Korg DSS-1 Sampler : Making It Happen

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THE NINE SECTIONS OF THE DSS-1 BOOK

Chapter 1
SAMPLING BASICS AND THE DSS-1 AS A COMPUTER

Computers and Sound

Sampling is the digital recording of electronic (analog) sound signals. The signals are sent into a computer's audio input, where the analog signal is converted into numbers (digital data). This process is called a/d (analog-to-digital) conversion.

How to record samples is described in great detail in section 4. Here's a brief overview of the way the DSS-1 treats the sound after the a/d conversion.

All sound happens as a series of vibrations, called waveforms. The computer tracks the shape of a waveform and takes a reading many thousands of times per second. These readings are stored in the computer's memory. The lists of these numbers are called waves.

While in the computer's memory, the readings can be edited (the numbers can be manipulated). When you play the sound, the computer uses the digital data in the wave to produce an analog signal. This process is called d/a (digital-to-analog) conversion. The analog signal can sound like the original sample, or it can sound very different if it has been edited.

Two Sampling Definitions

1) Technicians use the word 'sample' to mean a single reading - it is a single computer measurement of a waveform. Thousands of "samples" are taken every second during computer recording (audio sampling).

2) Musicians use the word sample in a less specific way - to refer to complete collections of computer readings of a given "sampled" sound. When you see a "killer piano sample" advertised, you know that what is being sold is a complete computer recording of a piano sound, not just a single reading picked out of a sampled waveform. KORG uses the word SAMPLE in the musician's rather than the technician's sense. In this book, I'll go along with KORG.

What you've read so far describes the SAMPLING PROCESS in a nutshell. Read on, because this next section is crucial to your understanding of how the computer on the DSS-1works.

How the DSS-1 Computer Works

The computer remembers information in groups of bits called bytes. There are 8 bits per byte. Depending on the size of the computer, one or more bytes are organized into words.

On the DSS-1, there are 2 bytes per word. KORG decided to allocate 1-1/2 bytes (12 bits) for each sampled reading. To speed up data access in the wave, each reading is stored in a single word. (This means four bits are unused). But on the DISK, where space is more important than speed, no bits are unused. This means that the maximum number of readings that can be stored on the DISK can be calculated by taking the DISK capacity in words and dividing it by 3/4ths.

This is because a reading which takes a whole word in memory is stored in 3/4ths of a word on DISK. You can also figure out the maximum number of readings that can be stored on the DISK by taking the DISK capacity in bytes (which is the more common measure of DISK capacity) and dividing it by 1-1/2. This is because a reading which takes a whole word in memory is stored in 1-1/2 bytes on DISK.

What does this mean to you ???

The LCD keeps showing you numbers which are describing the lengths (sizes) of SOUNDS and MULTI SOUNDS. A "length" is really the number of readings that the computer has taken of that SOUND or MULTI SOUND.

When you SAVE SOUND or MULTI SOUND DATA to DISK, you must always be sure that there's enough room on the DISK. The MAGIC NUMBER is 525640 words. If you try to SAVE any more than this, you'll get a DISK FULL message.

If you're trying to SAVE PROGRAM information in addition to SOUND and MULTI SOUND DATA, the MAGIC NUMBER is a little less than 525640. How much less? That depends on how much PROGRAM information there is. PROGRAM information for 128 PROGRAMS (the maximum number of PROGRAMS that a DISK can contain) takes roughly 20,000 words.

For the mathematically minded: Here's where the MAGIC NUMBER comes from, plus some other quasi-magic numbers.

Example: 773 Kbytes per disk
773 Blocks per DISK
1Kbyte (1024 bytes) per Block
4 bytes reserved for data name
1020 bytes usable for sampled readings
1020 bytes divided by 1.5 bytes per reading = 680 readings per Block
680 readings per Block multiplied by 773 Blocks = 525640 readings per DISK.

In this book, I'll use the term *WORDS* when referring to the capacity of the internal memory (which is called RAM). I'll use the term *BYTES* when referring to capacity of the DISK.

Computer Memory Basics

Every computer can only remember a certain number of words before its memory is full. On the DSS-1, the computer must remember wave data (sound information), and information dealing with the actual operations of the computer, disk drive, etc.

When you turn the power on, the DSS-1 knows very little: It only remembers how to do certain things, but it doesn't know what to work on yet. As you feed it the wave data (sound information), it stores the information in its on-board memory. The on-board memory is called RAM (Random Access Memory).

On the DSS-1, you'll sometimes get a message **WAVERAM FULL**. This will happen if you're trying to feed more wave data into the RAM than there is room available.

The RAM contents are NOT PERMANENT. The information in RAM is lost when you either load other information that wipes out what was in RAM before, or when you turn the power off.

That's where the DISKS come in.

The RAM contents can be written onto a disk for future recall. This writing action is called *SAVING.*

The re-calling of information from a disk into RAM is usually called LOADING. On the DSS-1, it is also called GETTING. "Getting" and "Loading" are the same thing when dealing with a disk.

Another term that you'll deal with is the SAMPLING RATE. It is sometimes called SAMPLING FREQUENCY. The SAMPLING RATE is the number of times per second that the computer takes a reading of the shape of the WAVEFORM. Obviously, the more often this happens, the more accurate the readings will be.

But there is a trade-off involved.

It's called TIME *(length)* of SAMPLE versus ACCURACY *(frequency response).* Higher SAMPLING RATES *(more readings per second)* fill up the RAM more quickly. So, the readings will not cover as many seconds before the WAVERAM is full. But while higher SAMPLING RATES mean shorter SAMPLING TIMES, the sounds are sampled with greater accuracy.

Lower SAMPLING RATES *(fewer readings per second)* allow for longer SAMPLING TIMES *(more seconds),* but they'll capture the sounds less faithfully.

As a general rule, you can expect a FREQUENCY RESPONSE *(also called bandwidth or hi-fi specs)* of a little less than half the SAMPLING RATE. On the DSS-1, the highest SAMPLING RATE of 48kHz gives you an excellent bandwidth of around 20kHz, which is as good or better than much of the home stereo equipment currently available.

A few basic calculations are all that's needed to manage the available memory successfully.

A handy rule of thumb about RAM versus DISK capacity: A known amount of data in RAM (as shown on the LCD) will take up one-and-a-half times as many bytes on the DISK.

Example: You've used up half the RAM's capacity - 131072 words. How much DISK space will this occupy when you SAVE it? 131072 times 1-1/2 = 196608. This is in bytes and takes up roughly one quarter of the DISK.

A full RAM takes up roughly half a DISK. The blank tables (copy them from the back of this book) will simplify these bookkeeping chores. When it comes to SAMPLING, common sense and critical listening are the ultimate arbiters. After all, this is still a MUSICAL INSTRUMENT, and music should be FUN. Let's get to it.

SECTION 1
GETTING TO KNOW THE BEAST

Chapter 2
HOOKUP, POWER UP, REAR PANEL FEATURES

Before setting up the DSS-1, pick the right spot near your mixer/amplifier/stereo system, with AC power safely accessible. Run your cables away from stumbling feet - connectors are easily broken. If you accidentally disconnect the power cable, you'll lose everything currently in the RAM (memory) of your DSS-1.

To get started, all you'll need is the AC cord that comes with the unit, and a pair of headphones with a 1/4" stereo plug. To connect the DSS-1 to a sound system, you need two audio cables. These connect to the DSS-1 with 1/4" plugs.

The DSS-1 has two audio outputs: Stereo Left and Stereo Right. For Mono connections, use the Right output, which supplies a serial connection of the Left + Right. The DSS-1 is only truly stereo when its on-board Digital Delays are being used.

You can listen through any audio mixer, keyboard amplifier, or home stereo system. The output signal from the DSS-1 matches most system's input sensitivities. On a home stereo receiver, try the AUX or TAPE inputs. DO NOT USE PHONO or MICROPHONE inputs. In case of mismatch, adjust the DSS-1 output level using the HIGH/LOW switch on the DSS-1 rear panel.

Be aware that not every amp will do justice to the varied sounds from your DSS-1. So, choose a system with the flattest possible frequency response. A good home stereo amp, a PA mixer/amp, or one of the new generation of keyboard amps will work fine. If they're stereo - better still. Even the best guitar amp was not built for the multitude of sounds that you can get from your DSS-1.

First, make the connections with all power OFF. Then, pull down the DSS-l's MASTER VOLUME slider (far left panel) and turn down the volume on your stereo system. Now turn on the power on the DSS-1 (rear panel, far right), then power up your playback system. The DSS-1 will make no sound at this point. That's because you haven't loaded a DISK yet. Can you wait 'til the next chapter?

OPTIONAL EQUIPMENT: The rear panel has provisions for hooking up other gear.

1. MIDI cables: They look like some DlN-audio cables, but make sure you get MIDI cables from any music store. And not even all cables sold as MIDI cables are wired correctly. Insist on-cables that have the outside pins #1 and 3 not connected to any wires.
2. Stereo Headphones with a stereo 1/4" plug
3. A damper pedal: for piano-like sustain action.
4. A footswitch for program changes.
5. A microphone cable or audio cable: for audio input sampling. This connects to the DSS-1 with a 1/4" mono plug.

Chapter 3
PLAYING SOUNDS FROM THE FACTORY DISKS

1. Insert the factory disk labeled Piano KSD 001 into the disk drive, label side up. The metallic part with Korg MF 2DD showing goes in first. Make it a habit to push the Protect Tab (on the flip side of the disk) all the way towards the corner.

Silkscreened on the panel of the DSS-1, next to the each MODE tab, is a list of FUNCTIONS for that MODE. Every time the DSS-1 is powered up, it sets itself to SYSTEM MODE. This is confirmed by the red light under the word SYSTEM on the right side of the panel. The LCD (Liquid Crystal Display) window shows **SYSTEM MODE - Select (1-9)**

2. While in SYSTEM mode, select **F1 GET SYSTEM** (press 1 on the number pad). The LCD now invites you to select SYSTEM A; The CURSOR (the blinking underline character) is positioned under the letter A. At this point you could use the up/down arrow keys under DATA ENTRY A to select SYSTEM B, C, or D.

3. Also blinking is the light above the ENTER tab. Press ENTER. The LCD asks you to confirm your intention with the YES tab (**Are You Sure? Y/N**).

4. Press YES. The LCD shows that F1 (function #1) is being executed, and politely asks you to wait. As soon as the SYSTEM is loaded, the LCD tells you so and invites you to select another function (**Select 1-9**), because the DSS-1 is still in SYSTEM MODE.

5. Play the keyboard while raising the MASTER VOLUME slider on the DSS-1 and the volume control on your playback system. If needed, adjust the Hi-Lo switch next to the DSS-1 outputs. When the sound is at a comfortable level, you hear the sound of a piano. BUT - the LCD doesn't show anything about a piano. It still shows **F1 SYS A Completed - Select (1-9)**.

How Do You Get to Hear Another Sound??

6. You need to get into the Play Mode to hear all the PROGRAMS in this SYSTEM A. But - you can't find a tab that says PLAY MODE or PROGRAM SELECT, because there isn't one.

You're in Play Mode whenever none of the 8 Mode Tabs are lit.

7. So, press the SYSTEM tab. This will exit from the SYSTEM mode and enter the PLAY mode. The red light goes out and the LCD shows **SYS A** (System A) **P01** (Program 01) **: G.Piano**. This is what you've been playing so far.

To select the other 31 PROGRAMS from within SYSTEM A, you can

a) type in the desired number on the number pad (1-9 must be typed as 01-09), or
b) press the UP-ARROW and DOWN-ARROW tabs from either Data Entry A or Data Entry B.

Have fun playing the sounds. Test them high and low on the keyboard, play hard and soft, with Aftertouch pressure, and explore the effects available from the JOYSTICK.

Chapter 4
PANEL DESCRIPTION, SUMMARY OF MODES

1**.** On the far left you see the JOYSTICK. You use it while playing notes on the keyboard. The exact response depends on a number of pre-programmed settings that can be different for each sound. Section 6 of this book will teach you how to program these settings. In general, the following effects can be expected from the JOYSTICK:

Move it up towards OSC and, on most PROGRAMS, you'll hear VIBRATO (Pitch Modulation).
Move it left and the pitch goes down.
Move it right and the pitch goes up.
Move it down towards VCF and, on most PROGRAMS, the brightness of the sound changes.

2. Next to the joystick you find the DISK DRIVE. Treat it with respect - dust, shocks, heat exposure and other unkind treatment will take their revenge on you.

When transporting your DSS-1, make sure to leave the pre-cut piece of cardboard inserted into the drive (it came with the DSS-1). If you've lost it, use a disk that has no irreplaceable information on it. The idea is to stop the moveable parts from moving during transport.

To remove a disk, simply press the eject button.

Never insert or eject a disk while the red light above the slot is lit.

Keep your disks clean and store them away from magnetic fields (speakers!). Buy a box of 3-5" floppies (they aren't really "floppy", they're rigid). They must be labeled; doublesided, double density, double track, 135 TPI (tracks per inch). I've used Maxell MF2DD with good results. Stick with a name brand - better safe than sorry.

3. Above the keyboard, the first slider from the left is the MASTER VOLUME. When your DSS-1 is connected to professional equipment, this should be at 10.

Use the Hi-Lo adjustment and your mixer's input trim to match levels. If the signal is still too strong for your system when at the Lo position, then reduce the Master Volume to less than 10.

Be careful when listening through headphones - start on a low setting and increase until comfortable.

4. The next slider is MASTER TUNE. It's always active. So, be careful that you don't touch it once your DSS-1 is in tune with whatever other instruments you're playing with.

5. There are two sliders for data selection: DATA ENTRY A and DATA ENTRY B. Both have a pair of UP/DOWN arrow tabs. Often the slider will give you coarse movement through a wide range of values, while the UP/DOWN tabs step through the values one at a time. But at other times, Slider A might give you access to one thing, while the tabs change another. I can't generalize at this point - I'll cue you when the necessity arises.

6. The LCD (the readout window) is the way the DSS-1 ‘talks' to you.

7. Below it you find the DELETE/CANCEL tab: that often lets you get out of a sticky situation.

8. COMPARE lets you do just that at certain stages of your programming.

9. The remaining tabs below the LCD are labeled with the numbers 0 through 9. I'll refer to these tabs as the NUMPAD to distinguish them from the many other keys and tabs on the DSS-1.

10. To the right of the LCD, you have a knob that you can turn to adjust the CONTRAST (brightness)

11. Below the CONTRAST knob you find the ENTER tab, with a red light that will blink when you are supposed to press ENTER. This tab is similar to EXECUTE or RETURN on other computers - it makes something happen.

12. Below it, you see two tabs under the word CURSOR. The left one is NO/◀, the right one is YES/▶

They have a double function:

a) When you make selections within the LCD, they move the cursor (the blinking underline character) around according to the arrows.

b) When the DSS-1 asks you whether you want to choose a certain action or function, you answer YES or NO with these tabs.

13. Now we come to the MODE tabs. Each of these is like a Government Department. What you do in one department (MODE) may or may not have a direct effect on what goes on in another department (MODE).

14. While you're in a MODE, the DSS-1 will eventually come back to the main readout for that MODE. This main readout asks you to select one of the available functions in that MODE. These functions are conveniently numbered and listed to the right of each MODE tab. We'll refer to each function with the capital letter F and its corresponding number. Example: **SYSTEM F1 GET SYSTEM** means the first function in SYSTEM mode: "Get System".

15. SAMPLE mode. This is the mode you use when you want to sample (record) a sound. This mode does NOT apply to samples already stored on disk.

16. EDIT SAMPLE mode. Here you modify several kinds of wave data:

a) SOUNDS that you may have just sampled (see #15 above) or otherwise created (see #17 below).
b) SOUNDS isolated from an existing MULTI SOUND for use with another MULTI SOUND.
c) SOUNDS that you had kept on a storage disk of your library.

17. CREATE WAVEFORM. This gives you two alternatives to Audio Input Sampling.

a) instead of recording a sound, use the LCD as a "plotter pad" to create a single cycle waveform
b) stipulate the shape (sound color) of a single cycle of a waveform by the overtone amplitude

The waveforms you create using the "drawing" procedure will sound at the pitch of B1 which is around 62 Hz, 1 halfstep and 2 octaves below middle C on a piano. But that's not the only pitch these waveforms can produce: you'll use transposition and keyboard allocation functions to get the pitches that you'll actually need. (Details in Sections 3 & 6).

The waveforms that you create with Harmonic Synthesis are automatically duplicated and spread across the keyboard. You can instantly play them like a regular synthesizer sound.

18. MULTI SOUND mode. This is where you create a MULTI SOUND by calling up different SOUNDS. These SOUNDS can be *Sampled* (recorded as per #15 above), or *Created* (as per #17 above).

You combine the SOUNDS into a MULTI SOUND by spreading them across the keyboard. While you assemble the SOUNDS in this mode, you have a last chance to treat them individually before they get locked into a MULTI SOUND.

To unlock SOUNDS from within a MULTI SOUND, you have to go through a rather tedious procedure which unlocks them one at a time. But this can be rewarding if you want to extract SOUNDS from within factory MULTI SOUNDS. (See Chapter 16).

19. MIDI mode. Here you set the communication between your DSS-1and the rest of the MIDI world.

20. SYSTEM mode. This is where you organize your performance set-ups. There are 4 SYSTEMS possible per disk. Each SYSTEM can contain up to 16 MULTI SOUNDS. Within each SYSTEM, the MULTI SOUNDS are used, one or two at a time, by up to 32 PROGRAMS (keyboard set-ups/presets). The next few chapters explain this.

21. DISK UTILITY mode. This is the mode for formatting (initializing) disks, reading directories, and deleting SOUNDS and MULTI SOUNDS.

22. PROGRAM PARAMETER mode. This is the section that turns MULTI SOUNDS into the musically useful PROGRAMS (Presets) that you call up during performances. Individual MULTI SOUNDS can be drastically modified. The same MULTI SOUND can end up "sounding" many different ways. In the PROGRAM PARAMETER mode, you also set up keyboard response, digital delays and other effects.

Chapter 5
HIERARCHY OF THE ORGANIZATION OF SOUNDS, FROM SAMPLE TO SYSTEM

SAMPLE / WAVEFORM / SOUND

These are three names for the smallest unit of sound in the organization of the DSS-1. Why three?

No real reason, except that KORG decided that a SAMPLE is "sampled", and a WAVEFORM is "created". That's two names out of three. The third name, SOUND, is used by the brain of the DSS-1 to refer to both SAMPLES and WAVEFORMS. Once the SAMPLES / WAVEFORMS have been organized in RAM or on DISK, the DSS-1 always lists them as SOUNDS.

A look ahead: You'll be making up your own disks and charts full of these SOUNDS (SAMPLES / WAVEFORMS). You'll choose SOUNDS from your personal library and mix and match them into MULTI SOUNDS. Then you'll modify the MULTI SOUNDS to make performance PROGRAMS. You'll learn how MULTI SOUNDS and PROGRAMS are then organized into SYSTEMS.

A Bit More About SOUNDS

If you go looking for a SOUND on a disk, you won't find it unless that SOUND was SAVED by using one of the 3 following FUNCTIONS:

1. SAMPLE mode: **F5 SAVE SAMPLE**

2. EDIT SAMPLE mode: **F8 SAVE / RENAME SAMPLE**

3. CREATE WAVEFORM mode: **F3 SAVE WAVEFORM**

If SOUNDS are combined into a MULTI SOUND, and only the MULTI SOUND has been SAVED to a given disk, then the SOUNDS can't be brought up individually from that disk.

You and I know that they're on the disk, since they're part of a MULTI SOUND which we can play and hear. But the computer doesn't know that. Just try DISK UTILITY mode: **F4 SOUND DIRECTORY** while a factory disk is in the drive. You'll get a "No Sounds" message, because none of what you hear from that disk was saved on that disk as a SOUND. What you hear is there only because it was SAVED as part of the MULTI SOUNDS on that disk. These MULTI SOUNDS are listed under DISK UTILITY mode: **F3 MULTI SOUND DIRECTORY.**

A SOUND can be up to 262144 words long. This is how much room there is in the DSS-1 wave RAM.

1. At a sampling rate of 48 kHz, the longest recording you can make will be 5.5 seconds.
2. At 32kHz, 8 seconds.
3. At 24 kHz, 11 seconds.
4. At 16 kHz,16 seconds.

If a single SOUND is the maximum length of 262144 words, it fills the whole wave RAM.

MULTI SOUNDS

Up to 16 SOUNDS can be combined to form a MULTI SOUND. A MULTI SOUND organizes individual SOUNDS (SAMPLES/WAVEFORMS) into keyboard set-ups.

Normally, a MULTI SOUND contains several SOUNDS spread across the keyboard. The total length of all the SOUNDS in a MULTI SOUND can't exceed the capacity of the wave RAM (262144 words).

So, if we try to assemble a MULTI SOUND that uses a SOUND that's 262144 words long, we find that there's no room in the wave RAM for other SOUNDS. The MULTI SOUND can contain only that single SOUND, because the SOUND fills the entire wave RAM.

A look ahead: To use the DSS-1 as a performance instrument, SOUNDS must be organized into MULTI SOUNDS. The MULTI SOUNDS must then be called up by PROGRAMS, which load the playback oscillators. MULTI SOUNDS and PROGRAMS arc then organized into SYSTEMS.

You'll be making your own disks of MULTI SOUNDS, which you'll use to create performance disks.

PROGRAMS

In a performance set-up, you can't play MULTI SOUNDS unless they're called by PROGRAMS. But PROGRAMS don't contain MULTI SOUNDS. All a PROGRAM does is remember a list of settings for a variety of PROGRAM PARAMETERS.

The most important PROGRAM PARAMETERS are **F12 OSC1 MULTI SOUND** and **F13 OSC2 MULTI SOUND**. These parameters tell the DSS-1 which of the MULTI SOUNDS in the current SYSTEM are to be loaded into each of the two oscillators. The same MULTI SOUND can be used in both oscillators.

The next section will show you how drastically the other PROGRAM PARAMETERS can affect the MULTI SOUNDS. By analyzing FACTORY DISKS, you'll see that several PROGRAMS that sound very different call up the identical MULTI SOUNDS in the oscillators. The reason why the PROGRAMS sound so different is because the PROGRAM PARAMETER mode on your DSS-1 gives you access to a very sophisticated synthesizer. How to use this synthesizer takes up the whole section 6 of this book.

The PROGRAM doesn't care about the size of the MULTI SOUND or the number of SOUNDS in the MULTI SOUNDS. The PROGRAM PARAMETERS affect the MULTI SOUND(S) as a whole.

SYSTEMS

When a PROGRAM loads the oscillators with MULTI SOUNDS, it can't pick just any MULTI SOUND. The MULTI SOUNDS can only come from among the group of MULTI SOUNDS that are assigned to the current SYSTEM.

A SYSTEM is the highest level of organization on the DSS-1 Here's how SYSTEM is put together: The DSS-1 requires SOUNDS to be organized into MULTI SOUNDS. MULTI SOUNDS are called up and modified by PROGRAMS. 32 PROGRAMS and a maximum of 16 MULTI SOUNDS form a SYSTEM. The total length of the MULTI SOUNDS in a single SYSTEM can't exceed 262144 words.

This means that the DSS-1 wave RAM can only hold 1 SYSTEM at a time. Up to four SYSTEMS can be stored on a disk - BUT no matter how the MULTI SOUNDS are distributed across the 4 SYSTEMS, the total amount of sound information for all the SYSTEMS on the DISK can't exceed the DISK's capacity: 773 kbytes. (Remember the RULE OF THUMB: RAM times 1-1/2 = DISK).

This is why the SHARING OF MULTI SOUNDS by several SYSTEMS is important. Check with the SYSTEM tables at the back of this book to get a better understanding of all this.

If a MULTI SOUND fills the whole wave RAM (262144 words), then a SYSTEM containing this MULTI SOUND can't contain any other MULTI SOUNDS.

But there could still be 32 variations of that MULTI SOUND available in this SYSTEM. These variations would be created in the 32 PROGRAMS. With enough synthesizer programming wizardry, these PROGRAMS could be made to sound very different from each other.

DISKS

You'll need to differentiate between two types of DISKS:

a) SYSTEM DISKS which are used for performing. These include the Factory Disks, and whatever performance-ready DISKS you'll make.

b) STORAGE/REFERENCE/WORK DISKS where you collect SAMPLES and WAVEFORMS (saved as SOUNDS, for future editing and assembling into MULTI SOUNDS). You'll also collect finished MULTI SOUNDS for future editing into PROGRAMS and SYSTEMS.

Disks can hold 773 BLOCKS (773 kBytes) of SOUND / MULTI SOUND (wave) information. Whether you use all this space up with just two or three monster SAMPLES, or whether you accumulate lots of small SOUNDS or MULTI SOUNDS, <u>525640 wave words is the DISK LIMIT for waveform info</u>. And if you also SAVE PROGRAM PARAMETER data, the number will be a bit less.

Summing Up the Memory Hierarchy

SOUNDS (SAMPLES or WAVEFORMS) are the building blocks for KEYBOARD SETUPS called MULTI SOUNDS.

MULTI SOUNDS are collections of one or more (up to 16) SOUNDS. Each SOUND within a MULTI SOUND is assigned to certain keys on the keyboard

PROGRAMS act as the 'presets' or 'patches' that you call up and play during performances. Each PROGRAM uses one or two MULTI SOUNDS. When you make up your own SOUNDS into MULTI SOUNDS, you then set PROGRAM PARAMETERS to modify and synthesize the combined sound.

SYSTEMS are made up of 1-16 MULTI SOUNDS and 32 PROGRAMS. This is the organization needed for performance. Only 1 SYSTEM can be in the DSS-1 RAM at a time. A SYSTEM can't be larger than the wave RAM (262144 words).

DISKS can hold 1-4 SYSTEMS. The max amount of space on the DISK is 773 kBytes (773 Blocks).

Chapter 6
ANALYSIS OF FACTORY DISK KSD-001

KSD-001 is the first DISK of the original set of four Factory Disks. These first four disks have since been revised. Your first disk may have the number KSDU-001R, in which case there will be some differences. But once you've made your own tables by following the methods explained here, my methods will work just fine for you.

Insert the Factory Disk KSD-001 PIANO into the Disk Drive.

1. Press the tab for DISK UTILITY mode

2. Then select **F3 MULTI SOUND DIRECTORY** (press 3 on the numpad). This will list all the MULTI SOUNDS that are used and shared by the 4 SYSTEMS on this disk. The LCD confirms **F3 MULTI SOUND DIR** and asks to **Insert Disk & Enter**. You've already inserted the DISK.

3. The red light above the ENTER tabs is blinking, so press ENTER - the LCD shows **Searching for M.Sounds on Disk**. As soon as the names of the MULTI SOUNDS have been read into the DSS-1 RAM, the LCD invites you to step through the directory by using DATA ENTRY A.

It shows you the first **MULTI SOUND: A.PF**. Just the name - not its length or any other information. We'll get the specifics in just a moment.

4. Let's read the names one by one, using the UP arrow tab of DATA ENTRY A. I've listed these MULTI SOUNDS in TABLE 1, which also shows the distribution of these MULTI SOUNDS among the 4 SYSTEMS on this disk.

After reading all the MULTI SOUNDS on this disk, we want to find out which MULTI SOUNDS are used by which SYSTEMS. You have to do this one SYSTEM at a time.

5. Enter SYSTEM mode

6. Load SYSTEM A (just like you did in Chapter 3)

7. Select **F7 MULTI SOUND DIRECTORY / FREE SPACE** (press 7 on the numpad).

8. Use DATA ENTRY A - This shows you the MULTI SOUND DIRECTORY FOR SYSTEM A. The LCD also shows you that there are 214 unused words (free space) in the wave RAM. TABLE 1 confirms this: add up the lengths of all the MULTI SOUNDS that SYSTEM A is using. Subtract this total from 262144 and you'll get 214 (free space).

9. Repeat steps 5-8 to see what MULTI SOUNDS are used by SYSTEMS B, C, and D of DISK 001.

As you will see, many MULTI SOUNDS from this disk are being used by more than one SYSTEM.

The numbers you see to the left of the MULTI SOUND names in the LCD simply show in what order the MULTI SOUNDS are remembered as part of a given SYSTEM. The numbers don't tell us in what order the MULTI SOUNDS were saved to the disk. For that we need TABLE 1

We see that SYSTEM A contains 8 MULTI SOUNDS. When SYSTEM A was put together and saved to disk (using SYSTEM mode **F2 SAVE SYSTEM**), its 8 MULTI SOUNDS were the first to arrive on the disk. That's why they're the first 8 MULTI SOUNDS listed in the MULTI SOUND DIRECTORY (DISK UTILITY mode F3).

The first MULTI SOUND, **A.PF**, is made up of actual PIANO audio samples. It is large - 254790 words long, which takes up almost the whole wave RAM. But quite a few PROGRAMS in this SYSTEM use this PIANO MULTI SOUND, in either one or both oscillators. In a minute we'll check out how these audio samples were put together to form this MULTI SOUND.

Since there isn't much room left in this SYSTEM after the PIANO has been included, the other MULTI SOUNDS need to be very short. Remember - the total size of a SYSTEM can't exceed the size of the wave RAM: 262144 words.

You can see that the other MULTI SOUNDS in SYSTEM A are all just 1020 words long. They're not audio-input samples, but created waveforms, synthesized on the DSS-1.

Each of these waveforms is created in the <u>CREATE WAVEFORM</u> mode and is only 1 wavecycle long. In MULTI SOUND mode, each of these waveforms is then looped, and transposed across the keyboard. To make the many variations of the waveforms that you hear, each waveform is treated differently in the PROGRAM PARAMETER mode (the "synthesizer section" of the DSS-I). We'll see in a minute which PROGRAMS in SYS A use which of these created MULTI SOUNDS.

MS#	LENGTH	SYS A	SYS B	SYS C	SYS D
(01) A.PF	254,790	1. 254,790			
(02) HDDAW0	1,020	2. 1,020	1. 1,020	1. 1,020	1. 1,020
(03) HDPW50	1,020	3. 1,020	2. 1,020	2. 1,020	2. 1,020
(04) HDFW12	1,020	4. 1,020	4. 1,020	4. 1,020	4. 1,020
(05) ADCOMBO	1,020	5. 1,020	not used	not used	not used
(06) ADOrgan	1,020	6. 1,020	6. 1,020	6. 1,020	6. 1,020
(07> AD80RHD2	1,020	7. 1,020	not used	9. 1,020	9. 1,020
(08) HD pf	1,020	8. 1,020	10. 1,020	not used	not used
(09) HDPW25	1,020	not used	3. 1,020	3. 1,020	3. 1,020
(10) HDCOMBO	1,020	not used	5. 1,020	5. 1,020	5. 1,020
(11) ADMetal	1,020	not used	7. 1,020	7. 1,020	7. 1,020
(12) HDSIN0	1,020	not used	8. 1.020	8. 1,020	8. 1,020
(13) HD30RH02	1,020	not used	9. 1,020	not used	not used
(14) CLAY	30,001	not used	11. 30,001	not used	not used
(15) Harps!	30,001	not used	12. 30,001	not used	not used
(16) MB	44.001	not used	13. 44,001	not used	not used
(17) HARP	64,001	not used	14. 64,001	not used	not used
(18) EP	30,001	not used	15. 30,001	not used	not used
(19) DW03-A.P	1,020	not used	not used	10. 1,020	10. 1,020
(20) HD0-PW50	1,020	not used	not used	11. 1,020	not used
(21) HDO-SWXO	1,020	not used	not used	12. 1,020	11. 1,020
(22) HD0-PW25	1,020	not used	not used	13. 1,020	not used
(23) HDYAJI0	1,020	not used	not used	14. 1,020	12. 1,020
(24) HDO-PWXO	1,020	not used	not used	15. 1,020	13. 1,020

(25) HDTRG0	1,020	not used	not used	16. 1,020	14. 1,020
(26) ADVIBE	1,020	not used	not used	not used	15. 1,020
(27) ADVOIC.N	1,020	not used	not used	not used	16. 1,020
TOTAL	474,215	261,930	208,205	16,320	16,320
	FREE	214+	53,939+	245,824+	245,824+
	Maximum RAM	262,144	262,144	262,144	262,144

Before we move on to SYSTEM B, let me say this again: *adding up the lengths of the MULTI SOUNDS that SYS A is using, plus the "free space" shown in the directory, brings us to the maximum wave RAM capacity of 262,144 words.*

To check the free space in RAM, select SYSTEM mode **F7 M.SOUND DIR/FREE SPACE**.

Look back at TABLE 1 again: The next 10 MULTI SOUNDS on this disk, HDPW25 to EP, arrived when SYSTEM B was put together and saved to disk.

SYS B makes use of quite a few of the created MULTI SOUNDS that are part of SYS A. But in SYS B, the MULTI SOUNDS #11-15 are new actual audio-input samples [editor note: I think he means #14-18]. At a glance, they only add up to around 198000 words. Even with the 10 new created MULTI SOUNDS that this SYSTEM is using, the total of used memory only comes to 208205. Add the free space (53939) and we're back to 262144 words.

SYSTEMS C and D contain no audio-input samples, only created waveforms. Some are new, and some are shared with SYSTEMS A & B. That's why SYSTEMS C & D use up very little memory - just 16320 words each in the wave RAM.

We now have a list of all the names and lengths of the MULTI SOUNDS on DISK 001 and can calculate how much space these MULTI SOUNDS occupy on the DISK.

Each MULTI SOUND needs to be on the DISK only once to be shared by the SYSTEMS. So, to calculate the total DISK space used by all the MULTI SOUNDS, we just add up the length of each MULTI SOUND once.

The MULTI SOUNDS on DISK 001 add up to 474215 words of wave data. Remember that the maximum amount of wave data that can be stored on DISK is 525640 words: the magic number. Subtract 474215 from 525640 and you get 51425. (The actual amount of space on the DISK is a little less than 51425 words, because PROGRAM PARAMETER information uses up some space.) This means we have room for another audio sample (or samples), or for several created waveforms.

Which SYSTEM could make use of that new audio-input sample (it would have to be made into a MULTI SOUND, of course) ? Any SYSTEM except SYS A, which has run out of free space.

CHAPTER 7
Analysis of Disk KSD-001 SYS A PROGRAMS 1-32

Again - if your first disk has the number KSDU-001R, then you might find some differences between your disk and the one I'm analyzing here (KSD-001 is Disk 1 of the first released set of 4 Factory Disks). Don't worry - the principle is the same.

Now that we know what MULTI SOUNDS are used by SYSTEM A, let's find out which PROGRAMS in SYSTEM A use which MULTI SOUNDS. And since every PROGRAM can assign the same or two different MULTI SOUNDS to the 2 oscillators, we'll look into the oscillators. And we'll also check out the **MIX RATIO**, the **OSCILLATOR OCTAVE**, and the **DETUNE/INTERVAL** settings for OSC 2.

PROGRAM 1 - G.PIANO1

1. Insert Disk KSD-001 PIANO
2. Press SYSTEM mode
3. Select **F1 GET SYSTEM** (press 1 on the numpad)
4. Select SYS A
5. Press ENTER
6. Press YES... and wait.
7. When loading is completed, enter PLAY mode (Cancel SYSTEM mode).
   Now you should have **SYS A P01:GPiano1** in the LCD.
8. Play it - it sounds quite a bit like an acoustic piano, at least when you hold single notes.
9. Enter the PROGRAM PARAMETER mode.
10. Select **F12 OSC1 MULTI SOUND** (press 1 and 2 on the numpad or use DATA ENTRY A).

The LCD informs us that:

   a. Oscillator 1 contains **MULTI SOUND 01 A.PF** and
   b. the length (L) of the MULTI SOUND is 254790 words.

Well, we already knew that (TABLE 1).

CAUTION: IGNORE THE PROMPT **Select & Enter** - DON'T PRESS ENTER

At this stage, both DATA ENTRY A & B are active. DATA ENTRYA allows you to step through the entire list of PROGRAM PARAMETERS.

DATA ENTRY B can have drastic results. It accesses all the MULTI SOUNDS in SYSTEM A and lets you replace the MULTI SOUND currently in OSC1 with another MULTI SOUND. If you move DATA ENTRY B, the LCD will show the name of a new MULTI SOUND, and the ENTER light will blink. If you press ENTER, you'll remove **A.PF** (the piano) from OSCl and replace it with whatever MULTI SOUND is showing in the LCD.

11. Now select **F13 OSC2 MULTI SOUND**. You can do this in one of two ways: either press 1 and 3 on the numpad, or use DATA ENTRY A.

The LCD informs you that Oscillator 2 contains the same **MULTI SOUND 01 A.PF**, still 254790 words.

12. Ignore the **Select & ENTER** prompt again – we'll substitute MULTI SOUNDS later. Instead, select **F14 MIX RATIO** (use the numpad or DATA ENTRY A). The LCD tells you how much we hear from each oscillator. The total is always 100%.

In this case we hear 100% from OSC1, nothing from OSC2. Does it matter? Are we missing anything? Not really, you might think, because there's nothing new in OSC2 - both oscillators contain **A.PF** (the piano). So what good would it do to hear a bit bom OSC2? Wouldn't it sound just the same ?

Not necessarily. There could be differences in the way the 2 oscillators play the same MULTI SOUND.

