rate			
A0-A8	Frequency Number (Lower 8 bits)							
B0-B8			KEY-ON	Block Number			F-Num (hi bits)	
BD	Trem Dep	Vibr Dep	PercMode	BD On	SD On	TT On	CY On	HH On
C0-C8	OutCh_D	OutCh_C	Right	Left	FeedBack Modulation Factor			SynthTyp
E0-F5						Waveform Select		

Chip-wide setting | Per operator | Per channel | Rhythm-mode only

* This applies only to port base+1

** This applies only to port base+3

For register bases A0, B0 and C0 there is one register per output channel. The primary register set holds the first nine channels (0-8) and the secondary holds last nine channels (9-17). For bases 20, 40, 60, 80 and E0 there are two registers per channel. Each register maps to one operator. Unfortunately the operator's register numbers are not continuous. The following table shows which operator maps to which register set and offset (in hex).

Op.	Set/Offset	Op.	Set/Offset
0	0/00	18	1/00
1	0/01	19	1/01

2	0/02	20	1/02
3	0/03	21	1/03
4	0/04	22	1/04
5	0/05	23	1/05
6	0/08	24	1/08
7	0/09	25	1/09
8	0/0A	26	1/0A
9	0/0B	27	1/0B
10	0/0C	28	1/0C
11	0/0D	29	1/0D
12	0/10	30	1/10
13	0/11	31	1/11
14	0/12	32	1/12
15	0/13	33	1/13
16	0/14	34	1/14
17	0/15	35	1/15

The following tables summarize which operators form a channel in various modes:

1. Two-operator Melodic Mode

Channel	**0**	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**	**14**	**15**	**16**	**17**
Operator 1	0	1	2	6	7	8	12	13	14	18	19	20	24	25	26	30	31	32
Operator 2	3	4	5	9	10	11	15	16	17	21	22	23	27	28	29	33	34	35

2. Two-operator Melodic and Percussion Mode

Channel	**0**	**1**	**2**	**3**	**4**	**5**	**BD**	**SD**	**TT**	**CY**	**HH**	**9**	**10**	**11**	**12**	**13**	**14**	**15**	**16**	**17**
Operator 1	0	1	2	6	7	8	12	16	14	17	13	18	19	20	24	25	26	30	31	32
Operator 2	3	4	5	9	10	11	15	-				21	22	23	27	28	29	33	34	35

3. Four-operator Melodic Mode

Channel	**0**	**1**	**2**	**6**	**7**	**8**	**9**	**10**	**11**	**15**	**16**	**17**
Operator 1	0	1	2	12	13	14	18	19	20	30	31	32
Operator 2	3	4	5	15	16	17	21	22	23	33	34	35
Operator 3	6	7	8	-			24	25	26	-		
Operator 4	9	10	11	-			27	28	29	-		

Channels 3, 4, 5 and 12, 13, 14 can't be used separately because their operators became part of channels 0, 1, 2 and 9, 10, 11 respectively. For instance a four-operator channel 1 consists of two two-operator channels number 1 and 4. (The second 2-OP channel number is always the first 2-OP channel number plus three.)

OPL3 allows you to enable/disable 4-OP mode separately for any of channels 0, 1, 2, 9, 10 and 11 (see register 104h in the reference below). This means for instance when you switch only channel 2 into 4-OP mode, channels number 0, 1, 3, 4, 6, 7, 8, 9, etc. will be still independent 2-OP channels.

Channels 6, 7, 8 and 15, 16, 17 are always two-operator ones. They can't be grouped to form four-operator channels.

