

DPM

SP



**SAMPLE
PLAYBACK
SYNTHESIZER**



written by
CRAIG ANDERTON





Intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.



Intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product.

CAUTION

RISK OF ELECTRIC SHOCK
DO NOT OPEN

WARNING

TO REDUCE THE RISK OF FIRE OR ELECTRIC SHOCK DO NOT EXPOSE THIS EQUIPMENT TO RAIN OR MOISTURE.

AVIS: RISQUE DE CHOC ELECTRIQUE-NE PAS OUVRIR.

Peavey DPM SP Manual

Chapter 1 SP Overview, User Interface, and Hookup

1.0 Basics.....	1-1
1.1 About This Manual: Learning the SP.....	1-2
1.2 SP Architecture.....	1-3
1.3 SP Operating System Basics	1-4
1.4 Screen Layouts.....	1-5
1.5 Navigating Within a Screen: The Arrow Buttons.....	1-5
1.6 Changing Parameter Values.....	1-5
1.7 EXEC Button Details.....	1-6
1.8 User Interface Fine Points & Power Tips.....	1-7
1.9 What's a Kiloword?.....	1-8
1.10 Conventions Used in this Manual	1-8
1.11 Basic Audio Hookup.....	1-8
1.12 Basic MIDI Hookup	1-10
1.13 Cascading Two or More SPs (Overflow Mode).....	1-10
1.14 SCSI Hookup.....	1-10

Chapter 2 Disk Menu

2.0 About Storage Devices.....	2-1
2.0A Floppy Disk Procedures.....	2-3
2.0B Care & Handling of Disks and Removable Cartridges.....	2-4
2.0C An Important Note About Banks!	2-5
2.1 Main Disk Menu Screens.....	2-6
2.1A Bank Auto-Load from Floppy Disk	2-6
2.1B Load Bank	2-6
2.1C Edit Bank Name	2-7
2.1D Save Bank to Disk.....	2-8
2.1E Add Wave, Tone, Map, Preset, or Multi to Memory.....	2-10
2.1F Get Info	2-13
2.2 Disk Utilities (Folder).....	2-15
2.2A Delete Bank on Disk (SCSI Devices Only).....	2-15
2.2B (Un)lock Bank (SCSI Devices Only)	2-16
2.2C Edit Volume Name (SCSI Devices Only).....	2-16
2.2D Initialize Device for SP Sound Storage	2-17

Chapter 3 Wave Menu

3.0 About Waves.....	3-1
3.1 Stereo Sampling Basics.....	3-1
3.2 Wave Management Screens.....	3-3
3.2A Wave Play Screen.....	3-3
3.2B Receive Sample Via MIDI.....	3-3
3.2C Wave Memory Status	3-4
3.2D Wave Info.....	3-5
3.2E Delete Wave	3-6
3.2F Copy Wave.....	3-6
3.2G Copy Name to Wave.....	3-7
3.2H Edit Wave Name.....	3-7
3.3 Wave Edit Functions (Folder).....	3-8
3.3A Wavepoint Audition.....	3-8
3.3B Wave Start/End.....	3-10
3.3C Trim Excess Memory	3-11
3.3D Dual Wave Audition.....	3-12
3.3E Stereo Merge.....	3-13
3.3F Wave Loop Points.....	3-14
3.3G Wave Loop Enable.....	3-15
3.3H Loop Length Fraction.....	3-15
3.3I Original Pitch.....	3-15
3.3J Wave Finetune.....	3-16

Chapter 4 Tone Menu

4.0 About Tones.....	4-1
4.0A The SP's "Synthesizer" Functions	4-1
4.0B DCO (Digitally Controlled Oscillator)	4-1
4.0C DCF (Digitally Controlled Filter).....	4-3
4.0D DCA (Digitally Controlled Amplifier).....	4-3
4.0E About SP Modulation Sources	4-3
4.0F Unipolar and Bipolar Modulation	4-5
4.0G Modulation Polarity.....	4-6
4.0H Positive and Negative Modulation.....	4-6
4.0I Modulation Subtleties.....	4-7
4.1 Tone Management Screens	4-8
4.1A Tone Play Screen.....	4-8
4.1B Create Tone.....	4-8
4.1C Delete Tone.....	4-9
4.1D Copy Tone.....	4-10
4.1E Copy Name to Tone.....	4-10
4.1F Edit Tone Name.....	4-11
4.1G Select Wave.....	4-12

4.2 Tone Modification Screens.....	4-12
4.2A Env 1 Times.....	4-12
4.2B Env 2 Times.....	4-14
4.2C DCA Envelope.....	4-15
4.2D DCA Modulation.....	4-15
4.2E DCA Aux Mod.....	4-16
4.2F DCF Cutoff.....	4-17
4.2G DCF Modulation.....	4-17
4.2H DCF Aux Mod.....	4-18
4.2I DCO Envelope.....	4-19
4.2J DCO Velocity Modulation.....	4-20
4.2K DCO LFO Modulation.....	4-20
4.2L DCO LFO Mode.....	4-21
4.2M DCO Aux Mod.....	4-21
4.2N DCO Auxiliary Modulation Mode.....	4-22
4.2O Pan Envelope.....	4-23
4.2P Pan Velocity Modulation.....	4-23
4.2Q Pan LFO Modulation.....	4-24
4.2R Pan LFO Mode.....	4-25
4.2S Pan Aux Mod.....	4-25
4.2T Pan Auxiliary Modulation Mode.....	4-26
4.2U LFO Waveshape.....	4-27
4.2V LFO Base Setup.....	4-29
4.2W LFO Speed Mod.....	4-30
4.2X LFO Amount Mod.....	4-31
4.2Y Wave Start Modulation.....	4-32
4.2Z Wave Start Modulation Quantization.....	4-33

Chapter 5 Map Menu

5.0 About Maps.....	5-1
5.0A Assigning Tones to Particular Zones.....	5-1
5.0B Transposition, Volume, and Pan.....	5-2
5.0C Individual Output Options.....	5-3
5.1 Map Management Screens.....	5-4
5.1A Map Play Screen.....	5-4
5.1B Create Map.....	5-4
5.1C Delete Map.....	5-4
5.1D Copy Map.....	5-5
5.1E Copy Name To Map.....	5-6
5.1F Edit Map Name.....	5-6
5.1G Delete Zone.....	5-7
5.1H Add Zone Into Map.....	5-7
5.2 Zone Edit Functions (Folder).....	5-8
5.2A Assign Tone to Zone.....	5-8

5.2B	Keyboard Mapping and Transposition.....	5-9
5.2C	Volume, Pan, and Output.....	5-10
5.2D	Zone Tuning Mode.....	5-10
5.2E	Zone Voice Allocation.....	5-11

Chapter 6 Preset Menu

6.0	About Presets.....	6-1
6.1	About Note Allocation.....	6-1
6.2	Preset Management Screens.....	6-2
6.2A	Preset Play Screen.....	6-2
6.2B	Create Preset.....	6-2
6.2C	Delete Preset.....	6-3
6.2D	Copy Preset.....	6-3
6.2E	Copy Name To Preset.....	6-4
6.2F	Edit Preset Name.....	6-4
6.2G	Preset Mode.....	6-5
6.3	Preset Modification Screens.....	6-6
6.3A	Assign Auxiliary Controllers.....	6-6
6.3B	Layer Detune.....	6-8
6.3C	Layer Detune Modulation.....	6-8
6.3D	Maximum Polyphony.....	6-9
6.3E	Allocation Priority.....	6-10
6.3F	Note Overflow.....	6-10
6.3G	Switch/Crossfade Controller.....	6-11
6.3H	Switch/Crossfade Pivot Point.....	6-12
6.3I	Layer Crossfade Region Width.....	6-13
6.3J	Select Maps.....	6-14
6.3K	Map Switch Control.....	6-15
6.3L	Map Switch Points.....	6-16
6.3M	Note Transpose.....	6-18
6.3N	Pitch Bend Range.....	6-19
6.3O	Output Assignment.....	6-19
6.3P	Keyboard Track Center.....	6-20
6.3Q	Pan Position.....	6-21
6.3R	Volume.....	6-21
6.3S	Filter Cutoff.....	6-22
6.3T	DCA Vel-Sens.....	6-22
6.3U	DCF Vel-Sens.....	6-23
6.3V	DCF Env Amount.....	6-23
6.3W	DCO Mod Range.....	6-24

Chapter 7 Global Menu

7.0 About the Global Menu.....	7-1
7.1 About Various MIDI Reception Modes	7-1
7.2 MIDI Screens.....	7-2
7.2A Omni/Poly Preset.....	7-2
7.2B Omni/Poly Channel & Volume.....	7-2
7.2C MIDI Receive Mode.....	7-2
7.2D MIDI Note Play Mode	7-3
7.3 Edit Multi Setup (Folder)	7-4
7.3A Copy Name To Multi.....	7-4
7.3B Edit Multi Name	7-4
7.3C Multi Enable.....	7-5
7.3D Multi Preset Assignment	7-6
7.3E Multi Volume.....	7-6
7.3F Multi Maximum Polyphony.....	7-6
7.3G Allocation Priority.....	7-7
7.3H Note Overflow	7-8
7.3I Store Multi.....	7-8
7.3J Recall Multi	7-9
7.3K Delete Multi	7-9
7.4 Preset Utility Screens.....	7-10
7.4A Preset Memory Status.....	7-10
7.4B Master Tuning	7-10
7.4C SCSI Device ID	7-11
7.4D Note Display Mode.....	7-11
7.4E Edit Selections	7-12
7.4F Clear All SP Memory.....	7-13

Chapter 8 Utility Menu

8.0 About the Utility Menu.....	8-1
8.1 About SMDI	8-1
8.2 Select Device/SX SCSI Number Screen	8-2
8.2A Select SMDI Slave Device/Set SX SCSI Number.....	8-2
8.3 DPM SX Configuration (Folder)	8-3
8.3A SX Firmware Version.....	8-3
8.3B SX Installed Memory	8-3
8.3C SX Sample Rate Selection.....	8-4
8.3D SX Sample Length Selection.....	8-4
8.4 Utility Sample Screens.....	8-5
8.4A Set SX Input Level.....	8-5
8.4B SX Sample Clip Count.....	8-6
8.4C Get Slave Sample	8-7
8.4D Send Wave.....	8-8

Chapter 9 SCSI and the SP

9.0 About SCSI	9-1
9.1 SCSI ID Numbers	9-1
9.2 Setting the SP SCSI ID	9-2
9.3 Initiators and Targets	9-2
9.4 Incompatible Hard Disk Formats	9-3
9.5 Incompatible SCSI Hard Disks	9-3
9.6 SCSI Connectors	9-3
9.7 SCSI Cabling	9-4
9.8 Termination	9-4
9.9 Typical SCSI Setups	9-5
9.10 Removing the SP's Terminators	9-7
9.11 Power-On Order	9-7
9.12 SCSI Troubleshooting	9-8
9.13 The Importance of Backup	9-8

Chapter 10 MIDI Supplement

10.0 MIDI Basics	10-1
10.1 MIDI Hardware	10-1
10.2 MIDI Message Basics	10-2
10.3 Channel Messages	10-3
10.3A Voice Messages	10-3
10.3B Mode Messages	10-5
10.3C Other Messages	10-5
10.4 System Common Messages	10-6
10.5 Books on MIDI	10-7

SP Tutorials Section

Introduction	
Tutorial 1: Playing Back Factory Disks/Setting Basic Parameters	1-5
Tutorial 2: Stereo Sampling with the DPM SX	6-10
Tutorial 3: Preset Edits	11-12
Tutorial 4: Tone Menu Edits	13-15

Chapter 1 SP Overview, User Interface, and Hookup

1.0 Basics

The SP is a high-fidelity, sample playback device. You can load sounds of superb quality (as developed by both Peavey and many third-party developers) into the SP, and trigger them with MIDI keyboard, bass, guitar, drum, or wind controllers, as well as sequencers. The wealth of features includes:

- **Expandable to 32 megabytes of memory.** The SP can be expanded using industry standard SIMM memory modules (units are shipped with 2 Megs; consult your Peavey dealer about additional memory installation). This allows for longer samples and less need for looping (which can create an overly-static sound), while providing both sonic and performance advantages over sample devices with less memory.
- **Four individual audio outputs.** These are useable as two stereo pairs, four individual outputs, or a stereo pair with two individual outputs.
- **Cascadable operation.** You can cascade as many SPs as desired to increase the amount of RAM, outputs, and voices. For example, two SPs can provide up to 64 Megs total RAM, eight individual outputs, and 32-voice polyphony.
- **MIDI interface and high-speed SCSI interface.** SCSI allows for transferring samples into the SP at 50 times the rate of the MIDI sample dump standard, as well as connection to hard drives, removable (cartridge) hard drives, CD-ROMs, computer-based sample editing programs, and the Peavey DPM SX sampling front end. Samples can also be transferred from any device supporting the MIDI sample dump standard, such as the award-winning DPM 3, or loaded from the internal disk drive.
- **Simple conversion to high-performance 16-bit sampling.** The SP is part of Peavey's modular synthesis/sampling system and was designed with modularity in mind. Adding the DPM SX, a 16-bit sample capture device, lets you record high-

fidelity sounds into the SX and transfer them over SCSI to the SP. Peavey offers several modules to trigger the SP: DPM C8 master keyboard, DPM 3 or DPM 2 synthesizer, or Peavey MIDI Bass.

- **High-density, 1.44 Megabyte disk drive.** High-density drives mean less disk swapping and fewer disks to store than traditional 720K drives, which offer half the storage.
- **Extensive editing features.** In addition to standard synthesizer features (envelopes, filtering, etc.), the SP's sophisticated modulation capabilities include bipolar and unipolar modulation, summed or scaled options for multiple modulation sources, and much more.
- **Upgrade paths.** Like other Peavey digital products, the SP can be updated simply and inexpensively. Peavey is always interested in your comments concerning future enhancements. Many updates are the direct result of user feedback.

1.1 About This Manual: Learning the SP

The SP's exceptional flexibility gives this instrument a future; you can develop a relationship with it, and discover new techniques in the months and years ahead. The SP is not a simple instrument—although it certainly can be treated that way if you just load in disks from the extensive SP sound library and play them back via a MIDI controller. To fully learn the SP will require some time, experimentation, and perhaps a little head-scratching. The reward will be mastery of an expressive and innovative musical instrument, but don't expect to do this over a weekend!

At this point no one really knows just how much the SP can do. New applications are being devised daily, which is great for SP owners, but makes it difficult for a manual to describe all aspects of SP operation. As a result, there are two manuals. The one you are holding is the reference manual, which describes all known features and functions. The second manual contains tutorials which take you through various operations, step-by-step. Doing these tutorials places the SP's functions in context.

To keep you up to date about the SP and other Peavey keyboard products, Peavey publishes *Key Issues*, a bimonthly publication

that features tips, applications articles, and news of third party developments. *Key Issues* is available from your local Peavey dealer and is highly recommended to get the most out of your investment.

Also make sure to send in your warranty card. Peavey's generous update policy will benefit you only if we know how to advise you of these updates.

1.2 SP Architecture

There are several menus of functions that you access to create a finished sound. Each menu has an associated front panel button and set of screens where you can perform various operations within that menu.

- The most basic level of sound is the *wave*, which consists of a raw sample. Processing options for its associated menu include looping, truncation, and similar sample-oriented operations.
- Waves can be processed by synthesizer-style processing options (envelopes, filtering, modulation, etc.) to create a *tone*.
- Tones are assigned to specific keyboard ranges, called *zones*, to create a *map*. This allows for multi-sampling and basic keyboard splits. Several map-related processing parameters, such as pan and volume adjustment, are also available.
- Finally, maps are used to create the two *layers* in a *preset*. A preset is the highest level of sound organization, and allows for such techniques as switching or crossfading between layers.
- In addition, a *multi* mode allows individual presets to be played over individual MIDI channels. Each preset in a multi responds to MIDI master volume commands (controller 7) and program changes. The SP stores 16 such multis, which are very useful in sequencing applications.
- The combination of all these elements creates a *bank*. When you save a bank to a storage device such as a hard disk, you store a "snapshot" of all DPM SP parameters.

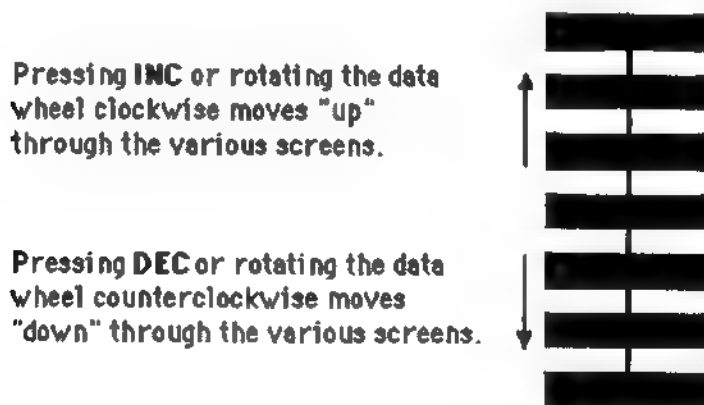
1.3 SP Operating System Basics

The SP operating system is logically laid out and easy to use.

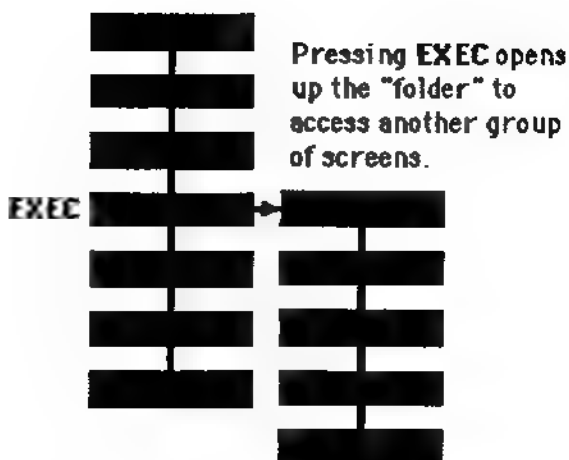
There are eight front panel menu buttons (PRESET, MAP, TONE, WAVE, DISK, GLOBAL, UTILITY, and EXEC). These select different editing *menus* except for EXEC, which is like the “enter” button on a computer. In some cases, an SP screen will ask you to press EXEC to complete or continue a particular operation.

Pressing a menu button such as MAP, TONE, etc. brings up one possible screen out of the many available screens in a menu. The *cursor*, a flashing block, will appear in the screen’s upper left hand corner. To call up different screens within the menu, the cursor must first be in the upper left hand corner (the “function select field”). Rotating the data wheel or pressing the INC/DEC buttons accesses other pages within the menu.

You can think of the menu screens as being arranged in a string, where pressing the INC button or rotating the data wheel clockwise takes you to the “higher” screen, and pressing the DEC button or rotating the data knob counterclockwise takes you to the “lower” screen:



In addition, some screens serve as “folders” that contain other screens. To open the folder, press EXEC (the screen will prompt you to do this). This opens up an additional string of screens, which you can also move through with the INC/DEC buttons or data knob:



1.4 Screen Layouts

We've already mentioned that the screen's upper left hand corner is the "sweet spot" for moving from one screen to another. Generally, where applicable, screens are laid out so that if you need to choose between different tones, zones, waves, maps, etc., these will be in the upper right hand corner. The lower line presents a screen's available options.

1.5 Navigating Within a Screen: The Arrow Buttons

The left and right arrow buttons select individual parameters within a screen. These buttons wrap around, so if you keep pressing one arrow button, you will eventually cycle through all available parameters.

Shortcut: If you press and hold the right or left arrow button and rotate the data knob, the knob controls cursor movement instead of the arrow button.

1.6 Changing Parameter Values

To change a parameter's value after selecting it, use the INC/DEC buttons or data knob to change the parameter's value. Press INC or DEC once to change the values one digit at a time; press and hold the button to scroll through values.

Shortcut: To accelerate the INC or DEC button scrolling speed, press the other button while holding the first button. *Example:* To speed up the INC button scrolling, while pressing and holding INC press the DEC button.

Several parameters are edited by entering numbers. If there is more than one digit, positioning the cursor under a digit changes that digit. This allows for large value changes without having to endlessly spin the data knob (or press the INC/DEC buttons until your fingers get short and stubby).

For example, suppose a parameter has a range of values from 0 to 255. Here's what happens with the cursor under specific digits.

Hundreds digit: Use the INC/DEC or data knob to change the parameter one hundred values at a time until it reaches the maximum value (e.g., 0, 100, 200, 255).

Tens digit: Use the INC/DEC or data knob to change the parameter ten values at a time until it reaches the maximum value (e.g., 0, 10, 20, 30, 40...255).

Ones digit: Use the INC/DEC or data knob to change the parameter one value at a time until it reaches the maximum value (e.g., 0, 1, 2, 3...255).

1.7 EXEC Button Details

On some screens, entering numbers requires pressing EXEC to complete a function. The artwork in this manual gives you a visual cue about when this is necessary by including a small line around the left and bottom of the screen:



```
lfo waveshape: t# 0
triangle wave
```

It is not necessary to press EXEC; the SP remembers these settings.



```
store multi to:
mu# 0 [MULTI NAME]
```

The outline indicates that it is necessary to press EXEC.

On wave menu screens where it is not necessary to press EXEC, pressing EXEC will trigger a note so you can do a quick audition of a sound. The sound's name flashes in the upper left corner.

1.8 User Interface Fine Points & Power Tips

- When you select a menu or folder, the last screen that was accessed in the menu or folder will appear. If no particular screen has been accessed since power-up, selecting a menu or folder selects the first menu or folder screen.
- When on a screen, you can return the cursor to the function select field in two ways: using the left/right arrow buttons, or by pressing the screen's menu button (e.g., if you're in a tone screen, press the TONE menu button).
- To exit a folder when the cursor is in the function select field, press the associated menu button. If the cursor is not in a screen's function select field, press the menu button twice to exit the folder. Or, use the arrow buttons to send the cursor to the function select field, then press the associated menu button to exit the folder.
- If you reach the end of a string of screens, trying to continue in the same direction will not take you to a new screen. In other words, if you reach the top screen, pressing INC will not wrap around to the bottom screen.
- Before leaving a screen, you can have the SP remember the cursor position on that screen by doing the following:
 - 1 Place the cursor under the desired parameter to be remembered.
 - 2 Press the menu button (*not* the arrow buttons) to send the cursor to the function select field, and select the next screen to edit with the data knob or INC/DEC buttons.
 - 3 When you want to return to the original screen, press the menu button to send the cursor to the function select field, but *hold the menu button down* as you use the data knob or INC/DEC buttons to select the original screen. Upon releasing the menu button, the cursor will jump to where it had been left on the screen.

As an example of how to use this, consider tweaking the attack times of two envelopes. By having the SP remember that the cursor should be on the attack parameter, it's easy to bounce back and forth between the two envelopes and adjust the attack times.

1.9 What's a Kiloword?

All memory readings (except for the available disk space screen) are given in kilowords. A word equals two bytes of memory. Thus, an SP with 2 Megabytes of RAM is equivalent to having 1 Megaword (1,024 kilowords) of storage.

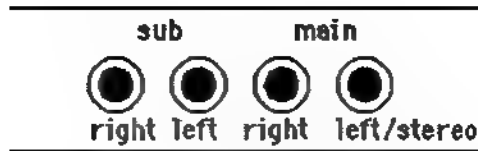
1.10 Conventions Used in this Manual

- Button names are indicated in **BOLD** and upper case letters.
- Each screen has its own heading. If the screen is a folder, then screens within the folder are given next, and their headings are preceded by a large dot (•) to remind you that they are part of a folder.

1.11 Basic Audio Hookup

The SP has two sets of audio outputs, Main and Sub. These can provide two stereo pairs, four individual outputs, or one stereo pair and two individual outputs. Please note that the front panel volume control affects the Main outs only (although the level of the signals feeding the Sub outs can be programmed within the SP). To hook up audio:

- 1 Unpack the SP. Save all packing materials in case the unit needs to be sent for updating or servicing.
- 2 Plug the line cord's female end into the SP's matching socket.
- 3 *With all devices in your system turned off and the volume controls turned down*, plug the audio outputs into an amplification system. For standard mono or stereo applications, use the main outputs as follows:



- **Mono:** Connect a mono cord from either the main right or left/stereo jack to a mono amplification system or individual mixer input.
- **Stereo (two cords):** Connect two mono cords from the main right or left/stereo jacks to a stereo amplification system or two mixer inputs. Pan the two channels hard left and hard right.
- **Stereo (single cord):** Connect a two-conductor stereo cord from the left/stereo jack to a stereo amplification system with a two-conductor stereo jack input. Pan the two channels hard left and hard right.
- **Stereo headphones:** Plug a set of stereo headphones into the left/stereo jack. With sensitive headphones, the volume should be sufficient for monitoring.

The two sub jacks can offer an additional stereo submix. In most applications involving both sets of outputs, you would send all four outputs to four channels of a mixer.

- 4 Turn on (in this order) the SP, any other MIDI gear or computers, mixer, and finally, the amplification system. Turn the mixer and amplifier volume controls up just a little bit; select normal volume levels once you're satisfied that the system is working properly. For lowest noise, it's a good idea to keep the SP's front panel volume control (which varies the overall level) up full.

This Page Left Intentionally Blank.

Chapter 2 Disk Menu

2.0 About Storage Devices

The memory in the SP is *volatile*, so anything stored in the SP disappears when the SP is turned off. It is therefore important to save SP banks to a non-volatile storage medium. Sound banks can also be loaded into the SP from various storage media.

The SP includes an internal high-density (HD) disk drive that stores up to 1.44 Megabytes of data. Since a bank often requires more memory than this, the SP can save a large bank to a set of multiple floppy disks.

Floppy disk drives are inherently slow. Therefore, the SP includes a SCSI port for hooking up fast mass storage devices such as hard disks. (For more information on SCSI, see Chapter 9.) These can hold hundreds of megabytes of data and transfer this data very rapidly to and from the SP.

Following is a summary of the types of storage devices useable with the SP. Prices are current “street prices” as of Fall 1991, but due to the volatile nature of the computer peripheral industry, prices fluctuate wildly (often downward) and these figures should be used only for relative comparisons.

- **High density floppy disk** *Maximum capacity:* 1.44 Megs. *Advantages:* low cost, ease of use. *Disadvantages:* Slow, limited storage capacity. *Cost:* Approximately \$0.50-\$1/megabyte.
- **Hard disk** *Maximum capacity:* 1.2 Gigabyte drives are commonly available as of this writing, but hard disk density continues to increase. 40 to 80 Meg hard drives are popular and economical. *Advantages:* Fast data transfer, large storage capacity. *Disadvantages:* High initial cost, failure causes large amounts of data to be lost. *Cost:* \$5/megabyte (80 Meg drive), \$2.50/Megabyte (1 Gigabyte drive).
- **Removable cartridge hard drive** *Maximum capacity:* 44 or 88 Megabytes, depending on type. *Advantages:* Unlimited storage capacity by buying additional cartridges, cartridges offer economical storage. *Disadvantages:* Initial cost, fragility of cartridges. *Cost:* \$500 for 44 Meg drive, \$1,500 for 88 Meg

then \$1.60/megabyte with 44 Meg cartridge, \$2/megabyte with 88 Meg cartridge.

- **Magneto-optical drive** *Maximum capacity:* 650 Megabytes typical. *Advantages:* Unlimited storage capacity by buying additional cartridges, very low storage cost. *Disadvantages:* Somewhat slower than hard drives. *Cost:* \$3,000 for drive then \$0.25/megabyte.
- **CD-ROM** *Maximum capacity:* 650 Megabytes. *Advantages:* Unlimited storage capacity by buying additional CD-ROMs. *Disadvantages:* Read-only operation, slow access time. *Cost:* \$500 for drive, CD-ROMs vary greatly in price from \$0.05 to \$1 (or more) per Megabyte, depending on the type of material included on the CD-ROM.

2.0A Floppy Disk Procedures

Like standard magnetic audio tape, floppy disks are relatively fragile since data is recorded as changes in the disk or tape's magnetic field and these fields are fairly weak. As a result, the SP verifies all saves to floppy disk.

- **Inserting disks** Insert a disk into the SP drive with the metal shutter going in first, label side up. Push the disk gently into the drive until it clicks in place, and the small eject button in the disk drive's lower right hand corner pops out.
- **Ejecting disks** To eject the disk, press firmly but gently on the eject button. The disk will come partway out of the drive; pull it out the rest of the way manually.
- **Write-protecting disks** Write-protecting a disk prevents recording data for as long as the disk is write-protected. Locate the small write-protect "slide switch" in the corner of the disk. Open it to write-protect, close to allow for recording.
- **Protecting the drive** During transport, insert a plastic or cardboard dummy disk (available at computer stores) into the SP disk drive. This prevents unwanted drive movement and protects from excessive vibration. Since disk drives are often considered the least reliable element in computer-based gear, avoid sharp blows, drops, or environmental extremes.
- **Only use High Density (as indicated with an HD logo) disks** Standard double density (2D) disks will not work reliably or properly with the SP. Use only high-quality HD disks.