NOTE: The following PROGRAM PARAMETERS affect each OSCILLATOR individually:

F11 Oscillator Octave
F12 Oscillator 1 Multi Sound
F13 Oscillator 2 Multi Sound
F14 Mix Ratio
F15 Oscillator 2 Detune & Interval
F17 Oscillator Modulation
F18 Autobend Mode

13. To change the **MIX RATIO**, use the DATA ENTRY slider B and set a percentage of 49/51.

14. Select **F11 OSC OCT**. (Use the DATA ENTRY A or the numpad). You see that both oscillators are set to the 8' range.

Ever since the days of pipe organs, pitch range has been measured in feet. Double a number and you move 1 octave down -- halve a number and you move an octave up.

15. The cursor blinks under the value for OSC1. Play a chord, then tap the UP arrow tab of the DATA ENTRY B once, changing OSC1 to the 4' range. Now play the same chord again - it sounds as if you played with both hands, 1 octave apart.

16. Now change OSC1 down to 16'. Play the same chord again – it sounds almost like a 12-string guitar, or like a two-handed piano. You hear the new chord 1 octave lower.

17. Reset OSC1 to 8'.

18. Select **F15 OSC 2 DETUNE & INTERVAL**. The LCD shows that OSC2 is tuned no differently than OSC1 - 00 value for Detune, 00 value for Interval.

F15 OSC 2 DETUNE & INTERVAL is different from **F11 OSC OCT**. Here you can detune OSC2 against OSC1, regardless of the octave ranges of the two oscillators. You can also set OSC2 to sound a specified number of half steps higher than OSC1 Let's try it.

19. While the cursor is under the value for DETUNE, tap the UP arrow tab B once and play. Repeat this a few times - at first it sounds like a flanger effect, but soon it gets HONKY-TONK, then dreadful.

20. Reset DETUNE to zero.

21. Move the cursor over (use the YES tab) and change the INTERVAL. Now you're dealing with set intervals, indicated in amounts of halfsteps.

NOTE: The changes to the piano program are not permanent - the original version is still on disk.

On the next page is a table with all the PROGRAMS from DISK 001, SYSTEM A. For each PROGRAM, it shows the MULTI SOUND for each OSCILLATOR, each OSCILLATOR's **OCTAVE** range, the **MIX RATIO**, and the **OSCILLATOR 2 DETUNE and INTERVAL** settings.

Now let's analyze this Piano some more. Surely it wasn't sampled (recorded) note by note -- there isn't enough memory for that, and only 16 SOUNDS can be part of one MULTI SOUND.

TABLE 5: FACTORY DISK KSD-001 SYSTEM A PROGRAMS, OSC, and MULTI SOUNDS

PROG #	PROG NAME	MS OSC1	MS OSC2	MIX %	OCT	DETUNE	INTERVAL
1	G.Piano1	A.PF	A.PF	100/0	8/8	0	0
2	G.Piano2	A.PF	A.PF	100/0	16/16	0	0
3	G.Piano3	A.PF	A.PF	100/0	4/4	0	0
4	G.Piano4	A.PF	A.PF	100/0	16/16	0	0
5	G.Piano5	A.PF	A.PF	100/0	8/8	0	0
6	G.Piano6	A.PF	A.PF	100/0	8/8	0	0
7	Piano7	A.PF	AD80RH02	50/50	16/16	1	0
8	G.Piano8	A.PF	A.PF	80/20	8/8	2	0
9	G.Piano9	A.PF	A.PF	54/46	16/8	0	0
10	G.Piano2	A.PF	A.PF	100/0	8/8	0	0
11	OrganCY3	HDSAW0	HDSAW0	52/48	16/16	4	0
12	G.Piano1	A.PF	A.PF	100/0	16/16	0	0
13	Voices1	ADCOMBO	HDSAW0	52/48	16/8	8	0
14	Voices2	ADCOMBO	HD pf	55/45	8/8	4	0
15	Voices3	ADCOMBO	ADCOMBO	65/35	16/16	5	0
16	Mix	A.PF	AD80RH02	60/40	16/16	0	0
17	BR3/Bell	HDSAW0	AD80RH02	0/100	16/16	5	0
18	BR3&Bell	HDSAW0	AD80RH02	60/40	16/4	1	0
19	E.PIANO1	AD80RH02	A.PF	85/15	16/4	1	7
20	E.PIANO2	AD80RH02	AD80RH02	72/28	8/8	5	0
21	STRINGS1	HDSAW0	HDSAW0	55/45	8/8	4	0
22	STRINGS2	HDSAW0	HDSAW0	55/45	16/16	2	0
23	BRASS	HDSAW0	HDSAW0	50/50	16/16	3	0
24	BRASS 0	HDSAW0	HDSAW0	100/0	16/16	3	0
25	ORGAN 1	ADOrgan	ADOrgan	60/40	16/4	0	0
26	ORGAN 2	ADOrgan	ADOrgan	52/48	16/4	0	7
27	Pianob0M	A.PF	AD80RH02	70/30	16/16	2	0
28	Sun Rise	A.PF	AD80RH02	40/60	16/16	2	0
29	Synth 1	HDSAW0	ADCOMBO	34/66	16/16	0	11
30	Synth 2	HDPW12	HDPW50	48/52	4/4	0	0
31	Synth 3	ADCOMBO	ADCOMBO	56/44	16/8	0	0
32	Synth 4	HDSAW0	A.PF	65/35	16/16	1	0

Chapter 8
ANALYSIS OF THE PIANO SAMPLE A.PF IN SYSTEM ON THE FACTORY DISK KSD-001

Having messed around with this piano sound in the last chapter, we'd better re-load it from disk IN ITS ORIGINAL VERSION. If you've been using the second-generation Disk KSDU-001R, then keep using it in the place of KSD-001, and make a note of the differences.

1. Insert FACTORY DISK KSD-001 PIANO.
2. Enter SYSTEM mode.
3. Select **F1 GET SYSTEM** (press 1 on the numpad).
4. Make sure that the LCD says **Get SYSTEM A**
5. Press ENTER.
6. Press YES and
7. Wait
8. Enter EDIT SAMPLE mode.
9. Select **F1 SELECT SAMPLE** (press 1 on the numpad).

10. The LCD asks **...from MEMORY or DISK?** We've already loaded it from disk into the memory, and the cursor is under **MEMORY**, so press ENTER. The LCD now shows **MULTI SOUND 01 A.PF** and its familiar length of 254790, asking us to select it, or another MULTI SOUND.

11. We want this PIANO, so press ENTER. Now the LCD asks us to select one of the SOUNDS that make up the MULTI SOUND A.PF. It doesn't have a name for it, just a number. Currently it shows 01.

12. Press ENTER. The LCD shows this first SOUND in more detail, asking us to check and confirm that it is what we want: **MULTI SOUND 01/Sound 01**. We see that this SOUND 01 is 64104 bytes long, and that it was sampled at a Sampling Rate of 32 kHz.

13. Let's GET it - press YES. For a brief second the LCD says **This will take a while**, but immediately we get the message **F1 M01;S01 Got**, and **ORG=C3 TOP=F3 TR**.

This translates into:

F1 is Function 1 SELECT SAMPLE in EDIT SAMPLE mode.
M01 is MULTI SOUND 01.
S01 refers to SOUND 01 in MULTI SOUND 01.

"Got" means that the DSS-1 has retrieved the SOUND for us by isolating it from the MULTI SOUND and by putting it into the WAVE RAM.

Remember that the WAVE RAM is a memory area that contains either the SYSTEM with its PROGRAMS and MULTI SOUNDS, or a SOUND (SAMPLE) that we want to treat individually.

ORG=C3 refers to the key that is currently assigned to play back the sampled pitch of the original recorded version of SOUND 01. On the DSS-1, C3 is the second C from the left of the keyboard.

TOP=F3 / TR means this: The DSS-1 is programmed to transpose the original pitch of SOUND 01 from C3 upwards in half steps, but only as far as F3. TR means TRANSPOSE. The opposite is NT – NOT TRANSPOSE. When NT is on, the TOP KEY ASSIGNMENT becomes meaningless: all keys will play back the original sampled pitch.

With TR showing, the DSS-1 will transpose down as far as the TOP KEY of the next lower SOUND.

Right now the SOUND in the WAVE RAM (SOUND 01) is "naked", the way it was sampled and edited BEFORE it became part of the MULTI SOUND.

Listen to the way the sound cuts off when you hold a note - the "analog" PROGRAM PARAMETERS that eventually give the completed MULTI SOUND its musical usefulness are missing.

14. Play C3 (the second C from the left) - can you tell by ear what the pitch of the original sample was ? Check it against another keyboard, or with a chromatic tuner - you'll find that it is A110, which on the piano would be the A an octave plus a minor third below middle C.

So this is our first SOUND from within the MULTI SOUND A.PF.

Now let's try to get the second SOUND. F1 SELECT SAMPLE seems the logical choice. But pressing 1 on the numpad gets us nowhere. Moving the DATA sliders only changes the values in the LCD - neither slider helps us select another SOUND. And the ENTER light is still blinking.

There seems to be only one choice:

Press ENTER -- the LCD proudly says SAMPLE SELECTED and invites you to select any Function from within EDIT SAMPLE mode. Again, the logical choice teems to be F1 SELECT SAMPLE.

But we can't use F1 to select the next SOUND from within this PIANO MULTI SOUND A.PF. If we try to select it with F1 and ...from MEMORY, we end up with a message: Select M.SOUND 01 !NO-NAME L=064104. This turns out to be the SOUND we have just been dealing with, instead of the next SOUND. Furthermore, when we try to use the DATA ENTRY sliders to display the other SOUNDS in MULTI SOUND 01 we get nowhere.

Why are we locked out ?

It turns out that the process of retrieving a SOUND from within a MULTI SOUND causes the DSS-1 brain to forget that it has the complete MULTI SOUND in memory once it has isolated the first SOUND.

So let's change tactics: try to select the next SOUND with F1 and ...from DISK. You'll get a No Sounds message. Why? Because Korg's programmers never saved these piano samples as individual SOUNDS on this disk, only as part of the MULTI SOUND A.PF, and the DSS-1 only recognizes them as such.

By placing a single SOUND/SAMPLE into the memory via EDIT SAMPLE mode, we have locked out all other SOUNDS or MULTI SOUNDS. Before we can get the next SOUND from within MULTI SOUND A.PF (or any other MULTI SOUND for that matter), we must re-load SYS A from the disk.

16. Enter SYSTEM mode.

17. Select **F1 GET SYSTEM** (press 1 on the numpad).

18. Use DATA ENTRY A to set the LCD to read **Get SYSTEM A**.

19. Press ENTER.

20. Press YES and wait

21. When loading is completed, enter EDIT SAMPLE mode.

22. Select **F1 SELECT SAMPLE** (press 1). The cursor suggests getting a sample from **MEMORY**.

23. Press ENTER. The LCD shows us the first MULTI SOUND - it is up to us to know which MULTI SOUND has the SOUND that we want to isolate. Here the first one happens to be the right one: **A.PF**

24. Press ENTER. Now the LCD shows the first of the SOUNDS (01) that make up this **MULTI SOUND 01 A.PF**. We need the second one.

25. Tap the UP-arrow tab of DATA ENTRY A to show **Select SOUND 02**.

26. Press ENTER. The LCD now displays this SOUND 02 in more detail: **F1 Get M01:S02 (Y/N)** with a LENGTH of 63095 and a Sampling Rate of 32kHz.

27. Press YES. This second SOUND of the piano MULTI SOUND also shows as **ORG=C3 / TOP=F3**.

28. Play C3 (the second C from the left) - you hear A220, which would be A below middle C on a piano.

So far we know that the piano strings A110 and A220 were recorded as part of this MULTI SOUND. Let's see what other notes were sampled.

29. Press ENTER to get out of this display.

30. Enter SYSTEM MODE

31. Select **F1 GET SYSTEM** (press 1).

32. Use the DATA ENTRY A to get the LCD to read **GET SYSTEM A**.

33. Press ENTER.

34. Press YES and...wait. When loading is completed

35. Enter EDIT SAMPLE mode.

36. Select Fl (press 1). We've just loaded the information into the MEMORY, so

37. Press ENTER while the cursor is under **MEMORY**. Again, our **A.PF MULTI SOUND 01** is showing

38. Press ENTER. We want the third SOUND from within this MULTISOUND

39. Tap the UP-arrow tab of DATA ENTRY A to show **SOUND No.=03**

40. Press ENTER. We're shown this SOUND in more detail: **Get M01:S03 (Y/N) L=64663 SAMPLING FREQUENCY=32 kHz**

41. Press YES.This also comes op as **ORG=C3/TOP=F3**

42. Play the second C from the left -- you hear tuning A440, which is the first A above middle C.

43. Repeat steps 29 through 41 from above to re-load SYSTEM A, except for step 39 where you choose SOUND 04, which is 30976 bytes long and, also at C3, plays the G at 784 Hz, which on the piano is 1 octave and a fifth above middle C

44. Repeat steps 29 through 41 again to get SOUND 05, which is 31952 bytes long, was sampled at 32kHz and, also at C3, plays G# at 1661 Hz. This is the last SOUND from MULTI SOUND **A.PF**

Adding together the lengths 64104 + 63095 + 64663 + 30976 + 31952 gives us 254790, which you knew all along to be the total length of this MULTI SOUND A.PF

So why were all the SOUNDS allocated to C3?

As you'll see when we get to audio-input sampling, a new sample gets automatically allocated to C3, no matter what (if any) its own musical pitch is. It's ORG (original) KEY can then be mapped anywhere on the keyboard (SAMPLE mode **F4 ORIGINAL / TOP KEY**). The TOP KEY (transposition) can be assigned to be no more than 1 octave higher than the key that plays back the original pitch (ORG KEY).

This lets you test keyboard set-ups at the time of sampling. But these assignments are not permanent.

You can save the sample (with its newly assigned ORG and TOP KEYS) as a SOUND. But when you call this SOUND back up from a disk, it will again be allocated to ORG C3.

HOWEVER: The interval that you had set between ORG and TOP will be remembered. Keyboard Assignment only becomes permanent when a SOUND becomes all or part of a MULTI SOUND (MULTI SOUND mode F3).

Chapter 9
ANALYSIS OF THE PIANO SAMPLE A.PF ON DISK KSD-001 AS A MULTI SOUND

Let's recap what we know about this MULTI SOUND

a) When loaded, it takes up most of the RAM, being 254790 bytes long.

b) While on disk, it takes up close to half the disk memory.

c) It consists of 5 SOUNDS that were all sampled at the rate of 32 kHz. The pitches sampled are:

A2 (110 Hz, MIDI Note #45)

A3 (220 Hz, MIDI Note #57)

A4 (440 Hz, MIDI Note #69)

G5 (784 Hz, MIDI Note #79)

G#6 (1661 Hz, MIDI Note #92)

Now it's time to find out how they're spread across the keyboard when we hear the whole MULTI SOUND. For this, we'll call up the MULTI SOUND **A.PF** into the DSS-1 RAM. It'll be complete, but without the PROGRAM PARAMETERS that it has attached when it's part of the loaded SYSTEM. This will show us the "naked" MULTISOUND.

We need to clear the DSS-1 memory.

1. Turn the power OFF and back ON. As always, the LCD shows **SYSTEM MODE (Select 1-9)**. This time we don't want to load a SYSTEM - just this MULTI SOUND **A.PF**

2. Select **F9 GET MULTI SOUND**. Make sure that DISK 001 is in the Drive, and

3. Press ENTER, The LCD says **Searching for MULTI SOUNDS on DISK**

4. When it's read all the MULTI SOUND names from its directory on the DISK, it wants you to make your selection, in two stages:

a. First you use DATA ENTRY A to bring the desired MULTI SOUND name into the LCD, then
b. you press ENTER, to tell the computer to get the information from the disk.

5. We want the first MULTI SOUND to appear in the LCD. Just tap the UP arrow tab A, verify **M.SND01A.PF** and press ENTER. The LCD shows you the LENGTH of the selected MULTI SOUND and asks you if this is really what you want.

6. Press YES, and the MULTI SOUND will be loaded from the disk. Note how long it takes: Almost as long as a SYSTEM, because it has almost the same length.

7. Now the LCD says **Loaded** and asks if you wish to continue, Yes or No. "Continue" means 'Getting more MULTI SOUNDS', as if we were building a SYSTEM. Well, we're not, so

8. Press NO.

9. This brings us back to the main screen of SYSTEM MODE, with our PIANO MULTI SOUND **A.PF** now in the wave RAM, ready to be played and analyzed.

10. Press the MULTI SOUND mode tab, and select **F3 ORIGINAL/TOP KEY** (press 3 on the numpad). The difference between this function and SAMPLE mode **F4 ORIGINAL/TOP KEY** is this: keyboard assignments made under MULTI SOUND MODE F3 are permanent within that MULTI SOUND.

As we saw in Chapter 8 - this isn't true for SAMPLE Mode F4.

11. We see SOUND 01 listed as **ORG = A2**, **TOP = F3**, with the **TR**anspose function enabled. Play the lowest A on the keyboard – it plays back the original SOUND #01 at the original pitch. Other notes from the bottom of the keyboard up to the second F are instant transpositions of this sampled A, transposed by the computer as you play them.

You'll also hear that this SOUND is looped to sustain at full volume for as long as you hold the keys down. As you'll see later on, this helps to conserve memory. In case you're wondering why we didn't hear the organ-like sustain when we played the sound from within the SYSTEM, it is because in a SYSTEM, the PROGRAM PARAMETERS add a lot of characteristics that were carefully programmed. Here we don't hear the PROGRAM, just the naked MULTI SOUND.

12. Select **F1 SELECT M.SOUND/SOUND** (press 1 on the numpad).

13. Move the cursor under the value for SOUND, and select **SOUND 02** with the DATA ENTRY A.

14. Select **F3 ORIGINAL/TOP KEY** (press 3 on the numpad), and you get the display for SOUND 02. As expected, the key for the original pitch is A3. Transposition of this sample is possible up to D4. But this sample also gets transposed DOWN, as far as F#4. We don't see that, but it follows on from the settings for SOUND 01 (see #11).

The DSS-1 allocates keys and transposition ranges from the bottom up. Any keys claimed by a lower SOUND can't be allocated to a higher SOUND within the same MULTI SOUND.

SOUND 02 in our MULTI SOUND stops at F#3 when you play it down from its original key of A3, because it runs up against F3, the TOP KEY of SOUND 01.

15. Check out the remaining SOUNDS 03, 04, 05, by using F1, cursor, Data Entry A and F3, just like we did before. You'll find their ORG keys / TOP keys to be A4/D5, G5/D6, G#6/F7.

This is part of the magic of digital sampling: Just 5 actual piano notes are digitally reproduced over the entire width of a 5-octave chromatic keyboard. Via MIDI control, this range would be even wider.

Chapter 10
FACTS AND FIGURES - AN INTERMEDIATE SUMMARY

Max WAVE MEMORY per DISK	525640 words
Max WAVE MEMORY in RAM (SYSTEM or SEMPLE)	262144 words
Max WAVE MEMORY per SYSTEM	262144 words
Max MULTI SOUNDS per SYSTEM	16
Max SYSTEMS per DISK (A/B/C/D)	4
Max PROGRAMS ("presets") per SYSTEM	32
Max MULTI SOUNDS per PROGRAM (1 MS per OSC)	2
Max KEYBOARD SPLITS (SOUNDS per MULTISOUND)	16
Max SAMPLE TIME @ 16Khz SAMPLING RATE	16 seconds
Max SAMPLE TIME @ 24Khz SAMPLING RATE	11 seconds
Max SAMPLE TIME @ 32Khz SAMPLING RATE	8 seconds
Max SAMPLE TIME @ 48Khz SAMPLING RATE	5.5 seconds
Approx HIGH FREQ RESPONSE @ 16kHz	6 kHz
Approx HIGH FREQ RESPONSE @ 24kHz	10 kHz
Approx HIGH FREQ RESPONSE @ 32kHz	13 kHz
Approx HIGH FREQ RESPONSE @ 48kHz	20 kHz
Default Status after Power On	SYSTEM MODE (Select 1-9)
Default PLAY MODE	All MODE lights OFF
Average SYSTEM LOADING TIME	30-40 seconds
Average SYSTEM SAVING TIME	2 minutes
Approx. DISK FORMATTING TIME	2 minutes

3 PROGRAM SELECTION METHODS:

a) FOOTSWITCH
b) ARROW TABS A & B
c) NUMERICAL "NUMPAD"

SECTION TWO:
BE CREATIVE - MODIFY THE FACTORYSOUNDS/DISKS

The first generation of Factory Disks (KSD-001/002/003/004) have a number of unfinished PROGRAMS labelled USER. That means you. It's your chance to make up your own PROGRAMS using the MULTI SOUNDS on Factory Disk.

You saw in the Analysis chapters how much a MULTI SOUND can be varied with the versatile synthesizer programming that's available on the DSS-1. Sure, sampling can be fun. But it's time-consuming and often tricky, so why not take advantage of the MULTI SOUNDS already on the Factory Disks?

But you'll want to preserve the original disks. You should only work on copies, so that any mistakes or mishaps won't spoil your day too much.

Chapter 11
HOW TO MAKE BACK-UP COPIES OF FACTORY DISKS

a) FORMATTING NEW DISKS - DISK UTILITY mode F0.

Just a reminder: Use only 3.5" Micro Floppy Disks of the double-density kind, usually labeled MF2DD. Before you can use a new disk, it needs to be formatted (initialized). The MEMORY PROTECT TAB must be away from the corner, in the "unprotected" position.

CAUTION: *FORMATTING ERASES ALL DATA ALREADY ON DISK.*

In other words, whether you format a brand new disk or a previously used disk, after formatting you'll have a blank, ready-for-work disk.

1. Insert the DISK to be formatted into the DISK DRIVE.
2. Enter DISK UTILITY mode.
3. Select F0 FORMAT DISK
4. Press ENTER.
5. Press YES.

At the end of the formatting operation, the LCD asks CONTINUE? Y/N

6a) Press YES if you have another disk to format.
6b) Press NO if you want to go on to something else.

If you press YES by mistake, the ENTER blinks and the LCD wants another disk. You madly press NO NO NO and nothing happens... But you're NOT locked out. Just press any MODE tab to escape.

Error Messages

If you get an ERROR message, but only once in a while, don't worry; It may have been a temporary hiccup. Just leave the disk in the drive and start again from #3 above.

If ERROR messages persist, then you need to check your disk. Is it warped? Does the metal part snap back after you pull it ride-ways? Is the MEMORY PROTECT tab properly away from the corner position? If nothing seems to work, give up on the disk rather than risking the drive or a loss of data later an. New disks from reputable sources rarely have problems and are usually covered by some kind of warranty.

Make it a habit to format new disks as soon as you bring them home. Attach a label, mark it with a small F somewhere - BE READY.

b) COPYING SYSTEM A

A Factory Disk can only be copied one SYSTEM at a time.

1. Insert the Factory Disk KSD-001 (PIANO) (or the newer version KSDU-001R).
2. Select SYSTEM mode.
3. Select **F1 GET SYSTEM** (press 1 on the numpad).
4. When **GET SYSTEM A** shows in the LCD, press ENTER.
5. Press YES.
6. When loading is completed, eject the Factory Disk and insert your new, formatted disk.
7. Still in SYSTEM mode, select **F2 SAVE SYSTEM** (press 2 on the numpad).
8. Verify that the LCD says **SAVE SYSTEM A**. (If for some reason you want SYSTEM A to be saved as another SYSTEM (B, C or D), you can use the DATA ENTRY slider A to make that change).
9. Press ENTER.
10. Press YES, and...wait.

What's being written onto the disk is this: All the MULTI SOUNDS that this SYSTEM uses, plus the PROGRAM information of SYSTEM A. Your attention is not required during the SAVING of this first SYSTEM. But during the following operation, you'll need to watch the LCD.

c) COPYING SYSTEM B

11. Now remove your back-up copy DISK and insert the Factory Disk.
12. Still in SYSTEM mode, select **FI GET SYSTEM** (press 1 on the numpad).
13. When the LCD shows **Get SYSTEM A**, use DATA ENTRY A to change it to **SYSTEM B**
14. Press ENTER.
15. Press YES and...wait.
16. When loading is completed, remove the Factory disk and insert your back-up copy.
17. Still in SYSTEM mode, select **F2 SAVE SYSTEM** (press 2), verify that it says **SAVE SYSTEM B**
18. Press ENTER.
19. Press YES, and watch the LCD.

When the DSS-1 comes across a MULTI SOUND in SYSTEM B that was already part of SYSTEM A, it stops SAVING, shows you the name of the MULTI SOUND in question, and asks if you want to DELETE the "old" version. It won't accept two MULTI SOUNDS with an identical name.

"Old" refers to the version that was saved with SYSTEM A.

Remember Table 1, where we analyzed which MULTI SOUNDS were shared by two or more SYSTEMS? There can't be two MULTI SOUNDS with the same name on a disk. So either the "old" one has to be replaced, or the "new" one won't get written.

With this Factory DISK we don't really care - they're the same MULTI SOUNDS under either SYSTEM. Answer YES or NO, depending on the color of your socks. But pressing NO saves time, because the new MULTI SOUND doesn't have to be written over the old one on the DISK - the DSS-1 just moves on.

At other times, however, whether you press YES or NO will make a difference. Namely, if you're copying SYSTEMS from different disks onto a new disk. (See Chapter 12).

On your various source-disks, you might have differently sounding, edited versions of the same MULTISOUND, or different MULTI SOUNDS which you've given the same name. That's not good housekeeping, but it's no problem as long as the MULTI SOUNDS are on separate DISKS.

But the DSS-1 won't SAVE two MULTI SOUNDS with the same name on the same disk. So it alerts you every time you try to SAVE a MULTI SOUND already named on the DISK. You have to make a decision which version of the MULTI SOUND you want to SAVE.

A word about what happens when you press NO in answer to the question **Delete MULTI SOUND (Y/N)?** one of two things will happen. Either the LCD will immediately show you another "shared" MULTI SOUND and ask you the same question, or it will notify you that the SAVING operation has been restarted. This means that a "new" (not shared) MULTI SOUND is being SAVED on DISK.

If you need the different versions of the same MULTI SOUND on the same disk, just give one of them a new name. You'll need to take time out for that -- rename/resave it in MULTI SOUND mode, then rewrite a PROGRAM for it, save the PROGRAM, resave your SYSTEM on the source disk and on the copy. But we're not there yet...

c) COPYING SYSTEMS C&D

Proceed as above. Now you have an exact copy of the factory disk. Keep the original in a safe place and use only the copy.

Chapter 12
COPYING SYSTEMS FROM SEPARATE DISKS TO A NEW DISK

Let's combine SYSTEMS from KSD-002 and KSD-001 onto a new disk.

1. Have a newly-formatted (blank) DISK handy.
2. Make sure that the MEMORY PROTECT tab on the new DISK is in the "unprotected" position.
3. Insert the Factory Disk 002.
4. Enter SYSTEM mode.
5. Select **F1 GET SYSTEM** (press 1 on the numpad).
6. Set LCD to read **Get SYSTEM B** (use DATA ENTRY A).
7. Press ENTER.
8. Press YES and...wait.
9. When loading is completed, eject the Factory DISK.
10. Insert the new blank DISK.
11. Still in SYSTEM mode, select **F2 SAVE SYSTEM** (press 2 on the numpad).
12. Use the DATA ENTRY A to change **...SYS:B** to read **...SYS:A**
13. Press ENTER .
14. Press YES and...wait.

This copies SYSTEM B from the FACTORY DISK to your disk, where it is SAVED as SYSTEM A.

When SAVING is complete, proceed with the next SYSTEM.

15. Eject your new DISK
16. Insert the FACTORY DISK KSD-001.
17. Still in SYSTEM mode, select **F1 GET SYSTEM** (press 1 on the numpad).
18. Set the LCD to read **GET SYSTEM:A** (use DATA ENTRY A).
19. Press ENTER.
20. Press YES and ...wait.
21. When loading is completed, eject the FACTORY DISK
22. Insert your new DISK
23. Still in SYSTEM mode, select **F2 SAVE SYSTEM** (press 2 on the numpad).
24. Use DATA ENTRY A to change the readout from **...SYS:A** to **...SYS:B**
25. Press ENTER.
26. Press YES and watch the LCD.
27. Every time SAVING stops and the LCD asks you whether you wish to **DELETE** the MULTI SOUND that appears in the readout, press NO. You know that on the original Factory Disks, MULTI SOUNDS with the same name are, in fact, the same MULTI SOUND.

In the future when you have your own library of customized DISKS, it will be up to you to know that the same data are hiding behind the same name.

Whenever you make a personalized version out of a factory SOUND or MULTI SOUND, you should at least add a symbol or letter to the name, to remind you in the future of the fact that you have made a change.

Getting the Third System into RAM

After the stop/start SAVING SYSTEM operation is done, you have to get the third SYSTEM into RAM

28. Eject your new DISK.
29. Insert the FACTORY DISK 002.
30. Still in SYSTEM mode, select **F1 GET SYSTEM** (press 1 on the numpad).
31. Using the DATA ENTRY A, set the readout to **GET SYSTEM A**.
32. Press ENTER.
33. Press YES and ...wait
34. When loading is completed, eject the Factory DISK.
35. Insert your new DISK.
36. Still in SYSTEM mode, select **F2 SAVE SYSTEM** (press 2 on the numpad).
37. Set the LCD to read **...SYS:C** (use DATA ENTRY A).
38. Press ENTER.
39. Press YES and watch the LCD.

PANIC ! DISK FULL !

Ah well - we should have done our homework, checking our tables before loading. These SYSTEMS contain MULTI SOUNDS that add up to more than the magic number of 525640 words that the DISK can hold. Although we know that these SYSTEMS share a lot of MULTI SOUNDS (all those that we had to "Delete"), those aren't the culprits, since they're only very short, synthesized MULTI SOUNDS - 1020 bytes long. It's the real audio-input samples that take up too much room.

What to do?

It's time to get the adding machine out. Adding up the MULTI SOUNDS from the various SYSTEMS will tell you in advance whether it is possible to SAVE a certain combination of SYSTEMS on a single DISK.

Remember Chapter 1? Did it put you to sleep when you first looked at it (or did you even look at it)? All that talk about bits and bytes and words and memory capacities? It now turns out you actually need to understand it...

So if you don't feel comfortable about figuring out whether a combination of SYSTEMS or MULTI SOUNDS will fit on a DISK, check out Chapter 1. It's really not as bad as it seems.

If you have Chapter 1 nailed, then you'll understand that there's a foolproof way to tell ahead of time whether a combination of SYSTEMS or MULTI SOUNDS will fit on a single DISK:

a) Add up the lengths of all the MULTI SOUNDS (if a MULTI SOUND is shared, include it once)
b) Multiply the total length of all the MULTI SOUNDS by 1.5.
c) If the result is greater than approximately 770,000, your combination won't fit.

The next few chapters show you different ways of dismantling and re-assembling MULTI SOUNDS, PROGRAMS, and SYSTEMS. With a bit of practice, you'll soon get the hang of it.

Chapter 13
FILLING IN A "USER" PROGRAM IN A FACTORY SYSTEM

Have you made a WORKING COPY of DISK KSD-002 BRASS? If not, chapter 11 tells you how.

Now we'll use your working copy of DISK KSD-002. Some Systems on this and other Factory Disks contain some PROGRAMS that either make no sound or produce the same sound as other PROGRAMS. They are labeled USER – that means YOU can bring them life.

Why do some of them make no sound? Because the programmer(s) of the 4 FACTORY DISKS have set certain critical parameters to values of zero. Why have they done this? To encourage the user to make his / her own PROGRAMS and SAVE them in spots currently occupied by USER PROGRAMS.

Three Ways to Bring a User Program to Life

There are basically three ways to bring a USER PROGRAM to life:

1. Every USER PROGRAM already has 1 or 2 MULTI SOUNDS assigned to the oscillators. You can raise the VCA and VCF levels above zero and go from there. Initially, the PROGRAM might sound very similar to other PROGRAMS in the SYSTEM. But as you adjust more and more PROGRAM PARAMETERS, you're bound to end up with a PROGRAM that has your own personal sound.

2. You can do what we're about to practice in this Chapter: modify any of the PROGRAMS that you've already heard in this SYSTEM. You'll save your new version where the old USER PROGRAM was.

3. You can make a PROGRAM from scratch. For that, you have to call up a USER PROGRAM and "initialize" it with **PROGRAM PARAMETER 00**. This replaces whatever PARAMETER values were set in the USER PROGRAM with a set of default values. If you make a PROGRAM from scratch, that PROGRAM must use MULTI SOUND(S) already in the SYSTEM. See Chapter 15 to find out what to do if you want to construct a PROGRAM that uses MULTI SOUND(S) not already in the SYSTEM.

Let's construct a USER PROGRAM as discussed above. We'll modify PROGRAM 26:ORGAN 2 of SYSTEM D (DISK KSD-002) and save it in PROGRAM 16:USER.

This will involve changing the values of a lot of the analog PROGRAM PARAMETERS. If you're not comfortable with this type of synthesis, just set all the values as I list them and save the new program.

1. Insert your back-up copy of FACTORY DISK KSD-002.
2. Enter SYSTEM mode.
3. Select **F1 GET SYSTEM** (press 1 on the numpad).
4. Use DATA ENTRY A to set the readout to **GET SYSTEM:D**
5. Enter PLAY mode (cancel SYSTEM mode).
6. Select **PROGRAM 26:ORGAN 2**

You'll be changing this PROGRAM beyond recognition. NOTE This change doesn't affect the PROGRAM 26-ORGAN 2 permanently, because you'll save it to PROGRAM 16:USER.

7. Enter <u>PROGRAM PARAMETER</u> mode.

Below you'll see a list of PROGRAM PARAMETERS that you'll need to select using either the numpad tabs or DATA ENTRY A. Each time you select a PARAMETER, certain values will appear in the LCD. Compare those values with the ones listed in the table and adjust them where necessary. For this you'll use the CURSOR and DATA ENTRY B. The CURSOR will get you around the LCD whenever you have more than one value to adjust.

You may have to press ENTER - keep an eye on it. If it blinks, hit ENTER. If you see the word "any" in my list of values, it means that you can set any value you want, including the value that's already there.

Play the keyboard intermittently - you'll hear the sound change as you go.

8. Using the numpad tabs, the CURSOR, and the DATA ENTRY tabs/sliders, set the following:

F11	16' / 4'	F12	07:ADMetal	F13	08:ADSIN0
F14	40% / 60%	F15	61 / 11	F16	OFF / 12bit
F17	OFF / 3x any	F18	BOTH/DOWN	F19	001 / 17
F21	00	F31	24dB / +	F32	096 / 33
F33	21 / 50	F34	any/00/any	F35	00/19/28/37/00/41
F36	63	F37	41	F38	00/29/00/00/00/51
F41	03	F42	19	F43	00/00/00
F44	32	F45	00/00/00	F46	OFF / any
F51	00	F52	any / 00	F53	00
F63	Poly-1 or -2	F65	+6 / +3	F71	09 / 21

Now you need to make sure that this becomes PROGRAM 16.

9. Still in PROGRAM PARAMETER mode, select **F01 WRITE/RENAME** (press 0 and 1 on the numpad).

10. Press YES, then use the DATA ENTRY B and the cursor to come up with any name you want.

11. Press ENTER. Now the DSS-1 wants to know whether this PROGRAM is to remain #26. This would wipe out and replace the current PROGRAM 26:ORGAN 2. The CURSOR is blinking under 26.

12. Change the number to 16 (with the DATA ENTRY tabs B). Next, the DSS-1 wants to know if we want this PROGRAM to be remembered in RAM. Sure we do.

13. Press YES.

14. Do we want to **Continue ?** No - we're using just this PROGRAM. Let's put it in the SYSTEM.

15. Enter SYSTEM mode.

16. Select **F6 SAVE ALL PROGRAMS** (you can't SAVE just a single PROGRAM). This will overwrite the current PROGRAM 16 USER with our modified PROGRAM. Since we're SAVING it as #16 it won't affect the old PROGRAM 26.

17. Make sure that your back-up copy of DISK KSD-002 is in the DISK DRIVE. Make sure that the prompt now reads **SAVE ALL PROGRAMS:D**
18. Press ENTER.

19. Press YES - Very quickly the PROGRAMS are SAVED to the disk.

20. Let's check the success of this operation. When SAVING is completed, Power OFF/ON.

21. Enter SYSTEM mode.

22. Select **F1 GET SYSTEM** (press 1 on the numpad).

23. Use the DATA ENTRY A to set the LCD to read **GET SYSTEM:D**

24. Press ENTER.

25. Press YES.

26. Enter PLAY mode (cancel SYSTEM mode).

27. Select Program 16, it should be whatever name you gave it

CONGRATULATIONS!

Chapter 14
PERMANENTLY MODIFYING A FACTORY PROGRAM

I've shown you how to modify a FACTORY PROGRAM and SAVE the modified version in one of the USER PROGRAM locations on your FACTORY DISK. Again, I'm using a Factory Disk from the first set of four that the DSS-1. If your first four Factory Disks are labelled KSDU-001R/002R/003R/004R, don't worry. The procedure is the same, except that your Disks don't have any USER PROGRAMS.

