Contents Authoring Guideline

For MA-3 Authoring Tool

MLD (L) Edition

Version 1.0.0

2004/09/30

YAMAHA Corporation

⊛ YAMAHA

Contents

Revision History

Version	Date	Content
1.0.0	2004/09/30	Newly Released

1. Overview

This document stipulates a guideline in order to create SMF (Standard MIDI File) that can pull out the performance of Yamaha's synthesizer LSI (MA-3) to the maximum extent when authoring the contents for terminals equipped with MA-3 by using MA-3 authoring tool.

Authoring tool read SMF which is according to this document and convert it to career format. The operation reading SMF except one described in this document is not guaranteed. Although MIDI sequencer application software for authoring SMF in accordance with this document is not designated, it is necessary that described events can be input.

Note: About the numerical notations

In this document, the data values are represented by decimal numbers or hexadecimal numbers. In case of hexadecimal numbers, a letter "***H***" (***Hexadecimal***) comes after the numerical values. Moreover, "***n***" expresses the arbitrary integers. When you input data values, refer to the following "***Table 1***."

Table 1 Corresponding Table between Decimal numbers and Hexadecimal numbers

Decimal	Hex	Decimal	Hex	Decimal	Hex	Decimal	Hex
0	00H	32	20H	64	40H	96	60H
1	01H	33	21H	65	41H	97	61H
2	02H	34	22H	66	42H	98	62H
3	03H	35	23H	67	43H	99	63H
4	04H	36	24H	68	44H	100	64H
5	05H	37	25H	69	45H	101	65H
6	06H	38	26H	70	46H	102	66H
7	07H	39	27H	71	47H	103	67H
8	08H	40	28H	72	48H	104	68H
9	09H	41	29H	73	49H	105	69H
10	0AH	42	2AH	74	4AH	106	6AH
11	0BH	43	2BH	75	4BH	107	6BH
12	0CH	44	2CH	76	4CH	108	6CH
13	0DH	45	2DH	77	4DH	109	6DH
14	0EH	46	2EH	78	4EH	110	6EH
15	0FH	47	2FH	79	4FH	111	6FH
16	10H	48	30H	80	50H	112	70H
17	11H	49	31H	81	51H	113	71H
18	12H	50	32H	82	52H	114	72H
19	13H	51	33H	83	53H	115	73H
20	14H	52	34H	84	54H	116	74H
21	15H	53	35H	85	55H	117	75H
22	16H	54	36H	86	56H	118	76H
23	17H	55	37H	87	57H	119	77H
24	18H	56	38H	88	58H	120	78H
25	19H	57	39H	89	59H	121	79H
26	1AH	58	3AH	90	5AH	122	7AH
27	1BH	59	3BH	91	5BH	123	7BH
28	1CH	60	3CH	92	5CH	124	7CH
29	1DH	61	3DH	93	5DH	125	7DH
30	1EH	62	3EH	94	5EH	126	7EH
31	1FH	63	3FH	95	5FH	127	7FH

2. Notes on Authoring SMF

2.1. SMF Format

Be sure to use the "**SMF Format 0**." In case of "**Format 1**", a conversion to format 0 is executed internally; however, it is not a thing which guarantees the absolute conversion.

2.2. MIDI Channel

MIDI channel from 1 to 16 are available.
MIDI channels of SMF read by MA-3 Authoring Tool are divided into four tracks in MLD as shown below.

Track1 → 1 to 4, Track2 → 5 to 8, Track3 → 9 to 12, and Track4 → 13 to 16

Note: "**Inside Omission**" of tracks is not acceptable in MA-3/MLD. For instance, data which use only MIDI channel9 outputs not only Track3 but also empty Track1 and Track2, surely. Therefore, in order to save a capacity of data to create, we recommend you to use from MIDI channel 1.

2.3. Synthesizer Mode and Number of Pronunciation

MA-3 Authoring Tool has the following two modes; FM32 tone mode and FM 16 tone mode.
- In FM 32 tone mode, up to 32 tones (all 2-operator) as FM synthesizer + up to 8 tones as PCM synthesizer can be used [Total 40 tones]
- In FM 16 tone mode, up to 16 tones (all 4-operator) as FM synthesizer + up to 8 tones as PCM synthesizer can be used [Total 24 tones].

Although data can be described with poly-mode in one MIDI channel, be sure to do not exceed the maximum simultaneous pronunciation. If tone exceeding the maximum simultaneous pronunciation is input, MA-3 Authoring Tool silences the notes of pronounced tones by giving priority to the ones arrived later.

Mode change is set up by the preference of Authoring Tool.

In addition, when FM 16 mode is designated in MA-3 Authoring Tool, and if the bank select MSB is designated as 124/125, all 4-operator voices are prepared as a default voice of FM. Moreover, when FM 32 mode is designated in MA-3 Authoring Tool, and if the bank select MSB is designated as 124/125, all 2-operator voices are prepared as a default voice of FM.

For details about "**Voice MAP**", see the additional document.

2.4. Tempo

The range of tempo is determined as up to 255 from 20.
Tempo change in music is supported in MA-3 Authoring Tool.
In MA-3 Authoring Tool, tempo can not be changed after SMF is read. When there is not tempo designation, MA-3 Authoring Tool treats a quarter notes as 120. (***Quarter-note = 120***)

2.5. Time Base

In MA-3 Authoring Tool, the "***tick***" number of MA-3/MLD can be calculated with the following formula as assuming that a time base of MA-3/MLD is the fixed value 48.