4. Four-operator Melodic and Percussion Mode

Channel	**0**	**1**	**2**	**BD**	**SD**	**TT**	**CY**	**HH**	**9**	**10**	**11**	**15**	**16**	**17**
Operator 1	0	1	2	12	16	14	17	13	18	19	20	30	31	32
Operator 2	3	4	5	15	-				21	22	23	33	34	35
Operator 3	6	7	8	-					24	25	26	-		
Operator 4	9	10	11	-					27	28	29	-		

Examples:

Two-operator channel #14 consists of operators 26 and 29 which occupy these registers (all are in the secondary register set):

12A	Operator 1	Tremolo/Vibrato/Sustain/KSR/Multiplication
12D	Operator 2	Tremolo/Vibrato/Sustain/KSR/Multiplication
14A	Operator 1	Key Scale Level/Output Level
14D	Operator 2	Key Scale Level/Output Level
16A	Operator 1	Attack Rate/Decay Rate
16D	Operator 2	Attack Rate/Decay Rate
18A	Operator 1	Sustain Level/Release Rate
18D	Operator 2	Sustain Level/Release Rate
1A5		Frequency Number (low)
1B5		Key On/Block Number/Frequency Number (high)
1C5		FeedBack/Synthesis Type
1EA	Operator 1	Waveform Select
1ED	Operator 2	Waveform Select

Four-operator channel #1 consists of operators 1, 4, 7 and 10. All registers except register 104h are in the primary set:

104	bit 1 = 1	Enable Four-Operator Synthesis in channel #1
21	Operator 1	Tremolo/Vibrato/Sustain/KSR/Multiplication
24	Operator 2	Tremolo/Vibrato/Sustain/KSR/Multiplication
29	Operator 3	Tremolo/Vibrato/Sustain/KSR/Multiplication
2C	Operator 4	Tremolo/Vibrato/Sustain/KSR/Multiplication
41	Operator 1	Key Scale Level/Output Level
44	Operator 2	Key Scale Level/Output Level
49	Operator 3	Key Scale Level/Output Level
4C	Operator 4	Key Scale Level/Output Level

61	Operator 1	Attack Rate/Decay Rate
64	Operator 2	Attack Rate/Decay Rate
69	Operator 3	Attack Rate/Decay Rate
6C	Operator 4	Attack Rate/Decay Rate
81	Operator 1	Sustain Level/Release Rate
84	Operator 2	Sustain Level/Release Rate
89	Operator 3	Sustain Level/Release Rate
8C	Operator 4	Sustain Level/Release Rate
A1		Frequency Number (low)
A4		Unused
B1		Key On/Block Number/Frequency Number (high)
B4		Unused
C1		FeedBack/Synthesis Type (part 1)
C4		Synthesis Type (part 2)
E1	Operator 1	Waveform Select
E4	Operator 2	Waveform Select
E9	Operator 3	Waveform Select
EC	Operator 4	Waveform Select

NOTE: If a register number is greater than 100h, then it belongs into the secondary register set. (I use this numbering to emphasize the fact that the particular register MUST be written to the secondary set.) See the example code below for details.

OPL3 Register Reference

Because the registers of OPL3 are almost the same as of OPL2, I have copied their descriptions from file ADLIB.DOC.

Status Register

D7	**D6**	**D5**	**D4**	**D3**	**D2**	**D1**	**D0**
IRQ	T1	T2	-				

- bit 7: IRQ Flag. Set whenever any timer has elapsed.
- bit 6: Timer 1 Flag. Set every time the preset time in Timer 1 has elapsed.
- bit 5: Timer 2 Flag. Set every time the preset time in Timer 2 has elapsed.

Timer interrupts are not wired to any IRQ (why??). The timers can be used to detect the OPL2/OPL3 chip (see Appendix B).

Data Registers

01: Test Register / Waveform Select Enable

D7	D6	D5	D4	D3	D2	D1	D0
Test Register		(WSE)	Test Register				

- bits 7-6: Test Register. Must be reset to zero before any operation.
- bit 5: Waveform Select Enable. If clear, all channels will use normal sine wave. If set, register E0-F5 (Waveform Select) contents will be used.
- bits 0-4: Test Register. Must be reset to zero before any operation.