Now, what if you want to replace a FACTORY PROGRAM with your modified version ? No problem.

Let's customize a FACTORY PROGRAM on your back-up copy of DISK KSD-001 PIANO by making two seemingly minor changes in the PROGRAM PARAMETER mode. The result will be a great lead sound for fast solo runs. It will also be well suited for chordal playing. The PROGRAM uses the DSS-l's
built-in DIGITAL DELAYS. This makes for a great stereo effect. If you're running your DSS-1 in mono (from only the RIGHT output), try listening through stereo headphones to capture the stereo effect.

1. Insert your back-up copy of DISK KSD-001 PIANO.
2. Load SYSTEM A.
3. Enter PLAY mode (cancel SYSTEM mode).
4. Select **PROGRAM 31:SYNTH 3**.
5. Enter PROGRAM PARAMETER mode.
6. Play the keyboard - only one note sounds at a time.
7. Select **F63 KEY ASSIGN MODE** (press digits 6 and 3 on the numpad).
8. The LCD shows **KEYBOARD ASSIGN MODE = UNISON**.
9. Use DATA ENTRY B to change that to **POLY-1.**

10. Play the keyboard. The sound is thinner, and you can play chords. During long notes you hear a part of the sound change from high to low pitch. You also hear that the harder you play, the higher the changing part of the sound seems to start out before it falls down.

11. Select **F16 SYNC MODE, D/A RESOLUTION** (press 1 and 6 on the numpad. **SYNC** is shown as **ON**, which locks the pitches of the two OSCILLATORS together. When the second OSCILLATOR is undergoing the AUTOBEND that you'll see in the next steps, instead of a true pitchshift we hear a change in the harmonics.

12. With the CURSOR under **ON**, use the DATA ENTRY B to set it to **OFF**. Play the keyboard - now you can hear the second OSCILLATOR independently from the first. When you play hard, the changing part of the sound starts a major seventh higher than it ends up after falling down.

13. Select **F41 AUTO BEND INT** (press 4 and 1on the numpad).

14. The LCD shows **A.BEND VEL-SENS. INTENSITY = 25**. This causes a downward pitch-shift for every note you play - the harder you play, the more the pitch shifts.

15. To reduce this effect, use the DATA ENTRY B to change the value to 05.

16. Select **F19 AUTO BEND TIME & INT** (press 1 and 9 on the numpad). The INTENSITY of the AUTO BEND is shown as 52, with the TIME = 23.

17. With the CURSOR under 52, decrease this value to 03, and reduce the value for TIME to 16. Now only the notes that you strike very hard have a small amount of pitchshift over a very quick time.

Before writing this personalized version of PROGRAM 31 into the SYSTEM, you should re-name it, so that you won't confuse it with the original PROGRAM 31:SYNTH 3 that you still have on the original FACTORY DISK KSD-001. Just add a little symbol (or your initials) to the PROGRAM name.

18. Select **F01 WRITE/RENAME** (press 0 and 1 on the numpad).
19. The LCD asks **Rename ? (Y/N)** - press YES.
20. Use the CURSOR and DATA ENTRY B to add a symbol or your initial.
21. Press ENTER.

22. Now the LCD wants to know under which number this new version should be filed in the PROGRAM memory. Since we decided that we'll overwrite the old PROGRAM #31, we'll leave 31 showing as the target Press ENTER.

23. Now the question is whether to write this into the MEMORY. This doesn't write it onto the DISK yet - just into RAM. Press YES.

24. "Continue" means do you want to do the same with another PROGRAM. Press NO.

Now you want to SAVE this new PROGRAM on DISK, where it will replace the old PROGRAM #31.

25. Make sure your back-up copy of DISK KSD-001 PIANO is still in the DISK DRIVE.
26. Enter SYSTEM mode.
27. Select **F6 SAVE ALL PROGRAMS** (you can't SAVE just a single PROGRAM).

28. The LCD shows that the DSS-1 is ready to SAVE ALL PROGRAMS currently in RAM to SYSTEM A. That's just what you want – press ENTER.

29. Just to make certain that you know what you are doing, you get the old **Are You SURE ?** - press YES.

SAVING PROGRAMS doesn't take long - remember that the PROGRAMS don't contain the WAVEFORM DATA - they only point to the MULTI SOUNDS to be used within that SYSTEM. This takes up very little space on the DISK.

Want to check the success of this operation?

30. While still in SYSTEM mode, select **F1 GET SYSTEM**
31. The LCD shows **GET SYSTEM:A** - press ENTER.
32. Press YES and...wait.
33. When LOADING is completed, enter PLAY mode (cancel SYSTEM mode).
34. Select **PROGRAM 31** - it should now read **SYNTH 3** and whatever symbol or initials you added.
35. It should sound polyphonic, with just a trifle of quick pitchshift when you play hard.

Chapter 15

INSERT A MULTI SOUND FROM ONE SYSTEM INTO ANOTHER ON THE SAME DISK

Adding a MULTI SOUND to a SYSTEM works fine as long as:

1. the SYSTEM has less than the allowable maximum of 16 MULTI SOUNDS, and

2. the total LENGTH of all the MULTI SOUNDS in the SYSTEM (including the one you're about to insert) won't exceed 262,144 words (the WAVE RAM maximum). CHECK YOUR TABLES!

After including the new MULTI SOUND in the SYSTEM, you'll want to write a new PROGRAM to use the new MULTI SOUND, or you'll never hear it. This means you'll have to sacrifice an already existing PROGRAM. Look for a PROGRAM that you can spare, or plan on using a USER PROGRAM number if you have any left. Make a note of your choice.

Let's take it from the top.

1. Enter SYSTEM mode and load the SYSTEM to which you want to add a MULTI SOUND (SYSTEM mode **F1 GET SYSTEM**).

2. Select SYSTEM mode **F9 GET MULTI SOUND** and use the DATA ENTRY A to read through the listing of all the MULTI SOUNDS on the DISK

3. Pick a MULTI SOUND. The number in the LCD will be one higher than the total number of MULTI SOUNDS already in the SYSTEM. Example: If your SYSTEM currently has 8 MULTI SOUNDS and you're selecting a ninth MULTI SOUND, the new MULTI SOUND will automatically be number 9.

4. When you've made your selection, press ENTER and YES.

At this stage, the SYSTEM contains the new MULTI SOUND , but you can't hear it yet. To SAVE it,

5. Select **F2 SAVE SYSTEM**

6. The CURSOR should blink under the same SYSTEM letter (A or B or C or D) that you had loaded in the first place. If the letter is different, use the DATA ENTRY A to change it.

7. Press ENTER - the LCD asks **Are You SURE? (Y/N)**

8. Press YES.

Here's how to set up a simple PROGRAM so that you can listen to the MULTI SOUND.

9. Enter PLAY mode (cancel SYSTEM mode).

10. Select the PROGRAM that you've chosen to sacrifice for this MULTI SOUND (use either the DATA ENTRY B tabs or the numpad).

11. Enter PROGRAM PARAMETER mode.

12. Select **F00 INITIALIZE PARAMS**

13. Press ENTER. This sets all the PROGRAM PARAMETERS to the initialized values discussed in SECTION 6.

14. Use either the DATA ENTRY A or the numpad to select **F12 OSC 1 MULTI SOUND**

15. Use the DATA ENTRY B to assign the new MULTI SOUND to OSCILLATOR 1.

Now you can play the keyboard and hear a "plain vanilla" version of the new MULTI SOUND. To refine and personalize this PROGRAM, you'll need to do some work in the PROGRAM PARAMETER mode. SECTION 6 will explain each PROGRAM PARAMETER in detail.

Before you finish this project, you need to SAVE this initialized PROGRAM. While you're at it, you can rename it. While still in PROGRAM PARAMETER mode,

16. Select **F01 WRITE/RENAME**. If you want to rename the PROGRAM to match the MULTI SOUND,

17. Press YES. Otherwise, press NO.

18. If you're renaming it, use the CURSOR and the DATA ENTRY B to write a new name.

19. After writing a name, press ENTER. The LCD asks you to confirm or reset the PROGRAM number. This should be the number of the PROGRAM that you initially chose to sacrifice.

20. Press ENTER. Now the DSS-I asks you whether you want this PROGRAM to be retained in RAM.

21. Press YES. Now you're ready to SAVE the PROGRAM as part of the SYSTEM.

22. Enter SYSTEM mode.

23. Select either **F2 SAVE SYSTEM** or **F6 SAVE ALL PROGRAMS**. It makes no difference except that SAVING PROGRAMS is much quicker.

Often you can't just simply add a MULTI SOUND to a SYSTEM. There may be no space: the SYSTEM may already have 16 MULTT SOUNDS, or the LENGTH of the new MULTI SOUND will push the total LENGTH of the SYSTEM's MULTI SOUNDS over the 262,144 word limit

In this case, you have to throw out a MULTI SOUND to make space for the new MULTI SOUND. However, aside from the above considerations about whether or not there is space for the new MULTI SOUND, there is a much more serious limitation to consider

When you use SYSTEM mode **F8 ERASE MULTI SOUND** to remove a MULTI SOUND from a SYSTEM, the SYSTEM's MULTI SOUND DIRECTORY "closes rank". Instead of leaving an empty space in the SYSTEM's MULTI SOUND DIRECTORY where the erased MULTI SOUND used to be, the DSS-1 "pulls up" the next MULTI SOUND. Then, like a chain reaction, the MULTI SOUNDS "below" move up the DIRECTORY by one postion. The end result of all this is:

1. the DIRECTORY has one less MULTI SOUND in it, and
2. there are no gaps.

Example: If we erase our piano MULTI SOUND **A.PF**, the first MULTI SOUND in SYSTEM A, all the PROGRAMS that played variations of this MULTI SOUND will now play variations of **HDSAW0**, the (formerly) second MULTI SOUND in this SYSTEM. MULTI SOUND **HDSAW0** has been "pulled up" to occupy the position in the DIRECTORY that used to be held by **A.PF**

And to make matters worse, the PROGRAMS still have their old names.

Example: In SYSTEM A, PROGRAM #01 is called G-Piano1. This PROGRAM uses MULTI SOUND **A.PF** in both OSCILLATORS. If you erase **A.PF** and call up PROGRAM #01, you'll hear the MULTI SOUND **HDSAW0** in both OSCILLATORS - even though the PROGRAM is still called **G.Piano1**

The MULTI SOUND DIRECTORY closed rank when you erased the first MULTI SOUND, but the PROGRAMS still do what they did before: load MULTI SOUNDS into OSCILLATORS by the MULTI SOUNDS positions in the DIRECTORY, not by the MULTI SOUNDS names.

Let's test this out:

1. Insert the FACTORY DISK KSD-001 PIANO.
2. Select SYSTEM mode.
3. Select **F1 GET SYSTEM** (press1 on the numpad).
4. Select **GET SYSTEM A**
5. Press ENTER.
6. Press YES and ...wait.
7. Still in SYSTEM mode, select **F8 ERASE MULTI SOUND** (press 8 on the numpad).
8. Use DATA ENTRY A to bring up **ERASE MULTI SOUND 01:A.PF L=254790**
9. Press ENTER.
10. The LCD wants confirmation **Erase 01:A.PF L=254790 (Y/N)** press YES.
11. After a quick **This will take a while**, it says **F8 A.PF Erased - Continue ? Y/N**. Press NO.
12. Enter PLAY mode (cancel SYSTEM mode).
13. Play the keyboard - the PROGRAM reads **SYS A P01:G.Piano1** But it doesn't sound like a piano.
14. Enter PROGRAM PARAMETER mode.
15. Select **F12 OSC1 MULTI SOUND** (press the digits 1 and 2 on the numpad).
16. The LCD shows that OSCILLATOR 1 now plays the MULTI SOUND **HDSAW0**
17. Select **F13 OSC2 MULTI SOUND** (press the 1 and 3 on the numpad).

OSCILLATOR 2 plays the same waveform as OSCILLATOR 1, and neither is playing our sampled piano sound, even though the PROGRAM is still called **G.Piano1**

This is because **HDSAW0** used to be the second MULTI SOUND in this SYSTEM. We "erased" the first one - our piano **A.PF** - and all the MULTI SOUNDS moved "up" by one position.

So. is there a way to replace a MULTI SOUND in a SYSTEM? Yes, here's what you have to do:

1. Make an accurate list of all the MULTI SOUNDS in the SYSTEM. List them in the exact order in which they appear when you use SYSTEM mode **F7 M.SOUND DIR/FREE SPACE**. To read the MULTI SOUND names one-by-one, use the UP-ARROW tabs of DATA ENTRY A. Be careful to tap the tabs lightly so that you don't skip any names.

2. Cross out the name of the MULTI SOUND that you're about to "erase".

3. In its place, write the name of the new MULTI SOUND.

4. Now the paperwork starts in earnest. Make a table 3 columns wide and 32 lines long. Label the columns "NAME", "OSCl", and "OSC2". Number the lines from 1 to 32. You'll fill in the table by doing the following for each of the 32 PROGRAMS in the SYSTEM:

a) Enter PLAY mode.
b) Select the next PROGRAM (the first time you do this, select PROGRAM #01. The next time, #02, and so on, until you finish with PROGRAM #32).
c) On your table, copy down the PROGRAM name as you see it in the LCD. The number that you see after the letter "P" in the LCD is the line number where you should write the PROGRAM name.
d) Enter PROGRAM PARAMETER mode.
e) Select **F12 OSC1 M.SOUND** (press 1 and 2 on the numpad, or use DATA ENTRY A).
f) In the "OSC1" column of the current PROGRAM, write the MULTI SOUND name in the LCD.
g) Select **FI3 OSC2 M.SOUND** (press 1 and 3 on the numpad, or use DATA ENTRY A).
h) In the "OSC2" column of the current PROGRAM, write the MULTI SOUND name in the LCD.

This takes care of one PROGRAM. Repeat a) thru h) until your table of 32 PROGRAMS is complete.

Note: All the PROGRAMS that use the MULTI SOUNDS you're about to throw out will sound very different after you've rebuilt the SYSTEM. They'll now call up the new MULTI SOUND. Eventually you may have to re-program them. Section 6 deals with programming.

Now you have the necessary information to rebuild your SYSTEM when the time comes.

5. Enter SYSTEM mode.

6. Select **F8 ERASE MULTI SOUND** (press 8 on the numpad).

7. Tap once on the UP-ARROW of DATA ENTRY A. This displays the name of the MULTI SOUND that is currently at the "top" of the SYSTEM's MULTI SOUND DIRECTORY.

8. Press ENTER.

9. Press YES. For a brief moment the LCD displays **This will take a while...** When the MULTI SOUND has been erased, the LCD asks **Continue? (Y/N).** To "continue" means to erase more MULTISOUNDS. We're going to erase all the MULTI SOUNDS in the SYSTEM because we'll have to rebuild the SYSTEM from scratch as we put the new MULTI SOUND in.

10. Press YES.

11. Repeat from step 7 thru 10 until the LCD displays **NO MULTISOUNDS**. You've erased them all.

Now you need to refer to your fist of MULTI SOUNDS for the new SYSTEM. That's the list you made following steps 1 thru 3.

12. Select **F9 GET MULTI SOUND** (press 9 on the numpad). You're about to re-assemble the MULTI SOUND DIRECTORY for this SYSTEM, now including your new MULTI SOUND.

13. Press ENTER. This reads the names of all the MULTI SOUNDS on the DISK into RAM. From this list you'll now select the MULTI SOUNDS for your new SYSTEM.

14. Use DATA ENTRY A to look for and display the name of the first MULTI SOUND on your list.

15. Press ENTER.

16. Press YES in response to **Are You Sure? (Y/N)**

17. After loading has been completed, press YES to continue.

Repeat 14 - 17 until you've loaded, in order, all the MULTI SOUNDS on your list in the new SYSTEM.

Next, you need to go through every PROGRAM in the SYSTEM and assign MULTI SOUNDS to the OSCILLATORS as per your table. Follow step 4, points a) thru h). Instead of writing down the MULTI SOUND names, use the DATA ENTRY B to display the correct MULTI SOUNDS for each OSCILLATOR in each PROGRAM. Each time your table lists the "old" MULTI SOUND, call up the new one. Remember that the new MULTI SOUND will make the PROGRAM sound different. Each MULTI SOUND selection needs to be confirmed by pressing ENTER.

This new SYSTEM must now be SAVED on the DISK.

18. Enter SYSTEM mode.
19. Select **F2 SAVE SYTEM** (press 2 on the numpad).
20. Use DATA ENTRY A to select the correct SYSTEM (A or B or C or D).
21. Press ENTER.

There is another way to get rid of a MULTI SOUND: DISK UTILITY mode, **F6 DELETE MULTI SOUND.** This is a more drastic move than erasing a MULTI SOUND from a SYSTEM. It wipes out the MULTI SOUND on the DISK. This will affect all SYSTEMS which share the deleted MULTI SOUND. In other words, you'd have to rebuild every SYSTEM. If it is worth the trouble for you to do it, be sure to use back-up copies of your DISKS.

Chapter 16
EXTRACTING SOUNDS FROM MULTI SOUNDS

One of the most exciting features of the DSS-1 is the possibility of re-using SOUNDS and MULTI SOUNDS in many new ways. On the FACTORY DISKS, we've already changed certain PROGRAMS to create new versions of existing MULTI SOUNDS.

Even more rewarding is the creative use of SOUNDS from within existing MULTI SOUNDS. It is a complex procedure, but well worth the effort. No set-up that comes from a factory is ever going to meet all your needs. Here is another chance to customize your MULTI SOUNDS and PROGRAMS.

Since we haven't done any audio sampling yet, we'll use SOUNDS from within two different MULTI SOUNDS on two different FACTORY DISKS.

From FACTORY DISK KSD-002 we'll extract the lower part (the first three SOUNDS) of the MULTI SOUND **BRASS ENSEMBLE**. Then we'll extract the middle and upper range (SOUNDS 2, 3, and 4) of the MULTI SOUND **HARP** from FACTORY DISK KSD-001. We'll combine these SOUNDS into a new MULTI SOUND that will play the BRASS in the left half of the keyboard. The right hand will play the HARP.

First you need to isolate the SOUNDS from within the two MULTI SOUNDS on the FACTORY DISKS. Each SOUND will have to be SAVED individually. In doing so, you'll lose the KEYBOARD ASSIGNMENT that the SOUNDS had in their old MULTI SOUNDS. But that's okay - you'll re-assign them when you assemble the new MULTI SOUND.

1. Clear the RAM (Power OFF/ON).
2. Insert DISK KSD-002.
3. Enter SYSTEM mode.
4. Select **F9 GET MULTI SOUND** (press 9 on the numpad).
5. Press ENTER.
6. Use DATA ENTRY A to select **MULTI SOUND BRASS ENS.**
7. Press ENTER.
8. After display and **Are you sure Y/N** - press YES.
9. Press NO in answer to **Continue?** "Continue" means "keep going with what you're doing right now" - here it is "getting MULTI SOUNDS". But we only need one MULTI SOUND, and we've got it. Let's look at the SOUNDS within this MULTI SOUND.

10. Enter EDIT SAMPLE mode.
11. Select **F1 SELECT SAMPLE** (press 1 on the numpad).
12.The cursor is under **MEMORY** - press ENTER.
13. **BRASS ENS.** is displayed - press ENTER.
14. **SOUND #01** (the first SOUND from MULTI SOUND **BRASS ENS**) is displayed – press ENTER.
15. Display and confirmation is requested - press YES.
16. **ORG = C3 / TOP = F3** are shown. We know that this is immaterial at this point. But what matters is the original musical pitch at which this SOUND was recorded/sampled. Unless you have perfect pitch, you'll need to compare it with another instrument or with a chromatic tuner. SOUND #01 plays a low B flat when you play the current ORG KEY of C3. We'll need to remember this later. Press ENTER.

17. Select **F8 SAVE/RENAME SAMPLE** (press 8 on the numpad).
18. **Rename ?** Press YES.
19. Using the cursor and DATA ENTRY A, write the name **BR'HRP#1** (solely for housekeeping purposes - it is destined to become SOUND #01 in the MULTI SOUND Brass/Harp).
20. Press ENTER.
21. Insert your formatted disk.
22. Your name **BR'HRP#1** is on display - press YES. The new SOUND is now being SAVED to DISK.

Remember, the overall plan is to extract SOUNDS 1, 2, and 3 from within the MULTI SOUND **BRASS ENS**. We've got #01, so now you need to isolate #02.

We started by loading the whole MULTI SOUND **BRASS ENS.** into RAM (steps 4-9 above). As far as we know, it should still be in RAM ("Memory" as opposed to "Disk").

But when you now select **F1 SELECT SAMPLE** and try to call up SOUND #02 "from Memory", you find only a **MULTI SOUND 01!NO-NAME** listed. The length given in the readout is, however, the length of SOUND #01 (25120 words). The whole MULTI SOUND is 114466 words long. This immediately tells you that there's something wrong. And it's not just a matter of the DSS-1 having "lost" the name of the MULTI SOUND. Moving either DATA ENTRY doesn't change the LCD. "ENTER" is blinking - you have no choice.

After pressing ENTER, you can't select any other SOUNDS than the one shown in the LCD, no matter which DATA ENTRY you use. You have to press ENTER again, and you find that all you've done is re-select SOUND #01. Very confusing, and not very helpful. KORG's engineers did a strange thing here. You have to start from scratch to get SOUND #02.

Just in case you think that there is a quicker way to get the next SOUND from within the MULTI SOUND **BRASS ENSEMBLE** by going directly to the DISK with one of the GET SOUNDS functions: This won't work, because KORG didn't SAVE the individual SOUNDS as SOUNDS / SAMPLES / WAVEFORMS on the FACTORY DISKS. They're only on the DISK as a part of the MULTI SOUND.

So - let's start from scratch.

23. Repeat steps 1- 13 from above.
24. Repeat step 14, but this time select **SOUND #02**
25.Repeat steps 15 - 18, and note the pitch of this SOUND: B flat one octave up from SOUND #01.
26. Repeat step 19 but make the last character a "2".
27. Repeat steps 20 - 22 from above.

Now you need SOUND #03. Again - yoo're stuck in trying to access SOUND #03 directly. The computer won't let you do it, although we suspect that the whole MULTI SOUND is still in the wave RAM. You have to re-load the MULTI SOUND from scratch , then isolate SOUND #03. Here goes.

28. Repeat steps 1 - 13 from above.
29. Repeat step 14, but this time select **SOUND #03**
30. Repeat steps 15 - 18, and note the pitch of this SOUND: B flat one octave up from SOUND #02.
31. Repeat step 19 but make the last character a "3".
32. Repeat steps 20 - 22 from above.

You now have on the work disk three SOUNDS (crude samples) from the MULTI SOUND **BRASS ENS.** Next you need to extract three SOUNDS from within the MULTI SOUND **HARP** on DISK KSD-001.

33. Repeat steps 1 - 13 from above, but use DISK KSD-001 and select MULTI SOUND **HARP**
34. Repeat step 14, but start with **SOUND #02**
35. Repeat steps 15 - 18, and note the pitch of this SOUND: G above middle C.
36. Repeat step 19 but make the last character a "4".
37. Repeat steps 20 - 22 from above.
38. Repeat steps 1 - 13 from above, but again use DISK KSD-001 and select MULTI SOUND **HARP**
39. Repeat step 14, but select **SOUND #03**
40. Repeat steps 15 - 18, and note the pitch of this SOUND: G one octave and a fifth above middle C.
41. Repeat step 19 but make the last character a "5".
42. Repeat steps 20 - 22 from above.
43. Repeat steps 1 - 13 from above, but again use DISK KSD-001 and select MULTI SOUND **HARP**
44. Repeat step 14, but select **SOUND #04**
45. Repeat steps 15 - 18, and note the pitch of this (last) SOUND: G one octave up from SOUND #03.
46. Repeat step 19 but make the last character a "6".
47. Repeat steps 20 - 22 from above.

This completes the preparation for the next task: the actual building of a MULTI SOUND. All the SOUNDS needed for the planned keyboard split are on your work disk. Take a break.

SECTION 3
BUILD AND EDIT A MULTI SOUND

Chapter 17
PLANNING THE INCLUSION OF A MULTI SOUND INTO AN EXISTING SYSTEM

Next you'll combine the SOUNDS you accumulated in the previous Chapter into a new MULTI SOUND. We'll look at the volume balance and the details of the loops of these SOUNDS. And you'll SAVE the completed MULTI SOUND. At that stage, it will be ready to be called up by a PROGRAM for inclusion in a SYSTEM. The plan is to incorporate this MULTI SOUND, with its PROGRAM, into your back-up copy of DISK KSD-002, where there is plenty of room in SYSTEM D. Or is there ? Let's see.

Before we go on, we'd better do our homework.

First Question:
Can the system accommodate a New Multi Sound? Let's keep an eye on the limitations of the memory.

a) The limit in words (the total of all assigned MULTI SOUNDS) is 262,144 per SYSTEM.
b) The limit in MULTI SOUNDS assignable to a SYSTEM is 16.

In our example, a) is not really a problem: SYSTEM D contains no real audio-input samples, only very short created waveforms that add up to just 16,320 words. There is plenty of room left for a big new MULTI SOUND.

Are you surprised, because you saw PROGRAMS called "Bassoon" and "Trombone" and "Flute"?

Sorry - they're only simple synthetic waveforms. And they sound like it, too. Just try listening to the "Bassoon": if you've ever listened closely to a real bassoon you'll wonder why this PROGRAM doesn't show the rich timbre changes throughout the instrument's range that we are used to hearing from the real instrument. The length of the MULTI SOUND used to lake this "Bassoon" is just 1020 words, which really is one wavecyde spread across the whole keyboard.

In our example, b) is a real problem: SYSTEM D on DISK KSD-002 has 16 MULTI SOUNDS assigned. That's the allowable maximum. So 1 MULTI SOUND will have to be "erased" from SYSTEM D before we can bring in our newly constructed **BR'HRP** MULTI SOUND.

Second Question:
What are the consequences of replacing a Multi Sound in a System?

As we saw in Chapter 15, MULTI SOUNDS "close rank" whenever one gets "erased". But the PROGRAMS, in calling up MULTI SOUNDS, still point at the same old positions in the DIRECTORY, unaware that some MULTI SOUNDS have changed places. As an example, let's pick (at random) a MULTI SOUND and see what would happen if we erased it from SYSTEM D to make room for our planned MULTI SOUND **BR'HRP**.

Let's pick MULTI SOUND 11 **HD0-SWX2**. The following PROGRAMS would have to be re-worked, because they are playing MULTI SOUNDS that would change position as soon as #11 was eliminated (old #12 moves up to become new #11, old #13 moves up to become new #12 etc.) In other words, any PROGRAMS that use MULTI SOUNDS 12-16 (numbers higher than 11) will be affected by the "erasing" of MULTI SOUND #11.

SYS:D PROG.01Bassoon	was **HD0-SWX2** (both OSC), now **HD0-PW10** (both OSC)
SYS:D PROG.02Trombone	was **HDSAW1** and **HDSAW3**, now **HDSAW3** and **HDTRG0**
SYS:D PROG.03Symphony	was **HDSAW0** and **HDSAW3**, now **HDSAW0** and **HDTRG0**
SYS:D PROG.04 Flute	was **HDTRG0** and **ADCOMB0**, now **HDPWMT1** and **ADCOM0**
SYS:D PROG.065thSYN-1	was **HD0-SWX2** and **ADMeta1**, now **HD0-PW10** and **ADMeta1**
SYS:D PROG.075thSYN-2	was **HDPWMT1** (both OSC), now nothing (there's no 16th MULTI SOUND)
SYS:D PROG.085thSYN-3	was **AD80RHO2** and **HDPWMT1**, now **AD80RHO2** and nothing.

All the other PROGRAMS play MULTI SOUNDS that have lower positions in the SYSTEM than the one we're hypothetically "erasing" - #11 **HD0-SWX2**. So these other PROGRAMS would not be affected.

In Section 9, you can see these PROGRAMS and MULTI SOUNDS in the tables for DISK KSD-002.

CONCLUSION: To make a tricky task a little easier, "erase" a MULTI SOUND that's used by the least number of PROGRAMS, or by the least important PROGRAMS. Or erase the MULTI SOUND that has the "highest" position (biggest number) in the SYSTEM.

In our example, the highest position in the SYSTEM is occupied by MULTI SOUND #16 **HDPWMT1**. If you eliminate it, then that position just remains vacant until you include another MULTI SOUND (there is no number 17 that can move up to take the place of the discarded number 16).

ALSO: MULTI SOUND **HDPWMT1** isn't really that important as a MULTI SOUND. Only PROGRAMS 7 and 8 play it. They're synthesizer programs, tuned in fifths. Cute, but not essential. Surely you can find another MULTI SOUND already part of SYSTEM D that can achieve a similar effect.

To assign a new MULTI SOUND to these two PROGRAMS you would enter the PROGRAM PARAMETER mode and use **F12 OSC1 MULTI SOUND** and **F13 OSC2 MULTI SOUND** to select new MULTI SOUNDS for the oscillators. Then you would follow Chapter 14 to SAVE the new versions of PROGRAMS 7 and 8, thereby overwriting the old versions.

Third Question:
Is there room on the disk for a New Multi Sound?

Even if we successfully deal with the limitations from within a SYSTEM, we mustn't forget that the 4 SYSTEMS on a DISK can't add up to more than around 525,000 words. To get at this number, add up all the MULTI SOUNDS of the four SYSTEMS, counting each MULTI SOUND only once (remember, SYSTEMS can share MULTI SOUNDS). For this you look up your tables. Then you add the new MULTI SOUND, and if the total is over the limit, you'll have to give up or find a new solution.

To know how big your new MULTI SOUND is before you plan on SAVING it on an existing DISK: Add up the SOUNDS within the MULTI SOUND. Here's the addition necessary for your new MULTI SOUND:

BR'HRP#1 = 25120
BR'HRP#2 = 26143
BR'HRP#3 = 28512
BR'HRP#4 = 16000
BR'HRP#5 = 16000
BR'HRP#6 = 8000

NEW M.SND 119775 total

This little addition shows us why we should give up on the idea: *There's no way that we can integrate a MULTI SOUND of 119,775 WORDS on DISK KSD-002 which is almost full: 510,889 words (763 blocks used, only 10 blocks free).*

We have two options:

a) Eliminate another, substantial MULTI SOUND from the DISK to make room for our new MULTI SOUND. This will take a lot of homework, because a lot of PROGRAMS will become unusable. Not just the ones that formerly called up the eliminated MULTI SOUND, but also those PROGRAMS that now call up MULTI SOUNDS that "close rank" and fill the gap left by the erased MULTI SOUND.

b) We can build a new DISK with 4 SYSTEMS that will be different from DISK KSD-002. The new SYSTEM with the new MULTI SOUND in place of the old **HDPWMT1** could become SYSTEM A on a new disk. We could then add SYSTEMS that would stay within the restrictions of available MEMORY, and create a new disk. An adding machine and the reference tables (Section 9) would be needed to plan this out.

CONCLUSION: All things considered, let's give up an the idea of incorporating the new MULTI SOUND into SYSTEM D *on DISK KSD-002.* Instead, let's start a new DISK. First you'll incorporate the new MULTI SOUND in SYSTEM D *in RAM* where there is room. Then you'll SAVE this SYSTEM to a new DISK, where you can add 3 more SYSTEMS later on.

So let's build this new MULTI SOUND.

Chapter 18
WHAT'S INVOLVED IN BUILDING A MULTI SOUND

FIRST you bring into RAM the SOUNDS (samples / waveforms) that you want to assemble into a MULTI SOUND. These SOUNDS can be assembled from various disks - just load them in the order that they'll be played from left to right on the keyboard once the MULTI SOUND is completed.

As you load the SOUNDS one by one, the LCD will show you the default Keyboard Assignment for the ORIGINAL key of each SOUND. If you didn't change the TOP key when SAVING the SOUNDS in EDIT SAMPLE mode, then the first SOUND will be at Orig=C3/Top=C3. If you leave it there, then the next SOUND (SOUND #2) will automatically come up as Orig=C#3 and Top=either C#3, or a higher note whose interval distance corresponds to the interval that you'd previously set in EDIT SAMPLE mode.

BUT, if you re-assign the first SOUND, then the next SOUND (SOUND #2) will have its Orig.Key automatically 1 halfstep higher than the key that you assigned as TOP key for SOUND #1. And SOUND #3 will automatically have its Orig.Key 1 halfstep higher than the TOP key of SOUND #2.

This is how the DSS-1 builds MULTI SOUNDS from left to right. Each SOUND is limited by its neighbors to either side. Only the first SOUND has no limit set to the left: Downward transposition of the first SOUND is possible from its assigned Orig.Key all the way to the left end of the keyboard. The maximum distance between the Orig. and the Top Key for each SOUND is 1 octave.

Sometimes you may want to load all SOUNDS and leave them at their default assignments until the MULTI SOUND is complete. Then you can call up each SOUND from RAM and change its Orig/Top key assignment, plus any of the other parameters that need adjustment in MULTI SOUND mode.

Before you load the sounds, make a list of the desired keyboard assignments, based on the original pitches of the samples, or based on other requirements in the case of non-pitched sounds (percussion or effects etc.)

After you've mapped the keyboard, there is a lot more work to do.

The individual SOUNDS can be balanced against each other by adjusting their loudness levels, their brightness and their tunings.

For sampled SOUNDS, individual playback start points and loop start/end points can be set. You single out the SOUND you want to treat (one at a time) and you decide how you want to treat it using one or more of the FUNCTIONS in the MULTI SOUND mode.

The following FUNCTIONS in MULTI SOUND mode affect the individual SOUND currently selected by **F1 SELECT MULTI SOUND/SOUND**:

F2 RELATIVE PARAMETERS: **Tune / Lev / Fc** -- **Tune** is self explanatory. **Lev** stands for LEVEL, which is the relative LOUDNESS level of one SOUND compared to the other SOUNDS in this MULTI SOUND. **Fc** stands for FILTER CUTOFF - it controls the brightness, similarly to a treble control. We'll cover **Fc** and FILTERS in detail in Section 6.

F3 ORIGINAL/TOP KEY. If you didn't re-assign each SOUND when you first loaded it, then this Function will give you another chance to do your Keyboard Assignment at any later stage.

F4 SOUND START & LENGTH. Each SOUND has a certain length. If you need less than the total length for playback within the MULTI SOUND, then you instruct the DSS-1 to "read" only a portion of the whole SOUND. This does NOT refund any "unused" memory - only the TRUNCATE function under EDIT SAMPLE mode can "recover" memory. This FUNCTION only means something for sampled SOUNDS. It has no audible effect on created waveforms.

F6 LOOP START & LENGTH. Here you choose which portion of a SOUND should be repeatedly "read" by the DSS-1 during playback. This is where you often need lots of patience, luck and keen cars. But the DSS-1 is very helpful about loops: **F7 LOOP PROCESS** and EDIT SAMPLE mode **F7 VIEW/EDIT SAMPLE DATA** can make a big difference. More about this in Section 3, Chapter 23. (Single-cycle created waveforms are not audibly affected).

F7 LOOP PROCESS (XFADE/B&F). This provides help in finding smooth loops. **XFADE** stands for CROSS-FADE, **B&F** stands for back-and-forth. More about this in Chapter 23. (Single-cycle created waveforms are not audibly affected).

MULTI SOUND mode FUNCTIONS **F5 LOOP ON/OFF** and **F9 SAVE/RENAME** affect all SOUNDS in the MULTI SOUND equally.

MULTI SOUND **F8 REPLACE SOUND** lets you load a SOUND into the already established slot of the SOUND number currently selected under F1. The new SOUND will take on the KEYBOARD ASSIGNMENT of the SOUND that it replaces.

When you have placed and treated all the individual samples in the MULTI SOUND, it'll be time to give this new MULTI SOUND an identity. You'll NAME and SAVE this MULTI SOUND. Store your MULTI SOUNDS on a separate DISK. When you're ready to incorporate a MULTI SOUND from your storage library into a SYSTEM, just load it into RAM, write one or several PROGRAM(s) and SAVE the RAM contents to a disk, using SYSTEM mode **F2 SAVE SYSTEM**.

An important point: From here on, you can't make changes to individual SOUNDS within this MULTI SOUND. Once you have assembled the MULTI SOUND, the future processing in the PROGRAM PARAMETER mode will affect the whole MULTI SOUND. Furthermore, the PROGRAM PARAMETERS affect the MULTI SOUNDS in both OSCILLATORS (except for a few FUNCTIONS where one or the other OSCILLATOR gets singled out).

Chapter 19
ASSEMBLING THE SOUNDS FOR A NEW MULTI SOUND

For practice's sake, well load all the SOUNDS with their default keyboard assignments into RAM. Then well go back and assign each to its proper Orig/Top keys. There is another way of doing this: You could assign the Keyboard Range to each SOUND as you load it. But do it my way first – it gives me a chance to show you the whole procedure.

1. Clear the RAM (Power OFF/ON)
2. Enter the MULTI SOUND mode.
3. Insert your DISK with the 6 SOUNDS **BR'HRP#1-6**
4. Select **F0 GET SOUNDS** (press 0 on the numpad).
5. Press ENTER. The LCD displays **LOOP ON? (Y/N).** What you select will affect all the SOUNDS. At this point, this is not a final decision. Since you'll need to test each SOUND for "loopability".