(tick number of MA-3/MLD)   = ((tick number of SMF) * (Time base of MA-3/MLD)) / (Time base of SMF)

Note: In MA-3 Authoring Tool, operations of SMF which has no value of time base that is dividable by 48 are not assured.

2.6. Channel Attribute

As a channel attribution, there are Normal channel and Drum channel. These attributes can be changed by bank select.
When there is no designation by bank select specifically, the channel 10 is treated as Drum channel, and other channels are treated as Normal channel.

3. Applicable MIDI Event

MA-3 Authoring Tool targets the following MIDI events. In addition, it ignores the events except one shown in the following table. So, be sure to insert the note-event on it. The following description of initialization setting value shows a default value used by MA-3 Authoring Tool when there is no specification in SMF.

MIDI events to be used are shown in the following Table 2.

Table 2 List of Used MIDI Event

MIDI Event Name	Format
Note-ON	**9nH kkH vvH**
Note-OFF	**8nH kkH vvH**
Program Change	**CnH ppH**
Bank Select	**BnH 00H mmH (MSB)** **BnH 20H llH (LSB)**
Modulation Depth	**BnH 01H vvH**
Channel Volume	**BnH 07H vvH**
Panpot	**BnH 0AH vvH**
Expression	**BnH 0BH vvH**
Hold1 (Dumper)	**BnH 40H vvH**
Data Entry	**BnH 06H mmH (MSB)** **BnH 26H llH (LSB)**
RPN	**BnH 64H llH (LSB)** **BnH 65H mmH (MSB)**
Pitch Bend Sensitivity	**BnH 64H 00H / BnH 65H 00H (RPN Parameter Designation)** **BnH 06H mmH / BnH 26H llH (Data Entry)**
Mono Mode ON	**BnH 7EH 01H**
Poly Mode ON	**BnH 7FH 00H**
Pitch Bend	**EnH llH mmH**
Tempo	**FFH 51H 03H ttH ttH ttH**
Play Start-up End Position	**FFH 06H 03H 51H 30H 30H (Q00)** **FFH 06H 03H 51H 30H 46H (Q0F)**
Copyright Display	**FFH 02H llH ddH…ddH**
Cue Point	**FFH 07H 05H 53H 54H 41H 52H 54H (START)** **FFH 07H 04H 53H 54H 4FH 50H (STOP)**
Master Volume	**F0H 7FH 7FH 04H 01H llH mmH F7H**
Stream PCM Wave Panpot	**F0H 43H 79H 06H 7FH 0BH iiH ccH ddH F7H**
Stream PCM Pronunciation Designation	**F0H 43H 79H 06H 7FH 07H ddH F7H**

3.1. Note ON

9nH   kkH   vvH

n: Channel Number 0 to 15(0H to FH)
kk: Note Number 0 to 114 (00H to 72H) A=69 of 440Hz
vv: Key Velocity 1 to 127: When "0", it is interpreted as Note-off

In applicable channel, a pronunciation by key of designated note number is started up.
When an applicable channel is a Normal channel, the keys of note number 21 to 114 are pronounced. In addition, Note number 0 to 20 and 115 to 127 are ignored.
When an applicable channel is a Drum/StreamPCM channel, keys of note number 0 to 12 and 92 to 110 indicate a pronunciation starting of StreamPCM.
Note event of StreamPCM can be input from Authoring Tool (Stream PCM Edit View of Piano Roll Window).

3.2. Note OFF

8nH   kkH   vvH

n: Channel Number 0 to 15 (0H to FH)
kk: Note Number 0 to 114 (0H to 72H) A=69 of 440Hz
vv: Key velocity is ignored.

In applicable channel, a pronunciation is ended by key of designated note number. When an applicable channel is a Drum/StreamPCM channel, keys of note number 0 to 12 and 92 to 110 indicate a pronunciation end of StreamPCM.

Note: In MA-3 Authoring Tool, when the time interval (*Gate-time*) between Note-On and Note-Off is less than 1-tick, the gate time is determined as 1-tick and is pronounced. However, when a gate-time of StreamPCM is less than 1-tick, it is ignored.
In addition, be sure to do not generate a sound by overlapping a same note in same channel. The pronunciation of the note which overlapped at this time may not become as input SMF.

Note: Since a quantization is set in MA-3 Authoring Tool when processing, a note with a gate-time (around 10 ms or less) may not be pronounced. For the calculation formula of time (*msec*) in SMF and tick, see "*Volume Specification and Note-message*" later in this document.

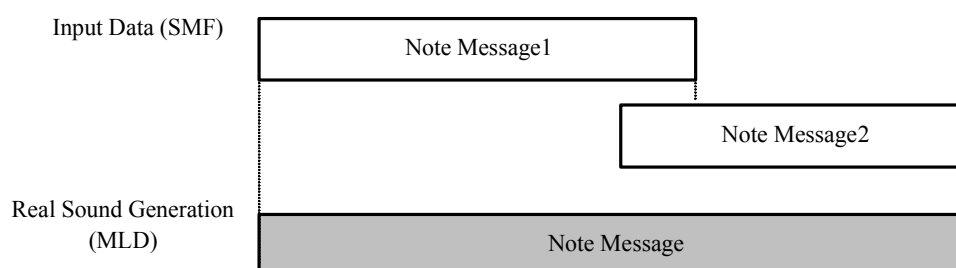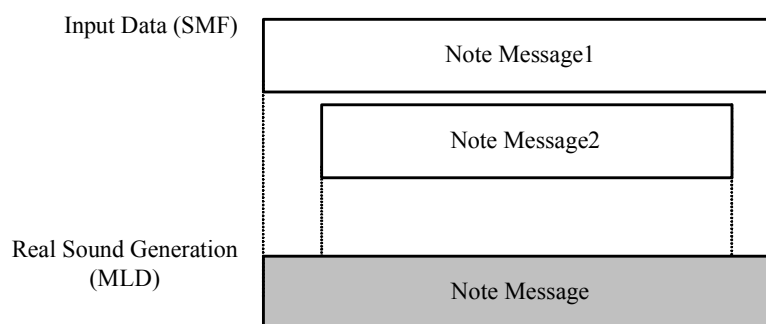**Figure A: Case 1 → When same notes are overlapped in same channel**