2.0B Care and Handling of Disks and Removable Cartridges

Diskettes and removeable cartridges all use magnetic material, which can become accidentally demagnetized. Observe the following:

- Don't subject removeable media to stray magnetic fields. In particular, avoid speakers, televisions, transformers, and magnets.
- Don't subject removeable media to temperature extremes, especially heat.
- Removeable cartridges should never be bulk-erased since the cartridges contain special tracks used by the cartridge drive. Even "empty" cartridges have data and should be treated with care.

Other tips include:

- Protect all removeable media from dust.
- Label all removeable media. Office supply stores offer two different types of labels, permanent and removable. Use the removable types. Never write on removeable cartridge labels with pencil or any other writing implement that can "shed," as this material could get into the disk drive mechanism.
- If you have any doubts about the integrity of a floppy disk, reformat before saving anything to it. If the formatting procedure fails, consider using the disk as a coaster.
- Always insert a removeable cartridge as indicated in the drive's instructions. Improper insertion can damage the drive.

2.0C An Important Note About Banks!

A bank consists of all material—waves, tones, maps, presets, and multis—stored in the SP's memory.

If the bank can fit on a single floppy disk, it will be saved to that disk. If the bank takes up more memory than a single disk, it needs to be saved to multiple disks (the display will prompt you how to proceed).

A floppy disk (or set of disks) cannot contain more than one bank. Therefore, there is no "delete bank" command for floppies, since saving a bank to a floppy automatically overwrites the existing bank. Furthermore, when saving and loading particular types of data, it is not necessary to specify the bank to which data will be saved (or the bank from which data will be loaded) since the floppy can have only one bank.

SCSI devices can hold multiple banks. Therefore, when saving or loading data to SCSI devices, a screen usually lets you specify the bank to which data will be saved, or from which data will be loaded.

2.1 Main Disk Menu Screens

2.1A Bank Auto-Load from Floppy Disk

If a disk containing a complete bank is in the SP drive on power-up, the bank will be loaded automatically. (Auto-load will not occur with multiple disk sets; these must be loaded manually.) The display says:



```
The Peavey DPM-SP
accessing disk ....
```

After loading, the bank resides in SP memory.

2.1B Load Bank



```
load bank from:flopd
--<press exec>--
```

Loads a bank of sound data from a floppy disk, set of floppy disks, or SCSI storage device (e.g., hard drive or CD-ROM). Loading overwrites any bank that may already be in the SP. There is 1 parameter.

- **from** Selects the device (floppy disk or SCSI device 0 through 6) containing the bank to be loaded.

Floppy Disk Instructions

A floppy can only contain one bank, so if you press EXEC at this point, the bank will be loaded. The display says loading during this process.

If the bank was saved to multiple floppy disks, insert the first disk of the set then press EXEC (or DISK to quit). If the display says **not the disk i need!** then you have not inserted the first disk of the set, but a subsequent one; press the **DISK** button, insert disks from the set, and press EXEC until you find the right one. (Next time, remember to label the disks properly when you save to them—see section 2.0B).

After the first disk has loaded, insert the second disk of the set and press EXEC. If you insert a disk out of order during the loading process, the SP will display the **not the disk i need!** message. Keep inserting disks, and pressing EXEC, until you find the right one (as indicated by the SP loading data from the disk).

The display will continue to prompt you to insert the next disk of the series. Press the EXEC button to continue the loading process, or the DISK button to quit.

The display flashes **operation complete** when loading is complete, then reverts to the **load bank from** screen.

SCSI Storage Device Instructions

SCSI devices can contain multiple banks. Therefore, pressing EXEC will access a bank selection screen. There is 1 parameter.

```
load bank from:scsi2
# 0 [BANK NAME]
```

- # Selects the individual bank to be loaded from the SCSI storage device.

Press EXEC to load the bank. The display flashes **operation complete** when loading is complete, then reverts to the **load bank from** screen.

2.1C Edit Bank Name

```
edit our bank name:
[BANK NAME]
```

Allows bank naming or renaming, up to 14 characters. This operation does not alter the name stored on a disk, however the new name will be stored with the bank the next time it is saved to disk. Naming banks is especially important with SCSI disks since this allows you to distinctly identify each of the many banks which can be stored on a single SCSI disk, but naming is also very good practice with floppies.

There are two screens. The first screen has 1 field.

- **(Bank name)** Shows the current bank name.

To access the second screen, press EXEC.

```
<exec to save edit>
      [BANK NAME]
```

The second screen has 1 field (for bank naming).

- **(Bank name)** Use the cursor buttons to choose the character to be edited, and the data knob to select the character. After naming, press EXEC to save the edits.

Turning the data knob fully counterclockwise selects a blank space. Turning the data knob clockwise (or using the INC button) from this blank space selects characters in the following order:

lower case letters	a-z
upper case letters	A-Z
symbols	` \$ % @ ^ _ + * / = []
	< > () " # & () : ? ! ' , . -
numbers	0-9

2.1D Save Bank to Disk

```
save bank to: scsi3
      <press exec>
```

Saves the current SP sound bank to a floppy disk, set of floppy disks, or erasable SCSI storage device (e.g., hard disk). There is 1 parameter.

- **to** Selects the device (floppy disk or SCSI device 0 through 6) to which the bank will be saved.

Floppy Disk Instructions

A floppy disk contains only one bank, so if you press EXEC at this point, the bank will be saved to the floppy. However, a bank that exceeds approximately 1.4 Megabytes will need to be saved to a

multiple disk set. The display will prompt you as follows:

- 1 After pressing EXEC, you'll see how many disks are needed to save the bank and be prompted to insert the first disk.
- 2 If more than one disk is required, the display will prompt you to insert the first disk.
- 3 Press the EXEC button to begin the saving process, or the DISK button to quit.
- 4 If additional disks are required for saving the bank, the display will prompt you when to insert them. Press the EXEC button to continue the saving process.
- 5 The display will flash **operation complete** when saving is complete, then revert to the **save bank to:** screen.

Disk labelling Label each disk of the set (1st, 2nd, 3rd, etc.) so you can find the correct disk to insert during the loading process.

Resaving to a multiple disk set When resaving modified data to a multiple disk set, the SP does not care which disk is inserted at which time since all data on the disk—including the disk number—is overwritten during the saving process.

Format while saving During a save operation, the SP will automatically format (also called initialize) a floppy disk if it is brand new and unformatted. Therefore, it is not necessary to go through a separate formatting procedure with a new disk prior to saving. Note that the SP will not automatically reformat an already formatted disk except in the unlikely case that the format has been so damaged that the disk appears to be unformatted.

Normally, once a floppy disk has been formatted for use with the SP, it can be saved to repeatedly without reformatting. However, if you encounter disk problems (such as an error while saving or during the verify process), reformatting is indicated and this must be done separately using the initialize (format) disk function (section 2.2D).

Since the SP formats while saving, it may not seem useful to have a separate initialization command. However, formatting takes time. It's advisable to have some pre-formatted disks so that you don't have to wait around any more than is necessary during the saving process.

SCSI Storage Device Instructions

SCSI devices can contain multiple banks. Therefore, pressing EXEC will access a bank select screen. There is 1 parameter.

```
save bank to: scsi3
# 1 [BANK NAME]
```

- # Selects the bank number under which the bank will be saved. You will see one of two things to the right of the bank number:

A series of dashes indicates that there is no bank already stored under that number.

A bank name shows that there is already a bank stored under that number. *Caution: This bank will be overwritten if you save.* A small lock symbol appearing to the left of a bank name indicates that the bank is write-protected and cannot be overwritten (unless you unlock it, as described in section 2.2B).

Press EXEC to save the bank. The display says **saving** during this process.

2.1E Add Wave, Tone, Map, Preset, or Multi to Memory

```
add ..... from:scsi2
one wave
```

Adds to SP memory an individual wave, tone, map, preset, or multi; all waves, all tones, all maps, all presets, all multis; or everything in a bank from a floppy disk, set of floppy disks, or SCSI storage device such as a hard drive or CD-ROM. This function is useful for merging additional data to what is already in SP memory.

Note that maps, presets, and multis will only load into empty locations and not overwrite existing maps, presets, and multis. However, waves and tones can overwrite existing waves and tones.

There are several screens. The first screen has 2 parameters.

- **from** Selects the device (floppy disk or SCSI device 0 through 6) containing the data to be loaded.
- **data type** Select the data type—an individual wave, tone, map, preset, or multi; all waves, all tones, all maps, all presets, all multis; or everything in a bank.

Press EXEC. For SCSI devices, the SP reads the directory of available banks. There is 1 parameter. (With floppy disks, which only have one bank, pressing EXEC skips the following screen and goes directly to the next one.)

```
add wave - from bank :  
# 0 [BANK NAME]
```

- **#** Selects the bank containing the data to be loaded (this display shows that a single wave will be loaded).

Press EXEC. If all of a data type has been selected (e.g., all tones), they will be loaded into empty memory slots starting with the lowest-numbered empty slot and moving up from there. The operation is then complete.

If one of a data type has been selected (e.g., one map), the SP reads the directory of that data type. There is 1 parameter.

```
add wave - from b# 0  
# 0 [DATA NAME]
```

- **#** Selects the data to be added. In this example, a single wave is being loaded from bank 0. The bank number in the upper right corner is for reference only and cannot be changed once you reach this screen.

Press EXEC. A screen now asks you where to add the data. There is 1 parameter.

```
add wave - into loc :  
# 0 [DATA TYPE]
```

- # Selects the location where the data is to be added. In this example, a single wave is being loaded into SP memory wave location 0.

Press EXEC. The adding operation is now complete if you selected a wave. Otherwise, a screen will appear that asks if you want to add linked data. There is 1 parameter, **yes** or **no**. The following example shows the screen that appears if you selected a preset.

```
also include linked  
maps/tones/waves? yes
```

If you select **yes** with the INC/DEC buttons or data knob, all linked maps, tones, and waves associated with the preset will also be loaded. With a map, linked tones and waves can be loaded; with a tone, linked waves can be loaded. This is an "intelligent" function since it will not load maps, tones, and waves that are already in memory. Press EXEC to complete the operation. If you select **no**, then the preset is loaded as a "shell" without the associated maps, tones, and waves.

2.1F Get Info

```
get info about:scsi5
--<press exec>--
```

Displays information (device name and remaining memory) about a particular storage device. The first screen has 1 parameter.

- **about** Selects the device (floppy disk or SCSI device 0 through 6) to be examined.

To access the second screen, press EXEC. For floppy disks, the display shows:

```
bank: [BANK NAME]
disk number: 1 of 2
```

This screen shows the floppy disk bank name on the upper line. Use the INC/DEC buttons or data knob to select one of two options in the lower line:

- **Disk number** Shows the position of the disk in its disk set. Use this to sort out those unlabelled disks! If the disk is the first disk in the bank set, then the total number of disks in the bank will also be shown. Otherwise, ?? will appear.
- **Wavefile** Amount of memory used by the waves on the floppy. This is shown only if the disk is the first in its bank set. Otherwise, several question marks appear.

For SCSI devices, the display shows:

```
scsi5: [DEVICE NAME]
avail spc: 42624kb
```

This screen shows the SCSI device being examined and device name on the upper line. Use the INC/DEC buttons or data knob to select one of several options in the lower line:

- **Avail spc** Available free memory space in kilobytes.
- **Total spc** The drive's total memory capacity in kilobytes.

- **Blk size** This is for diagnostic purposes, and shows the size of the blocks created on a SCSI storage device. This will usually be 128K but this figure can change with very large hard drives. The SP assumes a maximum of 512 banks of 128K blocks each; if this is not sufficient to use up the entire capacity of the hard drive, the SP will use a larger block size.
- **Bank info** Press EXEC; the display shows:

```
scsi2: --- bank b# 0
disk space: --- 128kb
```

This screen shows the device being examined and bank number on the upper line. Selecting a new bank flashes the bank name on the upper line, as does pressing the EXEC button. Use the INC/DEC buttons or data knob to select one of several options in the lower line:

- **Disk space** How much disk space is occupied by that bank.
- **Wavefile** The memory used by the waves in that bank.
- **Bank stat** Shows whether the bank is locked or unlocked.

2.2 Disk Utilities (Folder)

```
disk utilities menu  
-----<press exec>-----
```

Press EXEC to access the pages in the "folder."

2.2A • Delete Bank on Disk (SCSI Devices Only)

```
delete bank on:scsi2  
-----<press exec>-----
```

Deletes a bank from an eraseable SCSI storage device (e.g., hard drive). With floppies, saving a bank automatically deletes the existing bank on disk.

There are two screens. The first screen has 1 parameter.

- **on** Selects the SCSI device (0 through 6) containing the bank to be deleted.

To access the second screen, press EXEC. The SP reads the directory of available banks. There is 1 parameter.

```
delete bank on:scsi4  
# 21 - [BANK NAME] -----
```

- **#** Selects the bank to be deleted. If there is a lock symbol to the left of the bank name, the bank cannot be deleted unless you unlock it (see next section).

Caution!!! A Bank cannot be retrieved once it has been deleted.
Press EXEC to delete the bank.

2.2B • (Un)lock Bank (SCSI Devices Only)

```
(un)lock bank: scsi2  
-- <press exec> --
```

Unlocks a locked (write-protected) bank, or locks an unlocked bank. There are two screens. The first screen has 1 parameter.

- **bank** Selects the SCSI device (0 through 6) containing the bank to be locked or unlocked.

To access the second screen, press EXEC. The SP reads the directory of available banks. There is 1 parameter.

```
(un)lock bank: scsi2  
# 0 [BANK NAME]
```

- **#** Selects the bank to be locked (as shown by a lock symbol to the left of the bank name, as shown) or unlocked (no lock symbol present).

Press EXEC to change the status of the selected bank—locked to unlocked, or unlocked to locked.

2.2C • Edit Volume Name (SCSI Devices Only)

```
edit vol name: scsi2  
-- <press exec> --
```

Allows naming or renaming (up to 14 characters) of an erasable SCSI device volume. There are two screens. The first screen has 1 parameter.

- **vol name** Selects the volume (SCSI device) to be named or renamed.

To access the second screen, press EXEC. The second screen has 1 parameter.

```
<exec to save edit>  
[VOLUME NAME]
```

- **(Volume name)** Use the cursor buttons to choose the character to be edited and the data knob to select the character. After naming, press EXEC to save the edits.

Turning the data knob fully counterclockwise selects a blank space. Turning the data knob clockwise (or using the INC button) from the blank space selects characters in the following order:

lower case letters	a-z
upper case letters	A-Z
symbols	` \$ % @ ^ _ + * / = []
	< > { } " # & () ; ? ! ' , . -
numbers	0-9

2.2D • Initialize Device for SP Sound Storage

```
initialize: scsi2  
for SP sound storage
```

This function formats a floppy disk or eraseable SCSI device for SP storage.

Caution: Initializing erases any banks on the selected device and they cannot be recovered! Be very careful when you use this function.

Extra caution: If the SP is connected to a SCSI bus, this command will wipe out whichever drive you specify—so don't specify, for example, the internal hard disk on a Macintosh by mistake!! If you're at all uncertain about what you're doing, with all power off disconnect the SP and its hard disk from the SCSI bus before doing a disk initialization.

With floppy disks, formatting is usually necessary only with new disks since once a floppy disk has been formatted for use with the SP, it can be saved to repeatedly without reformatting. However, if you encounter disk problems (such as an error while saving or during the verify process), reformatting is indicated. There is 1 parameter.

- **initialize** Selects the eraseable SCSI storage device or floppy disk to be initialized.

Press EXEC. Because this is a pretty drastic operation—especially where high-capacity SCSI devices are involved, since you could wipe out the entire hard disk—there are two warning screens. The first one alternates between four displays:

- this will wipe out everything on disk!
- are you fairly sure you want to do this?
- press <exec> button to continue
- press <disk> button to quit

Press EXEC, and the second warning screen alternates between:

- disk initialization cannot be undone!
- last chance for you to reconsider!
- press <exec> button to continue
- press <disk> button to quit

Press EXEC, and the initialization operation begins. Those of you who like to live dangerously can bypass the warning screens by simply pressing EXEC three times after the initialize screen appears.

Chapter 3 Wave Menu

3.0 About Waves

When a sample has been loaded into the SP, either from MIDI, SCSI device, or floppy disk, it is called a *wave*. The wave menu provides several wave-related functions:

- **Wave Information** Shows the number of available waves in memory, how much memory they take up, and various wave characteristics (mono or stereo, forward looped or not, wave length in samples, wave size in words, and wave sample rate).
- **Wave Management** Delete, copy, name, or rename waves.
- **Wave Editing** Trim unused portion of a wave to save memory, merge two mono samples into stereo, set wave pitch, looping, start and end points, etc. You can also determine how waves will be auditioned, and whether the loop is on or not.
- **Instant Audition** On wave edit screens where it is not necessary to press EXEC to complete a function, pressing EXEC triggers a note so you can do a quick audition of a wave. The wave name flashes in the upper left corner.

3.1 Stereo Sampling Basics

For stereo sampling, use samples with the same sample rate (Peavey recommends 44.1 kHz) to preserve proper pitch when played in stereo.

Stereo samples come in two flavors: two mono samples that can be loaded in individually and merged into a stereo sample within the SP, or a stereo sample that can be imported from a sample-editing program such as Sound Designer. These types of stereo samples are interleaved, so as far as the computer or SP is concerned, the sample is a single stereo sample instead of two mono samples.

The following describes the general approach for converting two mono samples into a stereo sample:

- 1 Load each side of the stereo wave into the SP. This puts two waves in memory.
- 2 Set up the dual wave audition parameters to hear two waves at once. Generally, *mix* is the preferred audition option since any timing differences between the two waves will sound like flanging or chorusing, making it easy to hear if they don't start at the same time.
- 3 Use the various edit options to tweak the left channel wave for the optimum start and end points, and looping.
- 4 Optimize the other sample's start point (looping and end point is always the same as for the other wave of the stereo pair).
- 5 Once both samples are optimized, invoke the stereo merge option to convert the two optimized mono samples into a single stereo sample.

Stereo sampling is a fairly "deep" topic. For more information, please see the Tutorials.

3.2 Wave Management Screens

3.2A Wave Play Screen



The wave menu's first screen is a "play screen" that shows the wave name and number on the top line, and monitors the number of voices being played on the lower line. Each note represents a voice that's in use; each dot represents a voice that's available for allocation.

3.2B Receive Sample Via MIDI

Whenever a MIDI Sample Dump Standard (SDS)-compatible signal starts and appears at the SP's MIDI in, the SP automatically receives the sample. This sample could come from a Peavey DPM 3 SE or other SDS-compatible synthesizer, DPM SX Sample Expander, or computer-based sample editor. *It is not necessary to select any particular screen on the SP, as long as MIDI is active.*

As the SP receives the sample, the screen shows:



The flashing numbers show a countdown of how many words are left before loading is complete. Pressing EXEC during the transfer cancels it, and insures the sample will not show up in the SP.

After loading, the display returns to whatever screen was selected prior to sample reception.

Samples received by the SP will retain assigned sample numbers.

Caution: If a wave in SP memory has the same sample number as the sample being received, the wave in SP memory will be overwritten. You will not be warned about this; to avoid conflicts with incoming samples, use the Wave Info screen to see which sample numbers are in use.

3.2C Wave Memory Status

```
wave memory status:
space used: --- 932k
```

Shows the status of the SP's internal memory. There is 1 parameter, located on the lower line.

- **memory status** Selects one of the following:

space used	The amount of memory used by all waves.
space avail	The amount of free memory.
num of waves	The number of waves in memory.

3.2D Wave Info



wave info: w# 7
type: mono

Reads out information about a selected wave (the parameters cannot be edited and are for reference only). There are 2 parameters.

- **w# (wave number)** Selects the wave to be examined.
- **info select (lower left)** Selects the type of wave info for display:

type	Mono or stereo.
loop	Forward or off.
memloc	Memory location, for diagnostic purposes.
size	Wave size, in kwords.
length	Wave length, in samples.
start	Wave start point, expressed as samples into the wave
end	Wave end point, expressed as samples into the wave
lp sta	Loop start point, expressed as samples into the wave
lp end	Loop end point, expressed as samples into the wave
lp fr	Loop fraction end point, expressed in hex. If this number is anything other than 0, it means that a fractional loop is in use (section 3.3H).
chksum	Checksum, for diagnostic purposes.
period	Period of the wave, in nanoseconds.
s rate	Wave sample rate, in kHz.
normlz	Shows the sample rate offset compared to 44.1 kHz.
pitch	Shows the pitch of the wave as a MIDI note number.
fine t	Shows whether any fine tuning, in cents, has been applied to the wave.

3.2E Delete Wave

```
delete wave -- w#125
-- [WAVE NAME] --
```

Deletes a single wave, or all waves. There is 1 parameter.

- **w#** (wave number) Selects the wave to be deleted (the lower line shows the wave name). Going beyond the highest-numbered wave selects All Waves for deletion.

Press EXEC to delete the selected wave. The display shows operation complete to confirm.

3.2F Copy Wave

```
copy wave -- w# 99
to w#100 use:all
```

Copies one wave to any empty wave. There are 3 parameters.

- **w#** (wave number) Selects the wave to be copied.
- **to w#** (empty wave number). Selects the empty wave to hold the copy. If a wave number cannot be selected, it is not empty.
- **use** There are two options:

all	Copies the entire wave.
st>end	Copies only that portion of the wave between the wave start and end points (as shown in the Wave Info and Wave Start/End screens). This is useful if you have a long sample and want to make several smaller samples out of it.

Press EXEC to copy the selected wave. The display shows operation complete to confirm.

3.2G Copy Name to Wave

```
COPY name ---> w#117  
w# 0 [WAVE NAME]
```

Copies a wave, tone, map, or preset name to a wave. This saves time if a suitable wave name already exists, since you don't need to enter the wave name from scratch. There are 3 parameters.

- **w#** (wave number) Selects the wave to which the name will be copied.
- **w# t# m# p#** (wave, tone, map, or preset) Selects whether the name to be copied is a wave, tone, map, or preset name.
- **name** Selects the actual wave, tone, map, or preset name.

Press EXEC to copy the selected name to the wave. The display shows operation complete.

3.2H Edit Wave Name

```
edit wave name w# 77  
--- [WAVE NAME] ---
```

Allows wave naming or renaming, up to 14 characters. There are two screens. The first screen has 1 parameter (for wave selection):

- **w#** (wave number) Selects the wave to be named/renamed.

To access the second screen, press EXEC. The second screen has 1 parameter (for wave naming).

```
<exec to save edit>  
--- [WAVE NAME] ---
```

- **(wave name)** Use the cursor buttons to choose the character to be edited and the data knob to select the character. Remember that you can hold the INC or DEC button down and use the data knob to select characters quickly. After naming, press EXEC to save the edits.

Turning the data knob fully counterclockwise selects a blank space. Turning the data knob clockwise (or using the INC button) starting from the blank space selects characters in the following order:

lower case letters	a-z
upper case letters	A-Z
symbols	` \$ % @ ^ _ + * / = [] < > { } " # & () : ? ! ' , . -
numbers	0-9

3.3 Wave Edit Functions (Folder)



Press EXEC to access the pages in the “folder.”

3.3A • Wavepoint Audition



This is the “control panel” for setting loop point and start/end point audition characteristics while on screens within this folder. The purpose of this function is to continuously trigger a sample to make it easier to hear the effects of adjusting a parameter. There are 3 parameters.

- **on/off** When on, triggering a sample with the EXEC button plays the wave back at the original pitch. As soon as you vary a parameter on the Wave Loop Point, Wave Start/End, or Loop Fraction screens, the note will sustain, and you can hear the effect of changing the selected parameter. On is generally the most common setting for quick loop editing.

Moving the cursor off the current parameter (or pressing the WAVE menu button) cancels the sustain but as always, notes can be triggered via MIDI.

When **off**, neither edits nor the **EXEC** button will trigger sounds. Sounds will be triggered by MIDI.

- **vol** Selects a volume level between **00** and **99**. If the audition parameter is **on**, varying the **vol** parameter plays back the sample at the displayed volume.
- **lead-in** If the audition parameter is **on**, the sample will play back a certain percentage of the way into the total sample, as specified by the “lead-in” (e.g., **20** means the audition will start **20%** back from the parameter being edited—sample loop end point, loop start point, or sample end). In other words, this parameter “leads in” to the edit point. If the audition parameter is **on**, varying the lead-in parameter plays back the sample with the specified amount of lead-in.

Examples: The default is **10%**. This is a good lead-in value for editing the sample end point since the audition starts **10%** of the sample length before the end point. Therefore, you don’t have to listen to the entire sample just to edit the end. If the end point is the same as the loop end, it will not be possible to change the end point; turn off the loop (section 3.3G) before proceeding. Note that playback stops when it gets to the end of the wave so you can hear where the end is.

When editing a loop, especially a short one, set the lead-in to **0** to hear the loop by itself. As you edit, the loop will continue to play.

When editing the loop end point of a long loop, to avoid hearing the full loop set a lead-in around **20%** or so. This will lead-in to the wave **20%** before the loop end.



With leadin set at 20% (dotted line) before the wave end, you will hear the end of the wave without having to listen through the beginning. This allows for faster editing.

3.3B • Wave Start/End

```
wave start/end w# 5
-- -- 0 <z> -- 31802
```

Sets the wave start and end points. Increasing the wave start value cuts off more of the wave's attack. Decreasing the wave end value shortens the length of the wave being played (but does not remove any of the wave; this requires using the Trim Excess Memory function, described in the next section). There are 4 parameters.

- **w# (wave number)** Selects the wave to be edited.
- **wave start (lower left)** Selects how many samples into the wave playback will begin.
- **<z/m>** Selects between:
 - (z)ero crossing search**
To avoid popping sounds, the SP will place start and end points only on zero crossings.
 - (m)anual direct adjust**
You can enter start and end points anywhere in the wave, not just on zero crossings.
- **wave end** Selects the point where wave playback ends.

When the cursor is in the top line, triggering a wave with the EXEC button plays it back normally. When the cursor is in the bottom line, triggering a wave with the EXEC button plays back according to the option selected on the Wavepoint Audition page.

Note: It is important that a wave start and end on a zero crossing (i.e., where the wave has no amplitude) to prevent clicks. If you adjust the start and/or end point using the manual mode, do at least one press of the INC or DEC button with <z/m> set for zero-crossing mode for both the start and end. This will make sure that the start and end points are forced to occur on a zero crossing.



In this example, the start point is set further into the wave to cut off the attack, and the end of playback occurs before the actual wave end.

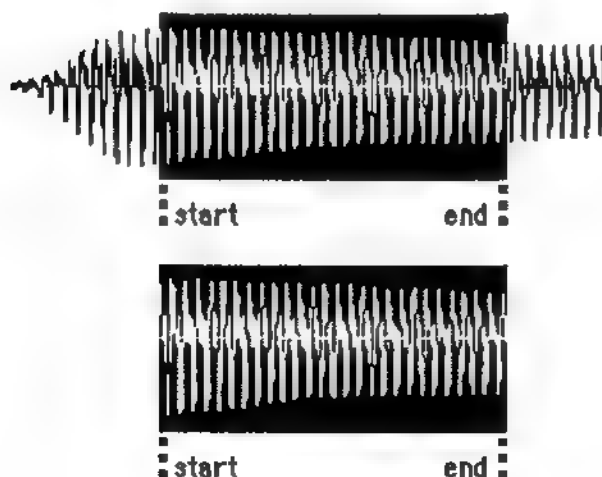
3.3C • Trim Excess Memory

```
trim xs mem: w# 5
      [WAVE NAME]
```

This erases the wave before the start point and after the end point to reclaim wave memory. There is 1 parameter.

- **w# (wave number)** Selects the wave to be trimmed. Going beyond the highest-numbered wave selects All Waves for trimming.

After selecting the wave to be trimmed, press EXEC. Any loops will retain their settings.



A wave before trimming (above) and after (below)

3.3D • Dual Wave Audition

```
dual wave aud: / mix  
lft:w# 62 rgt:w# 31
```

This function is an editing tool for the stereo merge function, not a program editing or MIDI response option. Its goal is to make it easy to match the right wave's start point to the left wave, which is particularly important if a stereo sample is imported as two mono samples rather than an interleaved stereo sample (as generated by sample-editing programs such as Alchemy or Sound Designer). Once the right wave's start point is optimized, the pair can be merged into a single, interleaved stereo sample.

With stereo merging, the left wave is the master wave; its sample rate is used for both waves, which is why both samples used in a stereo merge should have been sampled at the same rate. If the right wave changes pitch when auditioned with the Dual Wave Audition, then it was not sampled at the same rate as the left wave.

This screen determines how you will audition the waves during the stereo merging process. There are 3 parameters.