6. Press YES.
7. Use DATA ENTRY A to select the SOUND to be assigned to the far left on the keyboard: **BR'HRP#1**
8. Press ENTER.
9. Check the LCD and confirm your selection - press ENTER.
10. Check the LCD and, again, confirm your selection - press YES.
11. After loading the SOUND, the LCD shows the default **KEY ASSIGNMENT ORG =C3/TOP=C3**. We'll change it later, let's first get all 6 SOUNDS - press ENTER.

12. The LCD confirms that the first SOUND is **BR'HRP#1**. Press (**Satisfied**) YES
13. The LCD asks **More SOUNDS** - press YES. Because you may collect your SOUNDS from different DISKS, the LCD asks to insert a DISK. You have all our SOUNDS on the same DISK, which is already in the DRIVE.

14. Press ENTER. Use the DATA ENTRY A to select the next SOUND - it's automatically numbered SOUND#02 (because you already have a SOUND#l). You must select the SOUNDS in the order that you want them to end up on the keyboard from left to right.

15. Select **BR'HRP#2**
16. Press ENTER.
17. Check the LCD and confirm your selection - press ENTER. Confirm your selection again:
18. Press YES. After loading, the LCD shows the default KEYBOARD assignment **ORG = C#3/TOP=C#3** (= the next-highest from SOUND # 01 which was placed at C3/C3). You'll change it later, first, get the other SOUNDS up.

19. Press ENTER. The LCD displays the name of SOUND 02 and asks **Satisfied** - check for **BR'HRP#2**
20. Press YES.
21. The LCD asks **More SOUNDS** - press YES.
22. Your DISK is already in the DRIVE - press ENTER.
23. Use the DATA ENTRY A to select, as the 3rd SOUND, your **BR'HRP#3** (it automatically becomes SOUND #3, because you have two SOUNDS up so far).

24. Press ENTER.
25. Check the LCD and confirm your selection - press ENTER.

26. Check the LCD and, again, confirm your selection - press YES.
27. SOUND#03 comes up at the next available KEYBOARD assignment of D3/D3- you'll change it later - press ENTER.

28. The LCD displays **Satisfied** - press YES.
29. The LCD displays **More SOUNDS** - press YES.
30. Your DISK is already in the DRIVE - press ENTER.
31. Use the DATA ENTRY A to select, for your 4th SOUND, the SOUND **BR'HRP#4** (it automatically becomes SOUND #4, because you have three SOUNDS up so far).

32. Press ENTER.
33. Check the LCD and confirm your selection - press ENTER.
34. Check the LCD and, again, confirm your selection - press YES. The KEYBOARD assignment is the next-highest available: D#3/D#3. You'll change it later.

35. Press ENTER.
36. The LCD asks **Satisfied** - press YES.
37. The LCD asks **More SOUNDS** - press YES.
38. Your DISK is already in the DRIVE - press ENTER.
39. Use the DATA ENTRY A to select the fifth SOUND **BR'HRP#5**, for which the fifth slot is allocated.
40. Press ENTER.
41. Check the LCD and confirm your selection - press ENTER.
42 Check the LCD and, again, confirm your selection - press YES. The KEYBOARD assignment is again the next-highest available: E3/E3. You'll change it later, first get the last SOUND.

43. Press ENTER.
44. The LCD displays **Satisfied** - press YES.
45. Now the LCD asks **More SOUNDS** - press YES.
46. The next SOUND also happens to be on the DISK that's already in the DRIVE - press ENTER.
47. Use the DATA ENTRY A to select the final SOUND **BR'HRP#6**, for which slot six is allocated.
48. Press ENTER.
49. Check the LCD and confirm your selection - press ENTER.
50. Check the LCD and, again, confirm your selection - press YES.
51. The KEYBOARD assignment is F3/F3, which you'll change in a minute - press ENTER.
52. The LCD displays **Satisfied** - press YES.
53. The LCD asks **More SOUNDS** - press NO.

Chapter 20
ASSIGNING KEYBOARD RANGES TO SOUNDS IN A MULTI SOUND - ORG/TOP KEYS

You have just completed the assembly of a MULTI SOUND. You did this in the order in which you want to play the SOUNDS from left to right on the keyboard. But you haven't assigned the right ranges to the individual SOUNDS yet. Nor have you done any of the other adjustments to each SOUND that MULTI SOUND mode allows you to do.

This brings us back to the list of FUNCTIONS in the MULTI SOUND mode. You'll need to treat, one by one, each of the SOUNDS in the MULTI SOUND until your "Team" is complete, balanced and ready to be SAVED as a unit. First you want to allocate each of the SOUNDS to their correct KEYBOARD assignment. Start with #1 and "SELECT" it.

1. Select **F1 SELECT MULTI SOUND / SOUND** (press 1 on the numpad).

2. The LCD shows that you're ready to work on SOUND 01. (It calls the MULTI SOUND also #01, simply because you haven't called it anything else yet. Under **F9 Save/Rename** you'll decide on the name and number of this MULTI SOUND that you're building). SOUND 01 is indeed the one you want. If it wasn't, then you would have to change it by moving the CURSOR under SOUND and by using the DATA ENTRY A. As it's already the right SOUND, move on.

3. Select **F3 ORIGINAL/TOP KEY** (press 3 on the numpad).
4. For SOUND 01, set **ORG=A#1** and **TOP=F#2** (use the CURSOR and the DATA ENTRY A).
5. Select **F1 SELECT MULTI SOUND / SOUND** (press 1 on the numpad).
6. With the CURSOR under the value for SOUND, change it to **SOUND 02**.
7. Select **F3 ORIGINAL/TOP KEY** (press 3 on the numpad).
8. For SOUND 02, set **ORG=A#2** and **TOP=F#3**.

BUT THE TOP NOTE WON'T GO HIGHER THAN C#3. TIME OUT!

Why can't you assign SOUND 02 to a TOP key higher than C#3? You had no trouble in reassigning SOUND 01 because you didn't try to move it any higher than it already was. Instead, you moved it down from its default, into a range where there was no other SOUND as per default assignment.

But now, while trying to move SOUND 02 to a higher TOP key, you're running up against SOUND 03 which sits on D3. And SOUND 03 can't go up either, because SOUND 04 sits on D#3. And SOUND 04 can't go any higher either, because it's limited by SOUND 05 which sits on E3. Finally, when trying to move SOUND 05 up, you're blocked by SOUND 06 which sits on F3.

WHAT TO DO ?

You have to move the highest SOUND upwards, out of the way. Then the second-highest currently assigned SOUND has to move up, making room for the third-highest SOUND etc.

THE MORAL OF THIS STORY:

Key ORG/TOP Assignment can be done at two different times:

a) When the MULTI SOUND is being built, or
b) Anytime after the MULTI SOUND has been built.

While you're building a MULTI SOUND, each SOUND arrives with a "default" ORG/TOP Key Assignment. As you're building the MULTI SOUND from left to right (see Chapter 19), you can adjust the 'range' of each SOUND on the keyboard by changing its TOP key.

After building a MULTI SOUND, you can still change the "range" of one or more SOUNDS. But this time you need to work from right to left to avoid the kind of problems demonstrated above. So, let's work on our MULTI SOUND from right to left.

Working on the Multi Sound From Right to Left

9. Select **F1 SELECT MULTI SOUND / SOUND** (press 1 on the numpad).
10. With the CURSOR under the number of the SOUND, change it to **SOUND 06**.
11 Select **F3 ORIGINAL / TOP KEY** (press 3 on the numpad).
12. For SOUND 06, set the **ORG=G6 / TOP=C7**
13. Select **F1 SELECT MULTI SOUND / SOUND** (press 1 on the numpad).
14. With the CURSOR under the number of the SOUND, change it to **SOUND 05**.
15. Select **F3 ORIGINAL / TOP KEY** (press 3on the numpad).
16. For SOUND 05, set the **ORG=G5** / **TOP=C6**
17. Select **F1 SELECT MULTI SOUND / SOUND** (press 1 on the numpad).
18. With the CURSOR under the number of the SOUND, change it to **SOUND 04**.
19. Select **F3 ORIGINAL / TOP KEY** (press 3 on the numpad).
20. For SOUND 04, set the **ORG=G4 / TOP=C5**.
21. Select **F1 SELECT MULTI SOUND / SOUND** (press 1 on the numpad).
22. With the CURSOR under the number of the SOUND, change it to **SOUND 03**.
23. Select **F3 ORIGINAL / TOP KEY** (press 3 on the numpad).
24. For SOUND 03, set the **ORG=A#3 / TOP=F#4**.
25. Select **F1 SELECT MULTI SOUND / SOUND** (press 1 on the numpad).
26. With the CURSOR under the number of the SOUND, change it to **SOUND 02**.
27. Select **F3 ORIGINAL / TOP KEY** (press 3 on the numpad). Now we're back to our friend, SOUND #02. When you first tried to re-assign it, it wouldn't move away from it's TOP key of C#3.
28. Now set it to F#3 - no problem. The ORG key should still be at A#2.

SOUND #01 is still at **ORG=A#l / TOP=F#2**. This means that the G2, G#2 and A2 are sounding a downward transposition of SOUND #02. So the "range" of SOUND 02 is not just the keys from A#2-F#3, but rather from G2-F#3, because it extends downward from its ORG Key (A#2) until it hits the TOP Key of SOUND 01 (F#2).

Play the keyboard.

You'll hear the 'loops' that you haven't yet adjusted individually. Right now, because you answered YES to the earlier prompt **Loop ON Y/N**, each SOUND loops around the full length of its block of memory.

Those SOUNDS that have silence recorded before or after the actual sound portion are the most noticeable. This tells you that you could take these SOUNDS back to the EDIT SAMPLE mode, where you could shorten them (truncate) so that they wouldn't take up more space than necessary in a MULTI SOUND. You'll do this in the SAMPLING SECTION, Chapter 29.

In your BRS-HARP MULTI SOUND, you'll want the brassy SOUNDS to loop for long, sustained notes. But the harp SOUNDS should be as natural as possible, fading out whether you release the keys or not. So you'll have to find a silent portion of their SOUNDS to loop around.

If you want to take a break at this point, SAVE what you've accomplished so far. You'll be able to recall this (unfinished) MULTI SOUND later, and continue to refine it. For now, SAVE it as described in steps 1-5 of the next Chapter.

Chapter 21
NAME - SAVE - RELOAD THE NEW MULTI SOUND

1. Select **F9 SAVE/RENAME**.
The LCD shows your (unfinished) MULTI SOUND as **!NO-NAME**. You can fix that. **Rename ?**
2. Press YES.
3. Using the DATA ENTRY A and the YES/NO tabs (cursor), write the name **BRS-HARP**.
4. Press ENTER. Insert one of your (formatted) DISKS with at least 250,000 bytes (250 BLOCKS) of free space. Check LCD and confirm the name of your MULTI SOUND.
5. Press YES. This SAVES your MULTI SOUND to DISK.

From here on you'll be using your ears to make fine judgements while using the MULTI SOUND mode functions. Check your amplification equipment for flat EQ, because you don't want your tone controls to give you a false impression. When you're ready, get the MULTI SOUND from your work disk.

6. Insert your work disk with the MULTI SOUND **BRS-HARP** into the DISK drive.
7. Press the SYSTEM mode tab and **F9 GET MULTI SOUND**.
8. Press ENTER.
9. Use the DATA ENTRY A to select the MULTI SOUND **BRS-HARP**.
10. Press ENTER. The LCD shows **BRS-HARP** with a word length of 119,775. **Are You Sure ? (Y/N)**
11. Press YES.
12. Press NO in answer to **Continue? (Y/N)** since you don't need any more MULTI SOUNDS.
13. Enter the MULTI SOUND mode.
14. Select **F1 SELECT M.SOUND/SOUND**. The LCD displays **MULTI SOUND #01** and **SOUND #01**, with no names shown. Play the keyboard from the bottom up while watching the SOUND number in the LCD. As soon as you cross over into a new KEYBOARD ASSIGNMENT range, the SOUND number changes.

WHAT'S NEXT? Let's adjust "SOUND START & LENGTH" and "LOOPS".

Chapter 22
SOUND START & LENGTH IN A MULTI SOUND

To work on the START & LENGTH of a SOUND, you need to be in MULTI SOUND mode.

1. Select MULTI SOUND mode **F1 SELECT M.SND/SND**.
2. Select SOUND #01 in one of two ways: either play a key below F#2, or use the DATA ENTRY A.
3. Select **F5 LOOP ON/OFF** (press 5 on the numpad).
4. Use the DOWN ARROW of the DATA ENTRY A to turn the **LOOP OFF**. You'll turn it back ON later.

First, you need to check out a few things.

5. Play keys between C2 and F#2 - you hear a dirty "double-tongued" attack at the beginning of the SOUND. The other two brass SOUNDS don't have that, so you'll have to eliminate it from SOUND #01 for reasons of consistency throughout the 2 1/2 octaves of brass that you've built.

6. Select **F4 SOUND START & LENGTH** (press 4 on the numpad).

At this stage, the LCD doesn't show the number of the SOUND that you're working on. If you're not sure - select **F1 SELECT M.SOUND/SOUND** (press 1 on the numpad), check the number after SOUND: (currently #01) and press 4 on the numpad to get back to **F4 SOUND START & LENGTH**.

The LCD shows the current SOUND START (S.S) at 000000, which is the location of the very first computer reading. The SOUND LENGTH (S.L) is shown as 25,120. This means the SOUND is 25,120 words long - the same number as was displayed when you first brought this SOUND up from the DISK. This is what you would expect, since you haven't yet used **F4 SOUND START & LENGTH** to change the START or LENGTH of the SOUND.

You can instruct the DSS-1 to ignore a specified portion of the SOUND, either at the front, or at the end, or in both places. This is not the same as selecting a loop, which you have yet to do.

NOTE: **F4 SOUND START & LENGTH** doesn't return the unwanted portion(s) of the SOUND to the general "pool" of available RAM. Only EDIT SAMPLE mode **F3 TRUNCATE START/LENGTH** recovers memory by truly shortening a SAMPLE/SOUND.

For **F4 SOUND START & LENGTH**, both sets of DATA ENTRY sliders and tabs combine in this way.

Slider A steps through S.S and S.L numbers by 10,000s (10,000 / 20,000 / 30,000 etc.).
Arrow tabs A step by thousands (1,000 / 2,000 / 3,000 etc.).

Slider B steps by tens (this also takes care of the hundreds).
Arrow tabs B step by ones.

7. Play lots of short notes between C2 and F#2 while you experiment with these adjustments. You'll find that every time you change the START, the LENGTH automatically adjusts itself.

The computer subtracts from S.L the number of readings that you set in S.S. The SOUND LENGTH (S.L) is now shorter, because you've told the computer to ignore a specified number of readings at the beginning of the SOUND. When you play the keyboard, the computer starts reading from the newly-defined SOUND START.

8. When you're done with experimenting, set the START (S.S) to 300, which results in a LENGTH (S.L) of 24,820. (25,120 minus 300 = 24,820).

What you have now is SOUND #01 with a cleaner attack. You've told the computer to ignore the first 300 readings where the dirty attack used to be. How about cleaning up the end of this SOUND, where you hear a lot of nothing (noisy silence) ?

9. Bring the end in by reducing the current LENGTH from 24,820 to 8,000.

Now the SOUND is much shorter - it cuts off more abruptly, and even at high listening levels there is absolute silence beyond the new end-point. On playback, the computer is now starting the SOUND from the 301st reading. From there, it reads the next 8,000 values and stops.

Chapter 23
LOOPING SOUNDS IN A MULTI SOUND

WHAT ABOUT THE LOOPS? First of all, if you want the DSS-1 to LOOP, the LOOP most be turned on. MULTI SOUND mode is the only mode where that is possible, under two functions:

a) **F0 GET SOUNDS** - the LCD asks **LOOP ON Y/N** before actually getting the first of however many SOUNDS you're going to get

b) **F5 LOOP ON/OFF** - with the DATA ENTRY A you can always change the status of the LOOP.

NOTE: **F5 LOOP ON/OFF** affects all the SOUNDS in a MULTI SOUNDS. They either all loop, or none of them loop. But there are ways to get some SOUNDS to sustain, while other cut off - all in the same MULTI SOUND. We're not there yet.

MORE ABOUT THE LOOP

When you first load SOUNDS into a MULTI SOUND, the DSS-1 automatically sets the LOOP to repeat over the entire length of every SOUND.

If you want to LOOP on only a portion of the SOUND, you have to adjust LOOP START (L.S) and LOOP LENGTH (L.L), just like you did with SOUND START & LENGTH. You do this with **F6 LOOP START & LENGTH**.

Back to your SOUND #01, and I'll show you a quirk of the DSS-1. You have just shortened your SOUND drastically. If you now turn the LOOP **ON**, what do you expect to hear? (Remember, the LOOP is still set to repeat over the entire LENGTH of the original SOUND).

How about a short sound, a "glitch", and endless series of repeats of the entire original SOUND with the dirty silence between loops? The short sound would be your shortened SOUND (8,000 readings worth). The "glitch" would give away the jump that the computer has to make when it reaches the end of the short sound and begins reading from the very beginning of the original SOUND. The endless repeats with "dirty silence" would be the loops around the entire original SOUND. So, let's turn on the LOOP. Make sure you're in the MULTI SOUND mode.

1. Select **F5 LOOP ON/OFF** (press 5 on the numpad).
2. Use the DATA ENTRY A to change OFF to **ON**.
3. Play and sustain notes between C2 and F#2. Listen - no short SOUND, no "glitch". Instead you hear a clean attack followed by lots of repeats that each start with the dirty attack and end with the dirty silence. What's going on ?

Because of a quirk in the DSS-1, the end of the LOOP, as defined in **F6 LOOP START & LENGTH**, overrides whatever end point you set in **F4 SOUND START & LENGTH**. Strange but true.

This quirk causes the computer to ignore the end of the "short" 8,000 SOUND LENGTH (S.L) that you have just programmed. It does recognize the SOUND START (S.S) of 300, but it doesn't recognize the SOUND LENGTH (S.L) of 8,000.

The computer starts reading at sample 300 and reads all the way through the SOUND to the last sample (#25,120), ignoring the SOUND "end" which you set at sample #8300.

Because the LOOP is set to start on sample #00,000 and end on sample #25,120 (the entire LENGTH of the original SOUND), the computer then begins looping over the entire SOUND. This is why you hear repeated dirty attacks and dirty silences.

In this example the quirk doesn't really hang you up too much. If you want to eliminate the dirty attack or the dirty silence, just adjust the LOOP START & LENGTH.

But suppose you want to hear a late part of the SOUND before a LOOP starts, but you don't want this late part of your SOUND to be in the LOOP?

In this case, the end of the SOUND, as defined with **F4 SOUND START & LENGTH**, would have to be set to a larger number than the end of the LOOP. (Remember that on the DSS-1 you don't actually set the endpoints as numbers. The endpoints are the result of adding the START and the LENGTH numbers together). You can set the values, but you'll never hear the "late" part of your SOUND, not even on the first play before looping begins.

CONCLUSION: The LOOP end (the result of LOOP START plus LOOP LENGTH) always overrides the SOUND end (the result of SOUND START and SOUND LENGTH).

Rule of thumb: If you plan to have the LOOP **ON**, don't waste your time setting a SOUND LENGTH.

Getting Back to the Sound

Now let's get back to your SOUND and find a musically appropriate LOOP.

4. Select **F6 LOOP START & LENGTH**.

For **F6 LOOP START & LENGTH**, both sets of DATA ENTRY sliders and tabs combine in this way:

Slider A steps through L.S and L.L numbers by 10,000s (10,000 / 20,000 / 30,000 etc.).
Arrow tabs A step by thousands (1,000 / 2,000 / 3,000 etc.).

Slider B steps by tens (this also takes care of the hundreds).
Arrow tabs B step by ones.

The LCD shows the LOOP START (L.S) at 000000 and the LOOP LENGTH (L.L) at 025120 (the original LENGTH of the entire SOUND). As you know, the LOOP always defaults to the entire LENGTH whenever a SOUND is integrated into a MULTI SOUND with **F0 GET SOUND**.

The next logical move would be to adjust the LOOP START to a reading well beyond the attack part of the SOUND. That attack part needed cleaning up with SOUND START = 300, so we don't want to hear it during looping either. But here comes another quirk of the DSS-1:

5. The CURSOR blinks under L.S. Try changing the current value of 000000 to any other value (use any of the DATA ENTRY tabs and sliders). Can't do it ? The only thing that changes in the LCD is the upper line, which now reads **Press ENT to Auto** (don't press ENTER just yet - we'll get to it in a minute).

Notice that you can't adjust the LOOP START at this point, even though the CURSOR is under L.S. You must first reduce the value for LOOP LENGTH (don't forget to put the CURSOR under L.L)

There is no valid reason for this - it's just another quirk in the design of the DSS-1. Don't let it worry you, just remember it.

WHERE SHOULD YOU SET THE LOOP START & LENGTH ?

Some sounds are easier to loop than others. And some are downright impossible. Looping original audio samples is always a test for both your ears and your patience.

The basic idea is to find two portions of near-identical waveform characteristics. You'll use these two portions as the start and end of your loop. The closer they resemble each other, the smoother the loop.

Imagine joining two pieces of 2x4 end to end. To get the smoothest possible seam, both pieces must be exactly 2x4 with perfectly flat ends. If one is 2x3.5 or has a jagged end, the seam will be rough. When you loop a SAMPLE, you're trying to make perfect seams. This is why the waveform portions at the start and end of the loop need to be as similar as can be found.

With most kinds of samples, you're more likely to find two such portions in an area well into the sample. Attack characteristics generally don't loop well because a lot of quick changes occur at the beginning of most sounds. These changes are necessary for us to recognize the typical quality of a sound or instrument, but we don't want to hear them again during long notes. So look for portions of the sample where the sound has "settled" down from the attack characteristics.

Short segments of a sample often work best for avoiding glitches (looping clicks). If you pick a L.S and L.L value that are closely together, you stand a better chance of finding a reasonably glitch-free LOOP.

However, short glitch-free LOOPS don't always sound like the "real thing". This is because most kinds of sounds, even from instruments that can sustain long notes, change considerably over time. Just try saying a long "aah" without wavering... So, for LOOPS that sound like a real instrument playing sustained notes, try longer LOOPS.

AUTOMATIC SEARCH FOR ZERO-CROSSINGS

Once you've found two seemingly matching portions, mostly by ear, you can then ask the computer to find the exact readings that represent the "zero-crossings" of the waveform.

Since a waveform continuously goes up and down (positive and negative, peaks and valleys), the best seams (looping points) will sometimes occur at the point where the waveform crosses from negative over into positive. That point is called the "zero-crossing".

While you're in **F6 LOOP START & LENGTH**, the DSS-1 asks you to **Press ENT to Auto**. It does that every time you've changed either the L.S or the L.L value. If you press ENTER, the DSS-1 takes a few seconds to search for and memorize all the zero-crossings in the vicinity of both the current L.S and L.L.

If the DSS-1 can't find any zero-crossings, the LCD shows **Not found**. After you move to another area of the SOUND by changing one or both of the L.S and L.L values, you can press ENTER again to initiate another search for zero-crossings.

If the DSS-1 finds any, the LCD displays **Use CURSOR < >.** Every tap on cither the forward (YES) or backward (NO) tabs wil1 change the current values in the LCD. These values are the SAMPLE numbers of the zero-crossings in the vicinity of the current L.S and L.L

Keep playing the keyboard in the range assigned to the SOUND you're currently working on, and judge by ear the success or failure of the new L.S or L.L.

When the DSS-1 runs out of memorized cross-over points, the LCD shows **CANCEL to Manual**. As you press CANCEL, the upper line of the LCD returns to **LOOP START/LENGTH**. Move to another area of the SOUND by changing one or both of the L.S and L.L values. Again, **Press ENT to Auto** appears in the LCD, and your search for the perfect LOOP continues.

If you set L.S and L.L very close together, you'll hear the pitch and color of your SOUND change drastically. This happens when the distance between L.S and L.L is so short that you're asking the DSS-1 to loop around less than one complete wavecycle. If you want to explore this, you must ignore the **Press ENT to Auto** prompt, because you purposely want to hear the effect from a LOOP that doesn't match up two zero-crossings.

CROSS-FADE LOOP (MULTI SOUND mode F7)

Another kind of help is available for SOUNDS that are hard to loop. Anytime you have a serious mismatch between the points that you would like to select for L.S and L.L you can ask for a CROSS-FADE. We'll do this in a minute. Let me explain the concept of CROSS-FADE:

You select a L.S well into the SOUND, away from the attack portion. Then you select a L.L at a point further into the SOUND. Before the DSS-1 executes the CROSS-FADE, it asks for an amount of readings (FADE LENGTH). You set an amount that the DSS-1 will read in front of the L.S. The DSS-1 then jumps back into the LOOP and attenuates the amplitude of the SOUND for the same number of readings in front of the L.L point.

Then, starting at the jump-back point, it combines the amplitude of the portion of the SOUND that it read from in front of the LOOP with the portion of the LOOP from the jump-back point to the end of the LOOP. The idea is to change the end of the LOOP (by adding in amplitudes read from in front of the LOOP START) to make it look like the beginning of the LOOP. If you're trying to loop a SOUND that fades very rapidly in the original recording, then even CROSS-FADE may not achieve a smooth LOOP. When the levels of amplitude are too different at the L.S and L.L points, then you'll always hear a glitch during long notes.

BACK-AND-FORTH LOOP (MULTI SOUND mode F7)

Normally, the computer reads the LOOP from beginning (L.S) to end (L.L) and jumps back to the beginning (L.S), from where it continues reading. In other words, it reads the data only in one direction: a-b-c-a-b-c-a-b-c etc.

BACK-AND-FORTH (B&F) is a bi-directional way of reading the data within the LOOP; a-b-c-c-b-a-a-b-c-c-b-a etc This can make a big difference in the outcome of a LOOP, because it avoids the "jump" from L.L back to L.S.

BACK TO YOUR SOUND

Your brass ensemble shouldn't be too difficult to loop, because the color and the volume don't seem to change much over the duration of the original recording. This should allow you to pick a section from within the SOUND that the computer can repeat over and over without too much of a glitching noise at the "seams". Let's find out.

Beginning with L.L, experiment with different L.L and L.S values. Check all results by ear - remember that this SOUND plays from C2 to F#2. Re-read the above explanations carefully and explore the **Press ENT to Auto** and **Use CURSOR < >** features.

You'll find that the total of L.S and L.L can't exceed the original LENGTH of the SOUND - naturally.

Select **F7 LOOP PROCESS** and experiment with CROSS-FADE.

When you set the **FADE LENGTH**, make sure that your L.S is a fairly high number, because the FADE LENGTH signifies the amount of readings from in front of the L.S in your SOUND. The maximum FADE LENGTH is the number of readings between S.S and L.S. If your L.S is set too early in the SOUND then you're limiting the amount of readings available for CROSS-FADE.

In your **BR'HRP** SOUND 01, you've already heard that the first 300 readings contain a dirty attack that you've eliminated by setting the S.S to 300. To give yourself enough readings for a successful CROSS-FADE, you need to set the L.S at a number considerably larger than 300, say 2,500 or higher.

RED ALERT: If you plan to CROSS-FADE, don't set your L,S to be in front of (a smaller number than) your S.S. The DSS-1 gets very confused, gives you a very large number as your maximum FADE LENGTH, and basically trashes your SOUND.

After you set the FADE LENGTH, press ENTER and listen to the result. When the LCD asks **Make This Sound Permanent?** press NO - this cancels the CROSS-FADE and gives you a chance to try again with different values, or to change to a different Function.

While in **F7 LOOP PROCESS**, experiment with B&F. Again, press NO when asked **Make This Sound Permanent?** This cancels the B&F and gives you a chance to change functions.

HAVE FUN EXPERIMENTING - GET A FEEL OF WHAT ITS LIKE TO HUNT FOR THE ELUSIVE LOOP. IF YOU FIND ONE THAT'S PERFECT, WRITE DOWN THE VALUES OF L.S and L.L. (AND FADE LENGTH IF YOU USED CROSS-FADE).

Here's what I came up with:

1. Select **F6 LOOP START\LENGTH**
2. Set L.L to 005322 and L.S to 019774.
3. Play notes between C2 and F#2 and listen. This LOOP produces a slight tremolo that sounds quite natural. Should you keep this? Remember - this tremolo effect can't be cancelled - and chances are that you're not likely to find a matching tremolo in the other parts of your brass sound (SOUNDS #2 & #3). But keep it for the moment.

4. Select **F7 LOOP PROCESS** (the ENTER light is blinking, but this is one of the times you can ignore it).
5. Move the CURSOR under **B&F** (press YES tab).
6. Press ENTER - the LCD confirms that BACK & FORTH LOOP has been selected.

7. Press ENTER - the LCD flashes **This will take a while**, then it shows **Make This Sound Permanent? (Y/N)**
8. Play and listen - the B&F made the SOUND worse.
9. Press NO, because you don't want to keep this SOUND.
10. The LCD shows **Cancelled - Retry (Y/N)** Press NO.

WE NEED A BETTER LOOP!

11. Select **F6 LOOPSTART & LENGTH**

12. Set L.L to 535 and L.S to 12231.
13. This sounds dry, but the "analog" programming features should allow you to overcome this later.

Now that you have a reasonable LOOP for SOUND 01, move on to SOUND 02 and the rest of the SOUNDS in this MULTI SOUND.

14. Select **F1 SELECT M.SOUND/SOUND**.
15. Tap the UP-ARROW of DATA ENTRY A to select SOUND 02. Another way of selecting SOUNDS while in **F1 SELECT M.SOUND/SOUND** is to simply play the keyboard while watching the LCD. The SOUND NUMBER changes according to the keys you play - the DSS-1 is responding to the KEYBOARD ASSIGNMENTS.

16. Remember - SOUND 02 plays from G2 thru F#3. Select **F4 SOUND START & LENGTH** and watch the LCD. The S.S and S.L values confirm that this SOUND 02 is currently played in its entirety. The attack sounds fine to me - no need to change the START point. Let's go straight to the looping process.

17. Try finding your own loop and write the results down. I found one (L.L 267 and L.S 10876) that matches SOUND 01 in sound quality. Again - it sounds dry, but you can fix that later.

18. Select **F1 SELECT M.SOUND/SOUND**. Use the DATA ENTRY A to select SOUND 03, or play any keys between G3 and F#4.

19. Try finding your own loop and write the results down.

This completes looping the BRASS SOUNDS 01 & 02 & 03.

WHAT ABOUT THE HARP ?

Naturally, a harp sound doesn't sustain indefinitely, the strings fade out soon after being plucked. You now have a problem. According to **F5 LOOP ON/OFF**, either the whole MULTI SOUND loops or nothing loops. And you need to loop the BRASS SOUNDS for sustained notes. So, LOOP must be **ON**. And yet, you don't want the HARP SOUNDS to loop.

This is one of those times when you have to outsmart the computer. The DSS-1 doesn't care what part of a SOUND it LOOPS. If you pick a 'silent reading' from the end of each HARP SOUND, then you'll keep the computer happily looping, but you won't hear anything.

What is a silent reading? A group of readings with an amplitude of zero. How and where do you find such readings ? Either by ear, using **F6 LOOP START & LENGTH** (most likely near the end of the SAMPLE), or by using EDIT SAMPLE mode **F7 VIEW/EDIT SAMPLE DATA** (look for readings with zero amplitude).

Here's how I did it:

1. Select **F1 SELECT M.SOUND/SOUND**. Use the DATA ENTRY A or play the keys to select SOUND 04.
2. Select **F4 SOUND START & LENGTH**. Remove the attack delay by setting S.S to 1040. Don't worry about setting a S.L value, because the LOOP is **ON**, and the combined result of L.S and L.L overrides whatever you set for S.L (as discussed earlier in this Chapter).

3. Select **F6 LOOP START & LENGTH**. Set L.L to 1 (yes, just 000,001 -- a single sample).
4. Set L.S to13991.
5. Select **F1 SELECT M.SOUND/SOUND**, Use the DATA ENTRY A or play the keys to select SOUND 05.
6. Select **F4 SOUND START & LENGTH**. Remove the attack delay by setting S.S to 2380.
7. Select **F6 LOOP START & LENGTH**. After experimenting, set L.S to 15,759 and L.L to 000001.
8. Select **F1 SELECT M.SOUND/SOUND**. Use the DATA ENTRY A or play the keys to select SOUND 06.
9. Select **F4 SOUND START & LENGTH**. Remove the attack delay bysetting S.S to1,800.
10. Select **F6 LOOP START & LENGTH**. After experimenting, set L.S to 7,999 and L.L to 000001.

This is close to a natural HARP SOUND. It's not perfect. But since you're building a combined BRASS & HARP MULTI SOUND, a degree of compromise is necessary. If you wish to perfect the HARP SOUNDS by giving them a more natural decay characteristic, take each HARP SOUND to EDIT SAMPLE mode.

Using **F7 VIEW/EDIT SAMPLE DATA** you can tailor a downward slope of amplitude by adjusting each individual reading ("address"). Cancel your vacations and send out for pizza.

Chapter 24
FINE TUNING THE SOUNDS IN A NEW MULTISOUND

MULTI SOUND mode **F2 RELATIVE PARAMETERS** (**Tune/Lev/Fc**) gives you a last chance to treat the SOUNDS individually. The goal is to end up with a MULTI SOUND that doesn't let you hear the individual SOUNDS - you should just get an overall impression of a smooth sound (in your case, two sounds: BRASS & HARP).

You achieve this goal by balancing the timing, loudness level, and brightness of each SOUND in the MULTI SOUND.

Tune is self-explanatory. A chromatic tuner is handy at this stage.

Lev stands for loudness lewd. If your mixer has a reliable meter for amplitude measurements, then you'll get a more accurate reading on the levels than you can trust your ear to come up with.

Fc stands for Frequency Cutoff, which acts pretty much like a treble control.

1. Select **F1 SELECT M.SOUND/SOUND**. Use the DATA ENTRY A or play the keys to select SOUND 01.

2. Select **F2 REL.PARAMS**. The three parameters are now displayed side by side. Adjust the values with the DATA ENTRY A. Use the CURSOR to move from parameter to parameter.

3. Repeat steps 1 and 2 for SOUNDS 02 - 06. Keep in mind the KEYBOARD ASSIGNMENT for the SOUND currently selected. You can compare it to the other SOUNDS while adjusting values. Eventually you'll have a well-balanced BRASS and HARP section.

4. When you're done, you should SAVE this MULTI SOUND. Select **F9 SAVE/RENAME M.SOUND.** Re-name it if you wish - press YES in response to the prompt **RENAME?** and use DATA ENTRY A. Once you've dealt with Section 6 PROGRAM PARAMETER mode you may want to come back to this MULTI SOUND and add a lot of PROGRAM PARAMETERS to it.

SECTION 4 RECORDING AND EDITING YOUR OWN SAMPLES

In Chapter 1 you saw a general outline of the process of DIGITAL SAMPLING. Here we'll get specific.

The DSS-1 is very sensitive to the details of the sounds it records. Listen critically for noises and distortion in your input signals before they reach the DSS-1.

And when you use microphones, experiment with the positioning. Many vocal mics add boominess to sounds that you record from very close distances, so they may not be suitable for SAMPLING. And an air conditioner in the background can wreck your SAMPLE, even if you didn't think it was very loud. Whenever you record with a mic, you also record the room in which your sound source is placed. This can work to your advantage, if it is the "right" room.

Be adventurous.

I've gotten great results in the weirdest of circumstances. Get a drummer friend of yours to hit a drum in your bathroom, while you point the mic towards reflecting surfaces like the bathroom mirror (don't try to sample drums yourself - only drummers know how to make a drum "speak"). If you have a fireplace available - get a boom stand, stick the mic up the chimney, and place the drum in the fireplace (please - extinguish the fire first, I can't afford the liability insurance to cover readers' lawsuits...).

Should you record your SAMPLES with effects (reverb etc.) ?

Of course, as long as you realize that a SAMPLE with effect stands out in the company of other, non-effccted SAMPLES when you build a MULTI SOUND.

But you can get great results from this kind of trick: Record a sound with lots of reverb including early reflections. Then - in EDIT SAMPLE mode – chop off the reverberation after the sound itself has stopped playing (**F3 TRUNCATE START/LENGTH**). This can come out sounding like a gated reverb, the oh-so fashionable flavor of the month. And it saves you a reverb unit during mixdown.

The above examples are just there to whet your appetite for improvisation and adventure. The DSS-1 is very accommodating when it comes to editing SAMPLES. And the PROGRAM PARAMETER mode, where you treat MULTI SOUNDS before they become part of the performance SYSTEMS, lets you add new characteristics to your SAMPLES, including a dual DIGITAL DELAY.

So, let's quickly learn the basics to get you started. After that, the sky's the limit.

Chapter 25
PREPARE THE RECORDING OF A SINGLE SAMPLE

As a first learning example I'll get you to use a drum machine to provide a TOM sound. If that's not what you have available, then simulate it from a synthesizer, or hook op a microphone and record a similar sound. It's just for practice.