OPL3 does not implement WSE bit, and it should be left clear. In OPL2 mode all four waveforms are always available, and in OPL3 mode all eight waveforms are always available. For compatibility with OPL2, when a program is finished using an OPL3, the waveform select registers should be set to 0, if the next program assumes that by setting WSE bit to 0 the waveform is always sine like on a real OPL2.

02: Timer 1 Count

Upward 8 bit counter with a resolution of 80 µsec. If an overflow occurs, the status register bit is set, and the preset value is loaded into the timer again.

03: Timer 2 Count

Same as Timer 1, but with a resolution of 320 µsec.

004 (port: base+1): IRQ-Reset / Mask / Start

D7	D6	D5	D4	D3	D2	D1	D0
Rst	T1M	T2M	-			T2S	T1S

- bit 7: IRQ-Reset. Resets timer and IRQ flags in status register. All other bits are ignored when this bit is set.
- bit 6: Timer 1 Mask. If 1, status register is not affected in overflow.
- bit 5: Timer 2 Mask. Same as above.
- bit 1: Timer 2 Start. Timer on/off.
- bit 0: Timer 1 Start. Same as above.

104 (port: base+3): Four-Operator Enable

D7	D6	D5	D4	D3	D2	D1	D0
-		ChB	ChA	Ch9	Ch2	Ch1	Ch0

- bit 5: Enable four-operator synthesis for channel pair 11 - 14 (decimal).
- bit 4: Same as above for channel pair 10 - 13.
- bit 3: Same as above for channel pair 9 - 12.
- bit 2: Same as above for channel pair 2 - 5.
- bit 1: Same as above for channel pair 1 - 4.
- bit 0: Same as above for channel pair 0 - 3.

If reset to zero, OPL3 can produce 18 two-operator sounds at a time. If nonzero, OPL3 produces four-operator sound in appropriate channel pair.

105 (port: base+3): OPL3 Mode Enable

D7	D6	D5	D4	D3	D2	D1	D0
-							OPL3

- bit 0: OPL3 Mode Enable. When set, OPL3 extensions (36 operators, 4-OP synthesis, 8 waveforms, stereo output) can be used. When reset, the chip behaves as an ordinary OPL2. This bit is zero by default for compatibility with OPL2.

08: CSW / NOTE-SEL

D7	D6	D5	D4	D3	D2	D1	D0
(CSW)	N-S	-					

- bit 7: Composite sine wave mode on/off. All KEY-ON bits must be clear in order to use this mode. The card is unable to create any other sound when in CSW mode. (Unfortunately, I have no info how to use this mode :-<). The CSW mode is not implemented on an OPL3 and this bit is ignored.
- bit 6: NOTE-SEL. Controls the split point of the keyboard. When 0, the keyboard split is the second bit from the bit 8 of the F-Number. When 1, the MSb of the F-Number is used. (???)

The Note Select bit defines what kind of F-NUM values are used within an octave so the chip knows how to split an octave into two parts. When Note Select bit is 0, F-NUM values used within an octave all have their MSB bit (bit 9) set, ranging from 0x200 to 0x3FF, so the second most significant bit, bit 8 is used to determine if currently playing note belongs to lower (0x200-0x2FF) or upper (0x300-0x3FF) half within an octave. When Note Select bit is 1, the F-NUM value MSB bit, bit 9, is used to determine if the note belongs to lower (0x000-0x1FF) or upper (0x200-0x3FF) half within an octave.

20-35: Tremolo / Vibrato / Sustain / KSR / Frequency Multiplication Factor

D7	D6	D5	D4	D3	D2	D1	D0
Tre	Vib	Sus	KSR	Multiplication			

- bit 7: Tremolo (Amplitude vibrato) on/off.
- bit 6: Frequency vibrato on/off.
- bit 5: Sound Sustaining. When 1, operator's output level will be held at its sustain level until a KEY-OFF is done.
- bit 4: Envelope scaling (KSR) on/off. When 1, higher notes are shorter than lower notes.
- bits 0-3: Frequency Multiplication Factor (MULTI). Operator's frequency is set to (see registers A0, B0) F-Number * Factor.