- **off/mix/pan** There are three options:

off	Disables dual audition mode. You will hear only one wave at a time in wave start/end and loop screens.
mix	Blends together two waves. With Mix selected, there are 2 parameters on the lower line that select the two wave numbers to be mixed.
pan	Places one wave left in the stereo field, and the other right. With Pan selected, there are 2 parameters on the lower line that select the two wave numbers to be panned.

If Mix or Pan are on, the two selected waveforms will be the only waves that can be edited on the wave start/end screen.

- **lft:w# (wave number)** This is editable only with Mix or Pan enabled, and selects the left channel wave.
- **rgt:w# (wave number)** This is editable only with Mix or Pan enabled, and selects the right channel wave.

3.3E • Stereo Merge

```
stereo merge: ~ ~ ~ ~  
lft:w# 62 ~ rgt:w# 31
```

After adjusting the parameters of the individual mono waves to be turned into a stereo wave, use this screen to fuse the mono waves into stereo. The merging process forces the right wave to adopt all left wave parameters except for start point—loop points are acquired from the left wave, as is the sample length. If the right wave is longer than the left wave, merging automatically trims the right wave so it has the same length as the left wave. There are 2 parameters.

Stereo sampling is a fairly complex topic. For more information, please see the Tutorials.

- **lft:w#** (wave number) Selects the left channel wave.
- **rgt:w#** (wave number) Selects the right channel wave.

After selecting the waves to be merged, press **EXEC** to complete the operation. The right wave disappears, and the left wave becomes a single stereo wave which can be used in tones and loaded from/saved to disk, just like any normal wave. Merging also resets the Dual Wave Audition screen to **off**.

Caution: If the right wave is shorter than the left wave, you will get an error message. Swap waves so that the left wave becomes the right wave, and vice-versa.

If an error message reads “not enough free memory,” save the waves to disk then re-load them in the opposite order. In other words, if the right wave was loaded in first, load in the left wave first then load in the right wave. This should push around enough memory to free up sufficient space to do the stereo merge.

3.3F • Wave Loop Points

```
wave loop pts: w# 5
               8358 <z> 31802
```

Sets the wave's loop start and end points. There are 4 parameters.

- **w#** (wave number) Selects the wave whose loop points are to be edited.
- **loop start** Selects how many samples into the wave the loop will begin.
- **<z/m>** Selects between:

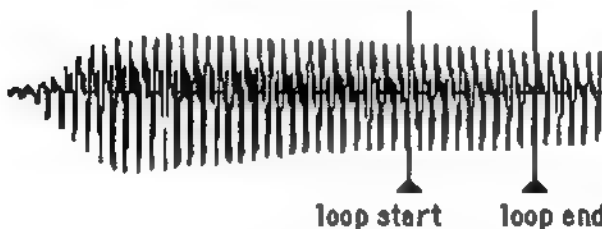
(z)ero crossing search

To minimize clicks when looping, the SP will place loop start and end points only on zero crossings.

(m)anual direct adjust

You can enter loop points anywhere in the sample, not just on zero crossings.

- **loop end** Selects the point where the loop jumps back to the beginning.



The two markers show the loop start and end points. When playback reaches the loop end point, it jumps back to the loop start and continues playback until it reaches the loop end, whereupon it jumps back to the loop start. This looping process repeats for as long as you keep your finger on the key.

3.3G • Wave Loop Enable

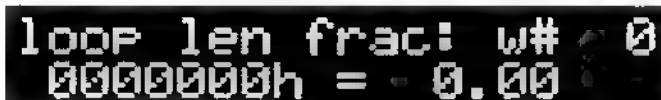


wave loop: on! w# 5

Determines whether the wave will loop, or simply play from the beginning to the end and stop. There are 2 parameters.

- **w# (wave number)** Selects the wave whose loop is to be turned on or off.
- **loop mode** Selects whether the loop is on or off.

3.3H • Loop Length Fraction



loop len frac: w# 0
00000000h = 0.00

Serves as a "fine tuning" control for the loop length, which allows for extremely precise loop tuning when using short length loops (the effect of changing this parameter will not be apparent with long loops). There are 2 parameters.

- **w# (wave number)** Selects the wave whose loop is to be fine tuned.
- **loop fraction** Shows the amount of change as a hexadecimal number. Don't let the hexadecimal numbering frighten you, since you'll probably end up adjusting the loop fraction by ear anyway for the best-sounding loop.

3.3I • Original Pitch



orig pitch: w# 34
60.00 (C 4)

This parameter "informs" the SP of the note's original (or root) pitch, expressed as a MIDI note number (the two digits to the left of the decimal place) and note name (in the example above, C4). This information is used at the map and preset levels to

determine what note will play the wave back at its original pitch when mapped. You will not hear any changes caused by adjusting this parameter when editing, since the sound is not being mapped; you need to be listening to a map or preset.

This parameter also “informs” the SP whether to offset the tuning by a certain number of cents (1/100ths of a semitone—the two digits to the right of the decimal place). Although fine tuning can be set on the next screen, this screen includes fine tuning for non-diatonic sounds. Thus, the primary pitch could be offset on this page; fine tuning can be used to create chorusing, flanging, and other special effects.

- **w#** (wave number, upper right) Selects the wave to be fine tuned.
- **detuning** (lower middle) Selects the amount of detuning, from 0 to 127 semitones and +99 to -99 cents.

3.3J • Wave Finetune



The screenshot shows a black rectangular area with white text. The top line reads "wave finetune: w# 34" and the bottom line reads "0 cents". There are small white arrows on the left and right sides of the bottom line, indicating navigation controls.

There may be cases where you want to drop a new sample into a multisample, and find that it is slightly out of tune with respect to the other samples. Wave finetune allows for changing the tuning. Any changes made to the pitch on this page are not transferred over SCSI, but pitch changes on the previous (Wave Pitch) screen are.

The wave's tuning can be changed in cents (1/100ths of a semitone). There are 2 parameters.

- **w#** (wave number, upper right) Selects the wave to be fine tuned.
- **detuning** (lower middle) Selects the amount of detuning, from +99 to -99 cents.

Chapter 4 Tone Menu

4.0 About Tones

A tone is a processed wave. The tone menu contains several synthesizer-like processing options such as response to keyboard dynamics, filtering, LFO modulation (vibrato, tremolo), etc. These turn the raw wave into a more musically-useful sound.

The tone menu also includes tone management functions such as delete, copy, name or rename tones, and the ability to assign a particular wave to a tone.

Note that on all screens that do not require pressing the EXEC button (i.e., those that perform voice editing functions), pressing EXEC shows the tone name in the upper line.

4.0A The SP's "Synthesizer" Functions

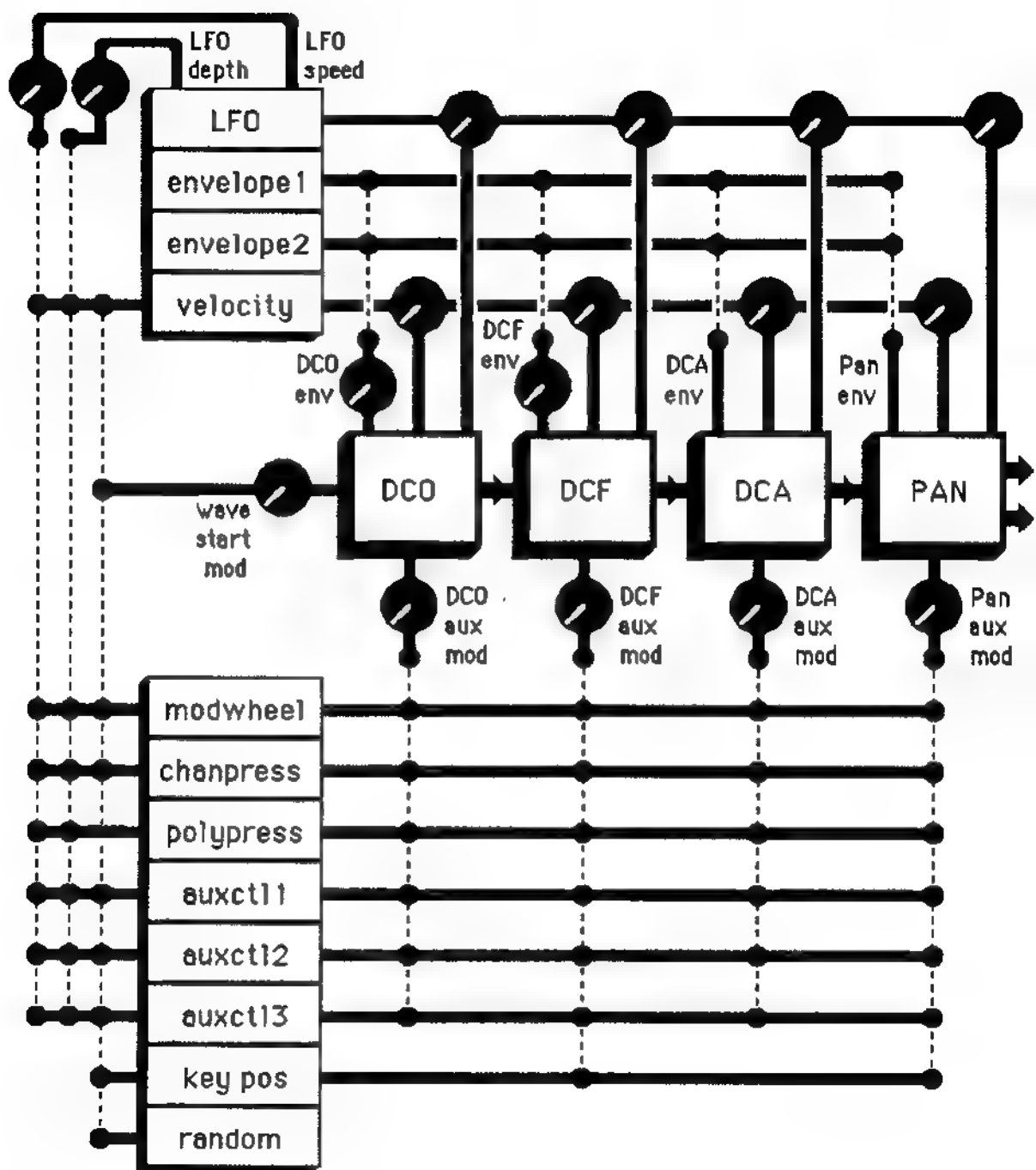
The diagram on the following page shows the signal flow for an SP tone, and the available "synthesizer" processing options.

The "knobs" indicate places in the signal flow where levels or amounts can be adjusted.

The dashed lines indicate that a function can be switched among various modulation sources (velocity, chanpress, keypos, etc.).
Examples: The DCO aux mod level can switch between modwheel, channel pressure, poly pressure, auxctl1 (auxiliary controller 1), auxctl2, or auxctl3; the DCO Env can switch between envelope 1 and envelope 2. The DCF aux mod can choose among the same options as the DCO, but can also use key position as a modulation source. Wave start modulation can use velocity, modwheel, chanpress, auxctl1, auxctl2, auxctl3, key position, or random.

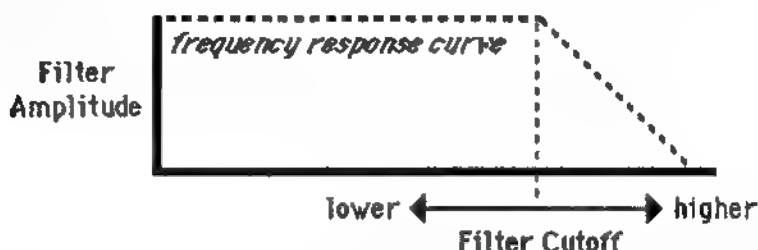
4.0B DCO (Digitally Controlled Oscillator)

The digital oscillator plays back a selected wave, chosen from any of the waves in memory.



4.0C DCF (Digitally Controlled Filter)

The DCF varies a signal's harmonic content, and therefore overall timbre, by attenuating frequencies above a specified *cutoff frequency*. Higher cutoff frequencies give a brighter sound; lower cutoff frequencies give a bassier sound since fewer harmonics are present.



Although static filter settings are useful, varying the filter cutoff dynamically over time can produce more interesting effects. Modulating with velocity produces brighter sounds with louder dynamics, which resembles the way some acoustic instruments respond. Modulating with an envelope creates a predictable change in harmonic structure, such as the increase in harmonics that occurs during the attack of a wind or brass instrument.

4.0D DCA (Digitally Controlled Amplifier)

The DCA (Digitally-Controlled Amplifier) controls the overall dynamics of the oscillator/filter combination.

Note that filter and DCA settings can interact. For example, if the filter cutoff is lower than the frequency of the note, very little signal will get through even if the DCA is on full. Also, if the DCA is set for a short decay, then you won't hear a long filter decay because the volume will reach zero before the filter decay finishes.

4.0E About SP Modulation Sources

Modulation modifies the sound over time, which helps create more expressive sounds. In the diagram above, dotted lines indicate when a module can choose any one of multiple modulation sources; the knobs represent the ability to change the modulation depth. The SP has several modulation sources:

- **Two envelope generators that produce a programmable modulation change over a period of time.** The DCO, DCF, DCA, and Pan sections can choose between either envelope. The DCO, DCF and Pan have level parameters that regulate the envelope amount.

The effect of an envelope depends on what it's controlling. For example, modulating the DCA with an envelope creates level changes. Amplitude that decays over time creates percussive sounds (plucked strings, drums, etc.); brass, woodwind, and some bowed instruments have amplitudes that increase over time. A note-on message triggers each envelope.

- **Modulation signals generated by keyboard dynamics (velocity).** Velocity produces data relating to how fast a key goes from the key up (note off) to the key down (note on) position, which correlates to the dynamics of your playing. The DCO, DCF, DCA, and Pan all have level parameters that regulate the effect of velocity.
- **LFO.** The LFO creates several periodic and aperiodic modulation signals. The LFO modulation amount can be constant and/or varied with the mod wheel. *Examples:* Applying a periodic square wave modulating signal to the oscillator produces a trill. Modulating the filter cutoff with a triangle wave produces a wa-wa effect or, if used subtly in the higher registers, a shimmering type of sound. The DCO, DCF, DCA, and Pan all have level controls for the LFO control signal. The LFO also includes an overall amount control (not shown) which sets the master LFO level.
- **Auxiliary modulation signals.** The DCO, DCF, DCA, and Pan can select any one of several auxiliary modulation sources. These include:

Keyboard pressure Pressing on the keys after they're down generates a control signal. Some keyboards generate *channel* pressure, which represents an average of all keys being pressed and affects all voices at once; others generate *poly* pressure, which produces individual pressure messages for each key. The SP can respond to either type of pressure message.

Modulation wheel This is normally a large knob or lever located to the left of most keyboards, and is traditionally

assigned to LFO amount so that rotating the wheel away from you introduces vibrato. However, it can be programmed for many other effects as well.

MIDI continuous controllers Three auxiliary MIDI controllers (auxctl), assigned to particular MIDI controller numbers at the preset level, allow for parameter control via MIDI continuous controller messages.

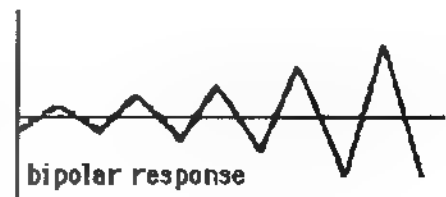
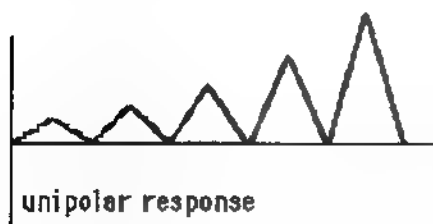
Note that the LFO rate and amount can also be modulated by various modulation signals.

Assigning more than one modulator allows for interaction between the modulators. *Example:* If a parameter responds to LFO and an envelope generator, the parameter will follow the general envelope shape but also be influenced by the LFO.

4.0F Unipolar and Bipolar Modulation

With some parameters, the modulation amount can be *unipolar* or *bipolar*. Unipolar modulation remains within a positive or negative quadrant; bipolar modulation is centered around zero and varies within two quadrants. The following diagrams show increasing modulation amounts for unipolar and bipolar triangle waveforms.

An a practical illustration of how this works, consider how guitarists add vibrato to guitar. If this is done by vibrating the strings by finger motion, the pitch can only be bent up. This is an example of unipolar modulation. With a “whammy bar,” the pitch can be bent either sharp or flat—an example of bipolar modulation.



With bipolar response applied to a mod wheel, the mod wheel becomes a “center off” device, like a pitch bend wheel. This is good for several applications, including pan modulation since the mod wheel can swing the sound across the stereo field, with the mod wheel center position corresponding to the center of the stereo

field. With a unipolar response, it would be necessary to program the sound to normally sit at one extreme or the other of the stereo field, then add or subtract modulation to swing it over to the other side.

4.0G Modulation Polarity

The polarity (also called phase) of the waveforms can also be changed for many parameters. The following diagram shows the same signals as above, but with reversed polarity. Note how the “peaks” and “valleys” are reversed.

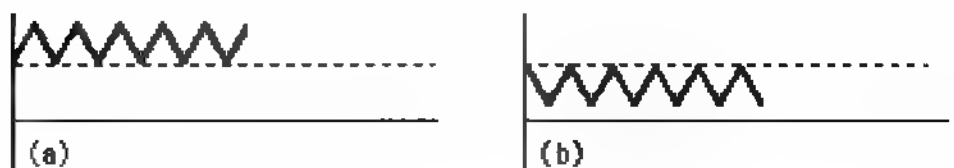
Polarity reversal is good for driving two modulation destinations from one modulation source so that the mod source can provide two different effects—such as having the filter cutoff go lower while the amplitude goes higher.



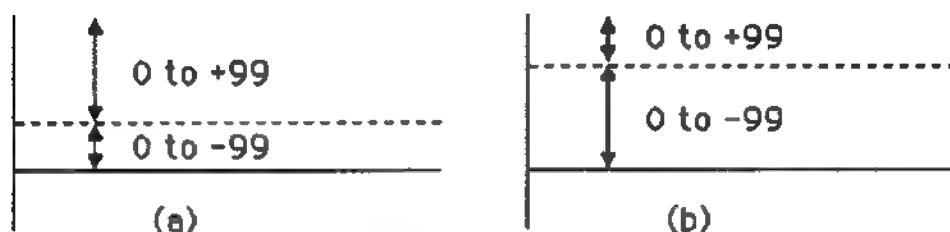
4.0H Positive and Negative Modulation

With some parameters, the modulation amount can be positive or negative. Generally, with positive amounts, an increasing control signal increases the value of the parameter being controlled. With negative amounts, an increasing control signal *decreases* the value of the parameter being controlled. A setting of 00 is equivalent to turning off the modulation source.

If a “baseline” setting exists, modulation amounts add or subtract values from that setting. *Example:* If the LFO amount is set to a certain level (as shown by the dashed line), positive unipolar modulation increases the amount from that level (a) and negative unipolar modulation decreases the amount from that level (b).



Modulation cannot force a value beyond its maximum or minimum value. However, the SP employs an innovative automatic scaling process that automatically adjusts the amount of modulation to fit the available range. No matter where the baseline is, whether low (diagram a) or high (diagram b), a modulation range of +99 extends from the baseline to the maximum parameter value, and a modulation range of -99 extends from the baseline to the minimum parameter level. The variation between steps changes to compensate.



4.01 Modulation Subtleties

There are several subtleties involving modulation. Although in most cases you will be adjusting parameters by ear until they sound good and not worry too much about the numbers, some of the following background material is important to those who want to develop their own SP sounds.

- With the DCF or DCA, the LFO always works downward from the maximum (unmodulated) level to some percentage of its unmodulated value. Therefore, the DCA and DCF are usually open all the way; applying the LFO lowers the DCA level or DCF cutoff. However, because of the interaction between positive and negative modulation, different polarities, and so on, this may appear to act differently depending on the modulation settings you've chosen.
- With the DCF, velocity modulation is scaled by the DCF envelope amount, *even if no envelope is selected*. Thus velocity will have no effect if the DCF envelope amount is set to zero, and will otherwise affect the DCF by scaling the envelope.
- With DCO and Pan, the velocity, aux mod, envelope, and LFO modulation options offer a "mode" choice that selects between *summed* and *scaled*. When summed, velocity is simply added to any other modulation sources. When scaled,

velocity scales the other modulation sources—in other words, greater amounts of velocity will increase the effect of other modulation sources, and less amounts of velocity will decrease the effect of other modulation sources.

- With the DCA, the auxiliary modulation controllers are scaled with other modulation sources. With the DCF, they are simply added in with other modulation sources.

4.1 Tone Management Screens


4.1A Tone Play Screen



The screenshot shows a two-line display. The top line contains the text "[TONE NAME] t# 1". The bottom line contains a series of musical notes (semibreves, minims, crotchets, quavers) followed by a series of dots, representing the allocation of voices.

The tone menu's first screen is a "play screen" that shows the tone name and number on the top line, and monitors the number of voices being played on the lower line. Each note represents a voice that's in use; each dot represents a voice that's available for allocation.

4.1B Create Tone



The screenshot shows a two-line display. The top line contains the text "create tone t#199". The bottom line contains the text "<press exec>".

Creates an empty tone. You can assign a wave to this tone, and vary several editable wave parameters (dynamics, filtering, etc.). There is 1 parameter.

- **t# (tone number)** Selects the number of the tone to be created. The default is the lowest available tone number.

Press EXEC to create the selected tone. Its default is to adopt the current edit wave, and take its name from that wave.

The display shows **operation complete** to confirm, then the default tone number increments to the next empty tone number.

Normally when you move around and select different tones, you will also be selecting different wave numbers invisibly. To take advantage of a shortcut where you create a tone and it pulls in a selected wave, once you're on the Create Tone screen, go to any wave screen and select a wave. Go back to Create Tone and it will pull in the wave.

4.1C Delete Tone

```
delete tone t#199  
[TONE NAME]
```

Deletes a single tone, or all tones. There is 1 parameter.

- **t# (tone number)** Selects the tone to be deleted (the lower line shows the tone's name). Going beyond the highest-numbered tone selects **All Tones** for deletion.

Press **EXEC** to confirm that you want to delete the selected tone. The display now asks:

```
also delete linked  
wave(s)? yes
```

There is 1 parameter, for yes or no. If you select **yes** then press **EXEC**, all waves associated with the tone will be deleted except for waves that are used elsewhere. If you select **no** then press **EXEC**, the tone "structure" itself is deleted, but the associated waves remain intact. In either case, the display shows **operation complete** to confirm.

The general rule is to answer **no** when you want to create a new tone based on the same set of waves, and **yes** when you want to remove a tone in its entirety to avoid using up memory on unneeded waves.

4.1D Copy Tone



A screenshot of a menu titled "COPY tone". It shows two options: "t# : 0" and "to t#199". The text is in a pixelated font on a dark background.

Copies one tone to any empty tone. There are 2 parameters.

- **t# (tone number)** Selects the tone to be copied.
- **To t# (empty tone number)** Selects the empty tone to hold the copy. If a tone number cannot be selected, it is not empty.

Press EXEC to copy the selected tone. The display shows **operation complete** to confirm.

4.1E Copy Name to Tone



A screenshot of a menu titled "COPY name". It shows two options: "t#199" and "[WAVE NAME]". The text is in a pixelated font on a dark background.

Copies a wave, tone, map, or preset name to a tone. This saves time if a suitable tone name already exists, since you don't need to enter the tone name from scratch. There are 3 parameters.

- **t# (tone number)** Selects the tone to which the name will be copied.
- **w# t# m# p# (wave, tone, map, or preset)** Selects whether the name to be copied is a wave, tone, map, or preset name.
- **name** Selects the actual wave, tone, map, or preset name.

Press EXEC to copy the selected name to the tone. The display shows **operation complete**.

4.1F Edit Tone Name

```
edit tone name - #199
      [TONE NAME]
```

Allows tone naming or renaming, up to 14 characters. There are two screens. The first screen has 1 parameter.

- **t# (tone number)** Selects the tone to be named/renamed.

To access the second screen, press EXEC.

```
<exec to save edit>
      [TONE NAME]
```

The second screen has 1 parameter (for tone naming).

- **(Tone name)** Use the cursor buttons to choose the character to be edited and the data knob to select the character. Remember that you can hold the INC or DEC button down and use the data knob to select characters quickly. After naming, press EXEC to save the edits.

Turning the data knob fully counterclockwise selects a blank space. Turning the data knob clockwise (or using the INC button) selects characters in the following order:

lower case letters	a-z
upper case letters	A-Z
symbols	` \$ % @ ^ _ + * / = []
	< > { } " # & () : ? ! ' , . -
numbers	0-9

4.1G Select Wave

```
select wave ~ t#199
w# 0 [WAVE NAME]
```

Assigns a wave to a tone (remember that when you create a tone, it adopts the current edit wave and name, so it will not always be necessary to select a wave). There are 2 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **w# (wave number)** Selects the wave to be assigned to the selected tone.

Press **EXEC** to assign the selected wave to the tone. The display shows **operation complete** to confirm.

4.2 Tone Modification Screens

4.2A Env 1 Times

```
env 1 times ~ t#199
a: 0 s:9t d:00 r:20
```

The SP has two envelopes, Env 1 and Env 2, that can be assigned to various tone menu modules.

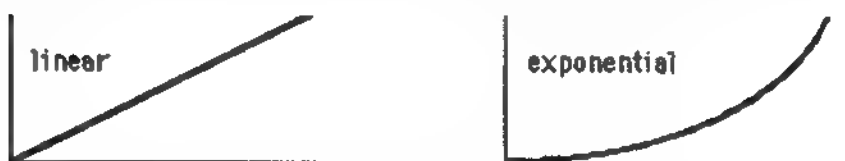
Envelope 1 controls the total duration of the voice, so when the envelope level returns to zero, the voice becomes available for allocation (for more on voice allocation, see section 6.1). If you don't assign envelope 1 to anything, its parameters still control the overall voice timing to keep the voice from being stuck on. Envelope 2 is an auxiliary envelope, and its settings have little or no impact on the voice allocation process.

Caution: If the envelope edits don't seem to affect the sound, make sure the envelope being edited is selected as the modulation source for the parameter you want to vary. For example, to hear how envelope edits affect the DCA, assign the envelope being edited to the DCA (on the DCA Envelope Select page).

There are 5 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **Attack.** Selects the Attack time, the time the envelope takes to go from 0 to its maximum level. Higher numbers give longer attack times.

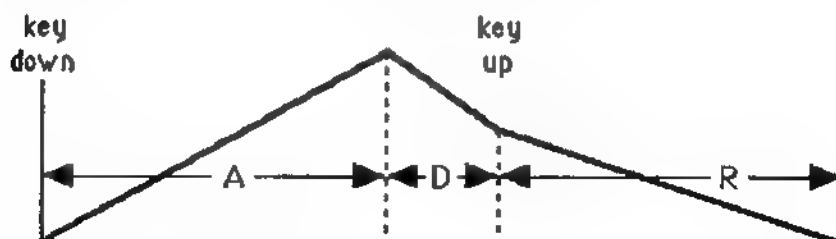
Positive numbers create a linear attack curve. Negative numbers create an exponential attack curve, which is useful for fadeins and backwards tape simulations.



Linear vs. exponential attack characteristics.

- **Sustain.** There are two different envelope modes, Gate and Trigger. Turning the data knob fully counter-clockwise selects **gt** (gate) mode, where the envelope decays according to the **d** (decay) parameter setting. If you keep your fingers on the keys, the sound will eventually decay to zero. If you release your fingers from the keys before the sound decays to zero, or during the attack, the envelope will continue to decay to zero according to the time set by the **r** (release) parameter.

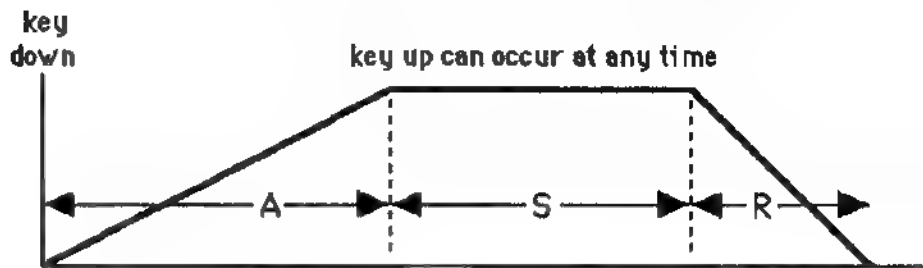
However, if **d** is set to maximum (the “infinity” setting, as shown by the symbol ∞), the envelope level remains at maximum for as long as its associated key is held down.



Gate mode envelope.

With the cursor under the **s** parameter, rotating the data knob clockwise (or pressing the INC button) selects a number from 00 to 99, which indicates trigger mode. The entire

envelope will play through from beginning to end, no matter when you release the key; the S parameter determines how long the envelope remains at maximum level before the release time starts. Higher S values give a longer sustain time.