Figure B: Case 2 → When same notes are overlapped in same channel

Input Data (SMF)

| Note Message1 |
| Note Message2 |

Real Sound Generation
(MLD)

| Note Message |

3.3. Program Change

CnH   ppH

n: Channel Number 0 to 15 (0H to FH)
pp: Program Number 0 to 127 (00H to 7FH)

Initial Setting Value: 0

Voice of designated channel is set up.
If an applicable channel is set up in Normal channel, a voice is selected from specified bank. In addition, if an applicable channel is set up in Drum channel, a drum set is selected. At this time, the following voices are selected by Program change.

Table 3 Bank Select Supporting Table (Drum/Stream Channel)

MSB	Category	Program Change				
		0	1	2	3 to 9	10 to 127
123, 125	Drum/ StreamPCM	Default Drum1	Default Drum2	User Drum	User Voice	Displace to User Drum

Voice which is available in MA-3 Authoring System is as follows.

- ・ Default Voice：Initial Setting Voice of MA-3 Authoring Tool
  - ➢ Default Melody
  - ➢ Default Drum1
  - ➢ Default Drum2

- ・ User Voice：Voice to be edited on MA-3 Authoring Tool
  - ➢ User Melody (Voice bank replaced from the Melody specification besides the range)
  - ➢ User Drum (Voice bank replaced from the Drum specification besides the range)
  - ➢ Other User Voices

Be sure to insert a program change to the next of a bank select of each channel head. Program change in music has no effect to the note which is under sound generation of the applicable channel. It becomes valid from the next Note-on.
Specially, in case of Mono-mode, a program change under sound generation is prohibited.

3.4. Control Change

3.4.1. Bank Select

BnH	00H	mmH (MSB)
BnH	20H	llH (LSB)

  n:　　　Channel Number 0 to 15 (0H to FH)
  mm:　　Bank Number MSB value 124, 125 (7CH, 7DH)
  ll:　　　Bank Number LSB value 0 to 127 (00H to 7FH)

  Initial Setting Value：　0/0

Bank of specification channel is set up.
As a general rule…
In case of Melody, be sure to use it with the following range.
BankSelectMSB：124, BankSelectLSB：0 to 9, ProgramChange：0 to 127
In case of Drum (*Percussion*), be sure to use it with the following range.
BankSelectMSB：125, BankSelectLSB：0, ProgramChange：0 to 9 (1 to 10)

For an interpretation about the bank which exceeds the above range, see the following table.

Table 4 Bank Select Supporting Table

MSB	Category	LSB										
		0	1	2	3	4	5	6	7	8	9	10 to 127
0 to 121	Depends on Channel	Except Channel 10：replace to User-Melody Channel 10：replace to User-Drum1										
122, 124	Normal	Default Melody	User Melody	User Voice								replace to User Melody
123, 125	Drum/ StreamPCM	No matter Bank LSB may be what numerical value, it is interpreted as 0. (*)										
126, 127	Depends on Channel	Except channel 10：replace to User-Melody Channel 10：replace to User-Drum1										

  (*) In case of Bank MSB 125, a voice to be used is decided by Program Change. For details, see "3.3 Program Change."

  In each channel, by designating a program change after specifying a drum bank, it becomes a Drum channel. In addition, by designating a program change after specifying a normal bank, it becomes a Normal channel. Moreover, even if a bank select is received, a voice of last program change will be effective until the next program change is received.
  Furthermore, in case of Drum/StreamPCM, a bank select depends on a Program change. For details about this correspondence, see "*Program Change*" later in this document.

  If two or more bank select exists, the newest message (back in time-axis) is processed by priority.

  By designating the Bank MSB 125, an applicable channel becomes Drum/Stream PCM channel. When a drum set is changed by Program change, an instrument of drum is switched into the instrument which corresponds to voice map. No matter what program change comes into Stream PCM, it supports uniquely like the relation between Note number and Stream Wave ID. However, the Stream Wave ID which can be registered into MLD is up to 32.
  For details about the voice which can be set up by Bank Select and Program change, see "3.3Program Change."

Table 5 Supporting Table of Drum/Stream PCM Bank Note#

Note #	Definition	Assign
0		Stream Wave ID :1
1		Stream Wave ID :2
2	Stream PCM	Stream Wave ID :3
:		:
12		Stream Wave ID :13
13		No Instrument
14		No Instrument
15	Drum Instrument	No Instrument
:		:
91		No Instrument
92		Stream Wave ID :14
93		Stream Wave ID :15
94	Stream PCM	Stream Wave ID :16
:		:
110		Stream Wave ID :32

3.4.2. Modulation Depth

BnH   01H   vvH

n: Channel Number 0 to 15 (0H to FH)
vv: Vibrato value 0 to 127 (00H to 7FH)

Initial Setting Value: 0

Designates the depth of vibrato (LFO pitch modulation) of a designated channel.
The relation between the value and depth is shown in "**Table 6**." The depth of vibrato here shows the multiple for vibrato depth that is set for each voice.