MULTI	Factor
0	0.5
1	1
2	2
3	3
4	4
5	5
6	6
7	7
8	8
9	9
10	10
11	10
12	12
13	12
14	15
15	15

40-55: Key Scale Level / Output Level

D7	D6	D5	D4	D3	D2	D1	D0
KSL		Output level					

- bits 6-7: Key Scale Level. Attenuates output level towards higher pitch:

KSL	Attenuation
0	-
1	3.0 dB/oct
2	1.5 dB/oct
3	6.0 dB/oct

The attenuation level is based on the 3-bit octave (block number) value and top four bits of the FNUM value, so each octave is subdivided into 16 parts.

- bits 0-5: Output Level. Attenuates the operator output level. 0 is the loudest, 3F is the softest. Attenuation range is 48dB with 0.75dB resolution.

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

- In additive synthesis, varying the output level of any operator varies the volume of its corresponding channel. In FM synthesis, varying the output level of the carrier varies the volume of the channel. Varying the output of the modulator will change the frequency spectrum produced by the carrier. The following table summarizes which operators' output levels should be updated when trying to change channel output level.

Mode	Op 1	Op 2	Op 3	Op 4
AM	Y	Y	N/A	N/A
FM	-	Y	N/A	N/A
FM-FM	-	-	-	Y
AM-FM	Y	-	-	Y
FM-AM	-	Y	-	Y
AM-AM	Y	-	Y	Y

60-75: Attack Rate / Decay Rate

D7	D6	D5	D4	D3	D2	D1	D0
Attack rate				Decay rate			

- bits 4-7: Attack Rate. Determines the rising time for the sound. The higher the value, the faster the attack. If value is 0, the sound will never attack, and if value is 15, the volume jumps directly from minimum to maximum.
- bits 0-3: Decay Rate. Determines the diminishing time for the sound. The higher the value, the shorter the decay. If value is 0, the sound does not decay towards sustain level and stays at maximum volume after attack.

80-95: Sustain Level / Release Rate

D7	D6	D5	D4	D3	D2	D1	D0
Sustain level				Release rate			

- bits 4-7: Sustain Level. Determines the point at which the sound ceases to decay and chages to a sound having a constant level. The sustain level is expressed as a fraction of the maximum level. 15 is the softest and 0 is the loudest sustain level. Note that the Sustain-bit in the register 20-35 must be set for this to have an effect. Otherwise the sound will continue with release phase after hitting sustain level.

SL3	SL2	SL1	SL0
-24dB	-12dB	-6dB	-3dB

There is an exception when all bits are set (value=15), the actual level is -93dB instead, matching as if the value were 31.

- bits 0-3: Release Rate. Determines the rate at which the sound disappears after KEY-OFF. The higher the value, the shorter the release. Value of 0 causes the sound not to release at all, it will continue to produce sound at level before KEY-OFF.

Some songs (e.g. `LAME intro tune for Xerox of INC.raw`) set the sustain and release values to the shortest (0xFF) while the note is still playing in order to immediately silence a note that would otherwise linger after keyoff.

A0-A8: Frequency Number

Determines the pitch of the note. Highest bits of F-Number are stored in the register below.

B0-B8: Key On / Block Number / F-Number(hi bits)

D7	D6	D5	D4	D3	D2	D1	D0
-		Note on	Block number			Frequency num.	

The **note on** bit is set to 1 to play a note on the channel, and 0 to silence the note. Setting this value to 1 when a note is already playing has no effect (it does not retrigger the note.)

The **block number** is often referred to as the octave, but this is a misleading description. The value of the block controls the frequency range spanned by the frequency numbers. A larger block allows higher frequencies (pitch) to be reached, at the expense of there being a greater difference in pitch of two adjacent frequency numbers. This works out well, since a low block number allows a very high pitch accuracy for deep notes where it is most noticeable, while high block numbers mean the coarse pitch settings need only be used at high frequencies where the coarseness is unnoticeable.