Trigger mode envelope

- **Decay.** In Gate mode, after the Attack phase the signal decays back to minimum level (or stays on indefinitely if the d parameter is set to ∞). The higher the d value, the longer the decay time. This parameter is not available in Trigger mode.
- **Release.** In Gate mode, if you release a key, the envelope decays from its existing level toward minimum level according to the r value. Higher numbers increase the release time.

In Trigger mode, the release value determines how long it takes for the envelope to return to minimum level after the sustain time has elapsed.

4.2B Env 2 Times

```
env 2 times t#199
a: 0 s:9t d:00 r:20
```

The Env 2 parameters are identical to Env 1. However, remember that Env 2 has little or no impact on the voice allocation process, whereas when Envelope 1's level returns to zero, the voice it controls is released for re-allocation.

4.2C DCA Envelope

```
dca envelope t#199  
env1
```

The DCA controls the overall tone level. There are three ways to control the DCA, and 2 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **env** Env 1 controls the DCA level with the Env 1 curve. Env 2 controls the DCA level with the Env 2 curve. With **none** selected, the DCA remains full on at all times.

4.2D DCA Modulation

```
dca modulation t# 0  
vel: -0 lfo: 0:nrm
```

When velocity modulates the DCA, the overall level tracks the dynamics of your playing. LFO signals can also modulate the DCA, which creates level variations (e.g., tremolo). There are 4 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **vel** With **vel** = 0, velocity doesn't affect the DCA level. Higher values lower the DCA level when you play softly. +99 gives the maximum dynamic range.

Negative values lower the DCA level with higher velocities. The more negative the value, the lower the DCA level will go. -99 gives the maximum negative dynamic range.

- **lfo** Higher values increase LFO modulation of the DCA, which varies the level according to the LFO page parameter settings.
- **nrm/inv** Changes the LFO waveform phase.

4.2E DCA Aux Mod

```
dca aux mod t#199  
src:modwheel amt:-30
```

In addition to velocity, envelope, and LFO, the DCA can be modulated by an auxiliary modulation source. There are 3 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **src (source)** Selects one of the DCA auxiliary modulation sources:

none	No auxiliary modulation.
modwheel	Modulation wheel.
chanpres	Channel (mono) keyboard pressure. The harder you press on the keys, the greater the degree of modulation.
polypres	Polyphonic (keyboard) pressure. Unlike channel pressure, which responds to the average pressure on all keys being held down, key pressure generates individual pressure messages for each key.
auxctl1	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
auxctl2	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
auxctl3	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
- **amt (amount)** Higher values allow more DCA modulation from the auxiliary source. This parameter can be inverted; with positive values, more modulation increases the DCA level and with negative values, more modulation decreases the DCA level.

4.2F DCF Cutoff

```
dcf cutoff:99 t#199  
envlof:env1 amt:+99
```

The DCF controls the tone's brightness by attenuating frequencies above a particular cutoff frequency. There are two main ways to control the DCF's cutoff, and 3 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **dcf cutoff** Edits the filter's initial cutoff frequency. Lower values give a duller sound; higher values give a brighter sound. Harmonically complex signals are most affected by the filter.
- **envelope** Env 1 controls the DCF cutoff with the Env 1 curve. Env 2 controls the DCF cutoff with the Env 2 curve. With **none** selected, the cutoff is affected only by the DCF cutoff parameter.
- **amt** (amount) Higher values allow more DCF modulation from the selected envelope. This parameter can be inverted; with positive values, more modulation increases the DCF cutoff frequency and with negative values, more modulation decreases the DCF cutoff frequency.

4.2G DCF Modulation

```
dcf modulation t# 0  
vel:+12 lfo:-20:nrm
```

When velocity modulates the DCF, the overall brightness tracks the dynamics of your playing. LFO signals can also modulate the DCF, which creates different types of periodic and non-periodic tonal changes. There are 4 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **vel** With **vel** = 0, velocity doesn't affect the DCF cutoff. Higher values lower the DCF cutoff when you play softly. +99 gives the maximum dynamic range.

Negative values lower the DCF cutoff with higher velocities. The more negative the value, the lower the DCF cutoff will go. -99 gives the maximum negative dynamic range.

- **lfo** Higher values increase LFO modulation of the DCF, which varies the level according to the LFO page parameter settings.
- **nrm/inv** Changes the LFO waveform phase.

4.2H DCF Aux Mod

```
dcf aux mod t#199  
src:polypres amt:+40
```

In addition to velocity, envelope, and LFO, the DCF cutoff can be modulated by an auxiliary modulation source. There are 3 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **src (source)** Selects one of the DCA auxiliary modulation sources:

none	No auxiliary modulation.
modwheel	Modulation wheel.
chanpres	Channel (mono) keyboard pressure. The harder you press on the keys, the greater the degree of modulation.
polypres	Polyphonic (keyboard) pressure. Unlike channel pressure, which responds to the average pressure on all keys being held down, key pressure generates individual pressure messages for each key.
auxctl1	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
auxctl2	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
auxctl3	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.

- **amt** (amount) Higher values allow more DCF modulation from the auxiliary source. This parameter can be inverted; with positive values, more modulation increases the DCF cutoff and with negative values, more modulation decreases the DCF cutoff.

4.21 DCO Envelope

```

dco envelope: t# 0
none:summed amt:+10

```

The DCO pitch can be controlled by several modulation sources. For example, with envelope control a note can start a little flat and rise up to pitch; LFO modulation can give vibrato effects. There are three ways to control the DCO pitch, and 4 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **none/env 1/env 2** Env 1 controls the DCO pitch with the Env 1 curve. Env 2 controls the DCO pitch with the Env 2 curve. With **none** selected, the DCO pitch is not affected by either envelope.
- **summed/scaled** When summed, the envelope is added directly to the pitch along with any other modulation sources; when scaled, the envelope is scaled by any other modulation source that is also in scaled mode. *Example:* If the LFO is also set to scaled mode, the envelope and LFO interact so that more DCO envelope level would increase the amount of LFO modulation.
- **amt** (amount) Higher values allow more DCO envelope modulation. This parameter can be inverted; with positive values, more modulation increases the DCO pitch and with negative values, more modulation decreases the DCO pitch.

4.2J DCO Velocity Modulation

```
dcv veloc mod: t# 0  
mode:summed amt:-17
```

When velocity modulates the DCO, the overall pitch tracks the dynamics of your playing. There are 3 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **mode** When summed, the velocity modulation is added directly to the pitch along with any other modulation sources; when scaled, the velocity modulation is scaled by any other modulation source that is also in scaled mode.
Example: If the envelope is also set to scaled mode, the velocity and envelope interact so that higher velocity values increase the envelope level.
- **amt** (amount) Higher values allow more DCO velocity modulation. This parameter can be inverted; with positive values, more velocity increases the DCO pitch and with negative values, more velocity decreases the DCO pitch.

4.2K DCO LFO Modulation

```
dcv lfo mod: t# 0  
phase:normal amt:+40
```

This screen determines how much the LFO varies the DCO pitch. There are 3 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **phase** Determines whether the LFO phase is normal or inverted (see section 4.0G).
- **amt** (amount) Higher values allow more LFO modulation. This parameter can be inverted; with positive values, a positive-going LFO waveform increases the DCO pitch and with negative values, a positive-going LFO waveform decreases the DCO pitch.

4.2L DCO LFO Mode

```
dc0 lfo mode: t# 0
summed:bipolar
```

This screen chooses between two ways in which the LFO varies the DCO pitch. There are 3 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **summed/scaled** When summed, the LFO is added directly to the pitch along with any other modulation sources; when scaled, the LFO is scaled by any other modulation source that is also in scaled mode. *Example:* If the envelope is also set to scaled mode, the envelope and LFO interact so that more DCO envelope level would increase the amount of LFO modulation.
- **bipolar/unipolar** Determines whether the LFO waveform will vary around a center point (bipolar) or remain in a single quadrant (unipolar); see section 4.0F.

4.2M DCO Aux Mod

```
dc0 aux mod t#199
src:polypres amt:+19
```

In addition to velocity, envelope, and LFO, the DCO pitch can be modulated by an auxiliary modulation source. There are 3 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **src (source)** Selects one of the DCO pitch auxiliary modulation sources:

none	No auxiliary modulation.
modwheel	Modulation wheel.
chanpres	Channel (mono) keyboard pressure. The harder you press on the keys, the greater the degree of modulation.
polypres	Polyphonic (keyboard) pressure. Unlike channel

pressure, which responds to the average pressure on all keys being held down, key pressure generates individual pressure messages for each key.

auxctl1 One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.

auxctl2 One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.

auxctl3 One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.

- **amt (amount)** Higher values allow more DCO pitch modulation from the auxiliary source. This parameter can be inverted; with positive values, more modulation increases the DCO pitch and with negative values, more modulation decreases the DCO pitch.

4.2N DCO Auxiliary Modulation Mode



```
dco auxm mode: t# 0
scaled:unipolar
```

This screen determines two ways in which the auxiliary modulation signal will vary the DCO pitch. There are 3 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **summed/scaled** When summed, the auxiliary modulation source is added directly to the pitch along with any other modulation sources; when scaled, the auxiliary modulation source is scaled by any other modulation source that is also in scaled mode.
- **bipolar/unipolar** Determines whether the auxiliary modulation source will vary around a center point (bipolar) or remain in a single quadrant (unipolar); see section 4.0F.

4.2O Pan Envelope

```
pan envelope: t# 0  
none:summed amt:+10
```

The panning (position of the signal in the stereo field) for the assigned output pair can be controlled by several modulation sources. *Examples:* With envelope control a note can start in one channel and end up over time in the other channel; periodic modulation causes a signal to move back and forth in the stereo field. There are three ways to control the pan position, and 4 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **none/env 1/env 2** Env 1 controls the pan position with the Env 1 curve. Env 2 controls the pan position with the Env 2 curve. With **none** selected, the pan position is not affected by the envelopes.
- **summed/scaled** When summed, the envelope is added directly to the pan position along with any other modulation sources; when scaled, the envelope is scaled by any other modulation source that is also in scaled mode. *Example:* If the LFO is also set to scaled mode, the envelope and LFO interact so that more pan envelope level would increase the amount of LFO pan modulation as well.
- **amt (amount)** Higher values allow more pan position envelope modulation. This parameter can be inverted; with positive values, more modulation changes the pan position from left to right and with negative values, more modulation changes the pan position from right to left.

4.2P Pan Velocity Modulation

```
pan veloc mod: t# 0  
mode:summed amt:-12
```

When velocity modulates panning, the overall pan position tracks the dynamics of your playing. There are 3 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **mode** When summed, the velocity modulation is added directly to the pan position along with any other modulation sources; when scaled, the velocity modulation is scaled by any other modulation source that is also in scaled mode.
Example: If the envelope is also set to scaled mode, the velocity and envelope interact so that higher velocity values increase the envelope level.
- **amt** (amount) Higher values allow more pan position modulation. This parameter can be inverted; with positive values, higher velocities move the pan position from left to right and with negative values, higher velocities move the pan position from right to left.

4.2Q Pan LFO Modulation

```

Pan lfo mod:  t#  0
Phase: normal  amt: +33
  
```

This screen determines how much the LFO varies the pan position. There are 3 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **phase** Determines whether the LFO phase is normal or inverted (see section 4.0G).
- **amt** (amount) Higher values allow more LFO modulation. This parameter can be inverted; with positive values, a positive-going LFO waveform changes the pan position from left to right and with negative values, a positive-going LFO waveform changes the pan position from right to left.

4.2R Pan LFO Mode

```
Pan lfo mode: t# 0  
summed:bipolar
```

This screen chooses between two ways in which the LFO varies the pan position. There are 3 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **summed/scaled** When summed, the LFO is added directly to the pan position along with any other modulation sources; when scaled, the LFO is scaled by any other modulation source that is also in scaled mode. *Example:* If the envelope is also set to scaled mode, the envelope and LFO would interact so that more pan envelope level would increase the amount of LFO modulation.
- **bipolar/unipolar** Determines whether the LFO waveform will vary around a center point (bipolar) or remain in a single quadrant (unipolar); see section 4.0F.

4.2S Pan Aux Mod

```
Pan aux mod t#199  
src:Polypres amt:+45
```

In addition to velocity, envelope, and LFO, the pan position can be modulated by an auxiliary modulation source. There are 3 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **src (source)** Selects one of the pan position auxiliary modulation sources:

none	No auxiliary modulation.
modwheel	Modulation wheel.
chanpres	Channel (mono) keyboard pressure. The harder you press on the keys, the greater the degree of modulation.
polypres	Polyphonic (keyboard) pressure. Unlike channel

pressure, which responds to the average pressure on all keys being held down, key pressure generates individual pressure messages for each key.

auxctl1 One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.

auxctl2 One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.

auxctl3 One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.

- **amt (amount)** Higher values allow more pan position modulation from the auxiliary source. This parameter can be inverted; with positive values, more modulation increases the pan position and with negative values, more modulation decreases the pan position.

4.2T Pan Auxiliary Modulation Mode



Pan auxm mode: t# 0
scaled:unipolar

This screen determines two ways in which the auxiliary modulation signal will vary the pan position. There are 3 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **summed/scaled** When summed, the auxiliary modulation source is added directly to the pan position along with any other modulation sources; when scaled, the auxiliary modulation source is scaled by any other modulation source that is also in scaled mode.
- **bipolar/unipolar** Determines whether the auxiliary modulation source will vary around a center point (bipolar) or remain in a single quadrant (unipolar); see section 4.0F.

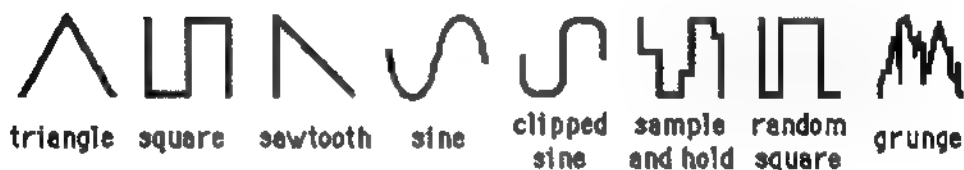
4.2U LFO Waveshape

```
lfo waveshape: t# 0  
triangle wave
```

The LFO (low frequency oscillator) provides any one of eight repetitive control signals (unlike an envelope, which produces a complex one-shot event when triggered). The polarity of these signals can be inverted on the DCO LFO mod page by setting the amount to a negative number. *Example:* The negative-going ramp wave becomes a positive-going ramp wave if the DCO LFO mod amount is set to a negative number.

There are 2 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **shape** Selects the LFO waveform. There are eight options.



triangle	The waveform has equal rise and fall times. Use for vibrato and tremolo effects.
square	Alternates between minimum and maximum values. Use for trills and special effects.
sawtooth	Ramp. Each cycle snaps instantly to a maximum level, ramps down to zero, then repeats the cycle. With negative amounts set an LFO destination mod screen, the waveform snaps to zero, ramps up to a maximum value, then repeats the cycle.
sine wave	Similar to a triangle wave, but with rounded peaks.
clipped sine	Somewhat like a rounded square wave. Use this for panning and other extreme modulation swings where you don't want clicks to occur from the sharp transitions normally associated with square wave modulation. Also good for air raid siren effects.

sampl/hold A randomized waveform with sharp transitions.

random sqr A square wave that may or may not change states when triggered by a keypress.

grunge A high-speed, noise-like waveform. Excellent when used to modulate the DCO for adding “rasp” to saxophone and other sounds.

4.2V LFO Base Setup

```
lfo base setup t# : 0  
speed:52      amt:45
```

Speed sets the LFO's rate of change; amount determines the minimum amount of LFO modulation that will be sent to all destinations. A non-zero amount will always generate LFO modulation (assuming there are modulation destinations assigned to LFO), even if you're not using the mod wheel to bring in the modulation. If you are using the mod wheel and want it to do all the controlling, leave the amount at zero. There are 3 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **speed** Selects the LFO waveform speed.
- **amt** Selects the overall LFO depth. Since each modulation destination can also vary the level received from the LFO, this parameter may seem redundant. However, it can be a "global" control. *Example:* If the LFO goes to several modulation destinations (DCO, DCA, DCF), you can reduce the LFO amount to reduce the amount going to *all* mod inputs.

This parameter allows for positive values only, since the LFO amount can be inverted at each LFO destination.

Caution: If the LFO amount doesn't seem to work, make sure LFO is selected as the modulation source for the parameter you want to vary.

4.2W LFO Speed Mod

```
lfo speed mod t# 0  
src:auxctl12 — amt:32
```

Several modulation sources can vary the LFO speed. There are 3 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **src** (source) Selects one of the LFO speed modulation sources:

none	No auxiliary modulation.
modwheel	Modulation wheel.
chanpres	Channel (mono) keyboard pressure. Press harder on the keys to increase the LFO speed.
polypres	Polyphonic (keyboard) pressure. Unlike channel pressure, which responds to the average pressure on all keys being held down, key pressure generates individual pressure messages for each key.
auxctl1	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
auxctl2	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
auxctl3	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.

- **amt** The top end of the modulation range will always be the nominal LFO speed set on the LFO base setup screen; more modulation moves the LFO speed downward to zero.

4.2X LFO Amount Mod

```
lfo amount mod t#199  
src:Polypres - amt:99
```

Several modulation sources can vary the LFO amount (depth). There are 3 parameters.

- **t#** (tone number) Selects the tone to be edited.
- **src** (source) Selects one of the LFO amount modulation sources:

none	No auxiliary modulation.
modwheel	Modulation wheel.
chanpres	Channel (mono) keyboard pressure. Press harder on the keys to increase the LFO amount.
polypres	Polyphonic (keyboard) pressure. Unlike channel pressure, which responds to the average pressure on all keys being held down, key pressure generates individual pressure messages for each key.
auxctl1	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
auxctl2	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
auxctl3	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.

- **Amount** (lower right) Higher values allow more LFO amount modulation.

4.2Y Wave Start Modulation

```
Wave start mod t# 0  
src:keybdpos amt:amt
```

Several modulation sources can modulate a wave's start point. This is useful in creating dynamic effects. *Example:* You could set a percussive wave's start point some distance into the sample to bypass the initial percussive transient. Having the start point track negative velocity (so that playing harder pushes the start point closer to the beginning of the wave) creates a more percussive effect when you play more forcefully. There are 3 parameters.

- **t# (tone number)** Selects the tone to be edited.
- **src (source)** Selects one of the wave start point modulation sources:

keybdpos	Start point correlates to the note being played, relative to the zone to which the wave is mapped. Playing on the lowest key of the zone gives the effect of zero modulation; at the high end of the zone, there is full modulation.
velocity	Start point correlates to velocity.
random	Start point varies randomly.
modwheel	Modulation wheel controls start point.
chanpres	Channel (mono) keyboard pressure. Press harder on the keys to change the start point.
auxctl1	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
auxctl2	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
auxctl3	One of three MIDI continuous controllers, assigned to a particular MIDI controller number in the preset menu.
- **amt (amount)** If set to **qnt** (for quantize), the start point modulation changes are set on the next screen. This setting is equivalent to zero amount if the next screen is not set up for start point quantization.

With positive values, increased modulation values move the start point further into the sample. With negative values, the start point is shifted into the sample, and increased modulation values push the start point closer to the beginning of the sample.

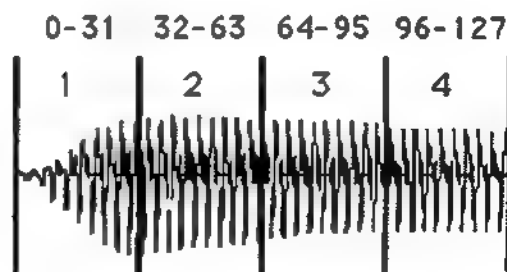
Example: Suppose you have a single sample (with a significant attack time) stretched across the keyboard. At the low end, the attack time may be too long. Using the keyboard to modulate the start point, and setting amount for a small negative value, will tighten up the attack time at the low end of the keyboard but leave the upper range unaffected.

4.2Z Wave Start Modulation Quantization

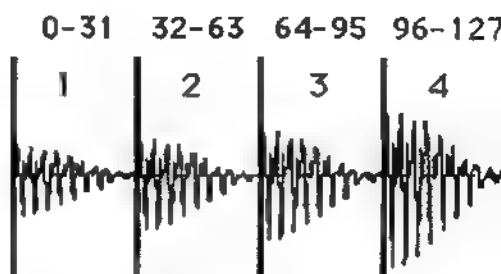
About Wave Start Modulation Quantization

Wave start quantization is a very specialized feature that is unique to the SP. It is designed for percussive samples, but as sound developers experiment with the SP, we expect to see other novel applications surface.

Perhaps the best way to explain this is with an application. With wave start quantization, a sample can be quantized into several equal parts, with different parts of the sample triggered by various values from different control sources (velocity, modwheel, auxiliary controller, etc.). Here is a waveform quantized into four sections (it could be quantized into other divisions as well); the upper line of numbers shows the range of values that trigger each part of the sample.



One application for waveform quantization is to create velocity-switched drum sounds from a single sample, without the need for switched layers. The following diagram shows four drum samples, with 1 being the softest-hit sample and 4 being the hardest-hit sample. If the controller source is velocity, then different velocity values will trigger samples with different dynamics.



Note that each sample must be exactly the same length for this trick to work. Generally, you would create the wave in a computer-based sample editing program to insure equal lengths for the samples, then transfer the sample over to the SP.

Programming Wave Start Modulation Quantization

When amount on the first Wave Start Mod screen is set to **qnt**, the second Wave Start Modulation screen is active. There are 2 parameters.

wave start mod t# 0
quantiz: 12 steps

- **t#** (tone number) Selects the tone to be edited.
- **quantiz** Selects the number of steps into which the sample will be quantized, or none.

There is one exception to **quantiz**: if the wave start modulation source on the previous screen is set to **keybdpos**, then the **quantiz** options on this screen are **none** and to **keyboard zone**. With **none** selected, quantization is off. With **keyboard zone** selected, the sample is quantized to the number of keys in the zone (the default zone, two octaves around middle C, is used when MIDI is assigned to play the current edit tone and no particular map is in effect).

Chapter 5 Map Menu

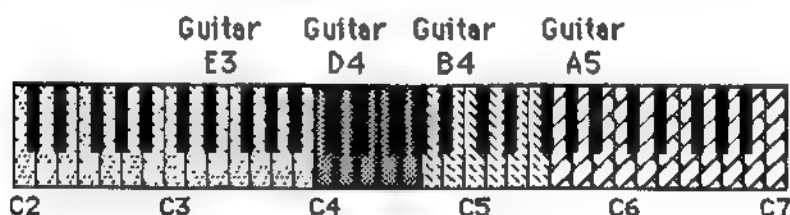
5.0 About Maps

Maps allow for easy keyboard splits and multisampling. Maps define which tones in memory will be assigned to the keyboard, as well as specify the following options for each tone:

- Keyboard note range
- Transposition amount
- Volume level
- Panning placement
- Output selection (main or sub stereo outputs)
- Zone tuning
- Voice allocation control

5.0A Assigning Tones to Particular Zones

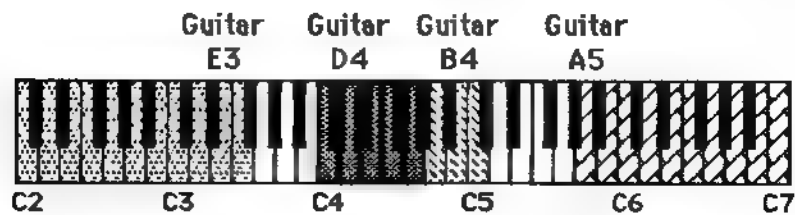
Here is an example of a map. Suppose you have four guitar samples—E3, D4, B4, and A5—that you want to play from a DPM 3 SE five-octave keyboard. After setting their wave and tone parameters, you can then assign the tones to particular note ranges, called *zones*:



Zones cannot be layered; the highest note of one zone is generally adjacent to the lowest note of the next higher-numbered zone (although there can be gaps between zones). However, you can open up as many zones as you want by simply changing a zone assignment to create more space. For example, suppose we want to add two more tones to the guitar map—one at A3 that covers the range of C3 to B3, and one at E5 that covers the range of D5 to G5.

First, edit the lowest zone so that it covers from C2 to F#3. This allows room to insert the A3-based tone between G3 and B3. Edit the highest zone to cover from G#5 to C7, and the next highest

zone to cover A4 to C#5. This opens up space for adding the E5-based tone between D5 and G5:



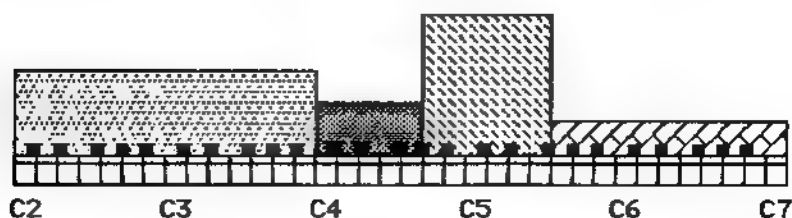
If you try to lower the lowest note of a zone and can't, select the next-lower zone and shift the highest note lower. *Example:* Suppose there are two zones, 0 and 1. Zone 0's highest note is D4 and zone 1's lowest note is D#4. If you try to change zone 1's lowest note to C4, you can't. However, if you change zone 0's high note from D4 to B3 it opens up sufficient space so that zone 1's note range can extend down to C4.

Caution: Zone numbers are not fixed if you add or delete zones, and will renumber themselves automatically to remain consecutive. *Example:* Suppose you have a keyboard map with four zones: 0, 1, 2, and 3. Now suppose you remove zones 1 and 3. Zone 0 will remain as zone 0, but zone 2 will renumber itself to zone 1.

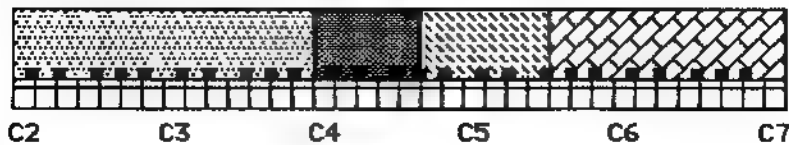
5.0B Transposition, Volume, and Pan

The *transposition* parameter lets you change the tone's tuning within the zone, which insures that playing a particular key triggers the correct pitch. *Example:* If you play within a zone and the notes that are triggered are too high in pitch, use the transposition parameter to transpose the zone downward.

If the volume level changes from zone to zone, use the *volume* parameter to compensate. For example, if we represent level differences in a map by looking at the keyboard from the front and drawing a graph that shows the relative levels (note how the lowest and second from highest zones are the loudest):



The volume parameter can even out these variations, to create a response that can be represented like this:



The *pan* parameter places a zone anywhere in the stereo field. The *output* parameter determines whether a zone feeds the main or sub outputs.

5.0C Individual Output Options

The four outputs can be treated as two stereo pairs or, with proper pan and output settings, as four individual mono outputs or one stereo pair and two individual outputs. Here's an example of how to use the latter with drums. Assign the kick to Main and pan it hard left, the snare to Main and pan it hard right, and assign the rest of the drum sounds as a submix to the Sub output. The kick, snare, and drum submix could go through different signal processors (EQ, delay, etc.) on their way to a mixer.

5.1 Map Management Screens

5.1A Map Play Screen



The screenshot shows a two-line display. The top line contains the text "[MAP NAME]" followed by some graphical elements and "m# 1". The bottom line contains a series of musical notes (semibreves and minims) followed by a series of dots.

The map menu's first screen is a "play screen" that shows the map name and number on the top line, and monitors the number of voices being played on the lower line. Each note represents a voice that's in use; each dot represents a voice that's available for allocation.

5.1B Create Map



The screenshot shows a two-line display. The top line contains the text "create map" followed by some graphical elements and "m#127". The bottom line contains the text "<Press exec>" followed by some graphical elements.

Creates an empty map, with the currently selected tone selected as a single zone. You can "map" various tones to specific keyboard ranges ("zones") and also determine the tone's panning, transposition, and level within the map. There is 1 parameter.

- **m# (map number)** Selects the number of the map to be created. The default is the lowest available map number.

Press EXEC to create the selected map. The display shows **operation complete** to confirm, then the default map number increments to the next lowest empty map number.

5.1C Delete Map



The screenshot shows a two-line display. The top line contains the text "delete map" followed by some graphical elements and "m#127". The bottom line contains the text "[MAP NAME]" followed by some graphical elements.

Deletes a single map, or all maps. There is 1 parameter.

- **m#** (map number) Selects the map to be deleted (the lower line shows the map's name). Going beyond the highest-numbered map selects **All Maps** for deletion.

Press **EXEC** to confirm that you want to delete the selected map. The display now asks:

```
also delete ~ linked
tones/waves? ~~~~~ yes
```

There is 1 parameter, for yes or no. If you select **yes** then press **EXEC**, all tones and waves associated with the map will be deleted except for tones and waves that are used elsewhere. If you select **no** then press **EXEC**, the map "structure" itself is deleted, but the associated tones and waves remain intact. In either case, the display shows **operation complete** to confirm.