Table 6 Relation between Vibrato value and Depth

Vibrato value	Depth of vibrato
0	OFF
1 to 31	x 1
32 to 63	x 2
64 to 95	x 4
96 to127	x 8

When the applicable channel is Drum/Stream PCM channel, it is invalid for note numbers 0 to 12 and 92 to110. (0 fixation)

3.4.3. Channel Volume

BnH   07H   vvH

n: Channel Number 0 to 15 (0H to FH)
vv: Control Value 0 to 127 (00H to 7FH)

Initial Setting Value: 100 (64H)

It aims to set up the volume balance between channels by the message which specifies the volume of an applicable channel.

Formula: Gain [dB] = $20*\log((vv)^2/127^2)$

When the applicable channel is Drum/Stream PCM channel, it is invalid to note numbers 0 to 12 and 92 to110.

Note: For the value of individual waveforms that are assigned to Stream PCM, use velocity.

3.4.4. Panpot

BnH   0AH   vvH

n: Channel Number 0 to 15 (0H to FH)
vv: Control Value 0 to 127 (00H to 7FH)

Initial Setting Value: 64 (40H) [Center]

Designates a position in the stereophonic sound field of designated channels. The positioning is made between the left end (vv=127) of the stereophonic sound field by using the following formulas.
When the applicable channel is Drum/Stream PCM channel, it is also valid for note numbers 0 to 12 and 92to 110 (Decimal numbers).

Formulas: Left Channel Gain[dB] = $20*\log(\cos(\pi/2*(vv)/127)$
                 Right Channel Gain[dB] = $20*\log(\sin(\pi/2*(vv)/127)$

Note: If a malfunction occurs when setting a panpot of same channel by same timing, the last panpot setting may be ignored; for the reasons, be sure to avoid such setting.

3.4.5. Expression

<u>**BnH   0BH   vvH**</u>

n: Channel Number 0 to 15 (0H to FH)
vv: Control Value 0 to 127 (00H to 7FH)

Initial Setting Value： 127 (7FH)

Designates a change of volume set up by channel volume of an applicable channel.
When the applicable channel is Drum/Stream PCM channel, it is invalid for note numbers 0 to 12 and 92 to110.

Note: <u>Although both of channel volumes and Expression control volume, the purposes differ. Channel volume is used for mixing-down with fader or whole-music volume which is set up in advance of playback of music data. In addition, Expression is used to control volumes to effecting expressions to the sound in the music etc.</u>

Formula: $Exp[dB] = 20*\log((vv)^2/127^2)$

3.4.6. Hold1 (Dumper)

<u>**BnH   40H   vvH**</u>

n: Channel Number 0 to 15 (0H to FH)
vv: Control Value 0 to 127 (00H to 7FH)

Initial Setting Value: 0

Designate ON/OFF of damper (Sustain Pedal) of application channels. Off is designated when the value is 0 to 63, or On is designated when the value is 64 to 127.
When the applicable channel is Drum/Stream PCM channel, it is invalid for note numbers 0 to 12 and 92 to 110. (0 fixation)

Note: <u>When Note-off is received in Note-on, Note-off may be held. When Dumper is switched from "ON" to "OFF", the delayed Note-off is implemented, and the volume-envelop is shifted to release.</u>

Note: <u>If Note-on is implemented to the note which has an especially short gate-time, a sustain may be ineffective. So, be sure to do something such as extending a release of voice, etc.</u>

Note: <u>When both Dumper-on and Note-off exist simultaneously, effects may no be works effectively. It is the convenience of hardware and is a reason that both Damper-on and Note-off cannot be processed simultaneously. In this case, be sure to designate a Damper-on 10msec before a Note-off.</u>

3.4.7. Data Entry

BnH   06H   mmH (MSB)
BnH   26H   llH (LSB)

n: Channel Number 0 to 15 (0H to FH)
mm: Data Value MSB 0 to 127 (00H to 7FH)
ll: Data Value LSB 0 to 127 (00H to 7FH)

Initial Setting Value: 0/0

It is used to the input of RPN value (MSB/LSB). For details, see "3.4.8 **RPN**."

3.4.8. RPN

BnH   64H   llH (LSB)
BnH   65H   mmH (MSB)

n: Channel Number 0 to 15 (0H to FH)
ll: Parameter Number MSB 0 to 127 (00H to 7FH)
mm: Parameter Number MSB 0 to 127 (00H to 7FH)

Initial Setting Value: 127/127 (7FH/7FH)

It is used to designate a parameter number of PRN.

3.4.8.1. Pitch Bend Sensitivity

BnH 64H 00H / BnH 65H 00H (RPN Parameter Designation)
BnH 06H mmH   / BnH 26H llH   (Data Entry)

n: Channel Number 0 to 15 (0H to FH)
mm: Data Value MSB 0 to 24 (00H to 18H)
ll: Data Value LSB (0 fixation)

Initial Setting Value: 2/0 (2-semitone)

   Sensitivity of Pitch Bend is performed. MSB of data entry shows a sensitivity of semitone units, and LSB of data entry shows a cent unit. For example, when MSB=1 and LSB=0, it becomes ±1 semitone (the total of change range becomes total two-semitones.)