The following numbers will be needed for quick reference when sampling.

Available Sampling Frequencies: 16kHz, 24kHz, 32kHz, 48kHz.

Available Sampling Times (maximum duration per sample):

At 16kHz: 16 seconds.
At 24kHz: 11 seconds.
At 32kHz: 8 seconds.
At 48 kHz: 55 seconds.

Maximum LENGTH per SAMPLE: The same as the maximum WAVERAM capacity - 262144 words.

Note that a SAMPLE of the maximum LENGTH will fill up a MULTI SOUND by itself, and that MULTI SOUND will have to be the only MULTI SOUND when you build a performance SYSTEM, since the maximum for a MULTI SOUND and for a SYSTEM is also the absolute WAVERAM maximum of 262144 words.

1. Connect the output from your drum machine to the DSS-1 AUDIO IN.
2. Turn the DSS-1 ON (no disk needed at this stage).
3. Select SAMPLE mode. The LCD prompts you to select the Sampling Frequency of 32kHz with the (blinking) ENTER tab.

TIME OUT. Do we want 32kHz? Or 16kHz, 24kHz, or maybe 48kHz?

These other choices are available by DATA ENTRY A. Which to choose? It's really a question of sound quality in the reproduction. At 32kHz SAMPLING FREQUENCY you can expect a frequency response (playback quality) of up to around 14kHz. Plenty for your average tom, but not enough for a cymbal or a high-hat.

The rule of thumb is this: The higher the frequencies in the original, the higher the necessary Sampling Frequency for high fidelity results during playback. In general, pick a SAMPLING FREQUENCY about two-and-a-half times greater than the maximum frequency you want to record.

When planning complex MULTI SOUNDS, you might have to compromise SAMPLING FREQUENCY versus available memory. Toms are not all that demanding in the higher frequencies, nor do they take up too much memory. 32kHz should be okay.

4. Set the LCD to read **32kHz** and press ENTER. The LCD asks you to choose from these TOTAL SAMPLING TIMES: 4 seconds or 8 seconds.

This is not the expected time duration of your sample - obviously a tom rings out long before 4 seconds are up. This just gives you a preliminary chance at using all or only half of the available memory. In the next step you'll further subdivide this into suitable memory blocks. Let's choose half the memory. 4 seconds.

5. With the CURSOR under **4.0** - press ENTER. The LCD shows **SMPL-NO.01 MEM.DIV.01**.

By now the DSS-1 has taken you automatically to **F1 SAMPLE NO./MEM.DIV**.

Since the DSS-1 lets you record several SAMPLES in one session - one straight after the other - the first SAMPLE always gets the #01. This number is not permanent and doesn't become attached to the SAMPLE NAME when you SAVE the SAMPLE to DISK. The number is only valid while the newly recorded SAMPLE or SAMPLES are in RAM.

MEMORY DIVISION refers to the 5 ways that you can subdivide the available memory in advance.

The choices are:

01 (the whole memory)
02 (half the memory)
04 (one quarter of the memory)
08 (one eighth of the memory)
16 (one sixteenth of the memory)

This means that in your example a selection of MEM.DIV.01 would work out to be 4 seconds per SAMPLE. MEM.DIV.02 would be 2 seconds. MEM.DIV.04 would be 1 second. MEM.DIV.08 would be half a second. MEM.DIV.16 would be a quarter of a second.

The chosen MEM.DIV. puts a limit in size on all the SAMPLES that you're taking in one session. You can't mix 'n' match sizes, so if you're not sure how long a SAMPLE will need to be, err on the side of generosity. You can always recoup memory in EDIT SAMPLE mode.

Example: If your tom sound lasts for up to half a second, but you record it in a MEM.DIV. that is 1 second long, then EDIT SAMPLE mode **F3 TRUNCATE START/LENGTH** will allow you to cut out the "air" or silence that you recorded before and/or after the actual sound of the tom.

Let's RECAP where we arc: you selected 32kHz, which gives you a total of 8 seconds to record. Then you cut that in half by selecting 4 seconds. You really only need 1 second for the tom, so

6. With the CURSOR under **MEM.DIV**, use the DATA ENTRY A to set a value of **04**.
7. Press ENTER. Now the LCD asks **SMPL-KEY ASSIGN AUTO SET ? (Y/N).**

This asks you whether you want to take advantage of a feature that automatically assigns new SAMPLES to certain keys on the keyboard.

If you answer **YES** then the original sample will be re-played from C3 with a TOP KEY of F3, and with TRanspose set to ON. You'll be able to change this later - for now it just saves time to answer YES.

If you answer **NO** then the LCD shows you the ORG KEY as C3, the TOP KEY as C3 and TR (TRanspose ON). You can then use the CURSOR and the DATA ENTRY A to set different KEY ASSIGNMENTS. The TOP KEY can be up to one octave higher than the ORG KEY.

Neither YES nor NO will result in a permanent KEY ASSIGNMENT – only in MULTI SOUND mode will it become permanent.

8. Press YES. Now the LCD confirms that you're dealing with **SMPL:01** and with a memory **DIV.04**. It invites you to select any of the 5 FUNCTIONS in SAMPLE mode.

There are two ways of initialing the actual recording of the SAMPLE.

a) **F0 SAMPLE START** lets you press ENTER when you're ready – then you'd better hurry to get the SAMPLE recorded in the short time that you had allocated (1 second in our case). For this you need to be sure that the level of the SAMPLE will be neither too weak nor too strong for a clean recording. This you'll do in a minute with **F2 ATTN/GAIN**.

b) **F0 SAMPLE START** can also be used after setting a **F3 TRIGGER LEVEL**. This puts the DSS-1 on stand-by until it senses an incoming signal of the preset level, which triggers the recording.

Next you need to do a level check to get the best possible signal-to-noise ratio during recording.

9. Select **F2 ATTN/GAIN** (press 2 on the numpad). The LCD now shows **ATTN=00dB** and **GAIN=00dB**, with an asterisk on the far left.

10. Hit the play-button for the tom sound on your drum machine. The LCD acts as a level meter with a momentary peak-hold marker.

As the drum machine plays, you see a bar graph shoot from left to right and back. The segment furthest to the right takes a second to fade away, leaving an easily readable marker as to the loudest level the signal got to.

The ideal level is one that causes the meter to shoot as far to the right as it can while still reading the peak with a marker consisting of three bars.

If this marker is a black, filled-in square, then the signal is too strong and distortion is likely to occur in your SAMPLE.

If the signal is too strong, then you can do ooe of two things:

a) Reduce the level of the tom on your drum machine, or

b) use the DATA ENTRY A to set ATTENUATION (level reduction) in increments of -2dB, up to a maximum of -10dB.

If the signal is too weak, then you can do one of two things:

a) Increase the level of the tom on your drum machine, or

b) use the DATA ENTRY B to set a higher GAIN in increments of 10dB, up to a maximum of 40dB. If stepping up by increments of 10dB is too coarse, then you can use the DATA ENTRY A to set some ATTENUATION to arrive at GAIN levels between the coarse settings.

The asterisk indicates the current TRIGGER LEVEL, which I'll describe in a minute. Let's SAMPLE the tom without re-setting the TRIGGER LEVEL.

Chapter 26
RECORD AND SAVE A SINGLE SAMPLE

1. Select **F0 SAMPLE START** (press 0 on the numpad). The LCD confirms that you're about to record SAMPLE 01 as the first of a series of 4 SAMPLES (whether you plan to SAMPLE any others or not), all of the maximum size of one quarter of the available memory. We know that this is 1 second each. The SAMPLING FREQUENCY is confirmed as 32kHz, and the KEY ASSIGNMENT will be C3 for playback of the original SAMPLE.

2. With one hand at the drum machine and the other hand at the DSS- 1, press, in quick succession, ENTER and the Play button on the drum machine. Watch the LCD while you listen to the hit: If the peak level is too high, then you'll have to retry. If it looked okay, then you can immediately judge the result by ear.

3. Play C3 (the second C from the left on the DSS-1 keyboard) and listen critically. The DSS-1 knows that the first attempt is not always successful - it already asks **Retry? (Y/N)**. If you press YES then you're back at the screen of **F0 SAMPLE START**, ready for another try. If you press NO then you get the main screen of the SAMPLE mode: **Select (0-5)**.

4. Press YES for a retry, so that you can explore the TRIGGER LEVEL option.

5. Select **F3 TRIGGER LEVEL** (press 3 on the numpad). The asterisk in the LCD is the current TRIGGER LEVEL. It is at the far left, and the numerical value is shown as 000.

If you leave the TRIGGER LEVEL at 000, then the slightest noise after selecting **F0 SAMPLE START** and ENTER will cause the recording to start. This is how you just recorded the tom in step 1. The reason you had to hurry after pressing ENTER is that there is always some hum and noise on the line, and at a TRIGGER LEVEL of 000, this noise is enough to start the recording as soon as you press ENTER. This ensures that you capture the very first milliseconds of your SAMPLE, which is particularly important with sounds that have a swelling attack characteristic.

If you raise the TRIGGER LEVEL (with the DATA ENTRY A), then you can go to **F0 SAMPLE START**, hit ENTER and nothing will happen until the DSS-1 senses a signal at that preset TRIGGER LEVEL. This works best for percussive sounds that are at their loudest in the first part of the sound. As you adjust the TRIGGER LEVEL, the readout gives you the new values, and the asterisk moves from left to right.

6. Use the DATA ENTRY A to set the TRIGGER LEVEL to 50.
7. Select **F0 SAMPLE START** (press 0 on the numpad).

8. Press ENTER. Now the LCD message is **Ready to Sample**, and the asterisk marks the TRIGGER LEVEL. There is no rush for you to hit the tom Play button on the drum machine. As long as no sound at or above the TRIGGER LEVEL reaches the DSS-1, it will remain in record-ready mode.

9. Hit the tom Play button on your drum machine and watch the LCD. As the level marker crosses the asterisk in the LCD, the SAMPLING starts. Then the LCD asks for an answer about a retry.

10. Play C3 and listen: Did you lose any part of the attack of the tom sound? If so, answer NO to the current prompt **Retry? (Y/N)**, go back to step 4 and set a lower TRIGGER LEVEL than the one you set under steps 5 and 6 above.

When you think you have a successful SAMPLE, I want to show you one more thing before saving the SAMPLE to DISK.

11. Select **F4 ORIGINAL/TOP KEY** (press 4 on the numpad). The LCD shows that you're dealing with SAMPLE 01, taking up a fourth of the memory, with an ORG (original) key of C3, a TOP key of F3, and with the TRanspose function set to ON.

12. Play the keyboard - all keys from the left up to F3 give you transposed versions of your tom.

13. Move the DATA ENTRY A and you'll find that the DSS-1 won't let you set ORG KEY or TOP KEY higher than B3. This is because you chose the AUTO ASSIGN option under step 7 in Chapter 25. The key of C4 is therefore reserved for SAMPLE 02 (which you haven't recorded yet). If you set the TOP KEY to B3, then the ORG KEY can't be lower than B2, because the interval between any ORG KEY and TOP KEY can't be more than one octave. Use the CURSOR and the DATA ENTRY A to adjust these values.

NOTE: The KEY ASSIGNMENT that you're setting here in SAMPLE MODE will not become a permanent part of this SAMPLE, even when you SAVE it to DISK with SAMPLE mode **F5 SAVE SAMPLE** or with EDIT SAMPLE mode **F8 SAVE/RENAME SAMPLE.** It will be stored with an ORG KEY of C3 - automatically. But the interval between ORG and TOP KEY will be memorized.

Only when you eventually use this SAMPLE as a SOUND in a MULTI SOUND does the KEY ASSIGNMENT become permanent (MULTI SOUND mode **F3 ORIGINAL/TOP KEY** and **F9 SAVE/RENAME MULTI SOUND**).

Next you need to SAVE this crude, yet unfinished SAMPLE. You should make it a habit to SAVE your SAMPLES at different stages, on DISKS set aside for this purpose, and labeled CLEARLY...

14. Select **F5 SAVE SAMPLE** (press 5 on the numpad). The LCD asks you to select the SAMPLE you wish to SAVE. If you had just SAMPLED more than one, you would use the DATA ENTRY A to step through the numbers. Right now you only have one SAMPLE #01, so

15. Press ENTER. Now the LCD shows **Input Name:!NO-NAME**. Input stands for Audio Input Sample, as opposed to a waveform that you might have created or drawn in CREATE WAVEFORM mode.

16. Use the CURSOR and the DATA ENTRY A to write the name "tom 1".

I have a reason for suggesting that you use all small letters. This SAMPLE is an original in its crude form. It is not even edited yet. If you make it a habit to SAVE unedited SAMPLES with small letter names and edited SAMPLES with CAPITAL letter names, then you'll be better organized. If you prefer, start the names with one of the many funny symbols for the unedited versions, and pick another symbol for the edited SAMPLES.

17. Insert a formatted DISK into the drive.

18. Press ENTER. The LCD confirms that it's "tom 1", which is 032,478 words long. It is SAMPLE 01.

19. Press YES. SAVING begins, then the LCD asks **Continue? (Y/N)**. Since you don't have any other SAMPLE to SAVE right now,

20. Press NO.

Why is this SAMPLE 032,478 words long?

At step 5, you selected 04 seconds as the initial memory division. Then, as step 6, you further subdivided this into 4 memory blocks.

At 32kHz SAMPLING FREQUENCY, the maximum WAVERAM capacity of 262144 words allows for a total SAMPLING TIME of 8 seconds.

Half of that, 4 seconds, works out to be 131072 words. Divide this into 4 memory blocks - each block should be 032768 words long. We're only 290 words off on our tom 1 SAMPLE - close enough for jazz, the DSS-1 obviously needs a few words for purposes other than wave data.

Chapter 27
RECORD AND SAVE A GROUP OF SAMPLES IN ONE PRE-PLANNED SESSION

The following is a practice run of a sampling session involving four drum sounds from a drum machine. You can substitute similarly short sounds from a synthesizer or any other source.

I'm assuming that you can adjust the levels on your drum machine/sound source, so that the ATTN / GAIN and the TRIGGER LEVEL will work for all four sounds.

If that doesn't work out in your case, simply go to **F2 ATTN/GAIN** and **F3 TRIGGER LEVEL** before selecting **F0 SAMPLE START** for each new SAMPLE.

I'm also assuming that all four sounds will fit into a MEMORY DIVISION of 1 second each. If that doesn't work out in your case, select 8 seconds (the whole WAVERAM) in step 4.

At the end of the recording, I'll get you to SAVE the four SAMPLES while still in SAMPLE mode. You should always do that, for reasons explained in the next chapter.

The four SAMPLES will automatically form a MULTI SOUND. I'll also show you how you can process and SAVE the four SAMPLES as four SOUNDS of a finished MULTI SOUND while in RAM.

1. Clear the RAM - turn the power OFF and ON.
2. Enter SAMPLE mode.
3. Press ENTER for a SAMPLING FREQUENCY of 32kHz. NOTE: YOU CAN'T MIX DIFFERENT SAMPLING FREQUENCIES AMONG SAMPLES RECORDED IN ONE SESSION.
4. Press ENTER for TOTAL TIME 4.0 seconds (half the memory).
5. Press the UP-ARROW tab A twice to set a MEM.DIV. of 04.
6. Press ENTER.
7. Press YES to confirm **ASSIGN AUTO SET** (this will transpose the 4 SAMPLES at intervals of 1 octave).

8. Select **F2 ATTN/GAIN** (press 2 on the numpad).
9. Play the four drum sounds and use both DATA ENTRY A & B to set the necessary level peak.
10. Select **F3 TRIGGER LEVEL** (press 3 on the numpad).
11. Use the DATA ENTRY A to set a TRIGGER LEVEL of around 20.

12. Select **F0 SAMPLE START** (press 0 on the numpad). The LCD confirms that SAMPLE 01 (of 04), assigned to C3, is ready to be recorded at the SAMPLING FREQUENCY of 32kHz.
13. Press ENTER.
14. Play the first drum sound.
15. Play C3 (the second C from the left on the keyboard). If you need to record it again, press YES for a retry, which gets you back to step 13 above. If it sounds okay, press NO.
16. Select **F1 SAMPLE NO./MEM.DIV**. (press 1 on the numpad).
17. Use the DATA ENTRY A to change the SMPL-NO. to 02.

18. Select **F0 SAMPLE START** (press 0 on the numpad). The LCD confirms that SAMPLE 02 (of 04), assigned to C4, is ready to be recorded at the SAMPLING FREQUENCY of 32kHZ.

19. Press ENTER.
20. Play the second drum sound.
21. Play C4 (the third C from the left on the keyboard). If you need to record it again, press YES for a retry, which gets you back to step 19 above. If it sounds okay, press NO.
22. Select **F1 SAMPLE NO./MEM.DIV**. (press 1 on the numpad).
23. Use the DATA ENTRY A to change the SMPL-NO. to 03.

24. Select **F0 SAMPLE START** (press 0 on the numpad). The LCD confirms that SAMPLE 03 (of 04), assigned to C5, is ready to be recorded at the SAMPLING FREQUENCY of 32 kHz.
25. Press ENTER.
26. Play the third drum sound.
27. Play C5 (the fourth C from the left on the keyboard). If you need to record it again, press YES for a retry, which gets you back to step 25 above. If it sounds okay, press NO.
28. Select **F1 SAMPLE NO./MEM.DIV**. (press 1 on the numpad).
29. Use the DATA ENTRY A to change the SMPL-NO. to 04.

30. Select **F0 SAMPLE START** (press 0 on the numpad). The LCD confirms that SAMPLE 04 (of 04), assigned to C6, is ready to be recorded at the SAMPLING FREQUENCY of 32 kHz.
31. Press ENTER.
32. Play the fourth drum sound.
33. Play C6 (the fifth C from the left on the keyboard). If you need to record it again, press YES for a retry, which gets you back to step 21 above. If it sounds okay, press NO.
34. Select **F5 SAVE SAMPLE** (press 5 on the numpad).

35. The current SMPL is still 04. Use the DATA ENTRY A to change it to 01.
36. Press ENTER.
37. Use the DATA ENTRY A and the CURSOR to give this first SAMPLE a name of your choice.
38. Press ENTER.
39. Insert a blank DISK with space for at least 130,000 words of WAVERAM (~200 DISK BLOCKS).
40. Check the displayed name and press YES to SAVE the first SAMPLE. If the name is incorrect, press NO, then YES to continue, ENTER, re-write the name, press ENTER, then YES. If you get a DISK FULL message, insert another DISK, press YES to continue, ENTER, ENTER and YES.
41. When SAVING is completed, press YES to continue.

42. The current SMPL is still 01. Use the DATA ENTRY A to change it to 02.
43. Press ENTER.
44. Use the DATA ENTRY A and the CURSOR to give this second SAMPLE a name of your choice.
45. Press ENTER.
46. Check the displayed name and press YES to SAVE the 2nd SAMPLE. If the name is incorrect, press NO, then YES to continue, ENTER, re-write the name, press ENTER, then YES. If you get a DISK FULL message, insert another DISK, press YES to continue, ENTER, ENTER and YES.
47. When SAVING is completed, press YES to continue.

48. The current SMPL is still 02. Use the DATA ENTRY A to change it to 03.
49. Press ENTER.
50. Use the DATA ENTRY A and the CURSOR to give this third SAMPLE a name of your choice.
51. Press ENTER. Check the displayed name and press YES to SAVE the third SAMPLE. If the name is incorrect, press NO, then YES to continue, ENTER, re-write the name, press ENTER, then YES. If you

get a DISK FULL message, insert another DISK, press YES to continue, ENTER, ENTER and YES.
52. When SAVING is completed, press YES to continue.

53. The current SMPL is still 03. Use the DATA ENTRY A to change it to 04.
54. Press ENTER.
55. Use the DATA ENTRY A and the CURSOR to give this last SAMPLE a name of your choice.
56. Press ENTER.
57. Check the displayed name and press YES to SAVE the fourth SAMPLE. If the name is incorrect, press NO, then YES to continue, ENTER, re-write the name, press ENTER, then YES. If you get a DISK FULL message, insert another DISK, press YES to continue, ENTER, ENTER and YES.

Although you've just SAVED all four SAMPLES, the LCD still asks if you wish to **Continue? (Y/N)**. This gives you a chance to SAVE one, some, or all SAMPLES again to another disk, as a back-up or for whatever reason. We're done, so PRESS NO.

Now you have these four drum sounds at your disposal for whatever purpose you might want to use them. When you'll call them up from DISK, the KEY ASSIGNMENT will again be at the default of C3.

IMPORTANT: DON'T ENTER THE "EDIT SAMPLE" MODE UNTIL YOU'VE DONE ONE OF THE FOLLOWING:

a) SAVED your SAMPLES while still in SAMPLE mode

b) TAKEN YOUR NEW SAMPLES STRAIGHT TO THE MULTI SOUND MODE AND SAVED THEM AS A MULTI SOUND.

Failure to do one of the above can result in you losing your SAMPLES. In particular, never go from the SAMPLE mode to the EDIT SAMPLE mode without SAVING your SAMPLES first (unless you have only one SAMPLE).

Why this drastic warning? Chapter 28 explains it all.

A word about Option b): right now, your four SAMPLES are still in RAM. As far as the DSS-1 is concerned, they now form a MULTI SOUND, because they were recorded in one pre-planned session. If you don't need the functions available from EDIT SAMPLE mode, then you might wish to keep this MULTI SOUND together the way the DSS-1 has automatically constructed it during your sampling session. You can use all the functions in the MULTI SOUND mode, including changing the KEY ASSIGNMENTS. Then you can give the MULTI SOUND a name and SAVE it.

IMPORTANT: When you enter MULTI SOUND mode with a newly recorded group of SAMPLES still in RAM, ignore **F0 GET SOUNDS**, because your SOUNDS are already in RAM. **F0 GET SOUNDS** requires that you load SOUNDS from a DISK, which is time consuming, unnecessary, and, if you haven't SAVED the SAMPLES, it could even be fatal. Instead, go straight to **F1SELECT M.SOUND/SOUNDS** and start working on the individual SOUNDS. For more about the MULTI SOUND mode, review Chapters 15-24.

Before you go on to the EDIT SAMPLE mode, you need to have one more SAMPLE on the DISK. Follow the above steps and record a lower pitched tom. Name it "tom 2" and SAVE it.

CHAPTER 28
THE EDIT SAMPLE MODE SELECT SAMPLES AUTO REPEAT

In this mode, you'll be treating individual SAMPLES. They can be audio-input SAMPLES, or they can be created or drawn WAVEFORMS. Remember that on the DSS-1 SAMPLES, WAVEFORMS, and SOUNDS are the same as far as their organization is concerned.

When looking for SAMPLES / WAVEFORMS on a DISK, you must look for SOUNDS. DISK UTILITY mode **F4 SOUND DIRECTORY**, **F5 DELETE SOUND**, and MULTI SOUND mode **F0 GET SOUNDS** are functions that list the SOUNDS that you had SAVED as SAMPLES or as WAVEFORMS.

When you enter the EDIT SAMPLE mode, the first function involves **F1 SELECT SAMPLE**. This can be an audio-input SAMPLE that you recorded yourself and SAVED as a SAMPLE like you did with the drum SAMPLES at the end of chapter 27. Or it can be a created or drawn WAVEFORM that you made up yourself (see next Section). Or it can be a SOUND / WAVEFORM / SAMPLE that you're isolating from within a MULTI SOUND on a DISK, just like you did in Chapter 16 with the Brass and Harp SOUNDS.

You'll be editing one SAMPLE at a time. If you go from a SAMPLING session straight to EDIT SAMPLE mode without switching off the DSS-1 or loading any other waveform data, then you should have your new SAMPLES still in RAM. This means that you can select them **from MEMORY.** But as soon as you do that with one of your new SAMPLES, you'll lose all the others, unless you had already SAVED them all.

CAUTION: IF YOU'VE JUST RECORDED MORE THAN 1 SAMPLE IN ONE SAMPLING SESSION, DON'T ENTER THE EDIT SAMPLE MODE UNTIL YOU'VE SAVED ALL YOUR SAMPLES INDIVIDUALLY WITH SAMPLE MODE **F5 SAVE SAMPLE** (AS SHOWN IN THE PRACTICE SESSION IN CHAPTER 27) OR AS A GROUP WITH MULTI SOUND MODE **F9 SAVE/RENAME MULTI SOUND**.

Reason: In EDIT SAMPLE mode **F1 SELECT SAMPLE**, you can choose any one of your newly recorded SAMPLES **from MEMORY** (from RAM) to be the first one to be edited. But when you're done and you want to access another SAMPLE with **F1 SELECT SAMPLE from MEMORY**, thinking that all your new SAMPLES should still be in RAM, you'll be disappointed. The DSS-1 will not let you select any other SAMPLES in this mode/function. Strange as it may seem, the DSS-1 loses the directory and gets stuck on SOUND #01, which is the SAMPLE that you first selected with EDIT SAMPLE mode **F1 SELECT SAMPLE.**

If you've already left the SAMPLE mode without SAVING your SAMPLES, but you haven't gone as far as EDIT SAMPLE mode **F1 SELECT SAMPLE**, and you haven't yet pressed YES in answer to the prompt **F01 Get M01:S01** (or S02 or whichever SOUND number) **(Y/N)**, then there is still one escape possible: Enter MULTI SOUND mode, select **F1 SELECT M.SOUND/SOUND** and play the default KEY ASSIGNMENTS while watching the LCD.

If all your SAMPLES play, and the SOUND number changes in the LCD when you cross over into new KEY ASSIGN ranges, then select MULTI SOUND mode **F9 SAVE/RENAME M.SOUND**, write a name for the whole MULTI SOUND and SAVE it. Do this even if you didn't intend to use all the new SAMPLES together in one MULTI SOUND.

To get back at the individual SAMPLES, you'll have to use SYSTEM mode **F9 GET MULTI SOUND**, then enter EDIT SAMPLE mode, isolate one of the SOUNDS from within the MULTI SOUND, edit and name, it, then SAVE it with **F8 SAVE/RENAME SAMPLE**. For the next SAMPLE from within this MULTI SOUND, you'll have to use SYSTEM mode **F9 GET MULTI SOUND** again and repeat the procedure. This is what we had to do in Chapter 16 to extract the BRASS and HARP SOUNDS from within their MULTI SOUNDS on the Factory DISKS, so you are already familiar with this cumbersome procedure.

For our practice example, well get the tom SAMPLES from the DISK that you used in the previous Chapter. Let's get the first SAMPLE.

1. Enter EDIT SAMPLE mode.
2. Select **F1 SELECT SAMPLE**. The LCD asks **from MEMORY** or **from DISK**?
3. Move the CURSOR under DISK (press the YES/RIGHT ARROW tab).
4. Insert your DISK with the SAMPLE "tom 1" into the drive.
5. Press ENTER. The DSS-1 searches for the names of the SOUNDS on your DISK.
6. Use the DATA ENTRY A until **SOUND: tom 1** is displayed.
7. Press ENTER. The DSS-1 gets the specific information on this SOUND and asks you to confirm with YES/NO. The info should read **Get tom 1, L=032478,SF=32kHz**. If it's correct,

8. Press YES, and the DSS-1 loads the SAMPLE tom 1. On completion, the KEY ASSIGNMENT is displayed as **ORG= C3, TOP =F3,TR**. The original playback key of C3 is by default - even if you bad tried to SAVE another ORG key in the previous Chapter. Play C3 to confirm that it is indeed the correct SAMPLE, then

9. press ENTER. Now the SAMPLE is ready for editing, and the LCD invites you to select any one of the 8 functions in EDIT SAMPLE mode.

For convenience, in EDIT SAMPLE mode the DSS-1 can repeatedly play a SAMPLE while you hold a key down, or, more conveniently still, while you keep your foot on the (optional) sustain pedal. This is not the same as looping which you assign only in MULTI SOUND mode.

10. Select F2 (press 2 on the numpad). The LCD shows **AUTO REPEAT = OFF**.
11. Press the UP ARROW tab of the DATA ENTRY A to change OFF to **ON**. Sustain a key between C2 (the lowest C on the keyboard) and F3 - the "tom 1" will hit repeatedly.

Chapter 29
THE EDIT SAMPLE MODE - TRUNCATING SAMPLES

Let's stay with the tom sample from the previous Chapter. At C3 the tom hits once every second, because that is the length of the memory block in which it was recorded. But the actual duration of the tom's sound is not a full second. This means that you need to eliminate the silent part of this memory block after the end of the tom's decaying sound. This is called TRUNCATE.

1. Select **F3 TRUNCATE START/LENGTH** (press 2 on the numpad). The LCD displays the SAMPLE START as happening at the very beginning of the memory block (S.S000000) and the SAMPLE LENGTH as the total length of the memory block (S.L032478).

For **F3 TRUNCATE START/LENGTH**, both sets of DATA ENTRY sliders and tabs combine in this way:

Slider A steps through S.S and S.L numbers by 10,000s (10,000 / 20,000 / 30,000 etc.)
Arrow tabs A step by thousands (1,000 / 2,000 / 3,000 etc.)

Slider B steps by twenties (this also takes care of the hundreds).
Arrow tabs B step by ones.

2. Move the CURSOR under the value for S.L.
3. Hold down C3 and tap the DOWN ARROW tab of the DATA ENTRY A, reducing the LENGTH by thousands. The LCD changes to **Press ENT to Auto**. You'll hear no change in the sound until you press ENTER.

4. Set the S.L to around 5,000 and
5. Press ENTER. The DSS-1 takes a short while to compute the new LENGTH, then, while you're still holding C3 down, the playback rate changes drastically. The tom sound is now too short, the decaying end is chopped off. Not to worry - the original is still on the DISK, and even the current change hasn't been made permanent yet.

6. Change the LENGTH to a higher value, then
7. Press ENTER.

Listen carefully far the end of the sound. If the repeat function is distracting you,

8. Select **F2 AUTO REPEAT** (press 2 on the numpad) and
9. Use the DOWN ARROW of the DATA ENTRY A to turn AUTO REPEAT **OFF**.
10. Select **F3 TRUNCATE START/LENGTH** and adjust the LENGTH value, then press ENTER.

If you're still not sure just where the LENGTH should be, then you should look at the individual readings of your SAMPLE (see the next Chapter).

Chapter 30
THE EDIT SAMPLE MODE – VIEW / EDIT SAMPLE DATA

Let's stay with the example from the previous Chapter.

1. Select **F7 VIEW/EDIT SAMPLE DATA** (press 7 on the numpad).

For **F7 VIEW/EDIT SAMPLE DATA**, both sets of DATA ENTRY sliders and tabs combine in this way:

AD numbers:

Slider A steps through the AD numbers by 10,000s (10,000 / 20,000 / 30,000 etc.).
Arrow tabs A step by thousands (1,000 / 2,000 / 3,000 etc.).

Slider B steps by twenties (this also takes care of the hundreds).
Arrow tabs B step by ones (this also takes care of the tens).

DATA values:

Slider A and Arrow tabs A step through the DATA by hundreds (plus or minus 100 / 200 / 300 etc.)
Slider B and Arrow tabs B step by ones (this also takes care of the tens).

AD stands for ADDRESS - this is the location of every single reading (word) from the beginning to the end of the memory block that contains the current SAMPLE. In your example, this goes from 000000 to 032477. Why not to 032478? Because computers always count the first of anything as zero, so the first reading of this "tom 1" SAMPLE is listed as AD 000000.

The DATA values reflect the amplitude of the waveform at a given ADDRESS, from a maximum negative value of 2048 to a maximum positive value of 2047.

2. Move the CURSOR under AD and step up to numbers above 30,000.

The DATA values should all be around zero, because the last few milliseconds of the memory block contain no more sound from the tom 1. There is probably just a little activity from the line noise that was recorded after the "tom 1" sound had decayed during the recording. This is similar to leaving the mic on and the tape rolling after recording in the studio - there is always a little noise or hiss in the circuits.

3. To find the point where the actual tom 1 sound ends, step through the SAMPLE ADDRESS numbers from 000000 up. As soon as both the positive and negative numbers diminish to less than 100 and stay there over a considerable number of AD points, you know that you've reached the end. Make a note of the AD number, then

4. Select **F3 TRUNCATE START/LENGTH** (press 3 on the numpad).
5. Move the CURSOR under S.L.
6. Set the S.L value to the AD number found under step 24.
7. Press ENTER. The DSS-1 computes the LENGTH and looks for adjacent points of zero amplitude.

If you repeatedly press ENTER, you see that the S.L values get readjusted to lower numbers. The DSS-1 "knows" where the best (smoothest) end-points are and sets a new one every time you press ENTER. But you can always change the S.L number with the DATA ENTRY sliders and tabs, then press ENTER and listen to the result. This is one of the occasions where you don't have to press ENTER - you can select another function anytime you wish.

So far you've explored four of the eight Functions in EDIT SAMPLE mode:

F1 SELECT SAMPLE
F2 AUTO REPEAT ON/OFF
F3 TRUNCATE START/LENGTH
F7 VIEW/EDIT SAMPLE DATA

Three of the remaining four functions change the SAMPLE drastically. Before we look at them, use the eighth Function (F8 SAVE/RENAME SAMPLE) to SAVE the truncated "tom 1".

Chapter 31
SAVE AND RELOAD THE EDITED SAMPLE

1. Select **F8 SAVE/RENAME SAMPLE** (press 8 on the numpad).
2. Press YES in answer to the **RENAME (Y/N)** prompt.

3. Using the CURSOR and DATA ENTRY A, write the name TOM 1 in CAPITAL letters. This is to indicate that it is an edited version, as opposed to the crude original SAMPLE which you named in small letters.

4. Press ENTER. Now the new name and the new LENGTH are displayed. Once you begin working with this new version you'll have recovered the "wasted" memory.

5. Insert a new but formatted DISK into the drive. This DISK should be labelled "Edited SAMPLES" or something similar, so that you won't confuse it with your DISKS containing the raw original SAMPLES.
6. Press YES in answer to the prompt **SAVE? (Y/N).**

Take a break. When you're ready to continue,
7. Insert your DISK "Edited SAMPLES" into the drive.
8. Enter EDIT SAMPLE mode.
9. Select **F1 SELECT SAMPLE**.
10. Move the CURSOR under DISK.
11. Press ENTER.
12. Use the DATA ENTRY A to display your edited SAMPLE TOM 1.
13. Press ENTER.
14. Check the LCD for the correct SAMPLE and press YES. The KEY ASSIGNMENT is displayed -
15. Press ENTER.

Chapter 32

THE EDIT SAMPLE MODE REVERSING SAMPLES

I'm sure that your imagination will ran rampant when it comes to the use of this feature. Many sounds come out surprisingly useful when reversed. Dog barks for one - the swelling-up-and-cutting-off nature of a dog bark, when reversed and truncated, can make weird-and-wonderful percussion sounds. And if you ever needed proof that your neighbor/girlfriend/teacher is "backwards" - here's your chance: Record his/her voice and reverse the SAMPLE.

For now, let's play with "TOM 1".

1. Select **F4 REVERSE SAMPLE**. The LCD prompts you to press ENTER to start reversing the SAMPLE.
2. Press ENTER. The LCD warns **This Will Take A While**, but soon it confirms **Sample Reversed**.
3. Play C3 on the keyboard. What you hear is what you get...

Before you experiment more with this reversed SAMPLE - SAVE it with **F8 SAVE/RENAME SAMPLE**. (Rename it "TOM 1 R").

Chapter 33
THE EDIT SAMPLE MODE - LINKING SAMPLES

One of the mare interesting things you do with the DSS-1 is the LINKING of SAMPLES. Imagine the attack of a trumpet finked to the voice of your favorite (or most hated) opera singer. Or the horn of a 1956 Chevy turning into a solo violin - this is what LINKING SAMPLES is all about Let's try it, while you still have the reversed version of the tom sound "TOM 1 R" in RAM.

1. Select **F5 LINK SAMPLES**. Immediately the DSS-1 wants to know which SAMPLE to call up. The SAMPLE you select will be attached to the end of the SAMPLE currently in RAM - the two SAMPLES will get played back-to-back.

2. Insert your DISK with the edited (truncated), but non-reversed version of TOM 1 into the drive.
3. Press ENTER.
4. Use the DATA ENTRY A to display TOM 1.
5. Press ENTER.

When the SAMPLE to be linked is displayed, its START and LENGTH are shown, and the CURSOR is under the START value. Should you wish to link this SAMPLE at a starting ADDRESS other than 000000, you would use the DATA ENTRY sliders and tabs A & B to set a new START value. In our example, let's leave START at 000000.

6. Press ENTER. Another display shows the **NAME**, **LENGTH** and **SAMPLING FREQUENCY** of the SAMPLE you've just chosen. If all is well,

7. Press YES - The SAMPLE gets loaded and LINKED to the SAMPLE currently in RAM. The LCD confirms the LINK and gives you a chance to adjust the LINK

8. Play C3 on the keyboard.

You hear the reversed SAMPLE followed by the forward SAMPLE. In your example, a swell up followed by a swell down. The DSS-1 looks for the smoothest possible LINK by automatically crossfading the two SAMPLES. If you press YES in response to the prompt **Adjust LINK? (Y/N)**, a new crossfade point is calculated - use your ears to judge. If nothing seems to work out, you can always use **F1 SELECT SAMPLE** to reload TOM 1 R. Then select **F5 LINK SAMPLES** and set a new START point for TOM 1.