The general rule is to answer **no** when you want to create a new map based on the same set of tones and waves, and **yes** when you want to remove a map in its entirety to avoid using up memory on unneeded tones and waves.

5.1D Copy Map

```
COPY MAP ~~~~~ m#127
to m# 0
```

Copies one map to any empty map. There are 2 parameters.

- **m#** (map number) Selects the map to be copied.
- **to m#** (empty map number) Selects the empty map to hold the copy. If a map number cannot be selected, it is not empty.

Press **EXEC** to copy the selected map. The display shows **operation complete** to confirm.

5.1E Copy Name To Map

```
COPY name ---> m#127  
w# 1 [WAVE NAME]
```

Copies a wave, tone, map, or preset name to a map. This saves time if a suitable map name already exists, since you don't need to enter the map name from scratch. There are 3 parameters.

- **m#** (map number) Selects the map to the name is copied to.
- **w# t# m# p#** (wave, tone, map, or preset) Selects whether the name to be copied is a wave, tone, map, or preset name.
- **name** Selects the actual wave, tone, map, or preset name.

Press EXEC to copy the selected name to the map. The display shows operation complete.

5.1F Edit Map Name

```
edit map name #127  
[MAP NAME]
```

Allows map naming or renaming, up to 14 characters. There are two screens. The first screen has 1 parameter.

- **m#** (map number) Selects the map to be named/renamed.

To access the second screen, press EXEC.

```
<exec to save edit>  
[MAP NAME]
```

The second screen has 1 parameter (for map naming).

- **(Map name)** Use the cursor buttons to choose the character to be edited and the data knob to select the character. Remember that you can hold the INC or DEC button down and use the data knob to select characters quickly. After naming, press EXEC to save the edits.

Turning the data knob fully counterclockwise selects a blank space. Turning the data knob clockwise (or using the INC button) selects characters in the following order:

lower case letters	a-z
upper case letters	A-Z
symbols	` \$ % @ ^ _ + * / = []
	< > { } " # & () : ? ! ' , . -
numbers	0-9

5.1G Delete Zone

```
delete zone: m#127
z# 0 [TONE NAME]
```

Deletes a zone from a map (but leaves the associated tones and waves intact). If only one zone is in a map, it cannot be deleted. There are 2 parameters.

- **m#** (map number) The zone is deleted from the selected map.
- **z#** (zone) Selects the zone to be deleted. The lower line shows the tone name associated with the zone.

Press EXEC to delete the selected zone. The display shows operation complete to confirm.

5.1H Add Zone Into Map

```
add into map: m#127
new zone at: z# 1
```

Adds a zone to a map. There are 2 parameters.

- **m#** (map number) The map to which the zone will be added.
- **z#** (zone) Chooses where the zone will be added. Entering a zone number inserts the new zone in place of the selected zone and pushes all the other zones up.

A zone can only be added in a “gap” in the map. If adding a zone fills up the map, the display shows (– no zone gaps –). If the number shows an existing zone, it will be pushed up when the new zone is added. By default, a new zone completely fills the gap into which it is inserted.

The entire map extends from C-1 to G9, so if a zone extends from C-1 to G9, it will not be possible to add a new zone.

Press EXEC to delete the selected zone. The display shows operation complete to confirm.

5.2 Zone Edit Functions (Folder)

```
edit zones:  m#127  
[MAP NAME]
```

Press EXEC to access the pages in the “folder.”

5.2A • Assign Tone to Zone

```
select tone  z# 0  
t# 7 [TONE NAME]
```

Selects which tone, from any in the SP’s memory, will be used for a particular zone. There are 2 parameters.

- **z# (zone number)** Selects the zone to which the tone will be assigned.
- **t# (tone number)** Selects the tone to be assigned to the zone in the first parameter.

When you select a zone with the INC/DEC buttons, the top line shows the map name for reference since you are locked in to a particular map on these edit screens.

5.2B • Keyboard Mapping and Transposition



A screenshot of a digital display showing keyboard mapping parameters. The text is in a monospaced font and is arranged in two lines. The first line shows 'keybd mapping: z# 2' and the second line shows 'C 3...C 5 tpos: 0'. The background is dark, and the text is light-colored.

Selects the key note range for the zone, and the degree of transposition. The zone high and low notes can be selected from the front panel, or via MIDI keyboard. There are 4 parameters.

- **z# (zone number)** Selects the zone to be edited.
- **C3... (range low note)** C3 is the default low note when a map is created, which creates a single zone. A note as low as C-1 can be selected.
- **...C5 (range high note)** C5 is the default high note when a map is created, which creates a single zone. A note as high as G9 can be selected.
- **tpos (transpose)** You normally want notes at particular pitches to sit on the appropriate keys (e.g., a pitch of C4 is on middle C). The parameter transposes the sound itself to match the appropriate keyboard key, with the displayed number showing the amount of transposition necessary to accomplish this. This should not have to be changed if the wave pitch parameter (section 3.31) of the wave being mapped is correctly set (assuming you're dealing with pitched sounds).

When editing map zone limits from the keyboard, please note:

- If the cursor is on the transpose field's top line, the prevailing play mode selected in the "midi notes play" screen of the Global menu remains in effect. However, moving the cursor to one of the zone limit parameters automatically sets the SP so it plays only the zone being edited.
- You can play any keyboard notes within the selected zone, and change the high and low limits within the constraints of the upper and lower note limits of adjacent zones. You will hear the note if you play within the selected zone; if you play a note that falls within the range of a different zone, the note limit setting will not be changed and you will not hear anything. This gives a very clear indication of a zone's current limits, and avoids confusion when editing zones.

- Finally, note that if there is room on either side of the zone, and you adjust the high limit lower than the lower limit or the low limit higher than the higher limit, the parameter not being adjusted will be “pushed” to insure that the high limit is never lower than the low limit. This also makes it easy to create a set of single-note zones, which is useful for some drum and sound effects applications. *Example:* Suppose a zone’s note range extends from C4 to C5, and there is no zone that contains notes below C4. If you change the upper limit from C5 to C3, the lower limit will automatically jump from C4 to C3 to insure that the high limit is not lower than the low limit.

5.2C • Volume, Pan, and Output

```
vol/pan/out: 0 z# 0
vl:99 pn:ctr ou:main
```

Selects a zone’s overall level, placement in the stereo field (panning), and output assignment. There are 4 parameters.

- **z#** (zone number) Selects the zone to be edited.
- **vl** (volume) Selects the zone’s overall volume level.
- **pn** (pan) Places the zone in the selected output connection’s stereo field. R99 is full right, ctr is center, and L99 is full left.
- **ou** (output) Sends the zone’s audio output to either the main or sub (auxiliary) stereo outputs. In either case, the pan parameter determines the stereo placement.

5.2D • Zone Tuning Mode

```
zone tuning: 0 z# 0
by note number
```

Selects whether a zone responds to incoming note data, or has its pitch set solely by the zone transposition amount. There are 2 parameters.

- **z#** (zone number) Selects the zone to be edited.

- **response** There are two choices:

by note number

The zone tuning responds to the incoming MIDI note number. As you play different notes, the tone used in the zone is transposed accordingly. This is the standard setting to use when playing keyboard.

fixed

The tone's pitch is fixed at the pitch determined by the zone transpose setting. This setting is recommended for percussion, sound effects, and vocal fly-ins where you want to keep a fixed pitch regardless of the keys to which the sound has been mapped.

5.2E • Zone Voice Allocation

```
alloc control: z# 0
group: 0 mode: normal
```

Note allocation is the process that decides which notes will be played when the SP's 16-note polyphony is exceeded. (If you require 32-note polyphony, run two SPs in overflow mode, as described in section 6.3F).

The SP has sophisticated note allocation options. Notes in a map's zone can be "bussed" (assigned) to any of 8 "groups" (see section 6.3D), and set to "normal" or "exclusive" mode (described next). Other group properties, such as maximum amount of polyphony, are set at the Preset level; see sections 6.3D and 6.3E.

There are 3 parameters.

- **z#** (zone number) Selects the zone to be edited.
- **group** The zone can be assigned to any one of eight groups, each of which can have its own allocation characteristics.
- **mode** There are two options:

normal Notes play as expected, up to the maximum polyphony limit set for that group.

exclusive Playing one note repeatedly plays notes as expected, up to the maximum polyphony limit set for that group. However, playing a different tone within the same exclusive group will cut off any other tones that are playing. The zone can be assigned to any one of eight groups, each of which can have its own allocation characteristics. *Example:* This is a popular option for hi-hat parts since playing a new hi-hat sound cuts off any decaying hi-hat sound. The most obvious example is that if you hit the hi-hat while open then close it down, the closed hi-hat sound will cut off the open hi-hat sound.

Most samplers and drum machines handle this situation by assigning the hi-hat sounds in such a way that only one note can play at a time. The SP can operate this way as well by setting a group's maximum polyphony to 1 at the preset level (section 6.3D). However, this approach gives an unrealistic sound if you strike an open hi-hat repeatedly; unlike a real hi-hat, which will continue to ring somewhat even after being struck again, hitting the hi-hat will abruptly cut off any note that was ringing.

By setting the group's maximum polyphony to a number greater than 1 and choosing exclusive for the foot-closed hi hat sound, you get the best of both worlds: repeated hits will not cause abrupt cutoffs, but playing a foot-closed hi-hat will.

Chapter 6 Preset Menu

6.0 About Presets

Presets are the overall “superstructure” for waves, tones, and maps. Each preset can be called up by like-numbered MIDI program change commands.

A preset has two layers, A and B, and each layer can contain up to four maps that can be enabled via crossfading and switching (however, only one map per layer can sound on any given note). With crossfading, as one layer gets softer, the other layer gets louder. The crossfade can be based on note position, velocity, any of three auxiliary MIDI controllers, or several other sources. Switching works similarly, but instead of a gradual crossfade, the layers jump from one to another. The same control sources are available as for crossfading. For example, you might want to switch a layer on if a particular velocity value is exceeded (e.g., to bring in a rimshot on high-velocity snare hits).

This is also the menu where MIDI continuous controllers are assigned to the auxiliary modulation sources used in this menu as well as the tone menu.

To speed up the programming process, several “master” parameter changes are also available in the preset menu. For example, to vary the volume of one layer compared to another, altering the layer volume offsets the volume settings in the map menu. Similarly, there’s a “master” filter cutoff control that offsets the filter settings in the tone menu.

6.1 About Note Allocation

Note allocation is the process that decides which notes will be stolen when the SP’s 16-note polyphony is exceeded. (If you require 32-note polyphony, run two SPs in overflow mode, as described in section 6.3F).

The SP has sophisticated note allocation options. Notes in a map’s zone can be “bussed” (assigned) to any of 8 “groups” (see section 6.3D). The group’s properties are set here, at the preset level.

Each group can be limited to a particular number of notes that can be played at one time. *Example:* A bass part might be limited to just one note if it is a single-line bass part. This prevents the part from unnecessarily stealing voices away from other sounds.

Each group can have a selected priority (absolute, high, medium, low) so that if the SP's 16-voice limit is exceeded, you can control which group will receive priority. Each group can also have individually selected overflow to a second SP.

6.2 Preset Management Screens

6.2A Preset Play Screen



The preset menu's first screen is a "play screen" that shows the preset name and number on the top line, and monitors the number of voices being played on the lower line. Each note represents a voice that's in use; each dot represents a voice that's available for allocation.

6.2B Create Preset



Creates an empty preset, which can layer/split up to four maps, and be selected by a like-numbered MIDI program change. There is 1 parameter.

- **p# (preset number)** Selects the number of the preset to be created. The default is the lowest available preset number.

Press EXEC to create the selected preset. The display shows **operation complete** to confirm, then the default preset number increments to the next lowest empty preset number. All map selections in both layers default to the currently selected map.

6.2C Delete Preset

```
delete preset p# 0  
[PRESET NAME]
```

Deletes a single preset, or all presets. There is 1 parameter.

- **p#** (preset number) Selects the preset to be deleted (the lower line shows the preset's name). Going beyond the highest-numbered preset selects **All Presets** for deletion.

Press **EXEC** to confirm that you want to delete the selected preset. The display now asks:

```
also delete linked  
maps/tones/wavs? yes
```

There is 1 parameter, for yes or no. If you select **yes** then press **EXEC**, all maps, tones, and waves associated with the preset will be deleted except for maps, tones, and waves that are used elsewhere. If you select **no** then press **EXEC**, the preset "structure" itself is deleted, but the associated maps, tones, and waves remain intact. In either case, the display shows **operation complete** to confirm.

The general rule is to answer **no** when you want to create a new preset based on the same set of maps, tones, and waves, and **yes** when you want to remove a preset in its entirety to avoid using up memory on unneeded maps, tones, and waves.

6.2D Copy Preset

```
copy preset p# 12  
to p# 21
```

Copies one preset to any empty preset. There are 2 parameters.

- **p#** (preset number) Selects the preset to be copied.
- **to p#** (empty preset number) Selects the empty preset to hold the copy. If a preset number cannot be selected, it isn't empty.

Press EXEC to copy the selected preset. The display shows operation complete to confirm.

6.2E Copy Name To Preset

```
COPY name ---> P# 11  
W# 10 [WAVE NAME]
```

Copies a wave, tone, map, or preset name to a preset. This saves time if a suitable preset name already exists, since you don't need to enter the preset name from scratch. There are 3 parameters.

- **p#** (preset number) Selects the preset to which the name is copied.
- **w# t# m# p#** (wave, tone, map, or preset) Selects whether the name to be copied is a wave, tone, map, or preset name.
- **name** Selects the actual wave, tone, map, or preset name.

Press EXEC to copy the selected name to the preset. The display shows operation complete.

6.2F Edit Preset Name

```
edit prst name # 0  
-- [PRESET NAME]
```

Allows preset naming or renaming, up to 14 characters. There are two screens. The first screen has 1 parameter.

- **p#** (preset number) Selects the preset to be named/renamed.

To access the second screen, press EXEC.

```
<exec to save edit>  
-- [PRESET NAME]
```

The second screen has 1 parameter (for preset naming).

- **(Preset name)** Use the cursor buttons to choose the character to be edited and the data knob to select the character. Remember that you can hold the INC or DEC button down and use the data knob to select characters quickly. After naming, press EXEC to save the edits.

Turning the data knob fully counterclockwise selects a blank space. Turning the data knob clockwise (or using the INC button) selects characters in the following order:

lower case letters	a-z
upper case letters	A-Z
symbols	` \$ % @ ^ _ + * / = [] < > { } " # & () : ? ! ' , . -
numbers	0-9

6.2G Preset Mode



The screenshot shows a two-line display. The top line reads 'Preset mode: P# 7' and the bottom line reads 'Play layer a only'. The text is in a monospaced font on a dark background.

Determines how the two layers respond when you play a key. There are 2 parameters.

- **p# (preset number)** Selects the preset to be edited.

The lower line shows the second parameter. There are five choices:

- **play layer A only** Only layer A will play. Layer B will be muted.
- **play layer B only** Only layer B will play. Layer A will be muted.
- **play both layers** Plays both layers A and B. This is like traditional synthesizer "layering."
- **layer A/B switch** Allows switching between layer A and layer B according to velocity, note position, mod wheel setting, etc. as described in section 6.3G. Switching occurs only at note-on; varying a controller while the note sustains has no effect.

- **layer A/B crossfade** Allows crossfading between layer A and layer B according to velocity, note position, mod wheel setting, etc. as described in section 6.3G.

Keyboard position, velocity, and random set a particular crossfade mix that does not change after a note has started. The other options allow for the crossfade to be changed while the note is playing.

6.3 Preset Modification Screens

6.3A Assign Auxiliary Controllers



```
aux ctrls for p# 43
au1: 10 (Pan)
```

In the tone menu, there are three auxiliary control signals that can modulate various parameters. The preset menu is where you link each auxiliary control signal to a particular MIDI continuous controller. (Continuous controllers allow for remote control of synth parameters by sending messages that reflect foot pedal position, mod wheel setting, or some other physical control device. See the MIDI supplement for more details.) There are 3 parameters.

- **p# (preset number)** Selects the preset to be edited.
- **au# (auxiliary controller number)** Chooses the auxiliary control number used in the tone menu to which a MIDI continuous controller will be assigned.
- **(controller assignment)** Shows which continuous controller is assigned to the selected auxiliary controller number, or off if not continuous controller is assigned. The following controller numbers have defined functions designated by the MIDI specification, as shown in the display:

- | | |
|---|-------------------------------|
| 1 | Modulation Wheel MSB (0-127) |
| 2 | Breath Controller MSB (0-127) |
| 4 | Foot Controller MSB (0-127) |
| 5 | Portamento Time MSB (0-127) |
| 6 | Data Slider MSB (0-127) |

7	Main Volume MSB (0-127)
8	Balance MSB (0-127)
10	Pan MSB (0-127)
11	Expression MSB (0-127)
16	General Purpose #1 MSB (0-127)
17	General Purpose #2 MSB (0-127)
18	General Purpose #3 MSB (0-127)
19	General Purpose #4 MSB (0-127)
33	Modulation Wheel LSB (0-127)
34	Breath Controller LSB (0-127)
36	Foot Controller LSB (0-127)
37	Portamento Time LSB (0-127)
38	Data Slider LSB (0-127)
39	Main Volume LSB (0-127)
40	Balance LSB (0-127)
42	Pan LSB (0-127)
43	Expression LSB (0-127)
48	General Purpose #1 LSB (0-127)
49	General Purpose #2 LSB (0-127)
50	General Purpose #3 LSB (0-127)
51	General Purpose #4 LSB (0-127)
64	Sustain Pedal (0 or 127)
65	Portamento On/Off (0 or 127)
66	Sostenuto Pedal (0 or 127)
67	Soft Pedal (0 or 127)
69	Hold 2 (0 or 127)
80	General Purpose #5 (0 or 127)
81	General Purpose #6 (0 or 127)
82	General Purpose #7 (0 or 127)
83	General Purpose #8 (0 or 127)
91	External effects
92	Tremolo Depth (0-127)
93	Chorus Depth (0-127)
94	Celeste Depth (0-127)
95	Phaser Depth (0-127)
96	Data Increment (0 or 127)
97	Data Decrement (0 or 127)
98	Non-Registered Parameter LSB (0-127)
99	Non-Registered Parameter MSB (0-127)
100	Registered Parameter LSB (0-127)
101	Registered Parameter MSB (0-127)

These names will appear in the display if their associated controller numbers are selected. However, all controller numbers in the range of 0-120 are available for use, including those which do not have names. All of these controller numbers are functionally equivalent; your choice of which controllers to use for aux 1 through 3 depends mostly upon which ones your keyboard or other controlling device is able to send.

Note that mod wheel (controller 1) is already available as a control source in the SP without assigning it as an aux controller. Also, controllers 1-10, 64, and 69 are commonly used according to their standard definitions, and should generally not be assigned as aux controllers so they can be used for their specified purposes.

6.3B Layer Detune



```
layer detune: p# 0
cents: 0 (-a +b)
```

Sets the detuning of one layer with respect to the other. This is useful for creating chorusing effects, as well as compensating for tuning inaccuracies in samples used in different layers. There are 2 parameters.

The **(-a +b)** in the lower right corner reminds you that positive values will detune layer A flat and layer B sharp.

- **p#** (preset number) Selects the preset with the layers to be detuned.
- **cents** Determines the amount of detuning, up to 99 cents. Positive values tune layer A flat and layer B sharp; negative values tune layer A sharp and layer B flat.

6.3C Layer Detune Modulation



```
lyr detun mod: p# 0
src: keybdpos amt: 0
```

Modulates the detuning of one layer with respect to the other. This creates dynamic chorusing and detuning effects. There are 3 parameters.

- **p#** (preset number) Selects the preset with the layers to be detuned via a particular modulation source.
- **src** There are nine control source options:

- | | |
|-----------------|--|
| keybdpos | The amount of detuning depends on the keyboard note being played. |
| velocity | Detuning occurs according to note velocity. |
| random | Selects random layer detuning. |
| modwheel | Assigns the modwheel to vary layer detuning. |
| chanpres | Assign keyboard channel pressure to vary layer detuning. |
| polypres | Assigns keyboard polyphonic (key) pressure to vary layer detuning. |
| auxctl1 | Assigns auxiliary modulation source 1 (as assigned to a MIDI continuous controller number on the Assign Auxiliary Controller page, section 6.3A) to vary layer detuning. |
| auxctl2 | Assigns auxiliary modulation source 2 (as assigned to a MIDI continuous controller number on the Assign Auxiliary Controller page, section 6.3A) to vary layer detuning. |
| auxctl3 | Assigns auxiliary modulation source 3 (as assigned to a MIDI continuous controller number on the Assign Auxiliary Controller page, section 6.3A) to vary layer detuning. |
- **amt** Determines the maximum amount of detuning effect (in cents) resulting from the controller input. Positive values tune layer A more flat and layer B more sharp as the controller value increases; negative values tune layer A more sharp and layer B more flat.

6.3D Maximum Polyphony



max Polyphony: p# 0
group: 0 Polyph: 16

Note allocation is the process that decides which notes will be stolen when the SP's 16-note polyphony is exceeded. See the beginning of this chapter for more information. This screen determines the maximum number of notes each group can play at one time. There are 3 parameters.

- **p#** (preset number) Selects the preset containing the group(s) whose maximum polyphony is to be set.
- **group** Chooses the group (0-7) whose maximum polyphony is to be set.

- **polyph** Choose the maximum number of notes (from 0, meaning the group will never play, to 16, meaning that the group can use all available SP voices).

6.3E Allocation Priority

```
alloc priority p# : 0
group:0 - - prio:low
```

Allocation priority determines which notes will be stolen first for playback if the SP's 16-voice polyphony is exceeded. This is usually based on a note's DCA value so that the lowest level notes are stolen first. Notes being held are always higher priority than notes that aren't being held.

There are four levels of priority (low, medium, high, and absolute) to determine how important the group of notes is. Absolute means nothing can steal the voice; you have to let go of the key and let it die off. Even another note in the same group can't steal it.

There are 3 parameters.

- **p#** (preset number) Selects the preset containing the group(s) whose allocation priority is to be edited.
- **group** Chooses the group (0-7) whose allocation priority is to be set.
- **prio** Chooses the priority (low, medium, high, absolute).

6.3F Note Overflow

```
note overflow: p# : 0
group:0 - overflo:off
```

Note overflow sets whether notes of a particular group will "overflow" to a second SP if the available polyphony is exceeded.

There are 3 parameters.

- **p#** (preset number) Selects the preset containing the group(s) whose allocation priority is to be set.
- **group** Chooses the group (0-7) whose overflow characteristics are to be set.
- **overflow** Selects between on (notes will overflow) and off (notes will not overflow). *Caution:* Do not set overflow to on when a group (or any of its “members”) is set for exclusive mode at the map level, as this can cause unpredictable results. See also section 5.2E (discussion of groups in the map section).

6.3G Switch/Crossfade Controller

```
lyr swit/xfade p# 0
ctrl source: keybdpos
```

Determines which controller causes layer switching or crossfading. There are 2 parameters.

- **p#** (preset number) Selects the preset containing the layers to be edited.
- **ctrl source** There are nine control source options:

keybdpos	Switching or crossfading occurs around the keyboard note specified on the switch/xfade pivot point page (section 6.3H).
velocity	Switching or crossfading occurs according to note velocity. <i>Example:</i> Assign soft instrument sounds to layer A and loud sounds to layer B. Switch between them so that low velocities trigger layer A and high velocities trigger B.
random	Switches or crossfades randomly between the two layers.
modwheel	Assigns the modwheel to switch or crossfade between the two different layers.
chanpres	Assign keyboard channel pressure to switch or crossfade between two different layers.
polypres	Assigns keyboard polyphonic (key) pressure to switch or crossfade between two different layers. When switching, this always plays layer A.
auxctl1	Assigns auxiliary modulation source 1 (as assigned to a MIDI continuous controller

number on the Assign Auxiliary Controller page, section 6.3A) to crossfade between layers.

Example: A MIDI controller value of 0 would select layer A; 127 would select layer B; 64 would play both layers at equal level if the pivot point is set to 64.

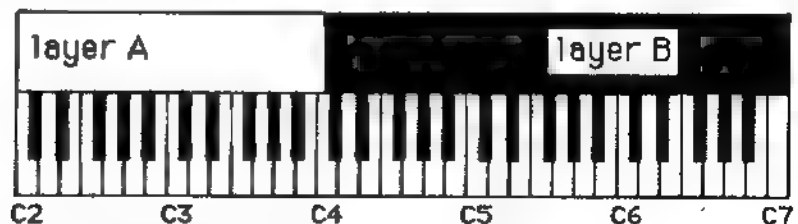
- auxctl2** Assigns auxiliary modulation source 2 (as assigned to a MIDI continuous controller number on the Assign Auxiliary Controller page, section 6.3A) to crossfade between layers.
- auxctl3** Assigns auxiliary modulation source 3 (as assigned to a MIDI continuous controller number on the Assign Auxiliary Controller page, section 6.3A) to crossfade between layers.

6.3H Switch/Crossfade Pivot Point

```
lyr swit/xfade P# 3  
Pivot Point: C 4
```

Sets the keyboard note, or decimal value associated with a non-note MIDI control source (e.g., mod wheel), at which the switching or crossfading occurs. (If display notes using in the Global menu is set to decimal number value instead of note name and octave, then the keyboard note will also be shown in decimal.)

Examples: Here's an example of the classic "keyboard split." If you select Middle C (C4, or note 60) as the pivot point and use keyboard note position-controlled switching, layer A will play at note 59 and below, while layer B will play from note 60 on up.



If you use keyboard position-controlled crossfading, the pivot point is where both layer levels are equal due to the crossfading action. Above note 60, layer B will fade in more and layer A will fade out more. Below note 60, layer B will fade out more and layer A will fade in more. The range over which the crossfade occurs is set on the Layer Crossfade Region Width page (next section).

There are 2 parameters.

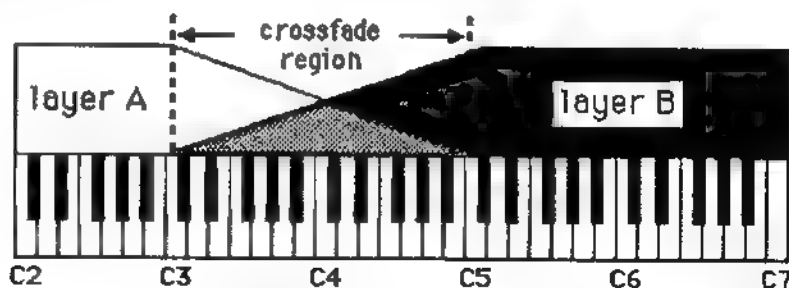
- **p# (preset number)** Selects the preset containing the layer pivot point to be edited.
- **pivot point** Chooses the desired MIDI note or controller value for the pivot point.

Special note: When random is used as the control source for layer A/B switch, the pivot point sets the relative probability of hearing each layer. Each layer will play equally with a value of 64; you'll hear layer B more frequently if the number goes below 64. With a setting of 0, you will almost always hear layer B.

6.3I Layer Crossfade Region Width

lyr crossfade p# 9
region width: 17

Sets the keyboard or controller range over which crossfading occurs if layer A/B is selected. The crossfade region is always centered upon the specified switch crossfade pivot point (see previous section). *Examples:* If you use the mod wheel to crossfade between layers and set the region width to 127, then rotating the mod wheel away from you will fade in one layer while fading out the other layer. If you use the keyboard to crossfade between layers, select Middle C (60) as the pivot point, and set the width to 25, the crossfading action will occur over a two octave range, which pivots on MIDI note 60 (middle C, or C4).



If you use keyboard position-controlled crossfading, the pivot point is where both layer levels are equal due to the crossfading action, and the crossfade region is the range over which the crossfade occurs.

There are 2 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **region width** Chooses a note range in semitones (or MIDI control source number range) around the pivot point.

Note that the region width can be set as wide as 255, even though note and controller values are limited to a 0-127 range. These larger values produce a more gradual crossfade where neither layer ever fades out completely.

Setting the width to 1 produces a very abrupt switching action, which is different from layer switching because you can switch back and forth while a note is sustaining.

6.3J Select Maps



The image shows a monochrome screen with the text 'select maps:' at the top. Below it, there are four parameters: 'p#', 'lyr', 'lev', and 'm#'. Each parameter has a value next to it, separated by a colon. The values are: 'p# 0', 'lyr: a', 'lev: 1', and 'm# 0'. There is a right arrow pointing to the 'm# 0' value.