The following table illustrates the effect of different block numbers.

Block	Note frequency		Difference between adjacent notes (Hz)
	F-num=1	F-num=1023	
0	0.047 Hz	48.503 Hz	0.048 Hz
1	0.094 Hz	97.006 Hz	0.095 Hz
2	0.189 Hz	194.013 Hz	0.190 Hz
3	0.379 Hz	388.026 Hz	0.379 Hz
4	0.758 Hz	776.053 Hz	0.759 Hz
5	1.517 Hz	1,552.107 Hz	1.517 Hz
6	3.034 Hz	3,104.215 Hz	3.034 Hz
7	6.068 Hz	6,208.431 Hz	6.069 Hz

This shows that block 7 is capable of reaching the highest note (6.2kHz) but since there are 6Hz between notes the accuracy suffers. Note A-4 is 440Hz but in this block, the two closest frequency numbers are 72 and 73, which create tones at 437Hz and 443Hz respectively, neither of which is particularly accurate. Blocks 3 and below are unable to reach as high as 440Hz, but block 4 can. With block 4, frequency numbers 579 and 580 produce 439.4Hz and 440.2Hz, considerably closer to the intended frequency.

In other words, when calculating notes, the best accuracy is achieved by selecting the lowest possible block number that can reach the desired note frequency.

The **frequency number** is split between registers A0 and B0, with the upper two bits being stored here. The full frequency number is a value between 0 and 1023 inclusive. The following formula is used to determine the frequency number for a given note frequency (in Hertz) and block:

f-num = freq * 2^(20 - block) / 49716

This formula can produce frequency numbers outside of the allowed 0-1023 range if the block number is too low, or the frequency is higher than the hardware can produce.

The sampling frequency of OPL2 and OPL3 chips is approximately 49716 Hz, leading to the constant in the formula. Occasionally this is rounded to 50000 leading to a small pitch variation if the wrong constant is used to play these songs. OPL2 cards like Adlib and early Sound Blasters used the 14.31818 MHz oscillator signal from ISA bus slot and divided it by four on card, thus running the OPL2 chip at 3.579545 MHz, which gets divided internally by 72 to get the sampling rate. OPL3 cards had to be compatible, and mostly the sound cards have separate 14.31818 MHz oscillator on board which the OPL3 uses directly, and this gets divided internally by 288 to get the sampling rate.

NOTE: In four-operator mode only the register value of Operators 1 and 2 is used, value of Operators 3 and 4 in this register is ignored. In other words: one channel uses only one frequency, block and KEY-ON value at a time, regardless whether it is a two- or four-operator channel.

Setting the frequency and/or block while the *note on* bit is set will result in the frequency changing immediately. By changing the values gradually, pitch bends can be achieved.

! Does changing the frequency just after a note-off cause a pitch bend for those instruments with lengthy release rates?

BD: Tremolo Depth / Vibrato Depth / Percussion Mode / BD/SD/TT/CY/HH On

D7	**D6**	**D5**	**D4**	**D3**	**D2**	**D1**	**D0**
Tre	Vib	Per	BD	SD	TT	CY	HH

- bit 7: Tremolo (Amplitude Vibrato) Depth. 0 = 1.0dB, 1 = 4.8dB.
- bit 6: Frequency Vibrato Depth. 0 = 7 cents, 1 = 14 cents. A "cent" is 1/100 of a semi-tone.
- bit 5: Percussion Mode. 0 = Melodic Mode, 1 = Percussion Mode.
- bit 4: BD On. KEY-ON of the Bass Drum channel.
- bit 3: SD On. KEY-ON of the Snare Drum channel.
- bit 2: TT On. KEY-ON of the Tom-Tom channel.
- bit 1: CY On. KEY-ON of the Cymbal channel.
- bit 0: HH On. KEY-ON of the Hi-Hat channel.

NOTE: KEY-ON bits of channels 6, 7 and 8 must be clear and their F-Nums, Attack/Decay/Release rates, etc. must be set properly to use percussion mode.