3.4.9. Mono Mode ON

BnH   7EH   01H

n: Channel Number 0 to 15 (0H to FH)

An applicable channel is switched to Mono-mode.
When an applicable channel is Drum/StreamPCM channel, this massage becomes invalid.
Be sure to do not designate it during Note-on.
Note of poly performs a process of slur (*legato*) when a channel is in Mono-mode.
Simultaneous-pronunciation of multi-voices in Mono-mode or the pronunciation designation with a short interval that seems simultaneous is prohibited.
The time interval it can be considered that is simultaneous is 1.5 or less msecs.

3.4.10. Poly Mode ON

BnH   7FH   01H

n: Channel Number 0 to 15 (0H to FH)

An applicable channel is switched to Poly-mode.
Mode change in music is prohibited.
Drum/Stream PCM channel is always pronounced irrespectively to the existence of this message by Poly-mode.

3.5. Pitch Bend

EnH   llH   mmH

n: Channel Number 0 to 15 (0H to FH)
ll: Bend Value LSB 0 to 127 (00H to 7FH)
mm: Bend Value MSB 0 to 127 (00H to 7FH)

Initial Setting Value: 0/64 (0H/40H) [Center]

Pitch of an applicable channel is changed ups and downs. The initial value of change width (Pitch bend sensitivity) is ±2 semitone. 0/0 makes the downward pitch bend maximum. 127/127 makes the upward pitch bend maximum. Pitch bend sensitivity can be set with 00H/00H of RPN.
When the applicable channel is Drum/StreamPCM channel, it is invalid for note numbers 0 to 12 and 92 to 110.

3.6. Meta Event

3.6.1. Tempo

FFH	51H	03H	ttH	ttH	ttH

  tt tt tt: Length of Quarter Notes (μ sec)

   MA-3 Authoring Tool supports a tempo change in music. A tempo change designated in given position is interpreted.

   Moreover, the range of tempo in MA-3 Authoring Tool is set as the minimum value 20(2DH C6H C0H) and the maximum value 255(03H 97H 1EH); in addition, the value smaller than the minimum value is converted to the minimum value 20 and the value bigger than the maximum value is converted to the maximum value 255.

3.6.2. Playback Startup Position/Playback End Position

FFH	06H	03H	51H	30H	30H	(Q00)
FFH	06H	03H	51H	30H	46H	(Q0F)

Both playback startup position and playback end position is described as a Marker of Meta-event.
MA-3 Authoring Tool converts the event to the playback start position and playback end position of MLD.
4 to 6 byte (51H 30H 30H) of playback start position is represented as "Q00" (Capital letter) by ASCII.
4 to 6 byte (51H 30H 46H) of playback end position is represented as "Q0F" (Capital letter) by ASCII.
Be sure to insert "Q00" and "Q0F" as a pair in music.
When there is no "Q00" and "Q0F", a playback start position is automatically inserted into the head of music; in addition, a playback end position is automatically inserted into the end of music. When authoring contents for mobile-terminals, it is not necessary to designate both playback start position and playback end position, especially.

3.6.3. Display of Copyright

FFH	02H	llH	ddH

  ll :   Byte number of text data (Variable length presentation)
  dd: Text Data

   By describing the copyright information, copyright can be input.

3.6.4. CuePoint

FFH   07H   05H   53H   54H   41H   52H   54H (START)

It is inserted into the startup position of music. When a setup bar is set up, be sure to insert it immediately after that. Data in a setup bar is inserted into the time interval which is same as start point.
In addition, when a playback startup position is left out, the start point becomes a playback startup position.
Be sure to do not designate it during Note-on.

FFH   07H   04H   53H   54H   4FH   50H (STOP)

It is inserted into the end of music.
Events exist after the stop point is ignored.
Be sure to do not designate it during Note-on.

3.7. Universal System Exclusive Message

3.7.1. Master Volume

Message	Contents
F0H 7FH	Universal real time exclusive header
<device ID>	ID of unit that becomes a target (127:ALL)
04H	Sub-ID number #1
01H	Sub-ID number #2
ll	Master Volume LSB
mm	Master Volume MSB
F7H	EOX

Initial Setting Value: 100 (64H)

The volume of synthesizer output in the last level is set up.
ll section is ignored.
When an applicable channel is Drum/StreamPCM channel, it becomes invalid to the Note-number 0-12 and 92-110.

Formula: $Gain[dB] = 20*\log((Data)^2/127^2)$

Note: We recommend you to raise volume control for the final contents to the maximum level which is not
      clipped.

F0H   7FH   7FH   04H   01H   ll   mm   F7H
ll :   Master Volume LSB
mm :   Master Volume MSB 0 to 127 (00H to 7FH)

Initial Setting Value: 100 (64H)

The volume of synthesizer output in the last level is set up.
Be sure to set LSB to "0." (LSB is ignored.）

3.8. Classified System Exclusive Message

Definitions such as voice setting of device specific or waveform setting is performed by exclusive.