Assigns up to four maps to layer A and up to four maps to layer B. These selections can be different for each of the two layers. Multiple maps in a layer are not stacked; rather, when a note is received, it will be played via *one* of the selected maps in a layer, as determined by the layer map switching setup. (By default, when a preset is created, only map level 1 will play; to get more map levels to play, the map level switching setup must be changed accordingly—see the next two sections.) There are 4 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** (layer) Selects the layer, A or B, to which maps will be assigned.
- **1-4** Selects one of the four layer “levels” to which you can assign a map.
- **m#** Assigns the selected map to the selected layer “level.”

As either the layer level or its selected map changes, the name of the associated maps flashes on the top line. This also happens if you press the EXEC button and the cursor is in the layer level number or map number parameter.

6.3K Map Switch Control

```
map switch ctl p# = 0  
lyr:a = src:velocity
```

When using multiple maps in a layer, this chooses the controller source (velocity, pressure, mod wheel, etc.) that will direct the switching between maps levels in that layer. The control source can be different for each of the two layers. The actual switching points are determined on the next screen.

The switching occurs only once at note-on, even if a continuous controller source is used. There are 4 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **src** There are eight control source options:

keybdpos	Switching occurs according to the keyboard note being played.
velocity	Switching occurs according to note velocity.
random	Selects random map switching.
modwheel	Assigns the modwheel to vary map switching.
chanpres	Assign keyboard channel pressure to vary map switching.
auxctl1	Assigns auxiliary modulation source 1 (as assigned to a MIDI continuous controller number on the Assign Auxiliary Controller page, section 6.3A) to vary map switching.
auxctl2	Assigns auxiliary modulation source 2 (as assigned to a MIDI continuous controller number on the Assign Auxiliary Controller page, section 6.3A) to vary map switching.
auxctl3	Assigns auxiliary modulation source 3 (as assigned to a MIDI continuous controller number on the Assign Auxiliary Controller page, section 6.3A) to vary map switching.

6.3L Map Switch Points

```
map switch pts p# : 0
lyr: a - 1<<<2<<<3<<<4
```

When using multiple maps in a layer, this screen specifies the controller ranges to which each map level in that layer will respond. As one example, this can create a very expressive piano preset by using velocity to switch between a map containing soft piano samples, a map containing loud piano samples, and one or two maps containing piano samples played at intermediate loudnesses. This recaptures some of the subtle, complex tonal variations which result from dynamic playing on a real piano, and is more realistic than relying upon simple DCA or DCF modulation to create dynamics.

For each layer, there are three switch point parameters which set up the “boundaries” between the four map levels in that layer. Each of the switch points is shown in the display as a value situated between two of the four map level numbers also appearing in the display (e.g., the value between 1 and 2, 2 and 3, and 3 and 4). Each switch point indicates the note velocity, controller value, or note number (as selected on the previously-described screen) which acts as the boundary between the two map levels. (Sometimes a set of arrows will appear in place of one or more switch point values. The meaning of this is explained below.)

The specified switch points can be different for each of the two layers. There are 5 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **1—2 switch point.** The field between 1 and 2 in the lower line sets the switch point between map levels 1 and 2. *Example:* If the switch point is velocity 64, map 1 will play at lower velocities and map 2 will play at higher velocities.
- **2—3 switch point.** The field between 2 and 3 in the lower line sets the switch point between map levels 2 and 3.
- **3—4 switch point.** The field between 3 and 4 in the lower line sets the switch point between map levels 3 and 4.

To set the limit, move the cursor under the appropriate arrow depending on which digit you want to edit, and use the INC/DEC buttons to set the limit points. By adjusting the switch points, you can activate any combination of the four map levels in a layer. To avoid confusion when doing this, keep the following in mind:

- *The 1-2 switch point can never be higher than the 2-3 switch point, and the 2-3 switch point can never be higher than the 3-4 switch point.* In programming terms, this means that level 1 will play at the lowest velocity, level 2 at higher velocities than level 1, and so on. The SP enforces this rule automatically. As you change one switch point, others may also change if this is required to keep them in correct order. (This can sometimes be used as an editing shortcut. For example, to restore the default switching setup where only level 1 is enabled, cursor into the 1-2 switch point field and increment this value to the maximum. This will also push the 2-3 and 3-4 switch points up to the maximum, leaving levels 2-4 disabled.)
- *Arrows pointing away from a level number indicate that this level is disabled and will never play.* The “active” range of switch points is 1-127. If a switch point is adjusted down to the minimum value of zero, it is displayed as right-pointing arrows to show that the map level to the left of this switch point is disabled. Likewise, if a switch point is adjusted up to the maximum of 128, it is displayed as left-pointing arrows to show that map levels further to the right are disabled. (This is the initial default value for all switch points in a newly-created preset, which is why you will often see nothing but left-pointing arrows in this screen. This indicates that only map level 1 is active and it will play at all velocities.) Finally, if two adjacent switch points are set to the same value, the map level which sits between them is bypassed during playback since the switch points have not left any “space” in which this level can respond. The leftmost of these two switch points will be displayed as left-pointing arrows to show that the map level between them is disabled.
- *There is never any overlap between levels, nor is it possible to create “dead zones” in these switching setups.* In other words, a single note cannot play through two or more map levels in a layer, nor can it “miss” completely. Regardless of how the switch points are specified, there is no velocity that will not get played through one of the specified layers.

The above examples assume that velocity is being used as the map switch control source. Remember that this is only one of several possibilities.

Examples: In the default setting, map level 1 is the only one that will play. Therefore, the 2<<<3 and 3<<<4 arrows all point toward map level 1.

If the setting is as follows: 1 60 2 100 3<<<4, then map level 1 will play from 0 to 59, level 2 from 60 to 99, and level 3 will cover all the remaining range because level 4 is disabled (as indicated by the left-pointing arrows).

6.3M Note Transpose



```
note transpose p# = 0
lyr:a t.pose:+12
```

Allows each layer to be transposed up to ± 99 semitones. This is different from the map zone transpose function because instead of transposing the actual sounds, as is done on the map menu, this simply changes the received note number before playing it through the preset, in effect redirecting the note to a higher or lower position on the keyboard. This is an easy way to create octave effects and other parallel harmonies. There are 3 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **tpose** Selects the amount of transposition. Positive numbers transpose up, negative numbers transpose down.

6.3N Pitch Bend Range

```
pitch bend: p# 0  
lyr:a bendrange:-12
```

Determines the maximum pitch bend range for a layer when receiving a full-range MIDI bend message. The range is variable from 0 (layer does not respond to pitch bend) to ± 24 semitones. There are 3 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **bendrange** Selects the pitch bend range, in semitones. With positive numbers, higher-value pitch bend messages raise the pitch. With negative numbers, higher-value pitch bend messages lower the pitch.

6.3O Output Assignment

```
output assign: p# 0  
lyr:a output:(map)
```

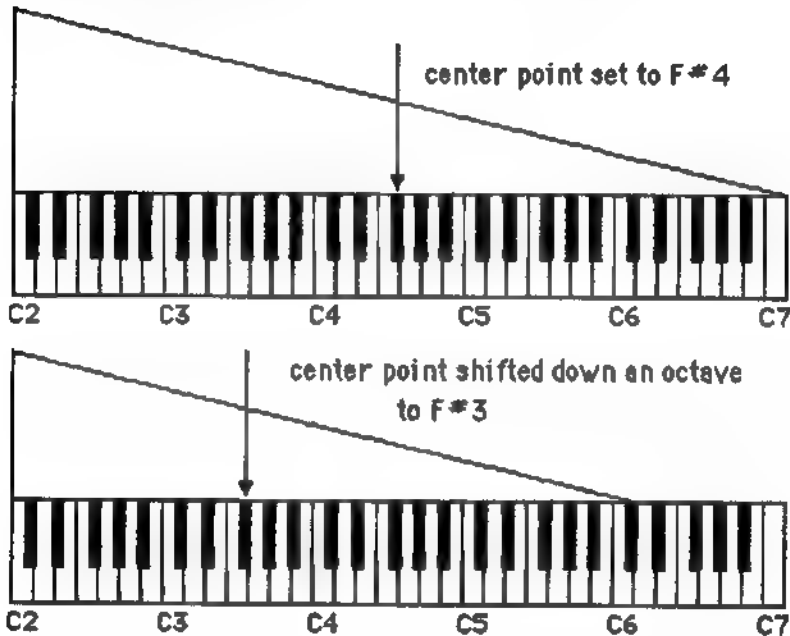
Determines whether a layer feeds the main output, sub output, or follows the output assignments from the map menu (section 5.0C). There are 3 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **output** Selects among:
 - (map) Follows the output assignments programmed in the map menu.
 - main The layer output feeds the main outputs.
Overrides map audio output assignments.
 - sub The layer output feeds the sub outputs.
Overrides map audio output assignments.

6.3P Keyboard Track Center

```
kbd trk center p# 0  
lyr:a note:C 4
```

This screen sets the keyboard tracking center point, which affects the Pan and DCF keyboard tracking functions. The following examples show how shifting the keyboard center point affects tracking.



There are 3 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **note** Chooses the center note for the keyboard tracking function.

6.3Q Pan Position

```
pan: [knob] p# 0  
lyr:a (map) kbd:+50
```

Places the audio output for each layer anywhere in the stereo field for the selected set of outputs, or follows the assignments in the map menu (section 5.0C). There are 4 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **pan** The data knob selects from hard left to hard right; the center position selects (map), where the layer follows the map menu's programmed pan assignments.



- **kbd** Sets how panning will be affected by keyboard tracking. With positive numbers, panning goes from left to right as you play higher up on the keyboard. With negative numbers, panning goes from right to left as you play higher up on the keyboard.

6.3R Volume

```
volume: [knob] p# 0  
lyr:a offset:+12
```

Varies the volume (level) for each layer by adding an offset to the map menu volume values (section 5.0B). There are 3 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **offset** Offsets the volume values programmed in the map menu. Positive numbers increase the map volume values by the amount shown in the display, and negative numbers lower the map volume values.

6.3S Filter Cutoff

```
filter cutoff: p# - 0  
lyr:a fc:+10 kbd:-20
```

Varies the filter cutoff for each layer by adding an offset to the tone menu filter cutoff values (section 4.2F). There are 4 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **fc** (filter cutoff) Offsets the filter cutoff values programmed in the tone menu. Positive numbers increase the tone filter cutoff values by the amount shown in the display, and negative numbers lower the tone filter cutoff values.
- **kbd** Sets how filter cutoff will be affected by keyboard tracking. With negative numbers, the filter cutoff decreases as you play higher up on the keyboard. With positive numbers, the filter cutoff increases as you play higher up on the keyboard.

6.3T DCA Vel-Sens

```
dca vel-sens: p# - 1  
lyr:b offset:-14
```

Varies the DCA velocity sensitivity for each layer by adding an offset to the tone DCA velocity sensitivity values (section 4.2D). There are 3 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **offset** Offsets the DCA velocity sensitivity values programmed in the tone menu. Positive numbers increase the tone DCA velocity sensitivity values by the amount shown in the display, and negative numbers lower the tone DCA velocity sensitivity values. *Exception:* The value here is used directly for the velocity sensitivity setting if the tone

DCA velocity sensitivity is set to 0. (When editing tones, you might want to set all the tone DCA velocity sensitivity values to 0 if all the vel-sens values are going to be the same, since you can set them all on this page by just adjusting one parameter.)

6.3U DCF Vel-Sens

```
dcf vel-sens:  P# 5  
lyr:a  offset: +7
```

Varies the DCF velocity sensitivity for each layer by adding an offset to the tone DCF velocity sensitivity values (section 4.2G). There are 3 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **offset** Offsets the DCF velocity sensitivity values programmed in the tone menu. Positive numbers increase the tone DCF velocity sensitivity values by the amount shown in the display, and negative numbers lower the tone DCF velocity sensitivity values. *Exception:* The value here is used directly for the velocity sensitivity setting if the tone DCA velocity sensitivity is set to 0. (When editing tones, you might want to set all the tone DCF velocity sensitivity values to 0 if all the vel-sens values are going to be the same, since you can set them all on this page by just adjusting one parameter.)

6.3V DCF Env Amount

```
dcf env amt:  P# 3  
lyr:a  offset: -20
```

Varies the DCF envelope amount for each layer by adding an offset to the tone DCF envelope amount values (section 4.2F). *Caution:* The net DCF envelope amount must be non-zero for DCF velocity sensitivity to be effective. There are 3 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **offset** Offsets the DCF envelope amount values programmed in the tone menu. Positive numbers increase the tone DCF envelope amount values by the amount shown in the display, and negative numbers lower the tone DCF envelope amount values. *Exception:* The value here is used directly for the velocity sensitivity setting if the tone DCA velocity sensitivity is set to 0. (When editing tones, you can set all the tone DCF envelope amount values to 0 if all the envelope amount values are going to be the same, since you can set them all on this page by just adjusting one parameter.)

6.3W DCO Mod Range

```

dco mod range: p# 0
lyr:a semitones: 12
  
```

Limits the range of DCO modulation (other than pitch bend) for each layer to a maximum number of semitones. With a small note range, DCO modulation produces relatively subtle effects. With a large note range, you can get some really strange effects. There are 3 parameters.

- **p#** (preset number) Selects the preset to be edited.
- **lyr** Selects between layer A or B.
- **semitones** Selects the maximum number of semitones over which DCO modulation will occur.

Chapter 7 Global Menu

7.0 About the Global Menu

Global menu parameters affect the SP's overall operation as opposed to particular subsections (e.g., map, preset, tone, wave, disk, etc.). This is where you adjust functions such as MIDI reception modes, different display options, master tuning, the SP's SCSI device ID, etc.

7.1 About Various MIDI Reception Modes

MIDI allows for several reception modes.

- In *Omni* mode, the SP receives all incoming MIDI information on all MIDI channels.
- In *Poly* mode, the SP receives incoming MIDI information from any *one* of the 16 MIDI channels.
- *Multi* mode is a special mode that combines Omni mode (since the SP "listens" to all 16 channels) and Poly mode (since data from individual channels can be routed to individual presets).

In Multi mode, an individual preset can be assigned to any one of the 16 available MIDI channels, and have its own volume level. Thus, a single SP can produce multiple instrument sounds, which is called *multi-timbral* operation. This is particularly useful when sequencing, since you can play back several tracks of parts with one device. Each multi channel responds to MIDI continuous controller 7 messages to vary the level, and to program change commands to switch preset numbers.

Multis are brought into a special memory buffer within the SP. Therefore, any changes that update the multi, such as program changes and volume level changes, are not saved with the multi *unless* you specifically save the multi to a particular multi number.

7.2 MIDI Screens

7.2A Omni/Poly Preset



omni/poly preset:
P# - 0 [PRESET NAME]

Selects the preset number the SP will play in MIDI Omni or Poly mode. This is the most common option when playing the SP from a MIDI keyboard. (In Multi mode, several different presets can be assigned to several different MIDI channels; see section 7.1). There is 1 parameter.

- **p#** Selects the preset number to be played.

7.2B Omni/Poly Channel & Volume



omni/poly channel: 1
main volume: 127

Selects the MIDI channel over which data is received in Poly mode. Also sets the master volume level for the SP in both Omni and Poly modes. There are 2 parameters.

- **channel** Selects the base channel number.
- **main volume** Selects the SP's master volume level in both Omni and Poly modes (0 is minimum level, 127 is maximum).

7.2C MIDI Receive Mode



midi rcv mode: omni

Selects the MIDI reception mode. There is 1 parameter.

- **rcv mode** Offers the following options:

omni	All 16 incoming MIDI channels play back through whatever preset you've selected.
poly	The SP responds only to data appearing on the base MIDI channel.
multi	Channel and program assignments are set in the Edit Multi Setup folder (section 7.3).

7.2D MIDI Note Play Mode

midi notes play:
what's on display

Selects whether a preset, map, tone, or wave will play in response to incoming notes. Generally you will choose **per MIDI rcv mode** or **what's on display**, but the other options are useful when editing at a specific level for locking the SP in to that level. There is 1 7.3 parameter.

per MIDI rcv mode

Plays normally in Omni, Poly, or Multi mode according to the current MIDI receive mode setting.

what's on display

Plays the displayed preset, map, tone or wave, depending on which menu is currently active. In other menus (e.g., disk, global, utility), reverts to current MIDI receive mode.

current edit preset

Locks in the preset being edited regardless of which screen is active.

current edit map

Locks in the map being edited regardless of which screen is active.

current edit tone

Locks in the tone being edited regardless of which screen is active.

current edit wave

Locks in the wave being edited regardless of which screen is active.

7.3 Edit Multi Setup (Folder)

```
edit multi setup  
--- <press exec>
```

Press EXEC to access the pages in the "folder."

7.3A • Copy Name To Multi

```
copy name ---> multi  
w# 0 [WAVE NAME]
```

Copies a wave, tone, map, or preset name to the multi. This saves time if a suitable multi name already exists, since you don't need to enter the multi name from scratch. There are 2 parameters.

- **w# t# m# p#** (wave, tone, map, or preset) Selects whether the name to be copied is a wave, tone, map, or preset name.
- **name** Selects the actual wave, tone, map, or preset name.

Press EXEC to copy the selected name to the multi. The display shows operation complete.

7.3B • Edit Multi Name

```
edit multi name:  
--- [MULTI NAME]
```

To edit the multi name, press EXEC.

```
<exec to save edit>  
--- [MULTI NAME]
```

The second screen has 1 parameter (for multi naming).

- **(Multi name)** Use the cursor buttons to choose the character to be edited and the data knob to select the character. After naming, press EXEC to save the edits.

Turning the data knob fully counterclockwise selects a blank space. Turning the data knob clockwise (or using the INC button) from the blank space selects characters in the following order:

lower case letters	a-z
upper case letters	A-Z
symbols	` \$ % @ ^ _ + * / = []
	< > { } " # & () : ? ! ' , . -
numbers	0-9

7.3C • Multi Enable



The screenshot shows a black background with white text. The first line reads 'multi enable: chn: 1' and the second line reads 'is: active'. There are some faint, blurry marks around the text.

Enables or disables individual channels for MIDI reception. There are 2 parameters.

- **channel** Selects the MIDI channel to be enabled or disabled.
- **is:** Selects between:

active	The preset assigned to this channel will respond to MIDI data.
off-line	The preset assigned to this channel will not respond to MIDI data.

Shortcut: On this screen, the EXEC button toggles between the active and offline settings. You can leave the cursor under the **chn** parameter to change channels rapidly, and use EXEC to switch the channel status.

7.3D • Multi Preset Assignment

```
multi preset: chn: 1  
p# 0 [PRESET NAME]
```

Assigns particular presets to particular MIDI channels. There are 2 parameters.

- **channel** Selects the MIDI channel to which an SP preset will be assigned.
- **p#** Selects the preset to be associated with the selected MIDI channel.

7.3E • Multi Volume

```
multi volume: chn: 1  
127
```

Sets the master level for each preset used in a multi. The volume parameter is updated if MIDI controller 7 (master volume) data appears on this channel.

There are 2 parameters.

- **channel** Selects the MIDI channel whose associated preset's level is to be adjusted.
- **(0-127)** Selects the volume level for the associated channel's preset. 0 is minimum volume, 127 is maximum.

7.3F • Multi Maximum Polyphony

```
multi max polyphony:  
group:0 polyph:16
```

Note allocation is the process that decides which notes will be stolen when the SP's 16-note polyphony is exceeded (see section 6.1 for more information). This screen determines the maximum number of notes each group (as assigned at the map level) can play at one time.

This screen is identical to the “multi maximum polyphony” screen in the preset menu. However, the global settings override settings made at the preset level since the polyphony settings of any presets active under the multi setup might conflict with one another. There are 2 parameters.

- **group** Chooses the group (0-7) whose maximum polyphony is to be set.
- **polyph** Choose the maximum number of notes (from 0, meaning the group will never play, to 16, meaning that the group can use all available SP voices).

7.3G • Allocation Priority



The screenshot shows a black background with white text. The first line reads 'multi alloc priority' and the second line reads 'GROUP:0 prio:low'.

Allocation priority determines which notes will be stolen first for playback if the SP’s 16-voice polyphony is exceeded. This is usually based on a note’s DCA value so that the lowest level notes are stolen first. Notes being held are always higher priority than notes that aren’t being held.

There are four levels of priority (low, medium, high, and absolute) to determine the importance of the group of notes. **Absolute** means nothing can steal the voice—you have to let go of the key and let it decay fully. Even another note in the same group can’t steal it.

This screen is identical to the “allocation priority” screen in the preset menu. However, the global settings override settings made at the preset level since the allocation priority settings of any presets active under the multi setup might conflict with one another. There are 3 parameters.

- **p#** (preset number). Selects the preset containing the group(s) whose allocation priority is to be edited.
- **group** Chooses the group (0-7) whose allocation priority is to be set.
- **prio** Chooses the priority (low, medium, high, absolute).

7.3H • Note Overflow



```
multi note overflow:  
group:0 overflow:off
```

Note overflow sets whether notes of a particular group will “overflow” to a second SP if the available polyphony is exceeded.

This screen is identical to the “multi note overflow” screen in the preset menu. However, the global settings override settings made at the preset level since the overflow settings of any presets active under the multi setup might conflict with one another. There are 3 parameters.

- **p#** (preset number). Selects the preset containing the group(s) whose allocation priority is to be set.
- **group** Chooses the group (0-7) whose overflow characteristics are to be set.
- **overflow** Selects between on (notes will overflow) or off (notes will not overflow). *Caution:* Do not set overflow to on when a group is set for exclusive mode at the map level, as this can cause unpredictable results.

7.3I • Store Multi



```
store multi to:  
mu# 0 [MULTI NAME]
```

An SP bank can hold up to 16 multis (0-15). This screen lets you save a multi setup within the bank.

Save a multi if you expect it to receive external MIDI volume or program change commands and want to recall the original version later. You can also save a multi that has been modified by MIDI volume or program change commands if you prefer the modified multi setup. There is 1 parameter.

- **mu#** Selects the multi number under which the current multi will be stored. The lower line shows the multi name (see section 7.3B).

Press EXEC to store the multi setup. *Note:* If you are only using one multi per bank, you needn't really concern yourself with the store, recall, and delete functions. The current multi buffer settings are saved to and loaded from disk (along with any stored setups) when a bank is saved to or loaded from disk.

7.3J • Recall Multi

```
recall multi from:
mu# 0 [MULTI NAME]
```

Loads one of the SP multis previously stored in a bank. There is 1 parameter.

- **mu#** Selects the multi number to be loaded. The lower line shows the multi name (see section 7.3B).

Press EXEC to load the multi.

7.3K • Delete Multi

```
delete stored multi:
mu# 0 [MULTI NAME]
```

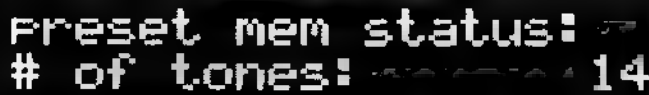
Deletes one of the SP multis previously stored in a bank. There is 1 parameter.

- **mu#** Selects the multi number to be deleted. The lower line shows the multi name (see section 7.3B).

Press EXEC to delete the multi. *Note:* This has no effect on the current multi buffer.

7.4 Global Utility Screens

7.4A Preset Memory Status



```
Preset mem status:
# of tones: 14
```

Shows information about what's currently in non-wave memory. The last two functions are mostly to see if you are having problems building new presets or maps due to memory limitations. There is 1 parameter.

- **(status parameter)** Selects the type of information to be displayed. Options are:

# of tones	Shows the number of tones in the bank (each tone requires 80 bytes).
# of maps	Shows the number of maps in the bank (each map requires from 48 to 6144 bytes, depending on the map data).
# of presets	Shows the number of presets in the bank (each preset requires 256 bytes).
# of multis	Shows the number of multis in the bank (each multi requires 96 bytes).
free bytes	The memory used to build presets and maps is separate from the wave memory. This shows how many bytes are left in preset memory.
map array	Shows the total amount of memory being used by all maps in the bank.

7.4B Master Tuning

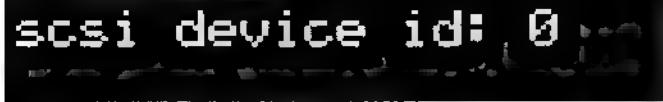


```
master tune: 0.00
```

Tunes the SP to other instruments or concert pitches other than A=440. There is one parameter.

- **master tune** Selects the tuning offset. Numbers to the left of the decimal point are in semitones; to the right, in cents (1/100th of a semitone).

7.4C SCSI Device ID

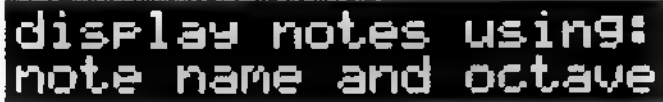


A screenshot of a menu item labeled 'scsi device id: 0'. The text is in a monospaced font on a black background.

Selects the SP's SCSI device ID so that it can be addressed by other devices connected to the SCSI bus. To have the SP always default to a particular SCSI ID on power-up, set the ID number with the hardware jumper setting (section 9.2). However, you can always override the default on this screen. There is 1 parameter.

- **SCSI device ID** Selects the SCSI ID number from 0 to 7. When set to 7, the SP cannot be addressed as a target device but can still act as an initiator device (e.g., to address a hard disk or DPM SX).

7.4D Note Display Mode



A screenshot of a menu item labeled 'display notes using: note name and octave'. The text is in a monospaced font on a black background.

Selects how notes are to be displayed. There is 1 parameter.

- **(display type)** Selects the note display mode. There are two options:

note name and octave

Shows a note name, such as C#, and its corresponding octave (*example: C#4, B2*).

decimal number value

Notes are shown as decimal numbers, according to the MIDI spec. *Examples: Middle C is 60, one octave lower is 48.*

7.4E Edit Selections

```
edit selections:  
track MIDI input
```

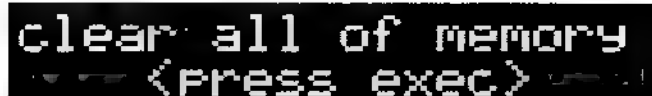
Determines whether the displayed map, map zone, preset, tone, or wave will change automatically in response to data from a MIDI keyboard or can only be changed manually. There is 1 parameter with two options:

- **track MIDI input** Screen parameters are updated from the data generated by a MIDI keyboard or other MIDI controller. This is a great time-saver for all types of editing since it allows you to select the items to be edited by simply playing notes on the keyboard. Track MIDI input also makes it easy to see which tone is playing on a given key, or exactly which map zones a specific note belongs to.

Note: If “MIDI notes play” (section 7.2D) is set to **what’s on display** then much of the advantage of this mode is lost because MIDI notes will simply play the displayed item, so no screen changes will occur. Instead, choose **per MIDI receive mode, current edit preset, or current edit map** for “MIDI notes play.”

- **change manually only** The screen won’t follow changes as you play, and items to be edited can only be selected manually. For example, this is useful when it necessary to edit a single tone or map zone while listening to it in relation to the complete map, or to edit a specific preset while a multi-timbral sequence is being played back through the SP. “Change manually only” keeps the screen from jumping around uncontrollably, which would make such an edit impossible.

7.4F Clear All SP Memory

A screenshot of a screen with a black background and white text. The text reads "clear all of memory" on the first line and "<Press exec>" on the second line. There are some faint, blurry marks around the text, possibly from the original document's scanning process.

This clears all SP memory—waves, tones, maps, presets, multis, everything.

To clear the memory, press EXEC. Because this is a pretty drastic operation, there is a warning screen that alternates between four displays:

- this will wipe out everything in mem!!
- are you fairly sure you want to do this?
- press <exec> button to continue
- press <global> button to quit

Press EXEC. The display shows **operation complete** to indicate that all SP memory contents have vanished.

This Page Left Intentionally Blank.

Chapter 8 Utility Menu

8.0 About the Utility Menu

The Utility menu provides for transferring samples between the SP and SCSI devices (including the DPM SX) using the SMIDI protocol. This menu also includes several SX setup screens to control the SX's sampling rate and sample length, and provides level indication for signals entering the SX.

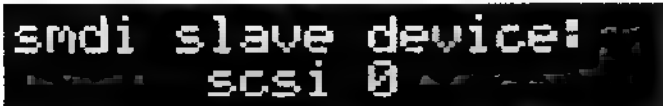
8.1 About SMDI

The original MIDI Sample Dump Standard provides a way to send digital audio over the MIDI line, but this is a slow way to transfer data. SMDI (SCSI Musical Data Interchange) is a new protocol that essentially specifies how to send MIDI Sample Dump Standard digital audio over SCSI, speeding up data transfer by a factor of 50 or greater compared to MIDI.

All other Peavey devices that support SCSI take advantage of SMDI, but Peavey is also approaching other manufacturers about adopting SMDI as a standard method of data transfer to allow for a greater degree of inter-instrument communication.

8.2 Select Device/SX SCSI Number Screen

8.2A Select SMDI Slave Device/Set SX SCSI Number



Chooses the SMDI slave device with which the SP will communicate. There is 1 parameter.

- **scsi** (SCSI device number) Selects a device to communicate with the SP over SCSI by specifying the device's associated SCSI ID number (0 to 6).