C0-C8: FeedBack Modulation Factor / Synthesis Type

D7	**D6**	**D5**	**D4**	**D3**	**D2**	**D1**	**D0**
OutCh_D	OutCh_C	R	L	Feedback			Syn

- bit 7: CHD, Output Channel D Enable. When set, channel output goes to fourth digital audio output channel.
- bit 6: CHC, Output Channel C Enable. When set, channel output goes to third digital audio output channel.
- bit 5: Right Speaker Enable. When set, channel output goes to second digital audio output channel, connected to right speaker.
- bit 4: Left Speaker Enable. When set, channel output goes to first digital audio output channel, connected to left speaker.
    - In OPL3 mode, at least one of the left/right bits must be set to hear the channel.
    - In OPL2 mode, these bits are ignored and audio is sent to all channels for compatibility reasons.
    - Left/Right bits can be used to realize sound "panning", but this method offers only three pan positions (left/center/right).
    - These bits apply only to operators producing sound (Carriers). Modulators are not affected by their setting.
    - Sound cards use only one stereo DAC, so only two of the four digital output channels are available.

- bits 1-3: FeedBack Modulation Factor. If 0, no feedback is present. If 1-7, operator 1 will send a portion of its output back into itself.

FeedBack	Factor
0	0
1	π/16
2	π/8
3	π/4
4	π/2
5	π
6	2π
7	4π

When in four-operator mode, the FeedBack value is used only by Operator 1, value of Operators 2, 3 and 4 is ignored.

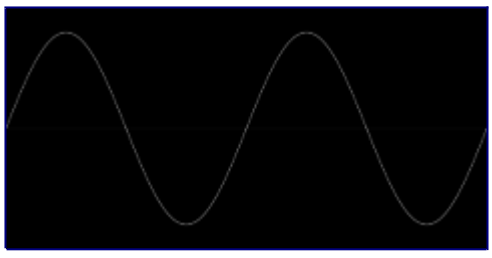
- bit 0: Synthesis Type. 1 = Additive synthesis, 0 = Frequency Modulation. In four-operator mode, there are two bits controlling the synthesis type. Both are the bit 0 of register C0, one of Operators 1 and 2 and the second of Operators 3 and 4.

Op 1&2	Op 3&4	Type
0	NONE	FM
1	NONE	AM
0	0	FM-FM
1	0	AM-FM
0	1	FM-AM
1	1	AM-AM

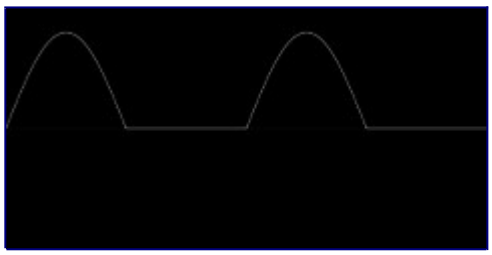
E0-F5: Waveform Select

D7	D6	D5	D4	D3	D2	D1	D0
-					Waveform		

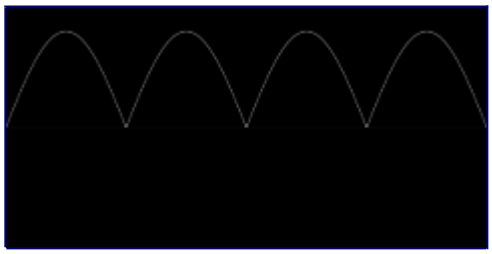
bits 0-2: WaveForm Select (WS):