3.8.1. MA-3 StreamPCM Wave Panpot

F0H   43H   79H   06H   7FH   0BH   iiH   ccH   ddH   F7H

ii: WaveID 1 to 32 (1H to 20F)
cc: Panpot designation → 0, Clear → 1, Pan-off → 2
dd: Panpot value 0 to 127 (00H to 7FH)

 Panpot for individual waveform in StreamPCM of applicable channel is designated. "Data=0" shows in left end, and "127" shows in right end.
 By receiving this message, a channel panpot is made to invalid. (Waveforms that not designated by this message uses the setting of channel panpot)
 After receiving this message, wave panpot designation is given priority only when clear is issued with this message.
 By designating "1" for CL, all of wave panpot settings that have been received are returned to channel panpot. Moreover, by specifying "2" to CL, turns off panpot assignment and pronounces by 0dB.

3.8.2. Designation of MA-3 StreamPCM Pronunciation Number

F0H   43H   79H   06H   7FH   07H   ddH   F7H

dd: StreamPCM Pronunciation Number 0 to 2 (00H to 02H)

Simultaneous pronunciation number of StreamPCM is designated.
Even if the StreamPCM is registered, pronunciation of StreamPCM is restricted when the number of pronunciation is limited by this message.
When StreamPCM is not used, and if "0" is designated, RAM in LSI can be used effectively.
When designating a Stereo Stream, be sure to designate "1."

4. Note in the Setting of Stream PCM

4.1. Maximum Number of Pronunciation

The maximum number of sound generation in Stream PCM is designated by MA-3 Authoring Tool ("Reserve setting" of Piano Roll/Stream PCM Edit View), and it is to two at the maximum.

In addition, the simultaneous pronunciation of the stream exceeding the reserve number was not guaranteed. The stream pronounced simultaneously should not exceed a setup of the reserve number.

Moreover, 1024 bytes of MA-3 RAM area (total 8176byte) is consumed by one stream (2048 bytes is consumed at the maximum).

4.2. Panpot

As means for setting panpot in stream PCM, two methods are available; setting it with channel panpot by using control change and setting it with MA-3 stream PCM wave panpot.

When the former method is used, when, for example, two stream PCMs exist in one channel at the same time, panpot with the same value is set in both of them. When the instrumental sound of drum exists in an applicable channel, this is also set in the panpot with the same value. When panpot of only one stream PCM is set at the same time, it is necessary to assign one stream PCM to one channel. At this time, panpot can be changed during generation of tones (between NoteOn and NoteOff).

When the latter method is used, even when, for example, two stream PCMs exist in one channel at the same time, panpot can be set for the stream PCMs individual. It can be set individually even if instrumental sound of drum exists. At this time, change of panpot is prohibited during generation of tones (between NoteOn and NoteOff).
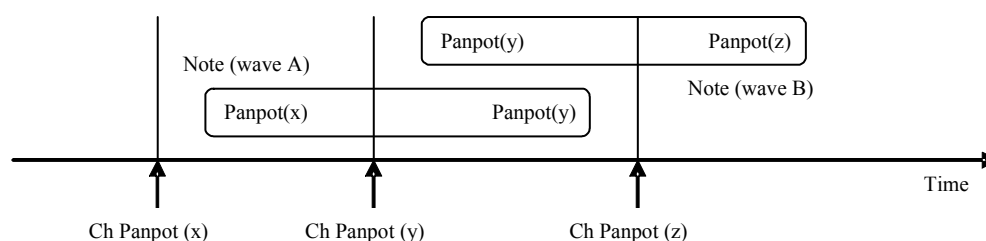

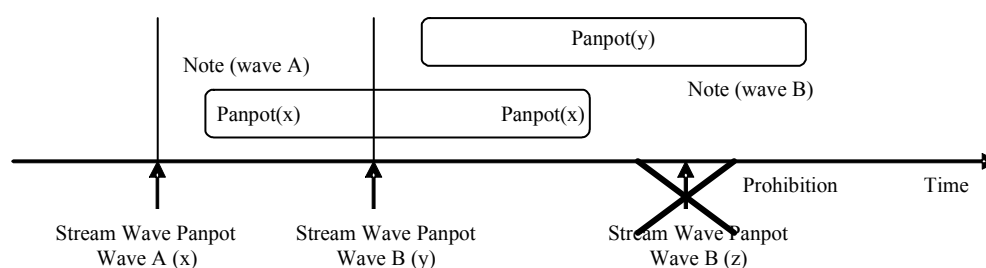
Figure C Setting with Channel Panpot

Figure D Setting with MA-3 StreamPCM Wave Panpot

5. Supplement

5.1. Volume Designation and Note-event

   In MA-3 Authoring Tool, please do not place a Note-event and volume specification at the same time because it may cause an outbreak of noises or lost an attack of sound. In order to avoid this, after volume specification should place a note event, after vacating the time more than 22 msec.
   The messages of the target volume specification are Master volume, Channel volume, Expression, and Pan pot. When especially volume change is large, it becomes easy to generate this problem.
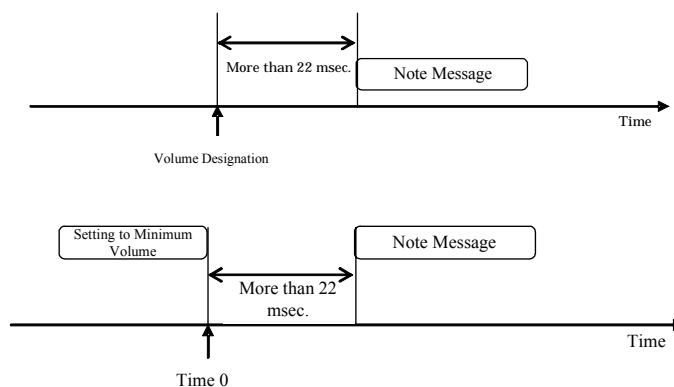

Figure E Volume Designation and Note-event

   Note: When the above mentioned problem is not corrected, extend the interval of volume specification and a note message further. The calculation formula for time (*msec*) and tick in SMF is shown below.