The DPM-SX sampling front end always powers up with a SCSI ID number of 0. To change this:

- 1 Remove power from the DPM SX.
- 2 Press and hold the SX's **ARM** button while restoring power to the DPM SX.
- 3 While holding the **ARM** button, press and hold the buttons indicated with an X in the table below to select the desired device number.

Device Number	Start/ Stop	MIDI Dump	Sample Rate
0			
1			X
2		X	
3		X	X
4	X		
5	X		X
6	X	X	
disabled	X	X	X

- 4 While continuing to hold these buttons, release the **ARM** button.
- 5 Release all remaining buttons. The selected device number will remain in effect until power is removed.

8.3 DPM SX Configuration (Folder)



```
DPM-SX configuration
<press exec>
```

Press EXEC to access the pages in the “folder.” Upon pressing EXEC, the SP will attempt to access the SX and read its current configuration info. If this is unsuccessful, the SP will not allow you to access the screens in the folder and instead will display an error message indicating the cause of the problem.

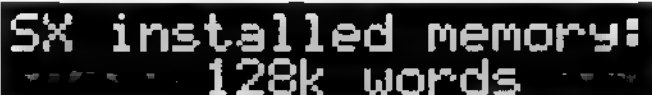
8.3A • SX Firmware Version



```
SX firmware version:
2.01
```

This reference screen shows the SX firmware version. The SP will work only with versions higher than 2.01. If you have an earlier version, contact your Peavey dealer for information on upgrading

8.3B • SX Installed Memory



```
SX installed memory:
128k words
```

This reference screen shows the amount of memory installed in the SX, in kilowords.

8.3C • SX Sample Rate Selection

```
SX sam rate: 44.10k  
<exec sends new val>
```

Controls the SX sample rate, from 12.00 kHz to 48.00 kHz. There is 1 parameter.

- **sam rate** Selects the sample rate.

To send this sample rate value to the SX, press EXEC. If the sample rate is 48 kHz, the SX's 48 kHz LED will light. If the sample rate is 24 kHz, the SX's 24 kHz LED will light. Other sample rates will cause the SX's ALT button to light. The SP display shows **operation complete** after the sample rate value has been sent.

Peavey recommends sampling into the SX at 44.10 kHz when the samples are to be transferred to the SP. Upon initially entering the configuration folder, the SP will automatically set the sample rate to 44.10 kHz. This can then be changed to any other value in the available range.

8.3D • SX Sample Length Selection

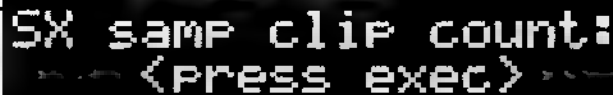
```
SX sam len: 128kw  
<exec sends new val>
```

Determines how much SX memory will be allocated for the sample you want to record into the SX. This applies only when sample is started according to input signal level trigger, instead of manually by hitting the SX start button. There is 1 parameter.

- **sam len** Selects the sample length, from 0 kilowords up to the amount of memory installed in the SX.

To send the sample length value to the SX, press EXEC. The display shows **operation complete** to confirm.

8.4B SX Sample Clip Count



```
SX samp clip count:
<press exec>
```

This function shows whether any samples were clipped (i.e., exceeded the available headroom) during the sampling process. These samples may clip for such a short period of time that it would be hard to see the 0db indicator light up; this screen provides an unambiguous indication of whether clipping occurred.

There are two screens. To access the sample clip count screen, press EXEC. The display's lower line shows how many samples were clipped. If this is a large number, as shown below, it is advisable to resample the signal since the clipping has probably resulted in audible distortion.



```
SX samp clip count:
2253 points clipped
```

8.4C Get Slave Sample

```
get slave samp# 0
using: scsi (smdi)
```

Transfers a sample from a slave SMDI devices, such as the DPM SX sampling front end, to the SP. There is 1 parameter.

- **samp# (sample number)** Selects the sample number to be transferred from the SMDI device. With the SX, this number will always be 0.

Press EXEC after selecting the sample to be transferred. Next, select an SP wave location to hold this sample, as shown on the following screen. There is 1 parameter.

```
put samp in waveloc:
# 0 -----
```

- **wave loc# (wave location number)** Selects the SP wave location into which the sample from the slave SMDI device will be transferred. If the wave location is empty, the display will show 14 dashes after the wave location number. Otherwise, the lower line will show the wave number's name. This wave will be overwritten if you transfer a sample into this location.

Press EXEC to transfer the sample from the SMDI slave device into the SP. The following screen appears. The top line shows the number of the current packet of data being transferred, and the size of the packet (in this case, 16384 kilobytes). The bottom line counts down the number of words as they are transferred from the slave device to the SP.

```
pknum: 16 16384
wds left: 131072
```

When the number of words left reaches 0, the sample has been transferred.

8.4D Send Wave

```
send wave [WAVE NAME] w# 5
```

Sends a wave from the SP to a SMDI slave device. There are 2 parameters.

- **w# (wave number)** Selects the SP wave to be transmitted to a SMDI slave device.

Press EXEC to bring up a second line with the remaining screen parameter.

- **sam#** Selects the sample number by which the wave will be identified when sent over SCSI.

```
send wave [WAVE NAME] w# 5  
to:scsi as sam# 0
```

Press EXEC to transfer the wave from the SP to the slave SMDI device.

Chapter 9 SCSI and the SP

9.0 About SCSI

SCSI (Small Computer System Interface, commonly pronounced “skuzzy”) is shrouded in legend and myth. Here’s what you need to know.

SCSI is a high-speed communications bus for computer data. It’s conceptually similar to MIDI, but operates at a much faster speed. MIDI is a serial interface that moves 10-bit data “words” one bit at a time, and can send a maximum of 31,250 bits per second; SCSI is a parallel interface that handles eight bits at a time instead of one, and can move up to around 12 *million* bits per second.

SCSI’s initial use was to move data between computers and external hard disks, which had to transfer much more data than floppies. A little while later, SCSI appeared in recording and music equipment. Rapid data transfer is important in digital audio systems, because digitized audio files use up lots of memory. Compared to SCSI, transferring samples via MIDI is very slow. However, the high speed of the SCSI bus means it’s much fussier with respect to cabling and setup than MIDI.

9.1 SCSI ID Numbers

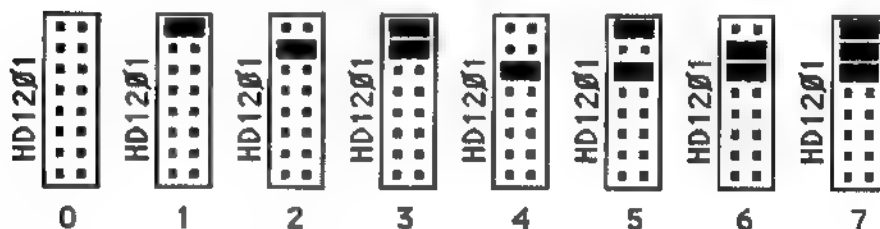
Up to eight SCSI devices can connect to the SCSI bus, and each must have its own unique ID number (from 0 to 7) so that it can be addressed individually. Most SCSI devices let you set the SCSI ID number, although some devices have fixed SCSI IDs that cannot be changed. *Example:* The Apple Macintosh has a fixed SCSI ID of 7; if it has an internal hard disk, the disk will have a fixed SCSI ID of 0.

Caution: Never set two SCSI devices to the same ID. Devices on the bus will become completely confused, which could lock up your system or cause other problems.

9.2 Setting the SP SCSI ID

The SP is shipped with a power-on default SCSI ID setting of 0, which can be changed on a temporary basis from the front panel (section 7.4C). However, the power-on default can be set to a different number by changing the position of jumper pins on a connector internal to the SP, as described in the following procedure.

- 1 With all power and cables disconnected from the SP, remove the 8 Phillips head screws that hold in the top plate.
- 2 Looking at the SP from the front, locate the large, square IC labelled as U501. Moving right from this IC, you'll find U202, U207, U208, RN1203, U1202, and finally, 16-pin header HD1201.
- 3 Use the one or more jumpers as necessary to short together different combinations of two pins, as shown below, to obtain one of the following default SCSI ID numbers.



- 4 Replace the top plate.

9.3 Initiators and Targets

The SP can serve either as a master device (initiator) in a system (i.e., it can issue commands to a Peavey SX, SCSI hard drive, etc.), or as a slave device (target) where it responds to external commands from a Macintosh or some other SCSI device.

9.4 Incompatible Hard Disk Formats

It is generally not possible to share a hard disk between a SCSI device like the SP and a computer, because each requires a different disk format. Therefore, you will generally need one hard disk formatted for the computer, and one formatted for the SP. These can be connected to the same SCSI bus, or operate independently.

One way around this is to use a hard disk with removable media, such as ones based on the Syquest drive mechanism. Then one disk cartridge could be formatted for the computer, and another, for the SP. To switch storage from one device to another, simply switch cartridges.

9.5 Incompatible SCSI Hard Disks

Although the SP has been tested with numerous hard disks, some hard disks do not follow the SCSI specification to the letter. Peavey cannot guarantee that a particular hard disk will work with the SP, unless it has been tested by Peavey. Before you buy any hard disk, test it with the SP to insure compatibility.

9.6 SCSI Connectors

There are two common types of SCSI connectors: standard 50-pin SCSI connectors, and smaller 25 pin DB-25 connectors. You may need to use an adapter cable with each end having its own connector type in order to hook the SP directly up to a SCSI device.

If a device has two SCSI connectors, you can use them interchangeably. The extra connector is there to pass the signal along to the next SCSI device.

9.7 SCSI Cabling

Rule #1: Always make any SCSI connections with the power turned off to all units. Severe damage could result to a device's SCSI controller chip if you do not follow this precaution.

Rule #2: Even with the power turned off, it is possible for you to build up a static electricity charge (like what happens when you walk on a carpet on a dry day) that can zap the SCSI circuitry. To prevent this from happening, touch something metal before connecting or disconnecting cables, and handle the cables only by their outer plastic sheaths. Do not touch any of the exposed metal pins on an SCSI connector if the other end is plugged into a SCSI device.

Rule #3: Avoid right-angle SCSI cable runs, or any types of bends that could damage the cable's internal wires. Because SCSI cables are far more delicate than MIDI or audio cables, if you use the SP live with a SCSI device, always carry a spare cable or two.

Rule #4: Use high-quality, properly shielded cables *designed specifically for SCSI operation (not printers or other non-SCSI devices)*. All cable runs should be as short as possible; problems can arise if the total length of SCSI cables is as little as 10 feet, and should be no more than 20 feet for all SCSI cables in a system. Cables should go in a daisy-chain from the first device in the chain (usually the computer), to other peripherals (hard disk, removable hard disk, CD-ROM drive, SP, etc.), to the last device in the chain. See "Typical SCSI Setups" for more information.

Extension cables are also available, but be careful to use the minimum length possible for SCSI cabling.

9.8 Termination

The SCSI bus works at such high speeds it resembles something more out of radio than audio. To insure proper impedance matches between various devices, it's crucial that the devices at each end of the chain—but *not the middle devices*—be *terminated*.

Termination provides the proper impedances and line characteristics, but having more than two terminators in the system can damage SCSI chips by drawing excessive current (a

system will not suffer damage if there aren't enough terminators, although instability may be a problem). Many SCSI devices come with internally-installed, but removable, terminators; other devices let you switch termination on or off. In some cases, termination is permanently installed. In other cases, termination is not included and an external termination device must be connected between the SCSI cable and the device to be terminated.

It is usually not obvious whether or not a device is terminated, so you must refer to the device's manual to determine whether or not it is terminated.

9.9 Typical SCSI Setups

Following are some typical SCSI setups.



This is the most common SP setup: a terminated hard drive for storage, connected by a single SCSI cable to either SP SCSI port. Since both units are terminated, the ends of the chain are properly terminated. If the hard drive is not terminated, it will require an external termination device to be connected between the cable and the hard disk SCSI connector.



Connecting the Peavey SX to the Peavey SP allows sampling your own sounds into the SP. Since both units are terminated, a single cable between the two is all that's needed.



Here's a system for those who need both storage and sample editing. A computer runs sample editing software. The hard drive

is a removable cartridge type; a cartridge with the SP format could be inserted to store SP sounds, and a cartridge formatted for the computer could store samples edited in the program along with other computer data.

The computer and SP are both internally terminated. Therefore, the hard drive must *not* be terminated to avoid having three terminators on the SCSI line.



This advanced system allows for sounds to be loaded into the SP from either a CD-ROM or hard drive. These units can also be used by the computer.

The computer and SP are both internally terminated. The CD-ROM and hard drive should have their terminators removed. If for some reason the CD-ROM or hard disk terminators could not be removed, you could remove the terminators from the SP and switch its position in the SCSI chain with the CD-ROM or hard disk.

9.10 Removing the SP's Terminators

Peavey recommends leaving in the SP's terminators and using it as the last device in a SCSI chain. However, if you must insert the SP in the middle of the SCSI chain, here's the procedure for terminator removal.

Caution: You must be familiar with electronic servicing techniques. Failure to perform this procedure properly will void your warranty. If you have any doubts about your ability to perform this operation, consult your Peavey dealer.

- 1 With all power and cables disconnected from the SP, remove the 8 Phillips head screws that hold in the top plate.
- 2 Locate the MIDI out jack.
- 3 There are two 14-pin IC packages close to the MIDI out jack, labelled RN1201 and RN1202. Carefully remove these from their sockets using an IC extraction device (available from stores such as Radio Shack) and store the terminators in a safe place in case you need them in the future.
- 4 Replace the top plate.

9.11 Power-On Order

The general rule is to turn on all SCSI peripherals first, then the computer (if present). Some SCSI devices, such as removable hard drives and CD-ROMs, may require a few seconds before they are available for operation (e.g., the hard disk may need to come up to speed). Let each SCSI device power up fully before turning on the next device.

In a system that includes the SP but does not include a computer, turn on all devices other than SP first, then turn on the SP.

Note that all devices connected to the SCSI line must be turned on whether they are being used or not. Otherwise, unreliable operation will probably result.

9.12 SCSI Troubleshooting

Like MIDI, SCSI has come a long way since it was introduced, and it shouldn't be too hard to get your system working reliably. If you encounter problems, check for:

- **Conflicting SCSI ID numbers.**
- **Device turn-on order.** If a SCSI system doesn't recognize a particular device, try turning devices on in different orders, and see if one order works better than others.
- **More than two terminators (fix this immediately and hope that no damage has occurred) or not enough terminators (there should be one on each end of the SCSI chain but none in the middle).**
- **Properly connected SCSI cables.** Make sure there are no kinks or radical bends.
- **Don't rule out the possibility that even a brand-new cable may be defective.** Always have a few spares handy you can substitute before assuming the problem is more serious.
- **With a Macintosh-based SCSI system, use a program such as Apple's HD SC utility (part of the System Utilities disk), SCSI Tools, or SCSI Probe to make sure that the bus recognizes all devices connected to SCSI.**

9.13 The Importance of Backup

Hard disks are generally very reliable, but they are still relatively fragile devices designed to sit demurely in the corner of a climate-controlled office. A hard disk can crash or fail, and with it goes your data.

To prevent this from happening, back up your sounds to floppy disk or if you can afford a second hard disk, use that for backup. A removable media hard disk is an excellent choice, as you can collect different types of sounds on different cartridges to better organize your sample library.

Chapter 10 MIDI Supplement

(This chapter is an abridged version of material from *Power Sequencing with Master Tracks Pro/Pro 4* and *The Complete Guide to the Alesis HR-16 and MMT-8*, copyright 1990 and 1989 respectively by AMSCO Publications, and is adapted with permission.)

10.0 MIDI Basics

Most current electronic instruments, including the SP, contain an internal computer. Computers and music have been working together for decades, which is not surprising given music's mathematical basis (consider frequencies, harmonics, vibrato rates, tunings, etc.). In the mid-70s, microcomputers became inexpensive enough to be built into consumer-priced musical instruments. They were used for everything from sound generation to storing parameters in memory for later recall.

In 1983, the MIDI (Musical Instrument Digital Interface) specification was introduced to better exploit the computers inside these new musical instruments, primarily to insure compatibility with equipment from different manufacturers. MIDI expresses musical events (notes played, vibrato, dynamics, tempo, etc.) as a common "language" consisting of standardized digital data. This data can be understood by MIDI-compatible computers and computer-based musical instruments.

Before electronics, music was expressed exclusively as written symbols. By translating musical parameters into digital data, MIDI can express not only the types of musical events written into sheet music, but other parameters as well (such as amount of pitch bend or degree of vibrato).

10.1 MIDI Hardware

MIDI-compatible devices usually include both MIDI in and MIDI out jacks, which terminate in 5-pin DIN-style connectors. The MIDI out jack transmits MIDI data to another MIDI device. As you play a MIDI controller such as the DPM C8 or DPM 3, data corresponding to what you play exits the MIDI out jack. *Example:*

If you play middle C, the MIDI out transmits a piece of data that says "middle C is down." If you release that key, the MIDI out transmits another piece of data that says "middle C has been released."

If the keyboard responds to the dynamics of your playing, the note data will include dynamics information too. Moving the modulation wheels and pedals attached to many synthesizers will also generate data associated with the wheel or pedal being used.

The MIDI in jack receives data from another MIDI device. In addition to the type of performance data described above, rhythmically-oriented MIDI devices (*e.g.*, drum machines) can often transmit and/or receive additional MIDI timing messages that keep other rhythmically-oriented units in a system synchronized with each other.

An optional MIDI thru jack provides a duplicate of the signal at the MIDI in jack. This is handy if you want to route MIDI data appearing at one device to another device as well.

Example: Suppose the DPM 3's MIDI out feeds the SP's MIDI in. Patching the SP's MIDI thru to a V3 rack synthesizer sends the signal present at the SP's input to the V3. Thus, playing on the DPM 3 triggers both the SP and the V3.

10.2 MIDI Message Basics

There are two main types of MIDI messages. *Channel* messages, which are channel-specific, consist of Voice and Mode messages. *System* messages, which do not have a channel number and are received by all units in a system, include Common, Real Time, and Exclusive messages.

10.3 Channel Messages

10.3A Voice Messages

A synthesizer or sampler's *voice* is the most basic unit of sound generation. Usually, each voice plays one note at a time, so the number of notes you can play at one time will be limited by the available number of voices. MIDI messages that affect voices include:

Note On Corresponds to a key being pressed down; values range from 000 (lowest note) to 127 (highest note). Middle C is 60.

Note Off Corresponds to a key being released; values are the same as note on.

Velocity Corresponds to dynamics; values range from 001 (minimum velocity) to 127 (maximum velocity). A velocity of 000 is equivalent to a note-off message.

Pressure Indicates the pressure applied to a keyboard after pressing a key.

Program Change Sending a program change command from a sequencer or other MIDI keyboard can change synth patches automatically, or in the case of the SP, call up different presets. There are 128 program change command numbers.

Note that not all units number programs consistently. Some number them as 000-127, others as 001-128, and still others arrange programs in banks of 8 programs (such as A1-A8, B1-B8, C1-C8, etc.).

Pitch Bend This "bends" a note from its standard pitch, which is excellent for creating lead guitar effects.

Continuous Controller Footpedals, breath controllers, and modulation wheels can vary sounds as you play, thus adding expressiveness. MIDI allows for 64 continuous controllers (these act like potentiometers in that you can choose one of many different values) and 58 continuous/switch controllers (these can act like continuous controllers but some are assumed to choose between two possible states, such as on/off).

Each type of controller is stamped with its own controller identification number. Not all controller numbers have been standardized for specific functions, but the following indicates the current list of assigned controllers. Numbers in parenthesis indicate the controller range.

- 1 Modulation Wheel (0-127)
- 2 Breath Controller (0-127)
- 3 Early DX7 Aftertouch (0-127)
- 4 Foot Controller (0-127)
- 5 Portamento Time (0-127)
- 6 Data Slider (0-127)
- 7 Main Volume (0-127)
- 8 Balance (0-127)
- 10 Pan (0-127)
- 11 Expression (0-127)
- 16 General Purpose #1 (0-127)
- 17 General Purpose #2 (0-127)
- 18 General Purpose #3 (0-127)
- 19 General Purpose #4 (0-127)
- 32-63 Least Significant Bits, Controllers 0-31 (0-127)
- 64 Sustain Pedal (0 or 127)
- 65 Portamento On/Off (0 or 127)
- 66 Sustain Pedal (0 or 127)
- 67 Soft Pedal (0 or 127)
- 69 Hold 2 (0 or 127)
- 80 General Purpose #5 (0 or 127)
- 81 General Purpose #6 (0 or 127)
- 82 General Purpose #7 (0 or 127)
- 83 General Purpose #8 (0 or 127)
- 92 Tremolo Depth (0-127)
- 93 Chorus Depth (0-127)
- 94 Celeste Depth (0-127)
- 95 Phase Depth (0-127)
- 96 Data Increment (0 or 127)
- 97 Data Decrement (0 or 127)
- 98 Non-Registered Parameter MSB (0-127)
- 99 Non-Registered Parameter LSB (0-127)
- 100 Registered Parameter MSB (0-127)
- 101 Registered Parameter LSB (0-127)
- 121 Reset All Controllers (0)
- 122 Local Control On/Off (0 or 127)
- 123 All Notes Off (0)
- 124 Omni Off (0)
- 125 Omni On (0)
- 126 Mono On (0-16; 0=Omni Off)
- 127 Poly On (0)

10.3B Mode Messages

There are two messages that determine the MIDI mode (*i.e.*, how the DPM 2 will receive MIDI data). The “omni” message determines how many channels will be recognized. Omni on means that data from all channels will be received; Omni off limits the number of channels, usually to one.

The “mono/poly” message deals with voice assignment within the synthesizer. In Mono mode, only one note at a time plays in response to voice messages; in Poly mode, as many voices can play notes as are available to play notes.

Using these two messages in various combinations produces the following mode messages.

Omni On/Poly (Mode 1) The SP’s voices respond to voice messages occurring on any channel.

Omni On/Mono (Mode 2) This mode is seldom implemented because playing one note out of the data occurring on all 16 channels is not real useful.

Omni Off/Poly (Mode 3) In this extremely common mode, the SP is tuned to a single channel; any incoming messages are assigned to synth voices, up to the maximum number of 16 voices.

Omni Off/Mono (Mode 4) Voice messages are received over several channels, but each channel plays monophonically. In other words, you could play one voice on channel 1, one voice on channel 2, etc. The SP implements an improved version of Mono mode called Multi Mode, which lets the synth receive *polyphonic* data over each channel.

10.3C Other Messages

Local Control On/Off With Local Control on, playing a master keyboard triggers the internal voices (if present) and sends data out the MIDI out jack. With Local Control off, the keyboard does not trigger the internal voices but does send data over MIDI out. The main use for Local Off is to play an expander module from a master keyboard, but not trigger the keyboard’s internal sounds.

10.4 System Common Messages

Intended for all units in a system, some of these messages are:

Song Position Pointer This indicates how many “MIDI beats” (normally a 16th note) have elapsed since a piece started (up to 16,384 total beats). It is primarily used to allow different sequencers and drum machines to auto-locate to each other so that if you start one sequencer, the other device will automatically jump to the same place in the song, whereupon both continue on together.

System Exclusive This message (called sys ex for short) is considered “exclusive” because different manufacturers send and receive data over MIDI which is intended only for that manufacturer’s equipment. *Example:* Sending a Peavey DPM 3 message to an Emu SP1200 won’t do anything but the message will be understood by other DPM 3s. This data often contains information about individual instrument patches.

Timing Clock A master tempo source (such as a sequencer) emits 24 timing messages (clocks) per quarter note. Each device synchronized to the sequencer advances by 1/24th of a quarter note when it receives the clock message, thus keeping units in sync after they’ve both started at the same time. Many devices subdivide this clock signal internally for higher resolution (e.g., 96 pulses per quarter note).

Start Signals all rhythmically-based units when to start playing.

Stop Signals all rhythmically-based units when to stop playing.

Continue Unlike a Start command, which re-starts a sequencer or drum machine from the beginning of a song each time it occurs, sending a continue message after stop will re-start units from where they were stopped.

10.5 Books on MIDI

The preceding does not substitute for reading a good book on the subject of MIDI. For further information, refer to the following:

MIDI For Musicians and *The Electronic Musician's Dictionary* by Craig Anderton; AMSCO Publications. The former was written specifically for musicians with no background in MIDI, and the latter defines terms related to musical electronics.

Music Through MIDI by Michael Boom; Microsoft Press. An excellent text for those just getting started with MIDI, synthesis, and related topics.

The Murphy's Law MIDI Book by Jeff Burger; Alexander Publishing. Emphasizes applications and problem-solving.

Using MIDI by Helen Casabona and David Frederick; Alfred Publishing. A general guide to MIDI with an emphasis on applications.

Understanding MIDI and *Understanding MIDI 2* by various authors; Amordian Press. A collection of MIDI-oriented articles from *Musician* magazine.

This Page Left Intentionally Blank.

SP Tutorials Section: Introduction

The following tutorials are intended to acquaint you with some useful SP functions, and to familiarize you with its “mindset.” This does not in any way substitute for reading the manual, but hopefully, will give you a non-intimidating, step-by-step introduction to the SP. However, it is recommended that you go through these tutorials to speed up the learning process.

Please go through the tutorials in order (although you can skip Tutorial 2 if you don’t have a DPM SX) since later tutorials assume more knowledge. For example, in the first tutorial, the text is fairly explicit about how many times you have to press an arrow button to place the cursor on a particular parameter. In later tutorials, you’re simply instructed to place the cursor on the parameter since it is assumed you know how to move the cursor around by then.

If you’d like more information about the subjects covered in these tutorials, look for section numbers in parentheses (such as 2.0A, 4.2Y, etc.). These refer to parts of the reference manual that have more details on the operations being discussed. Remember—the SP is a very sophisticated piece of gear that takes a while to understand. Work with it, play with it, and read the manual; everything will fall into place eventually.

Tutorial 1: Playing Back Factory Disks/Setting Basic Parameters

Auto-loading a Bank from Floppy Disk

- 1 *Read Chapter 1 of the reference manual* in order to understand the SP user interface, as well as how to patch the SP into your system. The SP MIDI in should be driven from the MIDI out of a controller (MIDI keyboard, MIDI guitar, MIDI drum pads, etc.). The SP audio outputs should be patched into an audio monitoring system.
- 2 With the SP turned off and monitoring system level turned down, insert the Ethereal Voices factory disk (2.0A).

- 3 Turn on the SP. The disk bank containing the Ethereal Voices sound will load automatically (2.1A).
- 4 The display shows:



The cursor will be flashing in the upper left-hand corner.

- 5 Play your MIDI controller, turn the SP's front panel volume control about 2/3 of the way up, then adjust the audio monitoring system to a comfortable level. You should be hearing something. If not, check that your controller is generating notes, has its MIDI out patched to the SP's MIDI in, and that the SP's audio outputs are properly hooked into your monitoring system.

Adjusting MIDI Channel and Main Volume

- 6 We will now make sure that the SP is set to respond to the desired MIDI channel, as well as adjust the main volume level. Even if the SP is working, it's advisable to go through the following steps to become more aware of the SP's MIDI functions.
- 7 Press the INC button once. The following screen appears (7.2B):



- 8 To adjust the MIDI base channel (the one over which the SP will receive data in Poly mode), press the right arrow button twice. The cursor should now be flashing and positioned over the channel parameter, which is 1.
- 9 Turn the data wheel clockwise to select any MIDI channel from 1 to 16. This should be set to the same channel on which your controller is sending data. You can also use the INC button to increase the channel number, or DEC button to decrease the channel number.

- 10 You can also adjust the SP's main volume from this screen. Press the right arrow button three times so that the cursor appears on the last digit of 127 in the lower line.
- 11 Rotate the data wheel counterclockwise to lower the overall volume level from 127, one value at a time. Confirm this by playing the keyboard as you change the volume level.
- 12 Now press the left arrow button once. The cursor appears on the tens digit of the volume display.
- 13 Rotate the data wheel. This alters the overall volume level ten units at a time (e.g., 10, 20, 30, 40, 50, etc.).
- 14 Rotate the data wheel counterclockwise until the main volume setting returns to 127.
- 15 Let's move to the next screen by pressing the right arrow or left arrow button until the cursor returns to the upper left hand corner. However, there is a shortcut. The SP powers up with the Global menu selected. Whenever you are in a screen in a particular menu, pressing the associated menu button always returns the cursor to the upper left hand corner of the current screen. In this instance, press the **Global** button and the cursor will jump to the upper left hand corner.
- 16 Press the **INC** button once (you could also rotate the data wheel clockwise, but this is more useful when you want to zoom past several screens) until the display shows:



This is the screen where you set the MIDI receive mode (7.2C). It is currently set to omni, which means that the SP will receive data coming in from any channel.