If you keep pressing YES for a better LINK adjustment, eventually the LCD will display **Can't Adjust Cross Fade? (Y/N).** If you press YES then you get a readout of the amount of words that you want to use for the CROSS FADE. Use the DATA ENTRY A to set a new **FADE LENGTH** (which can't be longer than the shorter of the two SAMPLES involved). Then press ENTER to execute. The LCD will ask you to retry the LINK. You'll be back where you started from at step 1.

If you press NO after the prompt **Can't Adjust Cross Fade (Y/N)**, then you're invited to immediately try the LINK again. When the DSS-1 has run out of options, it throws in the towel by displaying **Can't Adjust - Select (1-8).**

Rule of thumb: If you try to LINK two SAMPLES where the end of the first SAMPLE is very different from the beginning of the second SAMPLE - dont expect a smooth transition. Use all the Functions in EDIT SAMPLE mode, including **F7 VIEW/EDIT SAMPLE DATA**, to treat both SAMPLES in such a way that the "joint" will match up. This could require some drastic adjustments in **F7 VIEW/EDIT SAMPLE DATA**, for both SAMPLES. If you want a gunshot to turn smoothly into a gong, or a squealing tire to turn into a harp glissando - take your time and you'll get there.

Once you've SAVED a linked SAMPLE, you can always call it up again and edit it some more, including reversing it or linking it to yet another SAMPLE. The only limit is the RAM limit of 262,144 words.

Can you "un-link" a linked SAMPLE? Yes - very painfully. You'll have to find the "joint" and set the START or LENGTH under **F3 TRUNCATE START/LENGTH**. However, the "joint" will not return to the original end/start characteristics, because the "joint" has been cross-faded. The longer the CROSS FADE LENGTH, the more the original SAMPLES will have been altered.

But more than likely, you won't ever need to "un-link" two SAMPLES because you'll always have the individual SAMPLES on DISK, the way they were before you linked them. Another good reason for SAVING SAMPLES at every step of their development. Happy housekeeping!

Chapter 34
THE EDIT SAMPLE MODE - MIXING SAMPLES

To explore the last Function in EDIT SAMPLE mode, we'll try to MIX the two toms in their original "crude" versions.

1. Select **F1 SELECT SAMPLE.**
2. Insert your DISK with the original "crude" tom 1 SAMPLE.
3. Place the CURSOR under DISK
4. Press ENTER.
5. Use the DATA ENTRY A to display **tom 1**.
6. Press ENTER. When **tom 1** is displayed,
7. Press YES. When loading is completed,
8. Press ENTER.
9. Select **F6 MIX SAMPLES** (press 6 on the numpad). Immediately the DSS-1wants to know which SAMPLE you wish to mix with tom 1.
10. Insert your DISK containing tom 2.
11. Press ENTER.
12. Use the DATA ENTRY A to display tom 2.
13. Press ENTER. When tom 2 is displayed,
14. Press YES. As soon as loading is completed, the DSS-1 needs to know whether you wish the mix to be in equal proportions, and whether the tuning of the second SAMPLE needs adjusting. Default values are (mix) **RATIO=50%, TUNE=00**. Let's go with that.
15. Press ENTER. The LCD shows **This Will Take A While**. Eventually it says **Samples Mixed**.
16. Play C3 and listen.

At this stage, the mix is completed. You can't go back and change the **RATIO** or the **TUNE**. But you can always start at step 1 again.

If you're thinking that you might want to keep the present mix before trying another one, SAVE it with **F8 SAVE/RENAME SAMPLE**. And the finished, mixed SAMPLE can, of course, be mixed again with any SAMPLE. So if you don't quite hear enough from the lower tom 2 in the mix you just finished, go back to step 10 above, mix again and increase the RATIO of the second SAMPLE (tom 2) this time around.

A Look Ahead

A finished mixed SAMPLE can't be adjusted for balance ratio or any other factors - it is permanently mixed and will from now on be treated like any other SAMPLE. This is different from the Oscillator Mix Ratio in the PROGRAM PARAMETER mode (**F14 MIX RATIO**). There you can assign a balance between two MULTI SOUNDS, one in each Oscillator.

SECTION 5: CREATING WAVEFORMS

In this mode, the DSS-1 offers you two ways of creating sounds directly on the instrument, without the need for input sampling. You can actually build sounds from scratch. Not just by using some pre-packaged waveforms like on other synthesizers, but in a manner that was only possible on much more elaborate equipment until the DSS-1 came along.

With F1 DRAW WAVEFORM, you plot the shape of a single wave cycle. With F2 HARMONIC SYNTHESIS, you choose from a maximum of 128 harmonics ("overtones" or "partials") by setting the amplitude of each harmonic to values from zero (silent) to 255 (maximum loudness).

These created WAVEFORMS are SAVED on DISK as SOUNDS or as MULTI SOUNDS. When used as part of a SYSTEM, they work out to be very economical, because these WAVEFORMS take up very little memory, while giving you a lot of variety in programming. The great majority of MULTI SOUNDS on the FACTORY DISKS are created WAVEFORMS.

This gets us right into the nitty-gritty of a subject that you may or may not be familiar with: Basic Physics of Sound. Without this knowledge you'll have no chance of making efficient use of this MODE.

Chapter 35
BASIC PHYSICS OF SOUND

Fact 1: Sound is a series of vibrations.

When air molecules vibrate in response to a loudspeaker's motion or any other sound generator, your ears sense the vibrations and translate that sensation into the recognition of many things at once: the pitch of the sound you hear; the loudness; the timbre (color); the direction from which the sound reaches you; the reflections off walls and objects etc. The shape of the up-and-down movement of this vibration is called the WAVEFORM (a.k.a. WAVESHAPE)

Fact 2: Pitch is the result of the frequency of vibrations.

How often does a string have to vibrate to produce the tuning A above middle C?
Answer: 440 times per second.

A vibration consists of an "up" and a "down" motion, just like the peak and the valley of a wave. Together these two portions form a cycle. The word "frequency" itself means "how often" or "how frequently" a cycle completes itself. So the "tuning" A can be said to have a frequency of 440 cycles per second, abbreviated as cps. In honor of a scientist by the name of Heinrich Hertz, the frequency or cycles per second are often stated in hertz, abbreviated as Hz.

Frequency & cps & Hz all mean the same: a measurement of PITCH.

Fact 3: For every frequency (note, pitch) you produce on a musical instrument (called the Fundamental), and for every noise you generate on any object, you also get many other frequencies that you didn't actually play (called Overtones or Partials).

If you have access to a piano, try this: Gently press and hold down the key of E below middle C. Make sure that you don't "play" the strings – just press enough to lift the damper off the strings. Now play a hard and short note on the 3rd A below middle C. You'll hear the E swell up, almost as if you had played it. Now try the same with the E flat or the F next to the same E below middle C. Neither of these two notes will ring out like the E did when you strike the A. The note E is a Harmonic Overtone of A. The fact that the A causes the E to sound is called Sympathetic Vibration.

Fact 4: The Harmonic Overtones are multiples of the Fundamental Frequency.

Every higher octave is double the frequency of its lower octave. Most sounds also contain overtones that are not strict multiples of the Fundamental Frequency. The more of those non-harmonic overtones are present, the less we are sure about the exact pitch of a note. What gives a sound its color (timbre)? Three things: a) the amount of overtones, both harmonic and non-harmonic b) the relative loudness of the overtones, and c) the timing (envelope) of the overtones.

Fact 5: WAVEFORMS come in thousands of variations, but you realty only need to memorize one of them: the SQUARE WAVE.

It contains all of the odd-numbered harmonic overtones (no octave doublings). How loud the harmonics of a SQUARE WAVE are compared to the fundamental is easy to remember: Turn the number of the harmonic into a fraction under a 1 and you know the answer. If you know what notes make up a dominant 13th (#11) chord, then you know all you need to know - from the 7th harmonic upwards the numbers correspond to the positions in this chord. And while you think in terms of a chord, notice that the fifth in the chord is the third in the harmonic series, and the third in the chord is the fifth in the harmonic series. If ever in doubt, flip-flop these 3rd/5th functions and you can't go wrong.

In reality, this "chord of nature" isn't quite "right" to our ears, because the natural harmonics are slightly out of tune compared to the way we tune our musical instruments.

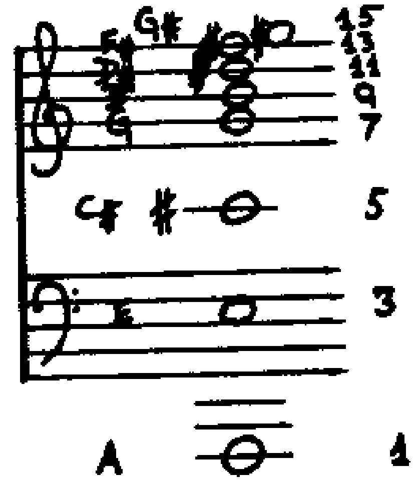
Here's the OVERTONE STRUCTURE OF THE SQUARE WAVE, shown in notes, based on an arbitrarily chosen Fundamental of low A.

THE HARMONIC SERIES
THE SQUARE WAVE

This example is based on the (arbitrarily chosen) FUNDAMENTAL of "A" two octaves plus a minor third below middle "C". The example lists the first 8 partials. Note that there are no octave duplications in the harmonic series of the SQUARE WAVE. The result: you only deal with ODD NUMBERS. The even numbers 2, 4, 6 , etc. are the result of octave duplications. Compare this example with the SAWTOOTH WAVE where all the harmonics re-appear at every octave.

The first eight partials of a SQUARE WAVE form a DOMINANT 13th CHORD with a #11. This is easy to memorize. But keep in mind that the way we tune musical instruments to the "tempered" scale is a compromise. Several notes in this example would sound quite out of tune if this chord were used in a musical context. Nature's "tuning" is different from the tuning used in the "European/western" culture.

Frequencies of the harmonic in this example, based on 'A' = 55 Hertz



15th: 825 Hz (15x55)
13th: 715 Hz (13x 55)
11th: 405 Hz (11x55)
9th: 495 Hz (9x55)
7th: 385 Hz (7x55)
5th: 275 Hz (5x55)
3rd: 145 Hz (3x55)
1st: 55 Hz (Fundamental)

Relative Loudness (Amplitude) of the harmonics of the
SQUARE WAVE compared to the FUNDAMENTAL's loudness

the 15th harmonic is 1/15 as loud as the FUNDAMENTAL
the 13th harmonic is 1/13 at loud as the FUNDAMENTAL
the 11th harmonic is 1/11 as loud as the FUNDAMENTAL
the 9th harmonic is 1/9 as loud as the FUNDAMENTAL
the 7th harmonic is 1/7 as loud as the FUNDAMENTAL
the 5th harmonic is 1/5 as loud as the FUNDAMENTAL
the 3rd harmonic is 1/3 as loud as the FUNDAMENTAL

Fact 6: The SAWTOOTH WAVE (aka RAMP WAVE) includes all the natural harmonics (the odd-numbered harmonics plus their octave doublings - all the even numbers).

This is the complete HARMONIC SERIES. It so happens that octave doublings always come out as even numbers, so you don't need to strain yourself trying to remember them all - just divide even numbers by half until you come to an odd number, and you'll know the pitch. Example: You wonder about the pitch of the 12th harmonic of A? Divide 12 by two = 6, still an even number. Divide it by two again = 3. The third harmonic you should have memorized as being one octave and a fifth above the fundamental. That makes it an E in our example. And since you divided twice, you came down by two octaves. To get back to the 12th harmonic, go up two octaves from the 3rd harmonic and you've got the twelfth nailed.

Here's the OVERTONE STRUCTURE OF THE SAWTOOTH WAVE, shown in notes, based on an arbitrarily chosen Fundamental of low A. The black note heads indicate octave doublings (even numbers).

THE HARMONIC SERIES THE SAWTOOTH WAVE

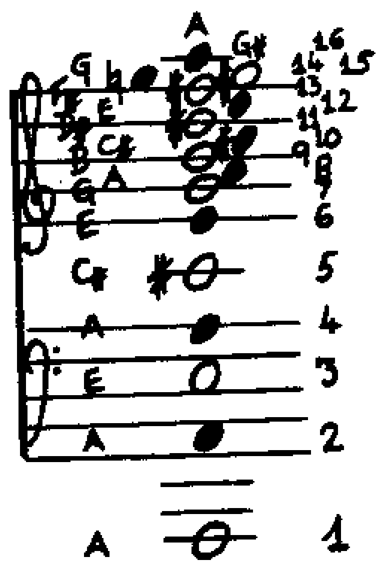
This example is based on the (arbitrarily chosen) FUNDAMENTAL of " A" two octaves plus a minor third below middle "C". The example lists the -first lfc partials. Note that the even numbers 2, 4, 6, etc. are the result of octave duplications. Compare this with the SQUARE WAVE where only the odd-numbered harmonics are present.

The first 13 partials of the SAWTOOTH WAVE (aka RAMP WAVE) form a DOMINANT 13th CHORD with a #11. This is easy to memorize. The upper part gets very crowded, due to the octave duplications of lower harmonics. And keep in mind that the way we tune musical instruments to the "tempered " scale is a compromise. Several notes in this example would sound quite out of tune if this chord were used in a musical context. Nature's "tuning" is different from the tuning used in the "European / western" culture.

Blank noteheads are the odd-numbered harmonics. Filled-in noteheads are octave duplications of lower harmonics. They always work out to be even-numbered harmonics.

Frequencies of the harmonics in this example, based on 'A' = 55 Hertz

16th: 880 Hz (16x55)
15th: 825 Hz (15x55)
14th: 770 Hz (14x55)
13th: 715 Hz (13x55)
12th: 660 Hz (12x55)
12th: 605 Kz (11x55)
10th: 550 Hz (10x55)
9th: 495 Hz (9x55)
8th: 440 Hz (8x55)
7th: 385 Hz (7x55)
6th: 330 Hz (6x55)
5th: 275 Hz (5x55)
4th: 220 Hz (4x55)
3rd: 165 Hz (3x55)
2nd: 110 Hz (2x55)
1st: 55 Hz (Fundamental)



Relative Loudness (Amplitude) of the harmonics of the SAW compared to the FUNDAMENTAL

the 16th harmonic is 1/16 as loud as the FUNDAMENTAL
the 15th harmonic is 1/15 as loud as the FUNDAMENTAL… etc

The above illustrations don't show higher numbered partials. But in nature there is only one limit to how many we can hear: Frequencies above around 16,000 (16kHz) are inaudible to the average adult human ear. The theoretical limit is usually stated as 20kHz. So if a sound's frequency is up there, whether it's an overtone of some lower fundamental or whether it is a fundamental, we can't hear it.

Harmonics above the 21st have to "share" notes in our chromatic scale. So they bunch up into quarter tones and even finer intervals. Most noticeably "out of tune" are the 7th and 11th harmonics.

Chapter 36
DRAWING WAVEFORMS

1. Enter CREATE WAVEFORM mode.
2. Select F1 DRAW WAVEFORM (press 1 on the numpad).
3. Watch the LCD and press ENTER.

For 8 seconds only you see the LCD as a "plotter pad". In the upper half you see blacked out squares. They represent the HEIGHT of an imaginary waveform, which is the wave's LEVEL OF AMPLITUDE. If you move the DATA SLIDER A during these 8 seconds, the number of black squares increases and decreases from left to right. The fewer squares, the less the LEVEL.

In the lower half of the LCD you see, from left to right, the word TIME and a set of arrows that crawl across 16 segments. This represents the time it takes to complete a wavecycle. By now the 8 seconds are over and the LCD says WAVEFORM Drawn - Retry? (Y/N).

4. Press YES. Be ready to move the DATA SLIDER A, then
5. Press ENTER and move the DATA SLIDER A during the 8 seconds before the LCD changes. As soon as the LCD changes to WAVEFORM DRAWN - Retry? (Y/N), test the new waveform.

6. Play the keyboard - the DSS-1 has just

a) computed your newly drawn waveform at 32kHz Sampling Frequency and at the pitch of B1
b) translated that shape into the necessary harmonics
c) transposed the single waveform into 8 waveforms
d) assigned these 8 waveforms across the keyboard (every F# is the start of a new waveform)
e) looped these 8 waveforms
f) combined these 8 waveforms into a MULTI SOUND

And it's ready to let you try again and again.
But every time you reply YES to the prompt Retry? (Y/N) the previously drawn waveform gets erased.

Remember - the result is always just a crude single wawcycle, without the refinements available from the MULTI SOUND and PROGRAM PARAMETER modes. So if the first results sound buzzy and a bit unpleasant, don't be surprised. As you'll get more experienced you'll begin to hear the potential of a waveform while it is still new.

7. Press NO in answer to Retry? Another prompt appears: Do you want to Edit the Waveform? (Y/N).

This is not the same as taking the waveform into EDIT SAMPLE mode. The currently offered Edit the Waveform feature is a shortcut to an edit function that can't be accessed at any other stage from within CREATE WAVEFORM mode. If you press NO then you'll need to enter EDIT SAMPLE mode to achieve the same edits that the current prompt lets you do. For now

8. Press YES. This takes you to a screen similar to the one that you saw in EDIT SAMPLE mode under F7 VIEW / EDIT SAMPLE DATA. But here you are still in CREATE WAVEFORM mode.

In this EDIT screen, the DATA ENTRY A changes the cursor-selected values by the hundreds, and the DATA ENTRY B changes them by the ones.

The ADDR values stand for the 512 sample points of this waveform, listed as 000-511. The LEVEL values are the amplitude of the wave, from negative (the "valley") -2048 through 0000 (the zero crossing) to positive (the "peak") + 2047.

However, strangely enough, my DSS-1 sets the maximum amplitude of drawn waveforms to + 2016, which I can manually edit to the actual available maximum of + 2047. Not that the difference between +2016 and +2047 would change the sound much in most cases. If your DSS-1 acts the same way as mine does, and if you're a stickler for perfection, use the CURSOR and the DATA ENTRY B to change the peaks from +2016 to +2047.

You may also notice that the smallest difference between any two DATA readings is 8. If you're familiar with Sampling Theory, you may be concerned that this decrease in accuracy from 12 to 9 bits may affect the quality of your created waveforms. But rest assured - the overall fidelity of sampling a dynamic range of -256 to +255 with 12-bit SAMPLES is basically the same as creating waveforms with a dynamic range of -2048 to +2047 using 9-bit data readings.

Step through the waveform one ADDR value at a time. Use the CURSOR to go from left to right. If you remember your drawing action, then you should see it confirmed here. Whenever you had a lot of black squares on the screen during the 8 seconds of drawing, you should now see high positive LEVEL values. Less than 8 black squares during drawing would mean negative LEVEL values. But you can draw a waveform within the negative range of values - the DSS-1 will treat it as a waveform having less overall loudness than those waveforms whose range of LEVEL extends to the + and - maximum.

What can you do next with this waveform ? You have several options.

Chapter 37
VARIOUS WAYS OF USING AND SAVING A DRAWN WAVEFORM

a) You can SAVE it with CREATE WAVEFORM mode **F3 SAVE WAVEFORM**

b) You can have the DSS-1 calculate and display the harmonics of the newly drawn waveform using the choice **Current** in **F2 HARMONIC SYNTHESIS**. (Caution: see below the severe limitations of this Option).

c) You can enter EDIT SAMPLE mode, call the waveform up **From Memory** and further edit it, then SAVE it with **F8 SAVE/RENAME SAMPLE**. (Caution: see below the severe limitations of this Option).

d) You can enter MULTI SOUND mode and edit the waveform as a group of 8 SOUNDS (remember - the newly drawn waveform is automatically a MULTI SOUND). Then you can SAVE/RENAME the MULTI SOUND with **F9 SAVE/RENAME MULTI SOUND**.

e) You can enter PROGRAM PARAMETER mode, write a PROGRAM for this waveform (remember - it is already a MULTI SOUND, but hasn't been SAVED yet), then enter SYSTEM mode and SAVE the SYSTEM (**F2 SAVE SYSTEM**) which automatically SAVES the MULTI SOUND and the PROGRAM as part of the SYSTEM.

f) **THERE'S ONLY ONE OPTION THAT SAVES THE NEWLY DRAWN WAVEFORM COMPLETE AS AN UNCHANGED MULTI SOUND: ENTER MULTI SOUND MODE, SELECT F9 SAVE/RENAME MULTI SOUND, GIVE IT A NAME AND SAVE IT.** This is similar to Option d), but without editing.

Each of these options has certain consequences. Let's look at them one by one.

OPTION a)
SAVE THE DRAWN WAVEFORM IN CREATE WAVEFORM MODE F3 SAVE WAVEFORM

After drawing the waveform, whether you answered YES to the prompt **Edit Waveform? (Y/N)**, you can

1. Select **F3 SAVE WAVEFORM** (press 3 on the numpad). The LCD asks you to **SELECT WAVEFORM**.

With the DATA ENTRY A you can select from the following:

Waveform:01 - Key:Bl
Waveform:02 - Key:B2
Waveform:03 - Key:B3
Waveform:04 - Key:B4
Waveform:05 - Key:B5
Waveform:06 - Key:B6
Waveform:07 - Key:B7
Waveform:08 - Key:B8

You can't move the CURSOR under the Key value - all you can do is select one of the 8 waveforms. Do I hear you say "what EIGHT waveforms – I only drew ONE"?

Remember - the DSS-1 automatically "multiplied" your drawn waveform, so that it would play over the entire range of the keyboard (and beyond if you address the DSS-1 via MIDI). Your original waveform is 512 words long and is assigned to B1. Why B1? TIME OUT.

If you try to play the key of B1, you'll find it's not even on the keyboard. The lowest key on the DSS-1's keyboard is C2. So what does **Key:B1** mean? It means that the sounding pitch of Waveform:01 is B1. What key on the DSS-l's keyboard should you play to hear that pitch of B1? Would you believe C3? Here's how this all works.

When you draw a waveform, the DSS-1 reads the amplitude of the DATA ENTRY A slider 512 times. It assumes that those 512 readings make up one complete wavecycle - no more, no less. The DSS-1 also assumes that this wavecycle is being "sampled" at 32kHz - in other words, 32,000 readings per second. Since it only has 512 readings, it assumes that the single wavecycle occurred at a frequency of 62.5 cycles per second. (62.5 cps x 512 readings per cycle = 32,000 readings per second). The pitch of B1 is roughly 62.5 cps or Hz.

Why does the key of C3 play the pitch of B1? Because C3 is the default KEY ASSIGNMENT for any newly "sampled" waveform. A drawn waveform is no exception.

Are you still wondering why you end up with eight waveforms when you only drew one? The answer lies in the automatic mapping of your drawn waveform across the whole keyboard. Basically, the DSS-1 makes an exact copy of your waveform for every octave above B1. Since the frequency of each higher octave is double compared to its lower octave, the number of samples required to represent the identical waveform is only half. So, you end up with the original drawn waveform at the original (default) pitch of B1, plus the next seven higher octaves as follows:

Waveform:01	Key:B1	512 words
Waveform:02	Key:B2	256 words
Waveform:03	Key:B3	128 words
Waveform:04	Key:B4	64 words
Waveform:05	Key:B5	32 words
Waveform:06	Key:B6	16 words
Waveform:07	Key:B7	8 words
Waveform:08	Key:B8	4 words
		1,020 words

All drawn or "synthesized" MULTI SOUNDS on the DSS-1 are 1,020 words long. This is because they're actually groups of 8 waveforms, mapped across the keyboard as shown above. Look at any tables showing the MULTI SOUNDS in the SYSTEMS on the Factory Disks and you'll see plenty of MULTI SOUNDS of this type.

By SAVING the waveform in the CREATE WAVEFORM mode using **F3 SAVE WAVEFORM**, you are breaking up this automatic grouping. You can only SAVE the eight transposed versions of your drawn waveform one at a time. (The same is true for "synthesized" waveforms).

If you want to SAVE the entire group of 8 waveforms instead of (or in addition to) SAVING them individually, enter the MULTI SOUND mode and use **F7 SAVE/RENAME MULTI SOUND** (see Option f).

Here's how you SAVE the 8 waveforms individually, in the CREATE WAVEFORM mode, using **F3 SAVE WAVEFORM**. You should still have the **Select Waveform** LCD display as per step #1from above.

2. Make sure that **Waveform:01 Key:B1** is displayed (use the DATA ENTRY A if necessary).
3. Press ENTER.
4. Using both the CURSOR and the DATA ENTRY A, write the name **a) 01-B1**.
5. Press ENTER . The LCD displays the new name, a LENGTH of 000512 I and the prompt **SAVE? (Y/N).**

6. Make sure that you have a practice DISK in the DRIVE. Press YES. When SAVING is completed, the prompt **Continue? (Y/N)** appears. To continue means to SAVE more waveforms.

7. Press YES. You're back at the prompt **Select Waveform**.
8. Use the DATA ENTRY A to display **WAVEFORM:02 KEY:B1**
9. Press ENTER.
10. Using both the CURSOR and the DATA ENTRY A, write the name **a) 02-B2**.
11. Press ENTER. The LCD displays the new name, a LENGTH of 000256 and the prompt **SAVE? (Y/N).**
12. Press YES. When SAVING is completed, the prompt **Continue? (Y/N)** appears. To continue means to SAVE more waveforms. Let's just SAVE one more - the tiny and highest #08.

13. Press YES. You're back at the prompt **Select Waveform**.
14. Use the DATA ENTRY A to display **WAVEFORM:08 KEY:B8**
15. Press ENTER
16. Using both the CURSOR and the DATA ENTRY A, write the name **a) 08-B8**.
17. Press ENTER. The LCD displays the new name, a LENGTH of 000004 and the prompt **SAVE? (Y/N).**
18. Press YES. When SAVING is completed, the prompt **Continue? (Y/N)** appears. To continue means to SAVE more waveforms. Instead, you want to see what happened to the waveforms you just SAVED.

19. Press NO.
20. To clear the RAM, turn the power OFF and back ON.
21. Enter EDIT SAMPLE mode.
22. Select **F1 SELECT SAMPLE** (press 1 on the numpad).
23. Move the CURSOR under **DISK**.
24. Press ENTER.
25. Use the DATA ENTRY A to display **SOUND: a) 01-B1**. Note that you SAVED this data as a WAVEFORM (**F3 SAVE WAVEFORM**). Under steps 2 and 3 above you selected it as a SAMPLE (**F1 SELECT SAMPLE**), and now, in the current display, it is listed as a SOUND (**F1 SELECT SOUND**).

26. Press ENTER. The LCD asks for confirmation **Get a) 01-B1? (Y/N)** and shows its LENGTH as 000512 and its SAMPLING FREQUENCY as 32kHz. (32kHz is the default SAMPLING FREQ in CREATE WAVEFORM mode).

27. Press YES. When loading is completed, the LCD displays the KEYBOARD ASSIGNMENT as **ORG=C3 TOP=F3**, with **TR**anspose ON.When you play the keyboard from F3 down, you hear only a click.

28. Press ENTER. Now the main menu of EDIT SAMPLE mode appears.
29. Select **F2 AUTO REPEAT ON/OFF** (press 2 on the numpad). The AUTO REPEAT is shown to be OFF.
30. Tap the UP ARROW tab A to set the OFF to **ON**.
31. Play a key from F3 down - now the sound sustains for as long as you hold the key down.

In this case, AUTO REPEAT ON works like the LOOP ON in MULTI SOUND mode. You know that this waveform is only one wavecycle (512 words) long - and the size of the memory block that it occupies is also 512 words long. So there is no air or silence before or after the waveform - by just repeating the entire memory block you get a perfect loop. But there's no way to SAVE this waveform with the loop - only MULTI SOUND F5 makes a loop permanent.

But what about the pitch of the waveform? Play the ORG KEY of C3 - you hear the pitch of B, and my chromatic tuner tells me that it is even a little flat. Keep this in mind - when you build a MULTI SOUND you'll have to

a) use MULTI SOUND mode **F3 ORIGINAL/TOP KEY** to assign drawn waveforms to ORG KEY=B (whatever octave you want them to be in), and

b) use **F2 REL.PARAMETERS (TUNE/LEV/Fc)** to check and correct any tuning discrepancies.

32. Select **F1 SELECT SAMPLE** (press 1 on the numpad).

33. Move the CURSOR under **DISK**
34. Press ENTER
35. Use the DATA ENTRY A to display **SOUND: 1) 08-B8**
36. Press ENTER. The LCD asks for confirmation to **Get a) 08-B8? (Y/N)** and shows its length as 000004 and its SAMPLING FREQUENCY as 32kHz. (32kHz is the default SAMPLING FREQUENCY in CREATE WAVEFORM mode).

37. Press YES. WARNING: TURN THE VOLUME WAY DOWN - NOW!
38. When loading is completed, you'll see the default KEY ASSIGNMENT of **ORG =C3 TOP= F3** with **TR**anspose ON. Play the keyboard from F3 down - you hear the "looped" high-pitched waveform, and C3 actually plays the pitch of B8 Why is it "looped"? Because the AUTO REPEAT is still ON.

CONCLUSION: When you draw a waveform, you obtain a MULTI SOUND made up of 8 transposing versions of the basic default waveform which is always at the pitch of B1.

If you SAVE a waveform in CREATE WAVEFORM mode (**F3 SAVE WAVEFORM**), you are asked to select from the eight transposing versions - you'll SAVE one at a time, or only those in the pitch areas that you need. Once SAVED, these versions of the drawn waveform will have a life of their own, stored on DISK as SOUNDS. This breaks up the automatically constructed MULTI SOUND. But straight after SAVING the 8 SOUNDS individually you can - while they're still in RAM - enter MULTI SOUND mode and SAVE them all together as a MULTI SOUND.

Later you can always call up all or some of the individual transpositions (now called SOUNDS) when you build a MULTI SOUND (MULTI SOUND mode **F0 GET SOUNDS**). You can then assign them to the keyboard in any order you like.

Remember although all functions in CREATE WAVEFORM mode deal with the name WAVEFORM, the 8 transposed versions of your waveform will be referred to as SOUNDS after you've SAVED them to DISK. The exception is in the EDIT SAMPLE mode **F1 SELECT SAMPLE**, where you can select your waveform under the name "SAMPLE".

OPTION b)
EDIT THE NEWLY-DRAWN WAVEFORM WITH F2 HARMONIC SYNTHESIS

This could be an interesting option, but I don't think it has been implemented yet. The DSS-1 lets you select **F2 HARMONIC SYNTHESIS** while a newly drawn waveform is still in RAM. When you're asked to select a waveform, you can select the "current" waveform and the DSS-1 then calculates and displays what you would hope to be the harmonic spectrum of your newly-drawn waveform. But if you step through the harmonics and look at their amplitudes, you'll see that the DSS-1 seems to have changed the waveform so drastically while "calculating" that the result can't be trusted.

Try it with a newly drawn waveform that you don't mind losing. You'll find that very high harmonics seem to have unrealistic amplitude levels. Even before analyzing the individual harmonics, your ears will tell you that your newly-drawn waveform is no longer the same.

I'll discuss the rest of HARMONIC SYNTHESIS (the part that works just fine) in Chapter 38.

OPTION c)
EDIT AND SAVE THE NEWLY-DRAWN WAVEFORM IN EDIT SAMPLE MODE

While you have a newly-drawn waveform still in RAM, you can enter EDIT SAMPLE mode and use all the functions available in this mode. You practiced all these moves in Chapters 28-34. Let's save time by just talking about what happens here with the newly drawn waveform, because you need to be aware of an annoying quirk of the DSS-1. If you wish, draw a waveform as described above, then step through the following suggestions. I'm not listing them in the usual way, since there is nothing really new for you to do.

First you need to enter EDIT SAMPLE mode, select **F1 SELECT SAMPLE**, and press ENTER with the CURSOR under **from MEMORY**. At this stage, the newly-drawn waveform is shown as MULTI SOUND #01 **!NO-NAME**, with a LENGTH of 1020. This is the result of the added LENGTHS of the 8 SOUNDS within this automatically created MULTI SOUND: 512 + 256 + 128 + 64 + 32 + 16 + 8 + 4 = 1020.

After confirming this MULTI SOUND by pressing ENTER, you can only edit one of the 8 SOUNDS from within this MULTI SOUND. You need to choose one of the 8 SOUNDS, by displaying it with the DATA ENTRY A and by pressing ENTER, YES and ENTER.

This (mandatory) isolation of 1 sound makes it impossible to come back to any of the other 7 SOUNDS from within this newly drawn waveform. You have, in fact, broken up the MULTI SOUND and forfeited access (in any mode) to the otter 7 SOUNDS. So if you didn't save the other 7 SOUNDS individually in the CREATE WAVEFORM MODE (**F3 SAVE WAVEFORM**), or the entire group of 8 as a MULTI SOUND in the MULTI SOUND MODE (**F9 SAVE/RENAME M.SOUND**), you'll have lost all but one of the 8 SOUNDS from your newly drawn WAVEFORM.

It's a strange thing: the other 7 SOUNDS are still in RAM, but the DSS-1 "loses" the directory as soon as you choose 1 SOUND. So it never lets you scroll through the 8 SOUNDS like it did the first time you selected **F1 SELECT SAMPLE**. Al subsequent selections of **F1 SELECT SAMPLE** or MULTI SOUND mode **F1 SELECT MULTI SOUND/SOUND** will result in the display of MULTI SOUND 01 SOUND 01, which will always be stuck on the SOUND you first selected (even if that wasn't SOUND 01 at the time).

You may remember the trouble we had in Chapter 16 when all we wanted to do was isolating the BRASS and HARP SOUNDS from within their MULTI SOUNDS. We had to keep re-loading the whole MULTI SOUND from DISK, then we had to isolate one SOUND at the time. Well, what got us out of trouble then was the fact that the MULTI SOUNDS were on DISK.

CONCLUSION: Before taking a newly-drawn waveform to EDIT SAMPLE mode, enter the MULTI SOUND mode, select **F9 SAVE/RENAME M.SOUND**, give the MULTI SOUND a name and SAVE it (Option f).

THEN: enter the EDIT SAMPLE mode, select the MULTI SOUND **from MEMORY**, isolate the first of the 8 SOUNDS, edit, name, and SAVE it. When you want to go on to the next of the 8 SOUNDS, select the whole MULTI SOUND from DISK with SYSTEM mode **F9 GET MULTI SOUND**. Then enter the EDIT SAMPLE mode, select the same MULTI SOUND again **from MEMORY** with **F1 GET SAMPLE**, isolate the desired SOUND from within this MULTI SOUND, edit the SOUND, name, and SAVE it. You can do this 7 times until all SOUNDS have been edited, named, and SAVED individually.

Whenever you want to re-build the MULTI SOUND the way it had been automatically constructed when you first drew the waveform, just load it up from the DISK. And if you need only the edited parts of it, then load them up as SOUNDS under their own names.

OPTION d)
EDITING AND SAVING A NEWLY-DRAWN WAVEFORM AS A MULTI SOUND

As soon as you've drawn a waveform, whether or not you've answered YES to the prompt **EDIT WAVEFORM? (Y/N)**, you can enter MULTI SOUND mode and use its functions to create a musically useful MULTI SOUND for future inclusion in a SYSTEM. Section 3 explained this mode in more detail. Here's what you'll do, in general.

After entering MULTI SOUND mode, ignore **F0 GET SOUNDS** – your SOUNDS are already in RAM. Go straight to **F1 SELECT M.SOUND/SOUND** and select the first of the 8 SOUNDS from within this MULTI SOUND. Because the default pitch of the drawn waveform is always B1, you'll need to assign the 8 SOUNDS to keys of B in whatever octaves you want them to play. You'll also need to check the tuning. The loop should, of course, stay ON, and you won't need to deal with **F6 LOOP START & LENGTH** - you already have ideal loops.

When you're done editing your new MULTI SOUND, use **F9 SAVE/RENAME M.SOUND** to SAVE it.

OPTION e)
TAKING A NEWLY-DRAWN WAVEFORM STRAIGHT TO PROGRAM PARAMETER MODE FOR IMMEDIATE INCLUSION IN A NEW SYSTEM

In General

As soon as you've drawn a waveform, whether or not you've answered YES to the prompt **EDIT WAVEFORM (Y/N)**, you can enter MULTI SOUND mode just to give this waveform a name. Then enter PROGRAM PARAMETER mode, write the desired parameters, and finally enter SYSTEM mode to SAVE the current Wave RAM (which, at this point, contains only your new MULTI SOUND) as a SYSTEM.

In More Detail

In MULTI SOUND mode, with **F9 SAVE/RENAME MULTI SOUND**, write a name, then you press NO to the prompt **SAVE? (Y/N)** and YES to the prompt **ABORT? (Y/N)**. If you wish to adjust other parameters while in MULTI SOUND mode, by all means do so. But instead of SAVING, you go on to the PROGRAM PARAMETER mode.

In PROGRAM PARAMETER mode, use all desired functions to write a PROGRAM, then give the PROGRAM a name under **F01 WRITE/RENAME PROGRAM**.

Then enter the SYSTEM mode and SAVE the SYSTEM (A or B or C or D). This writes the newly-drawn and edited MULTI SOUND plus its associated PROGRAM PARAMETERS to the DISK, as part of a SYSTEM. The 31 other PROGRAMS in this SYSTEM will play the same MULTI SOUND - it is so far the **only** MULTI SOUND in this SYSTEM.