Waveform 0: Sine

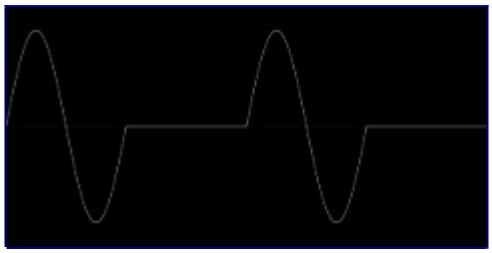


Waveform 1: Half-sine

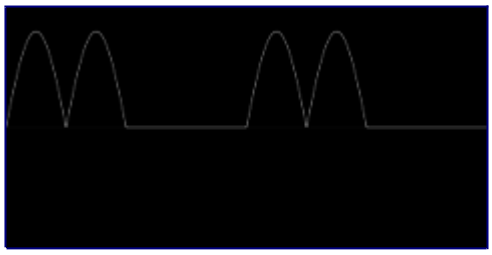


Waveform 2: Abs-sine

Waveform 3: Pulse-sine

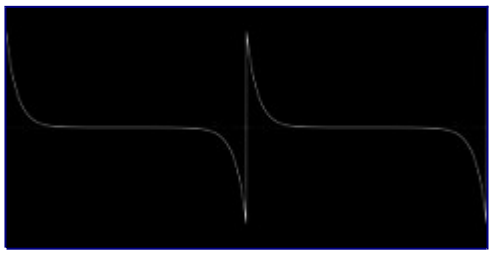


Waveform 4: Sine - even periods only



Waveform 5: Abs-sine - even periods only

Waveform 6: Square



Waveform 7: Derived square

- NOTE: Bit 5 of register 01 must be set to use waveforms other than sine. Waveforms 4-7 are available only on OPL3. See files WAVEn.GIF for real waveforms, where n is a number between 0 and 7.

Examples

These examples show a few working routines used in my MUS Player. They are written in Borland C++ 3.1 but should be easy to translate to any other language.

```
// I prefer using these Assembler-like types
typedef unsigned int  WORD;
typedef unsigned char BYTE;

/*
 * FM Synthesizer base port. SB Pro II - 0x220, Adlib 0x388
 */
WORD FMport = 0x220;

/*
 * Enables OPL3 extensions.
 */
WORD OPL3 = 1;

/*
 * Direct write to any Adlib/SB Pro II FM synthetiser register.
 *   reg - register number (range 0x001-0x0F5 and 0x101-0x1F5). When high byte
 *         of reg is zero, data go to port FMport, otherwise to FMport+2
 *   data - register value to be written
 */
BYTE FMwriteReg(WORD reg, BYTE data)
{
    asm {
        mov     dx,FMport
        mov     ax,reg
        or      ah,ah           // high byte is nonzero -- write to port base+2
        jz      out1
        inc     dx
        inc     dx
    }
out1: asm {
        out     dx,al
        mov     cx,6
    }
loop1:asm {                     // delay between writes
        in      al,dx
        loop    loop1

        inc     dx
        mov     al,data
        out     dx,al
        dec     dx
        mov     cx,36
    }
loop2:asm {                     // delay after data write
        in      al,dx
        loop    loop2
    }
    return _AL;
}

/*
 * Write to an operator pair. To be used for register bases of 0x20, 0x40,
 * 0x60, 0x80 and 0xE0.
 */
void FMwriteChannel(BYTE regbase, BYTE channel, BYTE data1, BYTE data2)
{
```

```
    static BYTE adlib_op[] = {0, 1, 2, 8, 9, 10, 16, 17, 18};
    static BYTE sbpro_op[] = { 0,  1,  2,   6,  7,  8,  12, 13, 14,
                              18, 19, 20,  24, 25, 26,  30, 31, 32};
    static WORD rg[] = {0x000,0x001,0x002,0x003,0x004,0x005,
                        0x008,0x009,0x00A,0x00B,0x00C,0x00D,
                        0x010,0x011,0x012,0x013,0x014,0x015,
                        0x100,0x101,0x102,0x103,0x104,0x105,
                        0x108,0x109,0x10A,0x10B,0x10C,0x10D,
                        0x110,0x111,0x112,0x113,0x114,0x115};

    if (OPL3)
    {
        register WORD reg = sbpro_op[channel];
        FMwriteReg(rg[reg]+regbase, data1);
        FMwriteReg(rg[reg+3]+regbase, data2);
    } else {
        register WORD reg = regbase+adlib_op[channel];
        FMwriteReg(reg, data1);
        FMwriteReg(reg+3, data2);
    }
}

/*
 * Write to channel a single value. To be used for register bases of
 * 0xA0, 0xB0 and 0xC0.
 */
void FMwriteValue(BYTE regbase, BYTE channel, BYTE value)
{
    static WORD ch[] = {0x000,0x001,0x002,0x003,0x004,0x005,0x006,0x007,0x008,
                        0x100,0x101,0x102,0x103,0x104,0x105,0x106,0x107,0x108};
    register WORD chan;

    if (OPL3)
        chan = ch[channel];
    else
        chan = channel;
    FMwriteReg(regbase + chan, value);
}
```