- 17 Press the right arrow button once. Press the **INC** button once so that the display shows **poly**.
- 18 Play your MIDI controller. The SP should still respond to incoming MIDI data. If not, then the controller is not transmitting on the same channel as set in steps 8-9. Return to the Omni/Poly Preset screen, then redo these steps to set the channel so that it receives data from the controller.

Adjusting Master Tuning

- 19 Not all instruments are tuned perfectly to concert pitch; it may be easier to tune the SP to an existing piece of music than tune all the other instruments. To access the tuning screen, first press the **Global** button to return the cursor to the upper left corner of the screen.
- 20 Rotate the data wheel clockwise until the following screen appears (7.4B):



The image shows a digital display with the text "master tune:" followed by a cursor (a vertical line) and the number "0.00". The cursor is positioned between the colon and the first zero of the decimal value.

- 21 Press the left arrow button. The cursor will jump to the last digit in the display. You could also get to this digit by pressing the right arrow button five times.
- 22 Rotate the data wheel clockwise to raise pitch one cent at a time, or counterclockwise to lower pitch one cent at a time. You must retrigger a note to hear the pitch change; you cannot hold a note, change the pitch, and hear a pitch change.
- 23 To change the master tuning in semitones, press the left arrow button twice so that the cursor is on the first digit to the left of the decimal point.
- 24 Rotate the data wheel clockwise to raise pitch one semitone at a time, or counterclockwise to lower pitch one semitone at a time. You must retrigger a note to hear the pitch change.
- 25 Return the master tuning to 0.00 by rotating the data wheel in the appropriate direction.

Selecting Different Presets

- 26 Press the **Preset** button. The following screen appears (6.2A):



Ethereal Breath P# 0

- 27 Press the left arrow button once, or the right arrow button three times, to select the last digit in the preset number.
- 28 Press the **INC** button to select the next-higher numbered preset. Play the MIDI controller and you will hear a variation on the Ethereal Voice sound. Note that the lower line indicates how many voices are being used by the SP as you play.
- 29 Repeat step 28 to select the next-higher numbered preset. Keep repeating this step until you select preset 5, the highest-numbered preset. You will have then auditioned all six available presets in the Ethereal Voices disk.

Loading a Different Sound Disk

- 30 Eject the Ethereal Voices disk by pressing the **Eject** button on the disk drive's lower right (2.0A).
- 31 Insert the Acoustic Piano disk.
- 32 Press the **Disk** button. The display shows (2.1B):



load bank from: flopd
<press exec>

- 33 Press the **EXEC** button. The new sound bank will be loaded.
- 34 In a manner similar to steps 26-28, audition the various presets on the Acoustic Piano disk.

Tutorial 2: Stereo Sampling with the DPM SX

Read the DPM SX manual for background on its operation before starting this tutorial, and sections 3.0 and 3.1 in the reference manual to learn about waves and stereo sampling. Note that the SX must have software version 2.01 or higher to work with the SP. We will assume that the SX has a SCSI interface, as this is recommended for communicating with the SP. MIDI can be used instead, but it is approximately 50 times slower, so this tutorial is based on a SCSI-equipped SX.

Hookup

- 1 Make sure power is turned off on the SP and SX.
- 2 Patch the SP's SCSI A connector to the SX's SCSI interface connector using a SCSI cable (9.6-9.11).
- 3 Choose a stereo signal source to sample. A stereo compact disc or tape track works well for experimentation. *Caution:* Commercial recordings are protected by copyright laws. Do not use samples of commercial recordings for anything other than private, non-commercial use unless you consult with a lawyer.
- 4 Patch one channel of the stereo source to the SX's line input. We will first record one channel, then the other channel.
- 5 Turn on the SP and SX. To prevent auto-loading, do not have a disk in the SP on power-up. This will insure that the maximum possible amount of SP memory is available for sampling.
- 6 Play the source material, and adjust the SX input level control to the highest possible level short of the clip threshold LED glowing red.

DPM SX Configuration

- 7 Press the UTIL button.
- 8 Press the INC button once. The display shows (8.3):



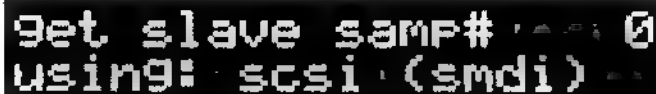
DPM-SX configuration
<press exec>

- 9 Press EXEC. The screen now shows the SX firmware version (8.3A). This must be 2.01 or above to work with the SP. If the screen shows something else, check with your Peavey dealer about upgrading your SX.
- 10 Press the INC button once. The screen displays the amount of installed memory in the SP (8.3B).
- 11 Press the INC button again. The display (8.3C) shows that the sampling rate is 44.10 kHz, the optimum sampling rate for the SP. Press EXEC to set the SX to this sampling rate.
- 12 Press the INC button again. The display (8.3D) shows the maximum available sample length. There's no need to change this for the purposes of this tutorial.
- 13 Press the UTIL button to exit the DPM-SX configuration folder.

Sampling

- 14 Press the INC button once (8.4A).
- 15 Press EXEC. The SP screen turns into a bargraph meter (8.4A). Play the source sample, and adjust the SX's input level control (or the signal source's output) so that the meter goes as high as possible without hitting the 0 dB reading, as indicated on the display's top line.
- 16 Either arm the SX as described in the SX manual, or press START/STOP then play one channel of the sound to be sampled.
- 17 Press the UTIL button to exit the bargraph meter screen.

- 18 Press the INC button once, then press EXEC. The SP screen shows the number of samples that were clipped (8.4B). If this is greater than 1, resample the signal by pressing DEC to return to the input signal level setting screen, then repeating steps 15-18. If 1 or fewer samples are clipped, proceed to step 19.
- 19 Press the INC button once. The display shows (8.4C):



```
Get slave samp# 0
Using: scsi (smdi)
```

With the SX, this number will always be 0 because the SX will hold only one sample in memory at a time. Press EXEC.

- 20 The display now shows (8.4C):



```
Put samp in waveloc:
# 0 -----
```

We will put the first sample of the stereo pair in wave location 0, and the second sample of the stereo pair in wave location 1. Press EXEC to put the first sample into location 0, which is displayed on the screen. The dashes indicate that the wave location is empty. The display will count down the samples as they are transferred to the SP.

- 21 Press the DEC button twice to select the SX input level screen.
- 22 Patch the other source signal channel into the SX's input jack, then repeat steps 15-19. After completing step 19, skip to step 23.
- 23 The display now shows:



```
Put samp in waveloc:
# 0 -----
```


Note that 0 is not followed by a series of dashes, indicating that the wave location is not empty. This is because we already placed the first sample in location 0. To put the second sample of the stereo pair in wave location 1, press the INC button so that the lower line shows 1-----.

- 24 Press EXEC to put the second sample into location 1. The display will count down the samples as they are transferred to the SP.

Fusing Two Mono Samples Into Stereo

- 25 Press the WAVE button.
- 26 Press the INC button or turn the data wheel until the display shows (3.3):



```
wave edit functions
<press exec>
```

- 27 Press EXEC. Press the INC button three times to select the Dual Wave Audition screen (3.3D).
- 28 Press the right arrow button, then press the INC button once to select mix.
- 29 Press the right arrow button six times so that the cursor ends up on the last digit of the **rgt:w#**. Press the INC button once to set this parameter to 1. At this point, wave 0 will be the left side, and wave 1, the right side.
- 30 Press EXEC to hear the two waveforms playing simultaneously. Chances are the attack times will not line up.
- 31 Press the WAVE button to send the cursor to the upper left hand corner, then press DEC twice to access the wave start/end screen (3.3B). Make sure the wave number (w#) is set to 0.
- 32 Use the right arrow button to place the cursor on the wave start point (the first parameter on the lower line). Use the

data wheel and/or INC/DEC buttons to optimize the wave's start time; press EXEC to hear the wave.

- 33 If you wanted to loop the stereo wave, now would be the time to loop the left wave. However, for brevity's sake we will assume for now that you do not want to loop the waves.
- 34 Place the cursor on the w# parameter, then press the INC button to select wave 1.
- 35 Use the arrow buttons to place the cursor on the right wave's start point (the first parameter on the lower line). Use the data wheel and/or INC/DEC buttons to optimize the right wave's start time; press EXEC to hear the wave.

Because the dual wave audition is in mix mode, you should hear flanging/chorus sounds as the two start points get closer to each other.

- 36 Once you have the start points set for the two waves, press the WAVE button to return the cursor to the wave start/end screen's upper left hand corner.
- 37 Press INC three times to reach the stereo merge screen (3.3E). The display shows:



The screenshot shows a black rectangular area with white text. The first line reads 'stereo merge:' followed by two small, faint waveforms. The second line reads 'lft:w# 0 rgt:w# 1'.

Check that the left wave is set to 0 and the right wave to 1.

- 38 Press EXEC, and the two mono waves are merged into a single stereo wave. You can now use this wave in the process of creating tones, maps, and presets. If you play your MIDI controller, you should hear the sampled stereo wave.

Tutorial 3: Preset Edits

The SP offers many levels of editing. Those who want to really tweak a sound can do so at the Tone or Wave level; however, there are many editing options at the preset level that affect the overall sound. These are some of the most useful edits when tweaking a factory disk to your particular specifications. Please read section 6.0 in the reference manual before proceeding to understand a bit about presets.

This tutorial shows how to adjust two common preset level parameters.

Transposition

- 1 Load the Acoustic Piano disk by turning off the SP, inserting the disk in the drive, and turning on power. You could also load from the disk menu, but this method insures a predictable set of parameters for the purposes of doing a tutorial.
- 2 Press the **PRESET** button.
- 3 Rotate the data wheel clockwise until the following screen appears (6.3M):



The screenshot shows a monochrome LCD screen with two lines of text. The first line reads 'note transpose p# 0' and the second line reads 'lyr:a -- tpose:-12'. The text is in a pixelated, digital font.

- 4 Preset 0, Acoustic Brite, has two layers. Each is transposed down one octave (-12 semitones). Press the left arrow button and the cursor will sit on the last digit under tpose.
- 5 Rotate the data wheel clockwise until tpose reads 0. As you rotate the data wheel, trigger the MIDI controller feeding the SP; you will hear layer A go up in pitch.
- 6 Press the left arrow button twice so that the cursor sits on the lyr (layer) parameter. Press the **INC** button to select layer B.
- 7 Press the right arrow button twice so that the cursor again sits on the last digit under tpose.

- 8 Rotate the data wheel clockwise until tpose reads 0. As you rotate the data wheel, trigger the MIDI controller feeding the SP; you will hear layer B go up in pitch.
- 9 Play the MIDI controller. Note that the entire piano sound has been raised up one octave compared to the original preset.
- 10 Maybe you'd like a bass piano that's transposed down an octave compared to the original preset. If so, follow a similar procedure as described above to set tpose to -24 for both layers.
- 11 Return to the original preset parameters by setting both tpose parameters to -12.

Set Pitch Bend Range

- 12 Press the PRESET button to send the cursor to the upper left-hand corner.
- 13 Press the INC button to select the next page (6.3N):



```
Pitch bend: F# 0
lyr:a - bendrange: +2
```

- 14 You can set the pitch bend range in a manner similar to transposition: place the cursor on bendrange for one layer, use the data wheel to set the amount of bend range (try +12, so that you can bend pitch up an octave), then choose the other layer and set its pitch bend range.
- 15 Return to the original preset parameters by setting both bendrange parameters to +2.

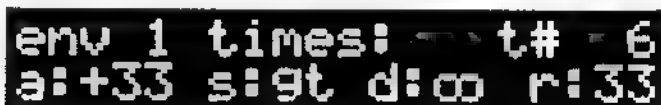
Tutorial 4: Tone Menu Edits

First, read over from section 4.0 up to 4.0F in the reference manual to understand what a tone is.

The Tone menu allows a great degree of sonic control over individual waves. Here are some simple ways you can modify the sound.

Attack and Release Times

- 1 Load the Ethereal Voices disk by turning off the SP, inserting the disk in the drive, and turning on power (2.1A). You could also load from the disk menu, but this method insures a predictable set of parameters for the purposes of doing a tutorial.
- 2 Press the **TONE** button. If you play a MIDI controller, this tone should stretch across the controller's note range, assuming that **MIDI notes play: what's on display** is selected in the Global menu (7.2D). For now, as long as the tone plays across the entire controller range, don't bother entering the Global menu; however, if you want to hear the tone in the context of the current preset, select **MIDI notes play: current edit preset**.
- 3 Rotate the data wheel clockwise until the following screen appears (4.2A):



```
env 1 times: t# 6
a: +33 s: 9t d: ∞ r: 33
```

- 4 Move the cursor to the **a:** (attack) parameter. Use the data wheel to set the value to 0. The sound should become more percussive.
- 5 Move the cursor to the **r:** (release) parameter. Use the data wheel to set the value to 20. The sound will have a short attack and release.
- 6 Press the **TONE** button to send the cursor to the upper left-hand corner.

Dynamics

- 7 Press the INC button three times to select the dca modulation page (4.2D):



```
dca modulation t# 6
vel: 0 lfo: 0:nrm
```

- 8 Move the cursor to the vel (velocity) parameter. Rotate the data wheel clockwise until vel reads +50. Note how the harder you play, the louder the sound becomes.
- 9 Now set vel to -50. The harder you play, the softer the sound becomes. Return vel to +50.
- 10 Press the TONE button to send the cursor to the upper left-hand corner.

Filtering

- 11 Press the INC button twice to select the dcf cutoff page (4.2F):



```
dcf cutoff: 99 t# 6
envlfo: none amt: 0
```

- 12 Move the cursor to the cutoff parameter. Rotate the data wheel counter-clockwise while triggering the MIDI controller feeding the SP; the lower the cutoff, the duller the sound. Return the value to 99.
- 13 Press the TONE button to send the cursor to the upper left-hand corner.

Vibrato

- 14 Press the INC button five times to select the dco lfo mod page (4.2K):



```
dco lfo mod: t# 6
phase: normal amt: 0
```

- 15 Move the cursor to the **amt** parameter and set the value to +5. You won't hear any vibrato yet, because we still need to edit the LFO parameters.
- 16 Press the **TONE** button to send the cursor to the upper left-hand corner.
- 17 Press the **INC** button 11 times to access the following screen (4.2V):



The screenshot shows a monochrome LCD screen with the text 'lfo base setup' at the top. Below it, there are three parameters: 't# 6', 'speed: 10', and 'amt: 0'. The 't#' parameter is on the right, 'speed' is on the left, and 'amt' is on the right. There are horizontal lines between the parameters.

- 18 Move the cursor to the **amt** parameter and set the value to +50. If you turnup the mod wheel on the controller feeding the SP, the note's pitch will fluctuate at a slow rate.
- 19 Move the cursor to the **speed** parameter and set the value to 75. The pitch variations will speed up.
- 20 While you're at it, press **DEC** once to access the LFO waveshape page and try some of the different LFO waveform options (4.2U). If the variations aren't very obvious, return to the dco lfo mod screen and increase the amount parameter.

This Page Left Intentionally Blank.

THIS LIMITED WARRANTY VALID ONLY WHEN PURCHASED AND REGISTERED IN THE UNITED STATES OR CANADA. ALL EXPORTED PRODUCTS ARE SUBJECT TO WARRANTY AND SERVICES TO BE SPECIFIED AND PROVIDED BY THE AUTHORIZED DISTRIBUTOR FOR EACH COUNTRY.

Ces clauses de garantie ne sont valables qu'aux Etats-Unis et au Canada. Dans tous les autres pays, les clauses de garantie et de maintenance sont fixées par le distributeur national et assues par lui selon la législation en vigueur.

Diese Garantie ist nur in den USA und Kanada gültig. Alle Export-Produkte sind der Garantie und dem Service des Importeurs des jeweiligen Landes unterworfen.

Esta garantía es válida solamente cuando el producto es comprado en E.U. continentales o en Canada. Todos los productos que sean comprados en el extranjero, están sujetos a las garantías y servicio que cada distribuidor autorizado determine y ofrezca en los diferentes países.

ONE-YEAR LIMITED WARRANTY/REMEDY

PEAVEY ELECTRONICS CORPORATION ("PEAVEY") warrants this product, EXCEPT for covers, footswitches, patchcords, tubes and meters, to be free from defects in material and workmanship for a period of one (1) year from date of purchase, PROVIDED, however that this limited warranty is extended only to the original retail purchaser and is subject to the conditions, exclusions and limitations hereinafter set forth:

PEAVEY 90-DAY LIMITED WARRANTY ON TUBES AND METERS

If this product contains tubes or meters, Peavey warrants the tubes or meters contained in the product to be free from defects in material and workmanship for a period of ninety (90) days from date of purchase; PROVIDED, however, that this limited warranty is extended only to the original retail purchaser and is also subject to the conditions, exclusions and limitations hereinafter set forth.

CONDITIONS, EXCLUSIONS AND LIMITATIONS OF LIMITED WARRANTIES

These limited warranties shall be void and of no effect if:

- a. The first purchase of the product is for the purpose of resale; or
 - b. The original retail purchase is not made from an AUTHORIZED PEAVEY DEALER; or
 - c. The product has been damaged by accident or unreasonable use, neglect, improper service or maintenance, or other causes not arising out of defects in material or workmanship; or
 - d. The serial number affixed to the product is altered, defaced or removed.
- In the event of a defect in material and/or workmanship covered by this limited warranty, Peavey will:
- a. In the case of tubes or meters, replace the defective component without charge;
 - b. In other covered cases (i.e., cases involving anything other than covers, footswitches, patchcords, tubes or meters), repair the defect in material or workmanship or replace the product, at Peavey's option;
- and provided, however, that, in any case, all costs of shipping, if necessary, are paid by you, the purchaser.

THE WARRANTY REGISTRATION CARD SHOULD BE ACCURATELY COMPLETED AND MAILED TO AND RECEIVED BY PEAVEY WITHIN FOURTEEN (14) DAYS FROM THE DATE OF YOUR PURCHASE.

In order to obtain service under these warranties, you must:

- a. Bring the defective item to any AUTHORIZED PEAVEY DEALER or AUTHORIZED PEAVEY SERVICE CENTER and present therewith the ORIGINAL PROOF OF PURCHASE supplied to you by the AUTHORIZED PEAVEY DEALER in connection with your purchase from him of this product.
- If the DEALER or SERVICE CENTER is unable to provide the necessary warranty service you will be directed to the nearest other PEAVEY AUTHORIZED DEALER or AUTHORIZED PEAVEY SERVICE CENTER which can provide such service.

OR

- b. Ship the defective item, prepaid, to:

PEAVEY ELECTRONICS CORPORATION
International Service Center
Highway 80 East
MERIDIAN, MS 39301

including therewith a complete, detailed description of the problem, together with a legible copy of the original PROOF OF PURCHASE and a complete return address. Upon Peavey's receipt of these items:

If the defect is remedied under these limited warranties and the other terms and conditions expressed herein have been complied with, Peavey will provide the necessary warranty service to repair or replace the product and will return it, FREIGHT COLLECT, to you, the purchaser.

Peavey's liability to the purchaser for damages from any cause whatsoever and regardless of the form of action, including negligence, is limited to the actual damages up to the greater of \$500.00 or an amount equal to the purchase price of the product that caused the damage or that is the subject of or is directly related to the cause of action. Such purchase price will be that in effect for the specific product when the cause of action arose. This limitation of liability will not apply to claims for personal injury or damage to real property or tangible personal property allegedly caused by Peavey's negligence. Peavey does not assume liability for personal injury or property damage arising out of or caused by a non-Peavey alteration or attachment, nor does Peavey assume any responsibility for damage to interconnected non-Peavey equipment that may result from the normal functioning and maintenance of the Peavey equipment.

UNDER NO CIRCUMSTANCES WILL PEAVEY BE LIABLE FOR ANY LOST PROFITS, LOST SAVINGS, ANY INCIDENTAL DAMAGES OR ANY CONSEQUENTIAL DAMAGES ARISING OUT OF THE USE OR INABILITY TO USE THE PRODUCT, EVEN IF PEAVEY HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

THESE LIMITED WARRANTIES ARE IN LIEU OF ANY AND ALL WARRANTIES, EXPRESS OR IMPLIED, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR USE; PROVIDED, HOWEVER, THAT IF THE OTHER TERMS AND CONDITIONS NECESSARY TO THE EXISTENCE OF THE EXPRESS, LIMITED WARRANTIES, AS HEREINABOVE STATED, HAVE BEEN COMPLIED WITH, IMPLIED WARRANTIES ARE NOT DISCLAIMED DURING THE APPLICABLE ONE-YEAR OR NINETY-DAY PERIOD FROM DATE OF PURCHASE OF THIS PRODUCT.

SOME STATES DO NOT ALLOW LIMITATION ON HOW LONG AN IMPLIED WARRANTY LASTS, OR THE EXCLUSION OR LIMITATION OF INCIDENTAL OR CONSEQUENTIAL DAMAGES, SO THE ABOVE LIMITATIONS OR EXCLUSIONS MAY NOT APPLY TO YOU. THESE LIMITED WARRANTIES GIVE YOU SPECIFIC LEGAL RIGHTS, AND YOU MAY ALSO HAVE OTHER RIGHTS WHICH MAY VARY FROM STATE TO STATE.

THESE LIMITED WARRANTIES ARE THE ONLY EXPRESS WARRANTIES ON THIS PRODUCT, AND NO OTHER STATEMENT, REPRESENTATION, WARRANTY OR AGREEMENT BY ANY PERSON SHALL BE VALID OR BINDING UPON PEAVEY.

In the event of any modification or disclaimer of express or implied warranties, or any limitation of remedies, contained herein conflicts with applicable law, then such modification, disclaimer or limitation, as the case may be, shall be deemed to be modified to the extent necessary to comply with such law.

Your remedies for breach of these warranties are limited to those remedies provided herein and Peavey Electronics Corporation gives this limited warranty only with respect to equipment purchased in the United States of America.

INSTRUCTIONS — WARRANTY REGISTRATION CARD

1. Mail the completed WARRANTY REGISTRATION CARD to:

PEAVEY ELECTRONICS CORPORATION
POST OFFICE BOX 2898
MERIDIAN, MISSISSIPPI 39302-2898

- a. Keep the PROOF OF PURCHASE. In the event warranty service is required during the warranty period, you will need this document. **There will be no identification card issued by Peavey Electronics Corporation.**
2. IMPORTANCE OF WARRANTY REGISTRATION CARDS AND NOTIFICATION OF CHANGES OF ADDRESS:
 - a. Completion and mailing of WARRANTY REGISTRATION CARDS — Should notification become necessary for any condition that may require correction, the REGISTRATION CARD will help ensure that you are contacted and properly notified.
 - b. Notice of address changes — If you move from the address shown on the WARRANTY REGISTRATION CARD, you should notify Peavey of the change of address so as to facilitate your receipt of any bulletins or other forms of notification which may become necessary in connection with any condition that may require dissemination of information or correction.
 3. You may contact Peavey directly by telephoning (601) 483-5365.
 4. Please have the Peavey product name and serial number available when communicating with Peavey Customer Service.

DANGER

EXPOSURE TO EXTREMELY HIGH NOISE LEVELS MAY CAUSE A PERMANENT HEARING LOSS. INDIVIDUALS VARY CONSIDERABLY IN SUSCEPTIBILITY TO NOISE INDUCED HEARING LOSS, BUT NEARLY EVERYONE WILL LOSE SOME HEARING IF EXPOSED TO SUFFICIENTLY INTENSE NOISE FOR A SUFFICIENT TIME.

THE U.S. GOVERNMENT'S OCCUPATIONAL SAFETY AND HEALTH ADMINISTRATION (OSHA) HAS SPECIFIED THE FOLLOWING PERMISSIBLE NOISE LEVEL EXPOSURES:

DURATION PER DAY IN HOURS

8
6
4
3
2
1

SOUND LEVEL (dBA, SLOW RESPONSE)

90
92
95
97
100
102
105
110
115

ACCORDING TO OSHA, ANY EXPOSURE IN EXCESS OF THE ABOVE PERMISSIBLE LIMITS COULD RESULT IN SOME HEARING LOSS.

EAR PLUGS OR PROTECTORS IN THE EAR CANALS OR OVER THE EARS MUST BE WORN WHEN OPERATING THIS AMPLIFICATION SYSTEM IN ORDER TO PREVENT A PERMANENT HEARING LOSS IF EXPOSURE IS IN EXCESS OF THE LIMITS AS SET FORTH ABOVE. TO INSURE AGAINST POTENTIALLY DANGEROUS EXPOSURE TO HIGH SOUND PRESSURE LEVELS, IT IS RECOMMENDED THAT ALL PERSONS EXPOSED TO EQUIPMENT CAPABLE OF PRODUCING HIGH SOUND PRESSURE LEVELS SUCH AS THIS AMPLIFICATION SYSTEM BE PROTECTED BY HEARING PROTECTORS WHILE THIS UNIT IS IN OPERATION.

CAUTION:

THIS MIXER/CONSOLE/AMPLIFIER DEVICE HAS BEEN DESIGNED AND CONSTRUCTED TO PROVIDE ADEQUATE SIGNAL (VOLUME) FOR PLAYING MODERN MUSIC. PROPER USE OF THE GAIN/EQUALIZER CONTROLS AND/OR IMPROPER USE OF INTERNAL/EXTERNAL MIXES MAY CREATE CLIPPING (SHARP WAVES) AND POSSIBLY CAUSE SUBSEQUENT DAMAGE TO THE LOUDSPEAKER SYSTEMS. EXTENDED OPERATION OF THE GAIN/EQUALIZATION CONTROLS IN THEIR MAXIMUM POSITIONS IS THEREFORE NOT RECOMMENDED. PLEASE BE AWARE THAT MAXIMUM POWER CAN BE OBTAINED WITH VERY HIGH SETTINGS OF THE GAIN/EQUALIZATION CONTROLS IF "THE BEST" SIGNAL IS VERY STRONG.

IT IS COMMON PRACTICE AMONG USERS OF SOUND REINFORCEMENT EQUIPMENT TO IDENTIFY THE INDIVIDUAL CHANNELS WITH A STRIP OF TAPE PLACED ABOVE OR BELOW THE ROW OF COLUMN HEADERS. MANY TYPES OR BRANDS OF TAPE HAVE VERY STRONG ADHESIVE WHICH CAN INHIBIT THE PAINT ON THE FACE PLATE AND USUALLY REMOVE THE PAINT WHEN THE TAPE IS REMOVED. WE STRONGLY RECOMMEND THAT SCOTCH TAPE NOT BE USED ON PAINTED SURFACES OR ANY OTHER TAPE THAT IS NOT SPECIALLY DESIGNED FOR SUCH APPLICATIONS. MEDICAL OR LIGHT ADHESIVE MARKING OR LABEL TAPE IS RECOMMENDED IF IT IS USED. ANY TAPE LEFT ON PAINTED SURFACE FOR EXTENDED PERIODS WILL BE DIFFICULT TO REMOVE. NEVER USE OLIVE OR SCOTCH TAPE FOR THESE APPLICATIONS.

1. Read all safety and operating instructions before using this product.
2. All safety and operating instructions should be retained for future reference.
3. Obey all cautions in the operating instructions and on the back of the unit.
4. All operating instructions should be followed.
5. This product should not be used near water, i.e. a bathtub, sink, swimming pool, wet basement, etc.
6. This product should be located so that its position does not interfere with its proper ventilation. It should not be placed flat against a wall or placed in a built-in enclosure that will impede the flow of cooling air.
7. This product should not be placed near a source of heat such as a stove, radiator or another heat producing appliance.
8. Connect only to a power supply of the type marked on the unit adjacent to the power supply cord.
9. Never break off the ground pin on the power supply cord. For more information on grounding write for our free booklet "Shock Hazard and Grounding."
10. Power supply cords should always be handled carefully. Never walk or place equipment on power supply cords. Periodically check cords for cuts or signs of stress, especially at the plug and the point where the cord exits the unit.
11. The power supply cord should be unplugged when the unit is to be unused for long periods of time.
12. If this product is to be mounted in an equipment rack, rear support should be provided.
13. Metal parts can be cleaned with a damp rag. The vinyl covering used on some units can be cleaned with a damp rag, or an ammonia based household cleaner if necessary.
14. Care should be taken so that objects do not fall and liquids are not spilled into the unit through the ventilation holes or any other openings.
15. This unit should be checked by a qualified service technician if:
 - A. The power supply cord or plug has been damaged.
 - B. Anything has fallen or been spilled into the unit.
 - C. The unit does not operate correctly.
 - D. The unit has been dropped or the enclosure damaged.
16. The user should not attempt to service this equipment. All service work should be done by a qualified service technician.

DLO Aux mod - modwheel

LFO Base setup - Speed 80, amount 0

LFO Amount mod - src - modwheel,
amount 20



Features and specifications subject to change without notice

Peavey Electronics Corporation • 1000 Shreve Memorial MS #300 • Fort Worth, TX 76104 • Tel: 817-335-3355 • Fax: 817-335-3356

©1995 Peavey

Printed in U.S.A.