Later, if you want to include other MULTI SOUNDS in this SYSTEM, you can add them as explained in Chapter 17.

Whenever you need to isolate parts of this MULTI SOUND (your drawn waveform), use SYSTEM mode **F9 GET MULTI SOUND**, then enter EDIT SAMPLE mode where you can select the MULTI SOUND **from Memory** and access the individual SOUNDS from within the MULTI SOUND.

OPTION f)
SAVING A NEWLY-DRAWN WAVEFORM AS A COMPLETE, UN-EDITED MULTI SOUND FOR ALL KINDS OF FUTURE USES

Option f) is the only Option that preserves your newly-drawn waveform unchanged, as a source for any kind of future editing.

After drawing a waveform, whether or not you reply YES to the prompt **EDIT WAVEFORM? (Y/N)**, enter MULTI SOUND mode. Go straight to **F9 SAVE/RENAME MULTI SOUND** and give it a name. SAVE it on a library DISK with other (finished or unfinished) MULTI SOUNDS.

Whenever you want to refine it as a MULTI SOUND, load it into RAM with SYSTEM mode **F9 GET MULTI SOUND**. Enter MULTI SOUND mode, ignore **F0 GET SOUNDS** (your SOUNDS are already in RAM), select **F1 SELECT M.SOUND/SOUNDS**, pick your first SOUND and start working on it. Remember that all 8 SOUNDS within the MULTI SOUND are pitched to B, so assign them accordingly. After editing all 8 SOUNDS in MULTI SOUND mode, you can rename it, SAVE it or go straight to PROGRAM PARAMETER mode and eventually SAVE it with a SYSTEM, as in option e).

CONCLUSION: Your best bet is to use both options a) and f) anytime you've drawn a waveform that you want to keep. Option a) stores the 8 transposed versions separately, and option f) stores the transposed versions as a MULTI SOUND. The trick is to SAVE the 8 transposed versions in CREATE WAVEFORM mode individually, and - while they're still in RAM - go to MULTI SOUND mode and SAVE them as a MULTI SOUND.

CHAPTER 38
HARMONIC SYNTHESIS

This rather unique feature of the DSS-1 lets you do two things:

a) You can choose from 5 pre-packaged waveforms, similar to the choices available from the oscillators on many synthesizers. But here you can analyze and alter the harmonic spectrum of up to 128 harmonics, one harmonic at a time.

b) You can build your own waveform from a "blank" by setting the amplitude of up to 128 harmonics, one harmonic at a time.

DATA ENTRY A steps through the 128 harmonics, DATA ENTRY B adjusts the amplitude levels. Minimum level is zero, maximum is 255.

The DSS-1 automatically assigns the waveform across the keyboard, as a MULTI SOUND consisting of 8 SOUNDS, with the LOOP ON. Check out Chapter 37 to see how this all happens.

You can SAVE the 8 SOUNDS individually, after which they'll be stored on DISK as single wavecycles. When you later select them from DISK in EDIT SAMPLE mode, they'll all come up at the default KEY ASSIGNMENT of C3. When later selected from DISK in MULTI SOUND mode, you can assign them anywhere on the keyboard, to re-group them as the original MULTI SOUND, or to incorporate them in a new MULTI SOUND.

While still in RAM, the 8 SOUNDS, with their original KEY ASSIGNMENTS, can also be SAVED as a complete MULTI SOUND.

Here's how you work with HARMONIC SYNTHESIS.

a) Selecting and Editing Preset Waveforms

After selecting **F2 HARMONIC SYNTHESIS**, you can step through the selections with the DATA ENTRY A. There can only be ONE selected and calculated waveform in RAM at any one time. Selecting and calculating a new waveform erases the previous waveform in RAM.

1. CURRENT

The main purpose of "CURRENT" is to retrieve the waveform you're currently working on if your work is interrupted. For instance: you can select **F3 SAVE WAVEFORM** to SAVE intermittent results, then press NO to the prompt **Continue? (Y/N)** and re-select CURRENT to get back to your waveform. When you first enter the CREATE WAVEFORM mode, all the harmonics in CURRENT are set to zero. At this stage, CURRENT is identical to BLANK, and you can use either one to start building a waveform.

2. BLANK

Just what it says - blank. All 128 harmonics have an initial level of zero. Once you've initiated the first calculation by setting at least one level value and pressing ENTER, you'll have to switch to CURRENT to keep working on your waveform if you select **F3 SAVE WAVEFORM** to SAVE intermittent results.

3. SAW

This is the classic SAWTOOTH wave, a.k.a. RAMP wave. In Chapter 35, I spelled out the harmonics of this waveform: the (odd and even) numbers. The relative amplitude of the harmonics is easy to figure out in this waveform. Just make the harmonic number into a fraction by placing it under a "1/", and you'll know how loud the harmonic is compared to its fundamental. For instance, the 4th harmonic is 1/4 as loud as its fundamental. Step through the SAW and compare the values of the harmonics to the value of the fundamental.

The SAWTOOTH wave is the basis for synthesizing virtually all BRASS and a lot of STRING sounds. If it sounds a bit rude and buzzy at first, realize that this will be easy to fix with the help of filters in the MULTI SOUND and PROGRAM PARAMETER modes.

4. SQUARE

This is the classic waveform for synthesizing reed instruments, particularly clarinet sounds, and many other useful sounds. As you saw in Chapter 35, this waveform only contains the odd numbered harmonics (no octave duplications). This, and the amplitude relationships between the harmonics and the fundamental, give the SQUARE WAVE its typical woody, slightly hollow sound. If you want to synthesize another typical classic waveform, the TRIANGLE wave, then you can start with the SQUARE and edit it as follows: Leave the even-numbered harmonics at zero level, and make the 3rd harmonic l/9th as loud as the fundamental, then the 5th harmonic l/25th as loud as the fundamental, the 7th harmonic l/49th as loud as the fundamental, the 9th harmonic 1/81st as loud as the fundamental, the 11th harmonic l/121st as loud as the fundamental. After that just set them all to zero - the higher harmonics are so soft that you needn't bother with them.

5. METAL

This is an attempt at synthesizing a waveform with a lot of non-harmonic overtones. Since, in HARMONIC mode, the DSS-1 can only deal with whole-numbered harmonics, the Factory programmers picked some "off-the-wall' harmonics and amplitude settings. Step through the values - there seems to be no rhyme or reason, but the result is quite successful in being a little 'clangorous'. In nature, metallic sounds typically have a lot of overtones "in the cracks" between the frequencies of the whole-numbered harmonics. This is what gives bells and gongs and other metallic instrument their typical sound. The more non-harmonic overtones with healthy amplitude levels that a sound has, the more non-pitched the sound becomes. Interaction between the non-harmonic overtones create new frequencies through ring modulation. This can make the difference between a gong (non-pitched) and a tubular bell (pitched).

6. CLAV

A kind of nasal narrow pulse wave, useful for all kinds of sounds when filtered and otherwise treated in PROGRAM PARAMETER mode. Notice the fairly high amplitude levels of harmonic #3 and its multiples.

7. ORGAN

Harmonics 1-4, 6, 8, 10, 12, 16 are all at maximum levels. All others are at zero. This creates a full and bright sound reminiscent of an "all-stops-out" organ, or a Hammond with the white drawbars fully drawn.

b) Building Your Own Waveform

The BLANK preset invites your creativity. If you or your neighborhood library have some books on acoustics and waveforms, read up on the subject. You can punch in your own values for harmonics and their levels, and hear what comes up. And don't forget - once you've SAVED individual waveforms as SOUNDS, you can always mix and link them with others in EDIT SAMPLE mode to create new monsters.

The only shortcoming of this process is the fact that you don't have control over the envelopes of the individual harmonics. But such a feature would, by far, exceed the capabilities of an instrument of the size and price of your DSS-1. And with skillful programming of the VCF in PROGRAM PARAMETER mode you can simulate a lot of these timings of harmonics - welcome to good old 'analog subtractive' synthesis. But more about that in the next Section.

There appears to be a third feature in HARMONIC SYNTHESIS: one which allows you to analyze a drawn waveform to obtain a readout of its harmonic spectrum, which you can then adjust. However, as I wrote in the previous Chapter - I don't think this really works (yet?).

The idea is that the DSS-1 calculates and displays the harmonic spectrum of a newly-drawn waveform. The mechanics work, but the results seem wrong. Try it out for yourself.

1. Enter CREATE WAVEFORM mode.
2. Select **F1 DRAW WAVEFORM** (press 1 on the numpad).
3. Watch the LCD and press ENTER.
4. Press YES. Be ready to move the DATA SLIDER A, then
5. Press ENTER and move the DATA SLIDER A during the 8 second before the LCD changes. As soon as the LCD changes to **WAVEFORM DRAWN, Retry? (Y/N)**, test the new waveform.

6. Play the keyboard and memorize the sound color of your newly drawn waveform.
7. Press NO to the prompt **Retry? (Y/N)**
8. Press NO to the prompt **Edit Waveform? (Y/N)**
9. Select **F2 HARMONIC SYNTHESIS** (press 2 on the numpad)
10. Ignore the blinking ENTER tab for now. Use the DATA ENTRY A to step through the available options: Current, Blank, Saw, Square, Metal, Clav, Organ.

11. Use the DATA ENTRY A to select Current.

The first choice in **F2 HARMONIC SYNTHESIS** is always CURRENT. This stands for the current waveform in RAM. In our case this is the waveform that you've just drawn. To transfer this waveform into the display of harmonics:

12. Press ENTER

The LCD first displays **Calculating...**, giving you the impression that some serious work is being done. The LCD then displays the amplitude level of the first harmonic of your newly-drawn/newly-"analyzed" waveform. Before you even go to the trouble of stepping through the harmonics, play the keyboard. What you hear bears little resemblance to your original waveform.

I find that this process alters the drawn waveform beyond recognition. I don't think that the values for the harmonics and their levels can be trusted - maybe we're not even meant to assume that the DSS-1 can actually do this function. So - on with the show.

F3 SAVE WAVEFORM

This is a review of this function - I explained it in the application examples in the previous Chapters.

The CREATE WAVEFORM mode builds looped MULTI SOUNDS based on a single wavecycle at the pitch of B1 with 7 transpositions. The total LENGTH of the MULTI SOUND is 1020 words.

The individual LENGTHS are:

SOUND 01: 512
SOUND 02: 256
SOUND 03: 128
SOUND 04: 64
SOUND 05: 32
SOUND 06: 16
SOUND 07: 08
SOUND 08: 04

When SAVING with **F3 SAVE WAVEFORM**, you have to break up this MULTI SOUND by naming and SAVING the 8 waveforms one at a time. They will then be individually stored on DISK and listed as SOUNDS, all with the default KEY ASSIGNMENT of C3.

After SAVING the 8 waveforms one at a time, they are still in RAM as the automatically constructed MULTI SOUND. It is a good idea to enter MULTI SOUND mode and, with **F9 SAVE/RENAME M.SOUND**, SAVE the entire looped MULTI SOUND with the current KEY ASSIGNMENTS for future reference.

SECTION 6
PROGRAM PARAMETER MODE

The many functions in this mode add up to a sophisticated SYNTHESIZER. Although it is fully digital in its operation, you need to be aware of the concepts behind the programming technique that goes back to the days when all synthesizers were analog: SUBTRACTIVE FILTER SYNTHESIS. Without a thorough knowledge of these concepts you'd be lost and you won't get your money's worth from the DSS-1. As you've seen in previous Chapters, the MULTI SOUNDS that contain your SAMPLES and WAVEFORMS are only as good as the PROGRAMS that let you hear them.

Chapter 39
SUBTRACTIVE FILTER SYNTHESIS AN OVERVIEW

Do you realize that any sound has only THREE BASIC QUALITIES? I call them the BASIC THREE:

1. PITCH   2. COLOR   3. LOUDNESS

That's it - no matter how many knobs, sliders and parameters a synthesizer has, you're only dealing with the BASIC THREE. This makes the programmer's life easy. Whatever you do to a sound, ask yourself: which of the BASIC THREE am I influencing?

Think of it this way: What are the BASIC THREE that are involved in driving a car? Acceleration, Deceleration and Changing Direction. That's all there is - never mind what happens at the Drive In.

But in today's cars, we have many features that help with the BASIC THREE: all kinds of gear / transmission configurations, power options, steering wheels that come in many shapes and sizes, etc. These features change from make/model to make/model. But this doesn't stop us from getting from A to B in a borrowed car, because we all have learned the basics and can adapt them to different cars.

The same is true for synthesizers. Every make and model has different features with different names, but there are only so many ways of changing the BASIC THREE: PITCH, COLOR, and LOUDNESS. Let's look at the BASIC THREE in more detail.

1. PITCH - THE FIRST OF THE BASIC THREE

On synthesizers, we rely on OSCILLATORS to give us our basic SOUND. On samplers like the DSS-1, we get that sound in the form of reproduced digital SAMPLES or as "created" waveforms. The DSS-1 has two OSCILLATORS that each can produce a separate "sound" in a PROGRAM. For the sake of organization, that sound must be what KORG calls a MULTI SOUND.

The OSCILLATORS give us the basic SOUND.
The KEYBOARD controls the PITCH of the OSCILLATORS' output.

But the keyboard is not the only pitch controller. The JOYSTICK and the AUTOBEND can "bend" the PITCH of the OSCILLATORS' output. So can the MODULATION GENERATOR, which is the name KORG chose for the LFO (Low Frequency Oscillator) as it is called on most other synthesizers. And if you play the DSS-1 via MIDI, you have all kinds of choices: sequencers, remote MIDI keyboards, MIDI drum pads, MIDI guitars, etc. send PITCH control data to the OSCILLATORS. So, for BASIC SOUND selection we use the OSCILLATORS. And for changes in PITCH we also go to the OSCILLATORS.

2. COLOR - THE SECOND OF THE BASIC THREE

Color (also called "timbre") is a result of the harmonic spectrum of a sound: what overtones are present, and how loud they are during the time (duration) of every note. In Section 5 CREATING WAVEFORMS I deal with the basic "textbook" waveforms.

When you prepare a PROGRAM in a SYSTEM, you decide the sound color in three stages.

a) The first decision is made when you select a MULTI SOUND for each OSCILLATOR. That MULTI SOUND has a certain basic COLOR that you then modify.

b) Once you've chosen a combination of MULTI SOUNDS for a PROGRAM, you can modify that basic COLOR with the help of the FILTER. The FILTER is really the heart of most synthesizers. On the DSS-1, you have a VCF, which stands for Voltage Controlled Filter. It acts like an LPF, which stands for Low Pass Filter: when you leave the VCF wide open (Cutoff Point at 127), then the MULTI SOUNDS pass through the FILTER without any alteration to the harmonic spectrum.

As soon as the VCF is partially closed (Cutoff Point lower than 127), the MULTI SOUNDS' overtones above the Cutoff Point are being faded out - the uppermost overtones are completely eliminated. The "highs" are filtered out, the "lows" pass through. The filtered MULTI SOUND has less brightness than it's original, unfiltered version.

The lower you set the Cutoff Point, the fewer overtones you hear, and the sound gets duller or "darker". When the Cutoff Point is near zero, the VCF "kills" the waveform, because not even the lowest overtones nor the Fundamental pass through. The result is silence.

c) So far, you could compare the action of the VCF to that of the tone controls on an amplifier, except that most tone controls don't change the COLOR quite as drastically as the VCF does. But this is where the similarity ends: when using tone controls, you mostly set-them-and-forget-them. You don't normally adjust tone controls during every musical note that gets played. But with filters on synthesizers, you want the Cutoff Point to change during every note played. Most sounds in nature undergo constant changes in brightness. Some of these changes happen so fast that we couldn't hope to move knobs and sliders fast and accurately enough to imitate these changes manually. You need electronically programmable help. For this you use an MG (which stands for Modulation Generator), or an EG (which stands for Envelope Generator).

The MG shifts the Cutoff Point in an ongoing, symmetrical, up-and-down fashion. This is known as "wah-wah" - say it and you'll know what I mean.

The EG also shifts the Cutoff Point during every note you play. But it doesn't do this in a symmetrical way. Instead, you program separate values for the different movements of the Cutoff Point: how fast the Cutoff Point gets shifted, and from what level to what level. This means that you have more control over the changes in COLOR when you use the EG. In addition to the "how fast" and "where-to-where" settings, you also set the overall EG INTENSITY: to what maximum value the EG moves the Cutoff Point from your set-it-and-forget-it level.

3. LOUDNESS - THE THIRD OF THE BASIC THREE

This sounds "kinda dumb": before and after every note there is silence.

So what dictates the level of loudness during every note ?

The answer is the VCA, which stands for Voltage Controlled Amplifier. Don't confuse this with the amplifier that the DSS-1's audio outputs get hooked up to. The VCA is an internal module of the DSS-1. You program a certain VCA level, which sets a limit on the loudness of the MULTI SOUND after passing through the VCF. So, it acts as an internal volume control.

Again, most sounds change loudness during every note played. Some swell up slowly, others fade out after an initial attack peak, etc.

There's no way that you could play the instrument and shape the loudness of every note with one hand on the MASTER VOLUME: you'd be too slow and inaccurate. So, you need electronically programmable help. For this, you use an EG (Envelope Generator) that is exclusively assigned to the VCA. The VCA EG changes the loudness of every note from zero (silence before the note gets played) to a programmable attack peak. If you're still sustaining the note, it can then further change the loudness while you go on sustaining the note.

Whenever you release the key, the VCA returns the loudness level to zero. You program exact RATES (how fast these changes occur) and LEVELS (how loud and soft the sound gets during the notes played).

The Order of Events in the BASIC THREE

The OSCILLATORS send a WAVEFORM, at a certain PITCH, to the FILTER (VCF), where the COLOR can be changed by filtering out OVERTONES. What's left of the WAVEFORM goes to the AMPLIFIER (VCA), where the LOUDNESS is regulated.

Seems easy enough, doesn't it?

What's so great on the DSS-1 is that there is much more control possible than what I listed above. Nothing needs to stay the same for long in terms of the BASIC THREE. KEYBOARD VELOCITY (how fast or hard you strike the keys) can affect the BASIC THREE constantly, which gives you a chance to change the programmed settings as you perform.

And once you've struck a key and "bottomed out", you have AFTERTOUCH available to further change the BASIC THREE (depending on how much pressure you put on the held-down keys). And before the sound leaves the DSS-1 on the way to your sound system, it passes two DIGITAL DELAYS that you can program for all kinds of echo and chorus effects in stereo.

Keep these underlying principles in mind as we step through all PROGRAM PARAMETERS on your DSS-1.

Chapter 40
F00-F21 PROGRAM BASICS & OSCILLATORS/NOISE

F00 INITIALIZE PARAMS

After you select this PARAMETER and press ENTER, the values of all the PARAMETERS arc set to the default values. Take the time to write these down for future reference.

This allows you to make up a PROGRAM without having to alter an existing PROGRAM with which you may not be familiar. Knowing all the values for all the PARAMETERS ahead of time will save you time, since a glance at your table is quicker than selecting PARAMETERS on the DSS-1 and reading (and trying to memorize) existing values.

F01 WRITE/RENAME

After selecting this PARAMETER, you get the LCD message **Rename? (Y/N)** - If you press YES, you then use the CURSOR and the DATA ENTRY B to write a new name for the current PROGRAM. When you've "written" the complete name, press ENTER. This takes you to the display that you would have gotten immediately if you had pressed NO in response to **Rename? (Y/N)**

This display **WRITE No. = ??, Select No. & ENTER** prompts you to enter the PROGRAM number you want the current PROGRAM stored as. Use the DATA ENTRY B to select a number 01 to 32, then ENTER.

Now you need to confirm whether you want this PROGRAM to be put into RAM. **Write in Mem? (Y/N)**

If you press NO, then you get **PGM Not Written, Continue? (Y/N)**
If you press NO, you get the general screen of the PROGRAM PARAMETER mode: **Select (00-96)**
If you press YES, you get back to the display **Rename? (Y/N)**

If you press YES in response to the previous message **Write in MEM? (Y/N)** then you get the display **Continue? (Y/N).** To "Continue" means renaming the current PROGRAM again: press YES and you get back to the initial message **Rename? (Y/N).** Press NO in response to **Continue? (Y/N)** and you get the general screen of the PROGRAM PARAMETER mode: **Select (00-96)**

The "renaming" and the assignment of a PROGRAM number are self explanatory. But maybe you're wondering what "writing in Mem" means? Simply this: the DSS-1 always assumes that you're about to build a SYSTEM. If you're indeed planning on SAVING the current RAM as a SYSTEM, or at least SAVING the PROGRAM data to a SYSTEM, then you should "write in Mem" - press YES.
See Section 2 for more details on manipulating PROGRAMS.

F11 OSC OCT

Each OSCILLATOR can produce its MULTI SOUND at one of three octaves. The lower setting is 16', the middle at 8', and the higher is at 4'. Use the CURSOR and the DATA ENTRY B to make changes. MULTI SOUNDS from the two OSCILLATORS can either be in unison, or one or two octaves apart.

You'll often use this PARAMETER in conjunction with **F15 OSC2 DETUNE & INTERVAL**.

F12 OSC1 M.SOUND

The LCD shows the name and length of the MULTI SOUND currently assigned to OSCILLATOR 1. To change this, use the DATA ENTRY B to select from the MULTI SOUNDS currently in RAM. When the desired MULTI SOUND appears in the LCD, press ENTER to complete the change of assignment.

Whether you can hear the newly assigned MULTI SOUND depends on

a) the values in **F14 MIX RATIO**
b) the value in **F32 VCF CUTOFF** and
c) the value in **F36 VCA TOTAL LEVEL**

F13 OSC2 M.SOUND

The LCD shows the name and length of the MULTI SOUND currently assigned to OSCILLATOR 2. To change this, use the DATA ENTRY B to select from the MULTI SOUNDS currently in RAM. When the desired MULTI SOUND appears in the LCD, press ENTER to complete the change of assignment.

Whether you can hear the newly assigned MULTI SOUND depends on

a) the values in **F14 MIX RATIO**
b) the value in **F32 VCF CUTOFF** and
c) the value in **F36 VCA TOTAL LEVEL**

F14 MIX RATIO

Using the DATA ENTRY B you can change the mix between the two OSCILLATORS, measured as a percentage ratio that always adds up to 100%. Notice that the default value is 100% for OSCILLATOR 1, which leaves nothing for OSCILLATOR 2. Adjust the mix before you can expect to hear any sound from OSCILLATOR 2 when constructing a PROGRAM from scratch. The MIX RATIO can be controlled in real time with the amount of KEYBOARD VELOCITY (how fast or hard you strike a note). For this you have to program **F41 VELOCITY SWITCH**.

F15 OSC2 DETUNE & INTERVAL

Here you can raise the tuning of OSCILLATOR 2 compared to the tuning of OSCILLATOR 1. Use the CURSOR and the DATA ENTRY B to change the values. With DETUNE, you can raise the pitch of OSCILLATOR 2 in 63 very small increments (l/64th of a halfstep each), up to a maximum of roughly a halfstep. With INTERVAL, you can raise the pitch of OSCILLATOR 2 in exact halfsteps, up to a value of 11 (an interval of a major seventh).

You can combine DETUNE and INTERVAL values to set any amount of "out-of-tune" in one octave.

Combine the settings from **F15 OSC2 DETUNE & INTERVAL** with those available from **F11 OSC OCT**, and you can go beyond the range of one octave.

For ensemble sounds ("section" sound as opposed to "solo" sound), tune the two OSCILLATORS to the identical octave (both 16', 8', or 4'), then set a small amount of DETUNE. Adjust the MIX RATIO. For "fanfare" brass in constant fifths, tune the two OSCILLATORS to the identical octave (both 16', 8', or 4') and set OSCILLATOR 2 to an INTERVAL of 07. Adjust the MIX RATIO. This works equally well for the headbanger's special: Low open fifths for heavy guitar drones.

To simulate one of the "percussion" stops on Hammond organs, assign an organ-like MULTI SOUND to OSCILLATOR 1 and a bell-like MULTI SOUND to OSCILLATOR 2. Then set OSCILLATOR 2 to play at least one "footage" octave higher than OSCILLATOR 1 (see **F11 OSC OCT**), and add an INTERVAL setting of your choice in **F15 OSC 2 DETUNE AND INTERVAL**. Again, adjust the MIX RATIO. This PARAMETER can also be effective when combined with AUTOBEND (see below).

F16 SYNC MODE, D/A RESOLUTION

Use the CURSOR and the DATA ENTRY B to change the values.

SYNC OFF - lets you hear each OSCILLATOR'S MULTI SOUND independently.
SYNC ON - forces the PITCH of OSCILLATOR 2 to lock onto OSCILLATOR 1.

Example: First you set an amount of AUTOBEND for OSCILLATOR 2 that is tuned differently from OSCILLATOR 1. While SYNC is ON, you'll hear OSCILLATOR 2 "fight" against OSCILLATOR 1. The result is often a shift in overtones, similar to a RESONANCE sweep in the VCF, but with a timing of its own. At other times, you get a ring modulator effect, particularly when INTERVAL and AUTOBEND are used for OSCILLATOR 2. The results depend on a lot of factors; this is one for the explorers - go for it.

D/A RESOLUTION stands for Digital-to-Anaiog Resolution. The DSS-1 works mostly with 12-bit resolution, except for "drawn" and "created" waveforms where the resolution is 9-bit. Basically, the more bits are used, the finer the details of the computed waveforms, as long as the dynamic range is the same. The choices in the LCD are 12-bit, 10-bit, 8-bit, 7-bit, and 6-bit. Why would you want to sacrifice the clean 12-bit sound for a lower resolution? There may be times, particularly if SYNC in ON, when a low resolution can add a welcome "dirty" element to a sound that needs a gritty quality. Again, explore.

F17 OSC MG MOD

This stands for OSCILLATOR MODULATION GENERATOR MODULATION. Yes, this is a mouthful. Let me explain.

KORG use the tern MODULATION GENERATOR (MG) for the Low frequency Oscillator (LFO) that you may know from other equipment. "Modulation" simply means change. And since the OSCILLATORS deal with the first of the BASIC THREE, PITCH (see Chapter 35), the MG can change ("modulate") the OSCILLATOR'S PITCH during every note played. It does so in an ongoing, symmetrical up-and-down manner. When used in moderation, this musical effect is called VIBRATO. When used in exaggeration, all kinds of sound effects can occur (sirens, r-r-r, etc.).

Note that the MG that affects the PITCH is independent from the MG described in Chapter 41 (**F34 VCF MG MOD**). PITCH and COLOR can each be modulated at their own frequency and intensity.

Use the CURSOR and the DATA ENTRY B to make changes to the following values in the LCD:

The top line of the LCD lets you decide which OSCILLATORS are having their PITCH modulated. The choices are BOTH, or only OSCILLATOR 1, or only OSCILLATOR 2.

The first PARAMETER on the bottom line of the LCD is the FREQUENCY (FRQ) of the MG. This sets the SPEED of the PITCH modulation. At **FRQ=00** (the slowest setting) it takes 2 seconds for one up-down cycle of PITCH modulation. At the maximum setting of 31 the effect is motor-like, too fast for truly musical uses but great for effects.

The second PARAMETER on the bottom line of the LCD is the INTENSITY (INT) of the PITCH modulation during every note played. It determines the distance that the PITCH travels during every up-and-down cycle of the MG. At a maximum of 15, the distance seems to be a little over a full tone up and a full tone down from the pitch of the note played.

Note: Setting the value far INT to 00 effectively cancels all PITCH MODULATION, unless you control the INTENSITY in real time via AFTER TOUCH (see F51 AFTER TOUCH SENSITIVITY / OSC MG MOD INT).

The third PARAMETER on the bottom line of the LCD is the DELAY (DLY). At **DLY=00** the PITCH modulation comes into effect as soon as you play a note. At the maximum of **DLY=15** the PITCH modulation doesn't start until approx. 4 seconds after you play a note. If you hold down the first note and add new notes, the modulation of the new notes will not be delayed once the PITCH modulation starts.

F18 AUTO BEND MODE

This PARAMETER is a preparation for **F19 AUTO BEND TIME & INTERVAL** and, optionally, also for **F 41 VELOCITY SENSITIVITY: AUTO BEND INTERVAL.**

The AUTO BEND affects the PITCH of every note played. Instead of getting the OSCILLATOR's true pitch every time you play a key, you first get a lower or higher pitch that glides towards the true pitch of the key you're holding down. As soon as the true PITCH has been reached, the AUTO BEND has "done its job" and causes no more PITCH change until you play another note. AUTO BEND works with every new key that you play, whether you're still holding down previous keys or not.

Use the CURSOR and the DATA ENTRY B to make changes to the following values in the LCD:

Next to "MODE" you select the OSCILLATOR(s) that have their PITCH affected by AUTO BEND. The first three choices are: BOTH OSCILLATORS, or only OSCILLATOR 1, or only OSCILLATOR 2. The fourth choice is OFF which cancels AUTO BEND. This overrides all other values in **F18 AUTO BEND MODE**, **F19 AUTO BEND TIME & INTERVAL**, and **F41 VELOCITY SENSITIVITY: AUTO BEND INTERVAL.**

POL (POLARITY) stands for the direction of the AUTO BEND. It's either UP or DOWN.

F19 AUTO BEND TIME & INTERVAL

See **F18 AUTO BEND MODE** for the explanation of AUTO BEND. Here you select the values as described below. They will only have an effect if you selected one of the first three choices (not OFF) in **FI8 AUTO BEND MODE**.

INT stands for INTERVAL. It controls the distance that the PITCH travels dining AUTO BEND. The actual musical intervals can be made to further depend on **F41 VELOCITY SENSITIVITY: AUTO BEND INTERVAL.**

When **F41 VELOCITY SENSITIVITY: AUTO BEND INTERVAL** is set to 00, a setting of **INT=000** under **FI8 AUTO BEND TIME & INT** cancels all AUTO BEND. Settings of **INT=001-127** set the distance (interval) from which the PITCH of every note starts to glide towards the true PITCH of the key you're playing.

The maximum PITCH distance (interval) is a major seventh in the DOWN polarity and two octaves plus a minor sixth in the UP polarity. There are roughly 4 INT value increments to every semitone (halfstep), as shown below. I tested the examples by ear and with a chromatic tuner, which is not entirely scientific. But the examples give you an idea of how the INT values relate to musical intervals.

AUTO BEND POLARITY: DOWN. Here are some examples of INT settings and the actual musical INTERVALS while **F41 VELOCITY SENSITIVITY: AUTO BEND INTERVAL** is set to 00.

INT=007 = roughly a full step above true pitch
INT=012 = roughly a minor third above true pitch
INT=018 = roughly a major third above true pitch
INT=027 = roughly a perfect fifth above true pitch
INT=036 = roughly a major sixth above true pitch
INT=040 = roughly a minor seventh above true pitch
INT=055-127 = roughly a major seventh above true pitch

All values higher than roughly **INT=50-60** result in the same interval: a major seventh. But there is a difference: the higher the value, the longer the PITCH stays at the interval of a major seventh before starting the glide down towards the true PITCH of the key you're playing. So, higher values act like a delay for the AUTO BEND in the DOWN POLARITY.

Examples: At any speed (**TIME=any**) and at **INT=45**, there is no delay – the PITCH glides down immediately on key attack. At the slowest speed (**TIME=31**) and at **INT=60** the PITCH stays at the interval of a major seventh for about 10 seconds before it starts to glide down. At the slowest speed (**TIME=31**) and at the greatest interval (**INT=127**) it takes over 40 seconds for the PITCH to start gliding down. The INT and TIME values interact and make for many variable effects. Take time to test them out.

AUTO BEND POLARITY: UP. Here are some examples of INT settings and the actual musical INTERVALS while **F41 VELOCITY SENSITIVITY: AUTO BEND INTERVAL** is set to 00.

INT=007 = roughly a full step below true pitch
INT=012 = roughly a minor third below true pitch
INT=018 = roughly a major third below true pitch
INT=027 = roughly a perfect fifth below true pitch
INT=036 = roughly a major sixth below true pitch

INT=040 = roughly a minor seventh below true pitch
INT=048 = roughly one octave below true pitch
INT=055 = roughly one ninth below true pitch
INT=069 = roughly one eleventh below true pitch
INT=096 = roughly two octaves below true pitch
INT=127 = roughly two octaves plus a minor sixth below true pitch

There is no "hidden" delay feature in the UP polarity.

Whatever INTERVALS you program with the values of INT= ?? in either POLARITY, can be made to change from the keyboard while you play. **F41 VELOCITY SENSITIVITY: AUTO BEND INTERVAL** can be set higher than 00, so that the amount of force in the attack of your playing can alter the INTERVAL from which the PITCH starts out at the beginning of every note you play.

TIME is adjustable (use the DATA ENTRY B) from 00-31. The word TIME is misleading, because these values (00-31) don't represent any measurable time (not milliseconds or seconds). What you're adjusting is a RATE OF SPEED. The actual time it takes for the PITCH to reach the true PITCH of the key you're playing depends on two factors: The INTERVAL (INT=??) and the RATE OF SPEED (TTME=??). A setting of TIME=00 cancels all AUTO BEND.

Examples: At a constant setting of TIME=31, it takes quite a while for the AUTO BEND to reach the true PITCH of the key you're playing:

TIME=31 INT=10 : 5 seconds
TIME=31 INT=20 : 10 seconds
TIME=31 INT=30 : 15 seconds
TIME=31 INT=40 : 20 seconds

When analyzing or building a PROGRAM, keep in mind that there are three ways to cancel the AUTO BEND effect.

a) Cancel AUTO BEND with **F18 AUTO BEND MODE**: set **MODE** to **OFF**

b) Cancel AUTO BEND with **F19 AUTO BEND TIME & INT**: set **INT** to **000**. This cancels AUTO BEND only if **F41 VELOCITY SENSITIVITY: AUTO BEND INTERVAL** is set to 00.
c) Cancel AUTO BEND with **F19 AUTO BEND TIME & INT**: set **TIME** to **00**.

Just how you end up hearing the effect of AUTO BEND also depends a lot on **F16 SYNC MODE**. With SYNC set to ON, the waveform of OSCILLATOR 2 is locked onto OSCILLATOR 1. You hear a kind of filter sweep or temporary chorusing if only OSCILLATOR 2 is affected by AUTO BEND. You don't actually hear the PITCH glide. Instead, you hear a change in the waveform. Try it at different INT and TIME values, and with different MULTI SOUNDS assigned to the two OSCILLATORS. There are too many possible results for me to list here. Explore and have fun.

F21 NOISE LEVEL

WHITE NOISE is a sound containing all frequencies at the same amplitude. In other words, no single frequency stands out in a way that would lead the listener to hear it as a fundamental of a series of overtones. That's why you don't hear an identifiable PITCH when listening to WHITE NOISE.

Use the DATA ENTRY B to add WHITE NOISE to any MULTI SOUNDS - the minimum NOISE level is 00 (OFF), maximum is 63.

WHITE NOISE is a welcome addition to many kinds of sounds. You might want to add "breath" to a wind instrument, or create wind and surf effects. Simulate the distortion from a Marshall stack behind a guitar sample. Create electronic percussion. Etc.

If you want to create a PROGRAM that uses only NOISE, you have to be cunning. The DSS-1 wants a MULTI SOUND in each OSCILLATOR at all times, before you can add NOISE. Here's how you can "cheat" the DSS-1 into giving you NOISE only:

1. Enter CREATE WAVEFORM mode.
2. Select **F2 HARMONIC SYNTHESIS** (Press 2 on the numpad). The LCD asks you to select a waveform.
3. Use the DATA ENTRY A to display **BLANK**.
4. Press ENTER.

This creates a MULTI SOUND with zero amplitude, which is the same as 1020 words worth of clean, looped silence. This silent MULTI SOUND is ready to be included in any SYSTEM, as long as the SYSTEM has space for it (see Section 3). Name the MULTI SOUND in MULTI SOUND mode, and SAVE it with the current SYSTEM. You can now assign any amount of NOISE and use all the other PROGRAM PARAMETERS - all you'll hear is NOISE.

Chapter 41
F31 - F35 VCF - THE FILTER SECTION

The FILTER is the heart of any synthesizer. In Chapter 39 I outlined the function and importance of the filtering process: how you affect the COLOR of sound by filtering the overtones. Here well get specific.

F31 VCF MODE & EG POLARITY

Use the CURSOR and the DATA ENTRY B to change the following values.

a) MODE

The choices for MODE are 24dB or 12dB (use the DATA ENTRY B). dB is short for Decibel, which is a unit for measuring loudness. What's measured here is the slope of attenuation above the FILTER's CUTOFF POINT (**Fc**). Let me explain.

In Chapter 39, I outlined how the VCF functions as a LPF: When the FILTER CUTOFF is set to lower than 127, the overtones below the CUTOFF POINT pass through unaltered, but the overtones above the CUTOFF POINT disappear, they are being filtered out. The term CUTOFF is well established, but it is also misleading. The VCF does not cut the overtones from the CUTOFF POINT upwards, but it rolls them off or fades them out. The disappearance of the filtered frequencies above the CUTOFF POINT is gradual. How gradual depends on the amount of dB chosen under MODE.