Formula：
   (tick number of SMF)   = ((Temp) *(Time of SMF)*(division of SMF)) / (60*1000)

   E.g.: (Tempo) = 160, (division of SMF) = 480, and Time in SMF (22 msec).
      It becomes (tick number of SMF) = 28 (*tick*).

5.2. Mono-mode ON and Restriction of the Maximum Simultaneous Pronunciation Number

In MA-3 Authoring Tool, when there is a channel in which uses Mono-mode ON, a MLD which exceeds the maximum simultaneous pronunciation number can not be created.
In which case, an error may be displayed if it exceeds the maximum pronunciation number using Mono-mode.
By not using Mono-mode ON or decreasing the simultaneous pronunciation number to be not exceeding the maximum simultaneous pronunciation number, be sure to avoid such problems.
In addition, the maximum simultaneous pronunciation number of WT voice changes according to the setting value of StreamPCM pronunciation number.

When StreamPCM is not used, the maximum pronunciation number of WT voice becomes 8.
When StreamPCM pronunciation number 1 is set up, the maximum pronunciation number of WT voice becomes 7.
When StreamPCM pronunciation number 1 is set up, the maximum pronunciation number of WT voice becomes 6.

5.3. Note-ON of same timing in Mono-mode ON

In a channel, which uses Mono-mode ON, be sure to do not place multi-Note-ON in same timing (Duration=0).
Although the last note is generated when there is multi-Note-ON existing in same timing in a channel which uses Mono-mode ON, there is a case in which the value may not reach to the total level (the volume becomes smaller).

5.4. Pronounceable Frequency of PCM Voice PCM

The pronunciation frequency range of a PCM voice is 1.5 kHz to 48 kHz. Please do not carry out pronunciation which is out of this range. When the frequency which reflected a pronunciation key, pitch bend, and LFO in Fs (frequency when flipping NoteNo.60 (C key)) exceeds this range, it processes as follows.

By the pronounced key

- When becomes smaller than 1.5 kHz : It is 1.5 kHz
- When becomes bigger than 48 kHz   : It is 48 kHz

By the pitch bend or LFO

- When becomes smaller than 1.5 kHz : 1.5 kHz or less is pronounced.
- When becomes more than 48 kHz: Un-expected value is carried out (Because of BTB)

5.5. Recommended Fs Setup Value of PCM Voice

If it uses for Fs of a PCM tone except the value of the following table "a recommendation Fs setting value list", gap may arise in a pitch. Table
Please set up the value of Fs like shown in the following table.

							(Unit Hz)
4125	10125	16125	22125	28125	34125	40125	46125
4500	10500	16500	22500	28500	34500	40500	46500
4875	10875	16875	22875	28875	34875	40875	46875
5250	11250	17250	23250	29250	35250	41250	47250
5625	11625	17625	23625	29625	35625	41625	47625
6000	12000	18000	24000	30000	36000	42000	48000
6375	12375	18375	24375	30375	36375	42375	
6750	12750	18750	24750	30750	36750	42750	
7125	13125	19125	25125	31125	37125	43125	
7500	13500	19500	25500	31500	37500	43500	
7875	13875	19875	25875	31875	37875	43875	
8250	14250	20250	26250	32250	38250	44250	
8625	14625	20625	26625	32625	38625	44625	
9000	15000	21000	27000	33000	39000	45000	
9375	15375	21375	27375	33375	39375	45375	
9750	15750	21750	27750	33750	39750	45750	

Table 7 Recommended Setting Values of Fs

5.6. Restriction of Available Voice Number

In melody voice, the total number of program change types which can be used in music is 127.
Be sure to use it by 127 or lower.
Moreover, in drum voice, the total number of note message types which can be used in music is 128. Be sure use it by 128 or lower.
The above mentioned restriction is a total of available voice types in music. When using a voice which exceeds those restrictions, a defect such as voices are not pronounced may occur.
Please understand the situation kindly.

5.7. Event Density Restriction

Event Density is a thing which defines the event numbers around unit time, and it can be calculated by Note-event (6-byte), Control-event (3-byte), Pitch-event (3-byte), and Exclusive Message (byte number of data section and 2-byte (F0, F7). The unit is "*byte/sec*."
The types of event density and the fiducial value to each in MA-3 Authoring Tool are described in table 9.

Table 8 Types and Fiducial Point of Event Density

Event Density	Definition	Fiducial Point {byte/s}
Average Event Density	Average event density through one song	500
Momentary Maximum Event Density	Event density at the point where a value is the biggest in a song.	1000

In MA-3 Authoring Tool, about the data which is bigger than the fiducial values shown in Table 9, restriction is prepared so that it cannot be saved.