Detection Methods

An official method of Adlib (OPL2) detection is:

1. Reset Timer 1 and Timer 2: write 60h to register 4.
2. Reset the IRQ: write 80h to register 4.
    - NOTE: Steps 1 and 2 can't be combined together.
3. Read status register: read port base+0 (388h). Save the result.
4. Set Timer 1 to FFh: write FFh to register 2.
5. Unmask and start Timer 1: write 21h to register 4.
6. Wait in a delay loop for at least 80 µsec.
7. Read status register: read port base+0 (388h). Save the result.
8. Reset Timer 1, Timer 2 and IRQ as in steps 1 and 2.
9. Test the results of the two reads: the first should be 0, the second should be C0h. If either is incorrect, then the OPL2 is not present.

- NOTE1: You should AND the result bytes with E0h because the unused bits are undefined.
- NOTE2: This testing method doesn't work in some SoundBlaster compatible cards.

OPL3 Detection

---

1. Detect OPL2. If present, continue.
2. Read status register: read port base+0.
3. AND the result with 06h.
4. If the result is zero, you have OPL3, otherwise OPL2.

- NOTE: This is NOT an official method. I have dug it out of a sound driver. I haven't tested it, because I haven't an OPL2 card (Adlib, SB Pro I). Nevertheless it "detects" my SB Pro II properly. ;-)

Another possible detection method for distinguishing between SB Pro I and SB Pro II would be to try to detect OPL2 at I/O port base+0 and then at port base+2. The first test should succeed and the second should fail if OPL3 is present. (Remember: SB Pro I contains twin OPL2 chips at addresses base+0 and base+2, while SB Pro II contains one OPL3 chip at I/O address base+0 thru base+3).

BLASTER Environment Variable

Perhaps the most recommended "detection" method. Reading this variable avoids blindfold I/O port scanning and possible device conflicts. The user is responsible for its proper setting.

The variable has this format:

```
BLASTER=Aaddr Iirq Ddma Ttype
```

- A: Base I/O address given in hex. For most Sound Blasters the default is 220.
- I: IRQ Number (decimal). Default 7.
- D: DMA Number (decimal). Default 1.
- T: Card Type (decimal):
    - 1 - Sound Blaster 1.5
    - 2 - Sound Blaster Pro I
    - 3 - Sound Blaster 2.0
    - 4 - Sound Blaster Pro II

Example:

```
BLASTER=A220 I7 D1 T4
```

References

The PC Games Programmers Encyclopedia, Mark Feldman and many others on Usenet and Internet, http://bespin.org/~qz/pc-gpe/

> ... you can find (almost) everything you need there

Sound Blaster - The Official Book, Richard Heimlich, David M. Golden, Ivan Luk, Peter M. Ridge, Osborne/McGraw Hill, ISBN 0-07-881907-5

> ... this is a number-one in my book-wishlist. If anyone wanted to get rid of the book, I wouldn't scorn it ... :-)

The SoundBlaster Developer Kit, Creative Labs Inc, Creative Technology PTE LTD

> ... I wonder if you can find something comprehensible in that.

This document was created from this article:
http://www.shikadi.net/moddingwiki/OPL_chip