5.8. Pronunciation Range in PCM Voice (WT Synthesizer)

The range of playback frequency is 1500Hz to 48000Hz.
If exceeding 48000Hz, a playback in a range from 48000Hz to 96000Hz is performed by 48000Hz.
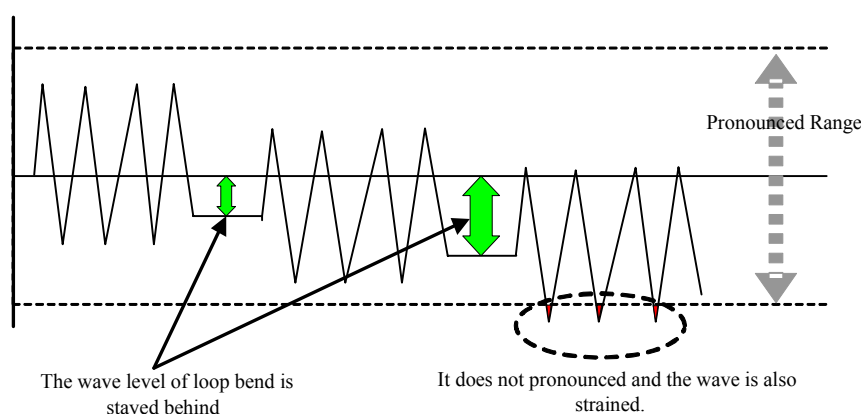However, a playback more than 96000Hz can not be guaranteed.

5.9. Note at the Time of Creating Voice Included PCM User Wave

When you create the tone using the PCM user waveform, be careful of below with the specification of MA-3 hardware.

When there is no waveform loop (a loop point and a loop end are the same value), the waveform value is continued and read with the specification of MA-3 hardware in the place where read-out of a waveform reached the loop end. For this reason, a tone parameter -- XOF = 1 -- and -- SR = 0 (or setup with long attenuation time) XOF = 0 [or] -- and -- If it is set as RR = 0 (or setup with long attenuation time), after note-off will be continued and this value will be maintained.

In this state, when much note-on of big volume overlaps, sound becomes easy to be distorted. Moreover, if the tone of such a waveform is pronounced repeatedly, it will become large by the number of times by which the value maintained in note-off was also pronounced, and will much more become easy to be distorted.

The wave level of loop bend is stayed behind

It does not pronounced and the wave is also strained.

In order to prevent such a condition, a loop and the waveform level which comes out are set to "0", or it recommends a loop and adjusting an envelope so that pronunciation may be lost more in front.
Please make a PCM voice according to.

With or without of wave	What kinds of color?	Wave level on loop bend	XOF	DR	SR	RR	SUS	Point
Without	One shot or Chunked	0	free	free	free	free	free	A problem is not produced.
		not 0	on	not 0 Sum of two lapsed time are shorter is better.	free	free	off	There is a possibility that a problem will arise. By DR and SR, please adjust so that pronunciation is lost a loop and before.
			off	free	free	not 0 Shorter is better	off	There is a possibility that a problem will arise. By RR, please adjust so that pronunciation is lost a loop and before.
With	Continuance	0	off	free	0	not 0	free	A problem is not produced.
		not 0	off	free	0	not 0	free	A problem is not produced.
	With Loop Decay	0	free	free Attenuation is generated.	free	free	free	A problem is not produced.
		not 0	free	free Attenuation is generated.	free	free	free	A problem is not produced.
Free : You may have what setup carried out.								

Table 9 PCM Voice Authoring Guideline

In addition, when the O.K. button is clicked on PCM Voice Edit, the following checks are performed.

1) Check of LoopPoint/EndPoint
When "LoopPoint/EndPoint" is out of range shown below, it's determined as an error.
・In case of 4bit ADPCM,
$0 \leqq$ LoopPoint $\leqq$ [Wave form sampling number -1]
$1 \leqq$ EndPoint $\leqq$ [Wave form sampling number]
・In case of 8bit PCM,
$0 \leqq$ LoopPoint $\leqq$ [Wave form sampling number -2]
$0 \leqq$ EndPoint $\leqq$ [Wave form sampling number -1]
・When LoopPoint pointes out the position exceeding EndPoint, it considers as an error.

2) Check of EG in case of Loop Point = End Point, and check of wave height at LoopPoint
When all the conditions of the following a. to b. are satisfied, it considers as an error.
a.Loop Point and End Point are specified as same position.
b.Evne in one side, 1000 or more are the wave high price of the following sample points by 16bit PCM conversion.
It is a sample in front of Loop Point (=End Point) and its one at the time of 4bit ADPCM use.
It is the sample of Loop Point (=End Point) and its one back at the time of 8bit PCM use.
c. A setup applicable to following either is made.
(a) XOF＝1 and SR$\leqq$1
(b) XOF＝1 and DR＝0 and SL$\neq$0
(c) XOF＝0 and RR$\leqq$1

5.10.Total Length after Conversion

The delta time in MLD is expressed for 1-byte fixed length numerical value. When exceeding 255, it is expressed by inserting two or more NOP events. Therefore, MLD needs the amount of data more nearly excessive than SMF, which is expressed by variable length numerical value of the delta time, in order to express the big delta time. The size of MLD file after conversion may enlarge.
Especially, if such a delta time exists in the channel 16, that influence will become remarkable because the inside omission of a track is not accepted. (see "2.2MIDI **Channel**.")

In conversion processing, the output buffer of a conversion file is secured based on the size of former SMF. However, the error of output buffer shortage may occur at the time of conversion of SMF with a big delta time by the above-mentioned reason. Please pay attention.
The example of file size after MLD conversion of SMF is shown below as reference data. The SMF consists of only two notes with an interval of the specified measure number.

Note interval	10 measures	20 measures	30 measures	40 measures	50 measures
SMF Size	82-byte	82-byte	82-byte	82-byte	82-byte
MLD Size	392-byte	516-byte	632-byte	756-byte	Error

(*When ch16 of SMF is only used.)