A setting of 24dB means that frequencies **one octave** higher **than** the CUTOFF POINT are 24dB softer than the frequencies right at the CUTOFF POINT.

A setting of 12 dB means that frequencies **one octave** higher **than** the CUTOFF POINT are 12 dB softer than the frequencies right **at** the CUTOFF POINT. More dB means more attenuation: At 24dB the filtering action of the VCF is more effective than at 12dB.

In most situations you'll make this choice while trusting your ears. As a rule of thumb: STRINGS and other sounds with smooth overtone roll-off curves come off better at 12dB. BRASS and other sounds that need to have a bite or edge to their color come off better at 24dB. A lot will depend on the values in **F32 VCF CUTOFF & EG INT** and in **F33 RESONANCE & KBDTRACK**.

b) POLARITY

During every note you can have the EG change the CUTOFF POINT from the value that you initially chose. For this, you assign a certain amount of ENVELOPE INTENSITY in **F32 VCF CUTOFF & EG INT**. The POLARITY is the direction in which the CUTOFF POINT is changed by the EG.

+ (positive) means that the EG (Envelope Generator) moves the CUTOFF POINT upwards during the attack of every note. This opens the VCF and produces a brighter COLOR.

- (negative) means that the EG (Envelope Generator) moves the CUTOFF POINT downward during the attack of every note. This closes the VCF and produces a darker COLOR.

Rule of thumb: When working with negative POLARITY of the VCF EG, start out with bigger than usual values for VCF CUTOFF, and also start out with lower than usual values for EG INTENSITY.

The reason for this: Negative polarity moves the CUTOFF POINT towards the CLOSED position of the VCF, where you here little or NO SOUND AT ALL.

F32 VCF CUTOFF & EG INT

Use the CURSOR and the DATA ENTRY B to change the following values.

a) CUTOFF

At the maximum setting of 127 the VCF is fully open: the OSCILLATORS' output passes through with all the overtones intact. The result is that you hear the MULTI SOUNDS of the OSCILLATORS at their brightest color.

At the minimum setting of 00 the VCF is closed: none of the OSCILLATORS' output passes through. The VCF blocks all sound - overtones and fundamental frequencies. The result is silence.

Before you settle on a VCF CUTOFF value, decide on whether you'll use ENVELOPE GENERATOR INTENSITY, RESONANCE and KEYBOARD TRACK to dynamically alter the brightness during every note you play.

b) EG INTENSITY

The Envelope Generator assigned to the VCF is an 'electronic helper' that you program to shift the CUTOFF POINT during every note you play. The way the EG shifts the CUTOFF POINT depends on speed and level values that you adjust as described under **F35 VCF EG**. Here we deal with the INTENSITY of the EG's job. INTENSITY means "by how much" does the EG shift the CUTOFF POINT during every note you play. **INT=00** means that the EG is not shifting the CUTOFF POINT at all. **INT=63** means that the EG opens (+ polarity) or closes (- polarity) the VCF the maximum amount from the initial CUTOFF POINT chosen with **F32 VCF CUTOFF**.

When the EG POLARITY is + (positive) and the CUTOFF POINT is at 127 (fully open VCF) then, *in theory*, the EG can't have any influence, no matter how high you set its INTENSITY. That's because the filter is already wide open - there simply is nothing left to do for the EG. But don't think that you can assign just any value for EG INT. On the DSS-1, high values of EG INT actually lower the initial CUTOFF POINT from settings at or near the maximum of 127. This is different from most VCF/EG combinations found on other synthesizers.

When the EG POLARITY is + (positive) and the CUTOFF POINT is low (partially closed VCF = dull or dark color), then the EG brightens up the color of the sound by raising the CUTOFF POINT during every note you play. The higher the INTENSITY, the more the sound brightens up.

When the EG POLARITY is + (positive) and the CUTOFF POINT is very low or even at 00 (closed VCF = silence), then the EG can "bring back to life" and brighten up the sound during every note you play. The amount of EG INTENSITY will dictate how much sound and brightness you hear during every note.

When the EG POLARITY is - (negative) and the CUTOFF POINT is at 127 (fully open VCF), then the EG can reduce the brightness of the sound during every note you play. INTENSITY values of more than 30 will lower the CUTOFF POINT all the way to the closed VCF position - the result: silence.

When the EG POLARITY is - (negative) and the CUTOFF POINT is low (partially closed VCF), then it takes very little EG INTENSITY to further close the VCF. The result, again: silence.

When the EG POLARITY is - (negative) and the CUTOFF POINT is very low or even at 00 (closed VCF = silence), then the EG can't have any influence, no matter what its INTENSITY is. The filter is already closed, there is simply nothing left to do for the (negative) EG.

F33 VCF RESO & KBDTRACK

Use the CURSOR and the DATA ENTRY B to change the following values.

a) RESONANCE

This works like a specialized volume control for a very narrow band of frequencies in the harmonic spectrum. It only acts on the frequencies at the CUTOFF POINT. The result of high values of RESONANCE is an edge in the color of the sound that can be heard as a whistling pitch of its own when RESONANCE is at maximum (63). The higher the values for RESONANCE, the more the FUNDAMENTAL frequency weakens while the RESONATED frequencies (those at the CUTOFF POINT) get louder.

When RESONANCE is at maximum and the CUTOFF POINT is moved slowly by the EG or by the MG, the frequencies at the CUTOFF POINT can be heard "spelling out" the harmonic spectrum of the current waveform. When you're comfortable with all the Functions described in this Chapter, come back to this little experiment: after clearing the RAM, get the SAW from F2 HARMONIC SYNTHESIS in the CREATE WAVEFORM mode. Enter the PROGRAM PARAMETER mode and set the VCF CUTOFF to around 60, the EG INT to 63, RESONANCE to 63, and, in the VCF EG, set the A (Attack) to 63. Play and hold a low note. You'll hear the pitches of the overtones in the order that I listed them for the SAWTOOTH wave in Chapter 38. Then repeat the above with the SQUARE wave and any other waveform of which you'd like to hear the overtones one by one.

b) KEYBOARD TRACK

When set to maximum (63), this feature adjusts the CUTOFF POINT for every note proportionately to its position on the keyboard. This maintains even brightness throughout the range.

When KEYBOARD TRACK is set to zero, your sounds will behave like most sounds in nature: higher pitches have relatively less brightness than lower pitches. This may sound contradictory, because higher pitches often seem to sound brighter. But when it comes to the harmonic spectrum, lower pitches tend to be richer sounding. The human ear can't hear the upper range of overtones when the fundamental is high.

Make it a habit to test your sounds at either end of the keyboard before settling on a final combination of filter settings. KEYBOARD TRACK may save your sound from being unbalanced throughout the musically useful range of pitches.

F34 VCF MG MOD

This stands for VOLTAGE CONTROLLED FILTER MODULATION GENERATOR MODULATION.Yes, this is another mouthful. Let me explain.

As we saw before, KORG use the term MODULATION GENERATOR (MG) for the Low Frequency Oscillator (LFO) that you may know from other equipment. Modulation simply means change. And since the VCF deals with the second of the BASIC THREE, COLOR (see Chapter 39), the MG can change ("modulate") the sound's COLOR during every note played. It does so by shifting the CUTOFF POINT in an ongoing, symmetrical up-and-down manner. The output of both OSCILLATORS is affected. You've heard this kind of effect before: wah-wah pedals are similar in that they continually change the COLOR from bright to dark as the player's foot moves up and down.

The center of the up-and-down shift is the initially chosen CUTOFF POINT (**F32 VCF CUTOFF**).

Note that the MG affecting the CUTOFF is independent from the MG described in Chapter 40 (**F17 OSC MG MOD**). PITCH and CUTOFF can be modulated independently, each at their own speed and intensity. Use the CURSOR and the DATA ENTRY B to make changes to the following values in the LCD:

a) **FRQ = 00** (slowest) **- 63** (fastest). This is the SPEED (frequency) of the effect.
b) **INT = 00** (cancels the effect) **- 63** (maximum CUTOFF shift). This adjusts the INTENSITY.
c) **DLY = 00** (effect starts immediately on key attack) - 63 (eight seconds of DELAY before effect).

Adding new notes to sustained notes bypasses the DELAY for the additional notes, only the first note experiences the DELAY (if any).

F35 VCF EG

The Envelope Generator assigned to the VCF is you program to shift the CUTOFF POINT during every note you play. Contrary to the MG, which shifts the CUTOFF continually and symmetrically around the initial CUTOFF POINT, the EG shifts the CUTOFF POINT from one precisely programmed level to another, at precisely programmed rates of speed.

KORG uses an ENVELOPE which allows you to program 6 different parameter values: 4 rates of speed and 2 levels. The 6 parameters are referred to in the LCD by the letters **A**, **D**, **B**, **S**, **S**, and **R**.

A stands for ATTACK RATE OF SPEED. It determines how fast the CUTOFF POINT is first shifted when a key is struck ('attacked'). It doesn't determine how far the CUTOFF POINT is shifted - that's determined by EG INTENSITY and, to a degree, also by the initial CUTOFF POINT (**F32 VCF CUTOFF & EG INT**).

The next two parameters, labeled **D** and **B**, need to be looked at together. **D** stands for DECAY RATE OF SPEED. **B** stands for BREAKPOINT LEVEL.

As soon as the CUTOFF POINT has been shifted to the first level (according to **F32 VCF CUTOFF & EG INT**), the D = DECAY RATE OF SPEED comes into play. It determines how fast the CUTOFF POINT is shifted to the level **B**.

If level **B** is at less than 63, then the value of **D** determines how long it takes to reach the level of **B**.

If **B** is at maximum (63), which is the same level that was first reached according to **F32 VCF CUTOFF & EG INT**, then there is no DECAY (change in level) happening at this point. In this case, the value for **D** determines how long it takes before the level is allowed to change towards the value of the second **S**. At **D = 63**, the first-reached level is frozen for about 6 seconds.

The next two parameters, both labeled **S**, need to be looked at together.

The first **S** stands for SLOPE It is a rate of speed.

The second **S** stands for SUSTAIN. It's a level. Once reached, this level remains unchanged while you keep holding a key or pressing the sustain pedal.

If the second **S**, the SUSTAIN LEVEL, is at a value that's different from **B**, then the value of the first **S** (SLOPE) determines how long it takes for the CUTOFF POINT to be shifted from the BREAKPOINT LEVEL (**B**) to the SUSTAIN LEVEL (the second **S**).

If the second **S**, the SUSTAIN LEVEL, is at the identical value of the value of **B**, then there is no change of level happening at this point. In this case, the value of the SLOPE (the first **S**) doesn't matter.

R stands for RELEASE RATE OF SPEED. Anytime you release the keys or the sustain pedal, the CUTOFF POINT returns to the level that it started out from. This level is the result of the value of CUTOFF and, to a degree, of EG INT, both of which you had set in **F32 VCF CUTOFF & EG INT**. The value of **R** determines how long it takes for this last shift of the CUTOFF POINT.

Note that a given value for any of the rates of speed (**A**, **D**, the first **S**, or **R**) can't be expected to correspond to a fixed amount of time (milliseconds or seconds). The actual time depends on both the RATE OF SPEED and the DISTANCE BETWEEN LEVELS.

The initial filter CUTOFF (and, to a degree, the value of EG INT) are set in **F32 VCF CUTOFF & EG INT**. This CUTOFF POINT is the first and last level of the EG. In **F35 VCF EG**, you set the levels for **B** and the second **S**.

For the levels **B** and the second **S**, the range of values is 00 to 63. A value of 00 is identical to the starting level (the initial CUTOFF POINT). A value of 63 is identical to the (maximum) level reached according to EG INT.

The VCF CUTOFF POINT can be influenced in real time by **F42 VELOCITY SENSITIVITY VCF EG CUTOFF**.

A, **D**, and the first **S** can all be influenced by VELOCITY SENSITIVITY **F43 VCF EG (ATK, DEC, SLP)**.

VCF CUTOFF POINT LEVELS REACHED BY THE VCF EG

Here's what happens to these levels during long, sustained notes.

VCF EG POLARITY: POSITIVE

From the initial CUTOFF POINT, the filter is opened up (at the speed of **A**) to a level that is the result of the EG INT value. If **B** is less than 63, the filter closes (at the speed of **D**) to the level specified by **B**. If **B** is 63, the filter remains open for a length of time related to **D**.

Once level **B** is reached, the filter opens or closes (at the speed of the first **S**) to the level specified by the second **S**. Levels **B** and the second **S** can be equal, in which case there is no movement from **B** to the second **S** (and the value for the first **S** is ignored). Whenever the key (or the sustain pedal) is released, the filter closes to the initial CUTOFF POINT. This happens at the speed of **R**.

VCF EG POLARITY: NEGATIVE

From the initial CUTOFF POINT, the filter is closed (at the speed of **A**) to a level that is the result of the EG INT value. If **B** is less than 63, the filter opens (at the speed of **D**) to the level specified by **B**. If **B** is 63, the filter remains closed (partially of fully) for a length of time related to **D**.

Once level **B** is reached, the filter closes or opens (at the speed of the first **S**) to the level specified by the second **S**. Levels **B** and the second **S** can be equal, in which case there is no movement from **B** to the second **S** (and the value for the first **S** is ignored). Whenever the key (or the sustain pedal) is released, the filter re-opens to the initial CUTOFF POINT. This happens at the speed of **R**.

EXAMPLE

1. Turn POWER OFF and back ON.
2. Enter the CREATE WAVEFORM mode.
3. Select **F2 HARMONIC SYNTHESIS** (press 2 on the numpad).
4. Tap the UP-ARROW of the DATA ENTRY A twice to select the SAWTOOTH waveform.
5. Press ENTER. After the waveform has been calculated,
6. Enter the PROGRAM PARAMETER mode.
7. Select **F32 VCF CUTOFF & EG INT** (press 3 and 2 on the numpad or use the DATA ENTRY A).
8. Use the DATA ENTRY B to reduce the value for CUTOFF to 60.
9. Play the keyboard and note the color of the sound.
10. Use the CURSOR and the DATA ENTRY B to set the value for EG-ENT to 20.
11. Play the keyboard and note the color of the sound.
12. Select **F35 VCF EG** (press 3 and 5 on the numpad or use the DATA ENTRY A).
13. Use the DATA ENTRY B to change the value under **A** to 25
14. Play the keyboard and note the shift of the CUTOFF POINT from the color first heard under step 9 above to the color next heard under step 11 above.

15. Select **F32 VCF CUTOFF & EG INT** (press 3 and 2 on the numpad or use the DATA ENTRY A).
16. Use the CURSOR and the DATA ENTRY B to change the value of EG-INT to 40.
17. Play the keyboard and note the increased change of color.

The increased INTENSITY now causes the EG to shift the CUTOFF POINT further away from the initially chosen setting of 60 - it ends up and sustains somewhere in the vicinity of a CUTOFF=95. To test this out,

18. Change EG-INT to 00 and use the CURSOR and DATA ENTRY B to change CUTOFF to 95.
19. Play the keyboard and note that the sound sustains at roughly the same level of brightness that it did when the EG shifted the CUTOFF POINT. What we've lost now is the slow shift in color that was caused by the slow **A** (ATTACK) RATE. Right now the EG is not involved (**INT= 00**).

Experiment with different values for CUTOFF, INTENSITY, RATES, and LEVELS. When you have gotten a sense of what's happening, re-read this Chapter (41) and switch the POLARITY to negative. Add RESONANCE, change the FILTER MODE from 24dB to 12 dB, and bring in some VCF MG MODULATION. Finally check out the difference that KEYBOARD TRACK can make to the brightness across the keyboard.

This should keep you amused for a few hours. But why stop there? Call up different waveforms from the CREATE WAVEFORM mode, and use some or all of the features described in Chapter 40 before you even get back to Chapter 41. You'll understand how it is possible to create all the many MULTI SOUNDS and PROGRAMS that you find on the Factory Disks that aren't even real SAMPLES. Then use all you've just learned on some of your own SAMPLES. See you in outer orbit.

Chapter 42
F36- 38 VCA - THE LOUDNESS REGULATOR

In Chapter 39, I outlined the overall function of the Voltage Controlled Amplifier. It sets the overall maximum loudness for every combination of MULTI SOUNDS that can be part of a PROGRAM. But to have only one level of loudness would be too restrictive. That's why the loudness can be further influenced, within the limits set by the overall loudness maximum, by the VCA EG, and by your playing style through VELOCITY and AFTER TOUCH.

F36 VCA TOTAL LEVEL

This determines the maximum level that the combined output of both OSCILLATORS can have. Whether your sound ever reaches this level depends on the following: **F32 VCF CUTOFF & EG INT** (remember - a closed filter can kill the sound before it ever gets to the VCA), **F44 VELOCITY SENSITIVITY – VCA EG LEVEL**, and **F53 AFTER TOUCH - VCA TOTAL LEVEL**.

F37 VCA DEC KBDTRACK

This stands for Voltage Controlled Amplifier Decay Keyboard Track. The range of values is from -63 to +63. +63 gives the maximum value, -63 the maximum inverted value, while 00 (center position) has no effect.

This feature influences two of the EG parameters described in the next paragraph: DECAY and SLOPE. These rates of speed in the VCA EG can be slowed down and sped up proportionally to the position of the notes played on the keyboard.

Increasing positive amounts of DEC KBDTRACK cause higher notes to have quicker DECAY and SLOPE times, while lower notes have slower DECAY and SLOPE times.

Increasing negative amounts of DEC KBDTRACK cause higher notes to have slower DECAY and SLOPE times, while lower notes have quicker DECAY and SLOPE times.

The DECAY and SLOPE times can further be influenced by VELOCITY SENSITIVITY. See **F45 VELOCITY SENS - VCA EG (ATK, DEC, SLP)**.

F38 VCA EG

This stands for Voltage Controlled Amplifier Envelope Generator. It is a programmable electronic "helper" that lets you determine what happens during every note in terms of LOUDNESS.

The LCD shows 6 parameters, each with a value from 00-63.

Four of these six parameters are RATES OF SPEED: **A** = ATTACK RATE OF SPEED, **D** = DECAY RATE OF SPEED, the first **S** = SLOPE RATE OF SPEED, and **R** = RELEASE RATE OF SPEED. For these RATES, a value of 00 is the fastest, 63 is the slowest possible speed.

Two of these six parameters are loudness LEVELS: **B** = BREAKPOINT LEVEL, and the second **S** = SUSTAIN LEVEL. For these LEVELS, a value of 00 means "zero loudness", which is silence. A value of 63 is the maximum available loudness as per **F36 VCA TOTAL LEVEL**.

Before you play a note, the level of loudness is zero. As SOON as you play a note, the loudness starts to rise towards the maximum level according to **F36 VCA TOTAL LEVEL**. How fast the signal reaches this level depends on your value for **A** = ATTACK RATE OF SPEED.

As soon as the signal has reached this first level of loudness (which, remember, you don't adjust from within this VCA EG screen - it is the **F36 VCA TOTAL LEVEL**), the signal wants to move on to the next level: **B** = BREAKPOINT LEVEL. If **B** is set to less than 63, then the value for **D** = DECAY RATE OF SPEED determines how soon level **B** is reached.

If, however, the level for **B** is set to 63, then no decay (decrease) of LEVEL occurs, since **B=63** is the same as the TOTAL LEVEL. In this case, the value of **D** determines how long the signal stays at the equivalent of the TOTAL LEVEL before it is allowed to move on towards the SUSTAIN level. A value of **D=63** "freezes" the loudness at maximum level for about 6 seconds.

As soon as the signal has reached the level **B** (if **B** is less than 63), or as soon as **D** has "unfrozen" the signal from the maximum level **B= 63**, the signal begins to move towards the level of the second **S**. How fast it gets there depends on the value of the first **S**. Once the signal has reached the level of the second **S**, the loudness will remain unchanged until you release the key or the sustain pedal.

If the value of the second **S** is identical to the value of **B** then the value for the first **S** is ignored.

Anytime you let go of the keys or the sustain pedal, the loudness returns to zero (silence). How fast this happens depends on the level of loudness at the time that you're releasing and on the value for **R**.

Note that a given value for a rate of speed can't be expected to correspond to a fixed amount of time (milliseconds or seconds). The actual time depends on both the RATE OF SPEED and the DISTANCE BETWEEN LEVELS.

The DECAY and SLOPE RATES OF SPEED can be influenced by **F37 VCA DEC KBDTRACK**.

The ATTACK, DECAY and SLOPE RATES OF SPEED can be influenced by VELOCITY SENSITIVITY **F45 VCA EG (ATK, DEC, SLP).**

The VCA EG LEVELS can be influenced by **VELOCITY SENSITIVITY F44 VCA EG LEVEL**.

The total loudness can be influenced by AFTER TOUCH SENSITIVITY **F53 VCA TOTAL LEVEL.**

The loudness mix between OSCILLATORS can be influenced by **VELOCITY SENSITIVITY F46 VELOCITY SWITCH**.

The efficiency of the VCA EG can further be influenced by the VCF. If any of the parameters that control the CUTOFF POINT cause the VCF to be closed, there will be no loudness passed on to the VCA.

Chapter 43
REALTIME CONTROLTHROUGH VELOCITY, AFTER TOUCH AND JOYSTICK.

The DSS-1 is a very expressive instrument in the hands of a caring musician. There is no excuse for mechanical, dead sounds which used to give synthesizers a bad name. Check out the following features one-by-one. Some will demand a great deal of adjustment and practice. Pianists and organists in particular need to acquire a whole new playing technique to make the most of this wealth of expressiveness that's at their fingertips. But it will be worth every minute of your time.

F41 VELOCITY SENSITIVITY AUTO BEND INT.
This stands for AUTO BEND INTERVAL. It gives you real-time control over the distance that the PITCH can travel when you play harder or softer. At a value of 00, your playing style has no effect on AUTO BEND. At 63 your playing style has maximum effect. Use the DATA ENTRY B for your selection. See Chapter 40 for the detailed description of AUTO BEND.

F42 VELOCITY SENSITIVITY VCF EG CUTOFF
The range of values is from 00 to 63. Use the DATA ENTRY B for your selection. At zero, your hard-or-soft playing style has no effect on the VCF CUTOFF. At higher values, you gain control over the initial CUTOFF POINT from which the VCF EG operates: the harder you attack the keyboard, the higher the CUTOFF POINT.

F43 VELOCITY SENSITIVITY VCF EG (ATK, DEC, SLP).
Your hard-or-soft playing style can make a difference to each of these parameters in the VCF EG: ATTACK RATE OF SPEED, DECAY RATE OF SPEED, SLOPE RATE OF SPEED (the second **S**). The range of values is from 00 (no effect) to 63 (maximum effect). Use the DATA ENTRY B for your selection. The higher the values and the harder your playing, the quicker the assigned RATES become.

F44 VELOCITY SENSITIVITY VCA EG LEVEL
The range is from 00 (no effect) to 63 (maximum effect). The higher the value and the harder your playing, the higher the loudness level in the VCA, proportional to the VCA EG LEVELS. Use the DATA ENTRY B for your selection.

F45 VELOCITY SENSITIVITY VCA EG (ATK, DEC, SLP)
Your hard-or-soft playing style can make a difference to each of these parameters in the VCA EG: ATTACK RATE OF SPEED, DECAY RATE OF SPEED, SLOPE RATE OF SPEED (the second **S**). The range of values is from 00 (no effect) to 63 (maximum effect). Use the DATA ENTRY B for your selection. The higher the values and the harder your playing, the quicker the assigned RATES become.

F46 VELOCITY SWITCH
This lets you switch between MULTI SOUNDS. Your hard-or-soft playing style can let you hear more of one or of the other OSCILLATOR. The range of VELOCITY SENSITIVITY is from 00 (OFF) to 63 (maximum). Use the DATA ENTRY B for your selection.

This is directly related to **F11 OSC MIX RATIO**. Whichever OSCILLATOR has the higher MIX RATIO percentage assigned is heard more prominently when you attack the keys lightly. As you play harder, you'll hear more of the other OSCILLATOR.

This feature counteracts the settings of **F11 OSC OCTAVE** and **F15 OSC DETUNE & INTERVAL**. Whichever OSCILLATOR is heard when you play lightly (the one with the higher MIX RATIO percentage) imposes its PITCH position on the other OSCILLATOR. This happens with **F16 SYNC** both ON and OFF.

This works best when you have a PROGRAM with two related but different MULTI SOUNDS in the OSCILLATORS. It works wonders for instruments that sound very differently when played at different volumes.

Example 1: Imagine a gong that only hums very darkly when struck lightly, but shatters metallically when struck hard. If you assign a 'dark' gong sample to OSCILLATOR 1 and a "shattering" gong sample to OSCILLATOR 2, then you can use VELOCITY SWITCH to get very realistic results by varying your key attack force.

Example 2: Sample a muted violin ("con sordino") and a harshly bowed, bright sounding violin. Assign the muted violin to OSC1, with a MIX RATIO of 90% approx. Assign the bright violin to OSC2. In addition to VELOCITY SWITCH, use **F44 VELOCITY SENSITIVITY VCA EG LEVEL** and **F45 VELOCITY SENSITIVITY VCA EG (ATK, DEC, SLP).** When you play lightly, you'll hear more of the muted violin, at low loudness, with the slowish EG RATES that you'll have to set with **F38 VCA EG**. When you play hard, you'll hear the bright violin, louder, and with accelerated EG RATES. If this is not enough, use the other features covered in this chapter for added expression and realism. See you at the Philharmonic...

F51 AFTER TOUCH SENSITIVITY OSC MG MOD INT

This ties in with the settings in **F17 OSC MG MOD**, as described in Chapter 35. After using the DATA ENTRY B to set a high value for the AFTER TOUCH, you can lean into held notes and use the added keyboard pressure to deepen the vibrato effect. This works even if there is zero INT assigned in **F17 OSC MG MOD**. But you can set a small amount of INT in F17 for permanent vibrato. A high amount (maximum = 15) in F51 gives you the chance to override the permanent setting by adding vibrato depth from AFTER TOUCH.

F52 AFTER TOUCH SENSITIVITY VCF CUTOFF/MG MOD

Here you choose between two ways of affecting the COLOR of your sound with AFTER TOUCH. The left half of the LCD displays the MODE. Use the DATA ENTRY B to toggle between **MG-MOD** and **CUTOFF**. The right half of the LCD shows a value between 00 (no effect) and 15 (maximum effect).

MG-MOD causes the "Wah-Wah" effect described in Chapter 41. The frequency will be that chosen in **F34 VCF MG MOD**. When using AFTER TOUCH for the VCF modulation, the DELAY setting is ignored.

CUTOFF causes a raising of the CUTOFF POINT when AFTER TOUCH is used.

F53 AFTER TOUCH SENSITIVITY VCA TOTAL LEVEL

This puts you, the player, in command over the total loudness level at the VCA, by using AFTER TOUCH pressure. At the maximum effect of 15, there is no volume when the keyboard is played very lightly. Find your own settings to suit the weight of your hands and your playing habits. Use the DATA ENTRY B for your selection.

F61 JOYSTICK PITCH BEND RANGE

The PITCH INTERVAL that you select for the JOYSTICK is always the same for the left (down/flat) and the right (up/sharp) direction. The maximum is a value of 12 halfsteps, which is one octave in either direction. Use the DATA ENTRY B for your selection.

The higher the value, the harder it is to control the PITCH when you don't want to move the JOYSTICK all the way. KORG very kindly add the word **WARNING** to values from 07 - 12.

Unless you're after some special effects you'll probably find that the default value of 02 is the most musically useful setting. But feel free to choose your own and to ignore the "Warning".

F62 JOYSTICK VCF SWEEP ON/OFF

When bending the JOYSTICK to the left and right, you can get a FILTER SWEEP in addition to the PITCH BEND. The choices are either ON or OFF (use the DATA ENTRY B).

Chapter 44
NON REAL TIME CONTROL AND EFFECTS

F63 KEY ASSIGN MODE

The DSS-1 normally operates as a 8-note polyphonic keyboard with last-note priority. This means that you can play up to 8 notes simultaneously. If you add a ninth note, then the first ("oldest") note will stop sounding.

Use the DATA ENTRY B to change from POLY-1or POLY-2 to UNISON.

POLY-1 is the normal default mode. Up to eight notes can sound simultaneously. The sequential voice assignment works like this: when you play a single note melody of more than eight notes, all eight voices are used. The first note gets the first voice, the second note gets the second voice etc. This means that no "note stealing" occurs until the ninth note silences the first note if the first note is still sounding. If your VCA envelope gives the sound a long release time, then, in POLY-1 mode, each note can ring out for the whole release time until you play more than eight notes.

POLY-2 works differently. Up to eight notes can sound simultaneously. But when you play a series of single notes, only the first voice gets used - repeatedly for every note. It therefore "chops off" its own release time when a new note is played after the previous note has been released. This makes for a drier, less swirling sound when you play a PROGRAM with a long VCA release time. Voices 2-8 only come into play when you press down several notes simultaneously.

UNISON turns the keyboard into a monophonic instrument. No chords are possible, only one note can sound at a time. This is a favorite setting with many players, particularly with those who learned their chops on the great monophonic synths of the 70s: MINI MOOGs etc. This keyboard mode makes for cleaner trills and articulation, because any new notes chop off lingering releases of the previous note. It is also more in keeping with many instruments that we tend to simulate: Single note guitar leads, bass parts etc.

Monophonic mode really shines when the next function is used:

F64 KEY ASSIGN UNISON DETUNE AND VOICES

You can assign 2 (= minimum), 4, 6, or all 8 voices of the DSS-1 to every single note you play. Obviously this makes for a bigger sound. But you can also assign the voices to be slightly out of tune with each other. This makes for an even bigger sound. Use the DATA ENTRY B to choose from DETUNE=01(minimum) to 08 (maximum).

P65 EQUALIZER (BASS, TREBLE)

As if you hadn't had enough chances to tailor the color of your sounds, here you have a straight tone control, with a range from -04 to +08 for BASS and for TREBLE. Use the CURSOR and the DATA ENTRY B to make your selections.

F71 - 96 DIGITAL DELAYS

This is truly a bonus: not one, but two digital delays built into the instrument, saving you hundreds of dollars and a lot of knob twiddling on what would otherwise have to be outboard equipment.

It is beyond the scope of this book to go into details of this exciting feature. Suffice it to say that **F91 DDL-2 INPUT SELECT** lets you choose between running the delays in parallel or serially (in a chain). Use the DATA ENTRY B to make your selection.

If you choose **DIRECT** then you're affecting the combined sound from both OSCILLATORS with DDL-2, in parallel with DDL-1. This ties in with the LEFT & RIGHT audio OUTPUT jacks on the back panel, where DDL-1 appears at the LEFT and DDL-2 at the RIGHT jack.

If you choose **DDL-1** then you can use DDL-2 to further enhance the already processed signal coming from DDL-1. This serial signal appears in mono at the RIGHT output jack.

Set the values of both **F83** and **F94 EFFECT LEVEL** to zero if you wish to ignore the DDLs.

Well, what do you think ? Did you figure on all this processing power when you bought this "SAMPLER" ? If you didn't already know it - I bet that you're quite sure by now that this is at least as powerful a synthesizer as it is a "sampler".

If you wish to practice, why don't you get the DISK with the BRASS/HARP MULTI SOUND that we so laboriously constructed in Chapter 16 onwards. I'm sure that you'll turn it into a masterpiece. You'll find that the good ol' analog SUBTRACTIVE FILTER SYNTHESIS still works like a charm.

Never lose sight of the BASIC THREE in your sounds, be they synthesized waveforms or audio-input samples: PITCH, COLOR, LOUDNESS.

SECTION7: DISK UTILITY MODE

Chapter 45
DISK FORMATTING AND PROTECTION

F0 FORMAT DISK

The DSS-1, like any computer, can only write data onto a DISK if the DISK DRIVE has had a chance to map out the sectors on the DISK during the FORMATTING process. It takes about 2 minutes and results in a blank, ready-for-work DISK. If you have several DISKS to format, press YES in reply to the prompt **Continue? (Y/N)**. If you're ready to go on to other things, press NO and you can choose any other FUNCTION or MODE.

CAUTION: FORMATTING ERASES ALL DATA ON DISK

F1 PROTECT DISK (SET/RESET)

This is a smart feature that can save you a lot of cold sweat. Although all commercially available DISKS have a MEMORY PROTECT sliding tab (see Factory Disks), this Function provides additional insurance against erasing/overwriting your data by accident.

After selecting F1, the first LCD screen invites you to press ENTER. After pressing ENTER, use the CURSOR to select either SET (protect) or RESET (unprotect) before pressing ENTER.

CAUTION: SET PROTECT DOES NOT PREVENT A DISK FROM BEING FORMATTED.

Chapter 46
THE DISK DIRECTORIES

F2 PROGRAM DIRECTORY

This reads into RAM the PROGRAM names from any one of the 4 SYSTEMS on the DISK currently in the DISK DRIVE. It is only an index of the names - the PROGRAMS don't actually get loaded (see SYSTEM MODE **F5 GET ALL PROGRAMS**).

Use the DATA ENTRY A to select a SYSTEM before pressing ENTER. Then use the DATA ENTRY A to display the PROGRAM names.

F3 MULTI SOUND DIRECTORY

This reads into RAM the MULTI SOUND names from the DISK currently in the DISK DRIVE. It is only an index - the MULTI SOUNDS don't actually get loaded (see SYSTEM MODE **F9 GET MULTI SOUND**).

After pressing ENTER, use the DATA ENTRY A to display the MULTI SOUND names. Note that they are not numbered. But, with the DATA ENTRY slider A moved from its lowest position upward, the MULTI SOUNDS are listed in the order in which they were SAVED to the DISK.

F4 SOUND DIRECTORY

If you try this with a Factory Disk, after pressing ENTER you'll get a display **NO SOUNDS**. If this seems paradoxical, remember this: SOUNDS on the DSS-1 are individual WAVEFORMS / SAMPLES, independent from MULTI SOUNDS. Only SOUNDS / WAVEFORMS / SAMPLES that are SAVED to DISK with the following MODE/FUNCTIONS are recognized and listed as SOUNDS in **F4 SOUND DIRECTORY**:

SAMPLE MODE **F5 SAVE SAMPLE**
EDIT SAMPLE MODE **F8 SAVE/RENAME SAMPLE**
CREATE WAVEFORM MODE **F3 SAVE WAVEFORM**

If your DISK contains any SOUNDS, after pressing ENTER use the DATA ENTRY A to display the SOUND names.

Chapter 47
DELETING AND DISK STATUS

F5 DELETE SOUND

This erases a selected SOUND (SAMPLE/WAVEFORM) permanently from the DISK currently in the DISK DRIVE. After pressing ENTER, use the DATA ENTRY A to display the SOUND you wish to DELETE. After pressing ENTER, you get a last chance to change your mind: pressing YES in reply to **Are You SURE? (Y/N)** deletes the SOUND. Press NO in reply to **Continue? (Y/N)** unless you have another SOUND that you wish to delete. If a SOUND is used in a MULTI SOUND on the same DISK, deleting that SOUND does not affect the MULTI SOUND.

F6 DELETE MULTISOUND

This erases a selected MULTI SOUND permanently from the DISK currently in the DISK DRIVE. After pressing ENTER, use the DATA ENTRY A to display the MULTI SOUND you wish to DELETE. After pressing ENTER, you get a last chance to change your mind: pressing YES in reply to **Are You SURE? (Y/N)** deletes the MULTI SOUND. Press NO in reply to **Continue? (Y/N)** unless you have another MULTI SOUND that you wish to delete.

F7 DISK STATUS

After pressing ENTER you get a display that shows the number of used and free blocks on the DISK currently in the DISK DRIVE. Sec Chapters 1 and 10 for details of the memory capacity.

SECTION 8: MIDI

Chapter 48
QUICK MIDI BASICS

MIDI stands for Musical Instruments Digital Interface. All recent keyboards and drum machines sold for professional use have MIDI built-in, as well as most such instruments marketed for semi-pros and home users. MIDI has revolutionized the music industry.

By connecting two or more MIDI instruments with the specialized MIDI cables, a lot of information can be shared. Before I list some of the things that are transmitted and received via MIDI, let me clear up a frequent misunderstanding: MIDI instruments still have to be connected to mixers / amplifiers / loudspeakers from the instruments' regular audio outputs. MIDI does *not* transmit sound signals.

MIDI transmits data about events. When you play a note on the keyboard, that's a *note ON* event. When you release a key, that's a *note OFF* event. When you step on the sustain pedal, select a new sound PROGRAM, move the JOYSTICK, etc - these are all events. Messages about events are sent by the instrument on which the events happen (the "master"), and receiving instruments (the "slaves") can respond by imitating your actions on the master: playing the same notes, releasing the same notes, switching to the same sound PROGRAM numbers from their own memories, etc.

The events are transmitted at high speed in the form of a computer language, the MIDI protocol. All events are translated into numbers, and can therefore be stored by computers. This is why MIDI sequencers are often called MIDI recorders. A sequencer can come in the shape of a box that does nothing but "sequencing", or a personal computer can be programmed to act like a MIDI sequencer. Once you've "recorded" several musical parts into the sequencer, it can then re-transmit these parts to several synthesizers, with the right part going to the right instrument simultaneously. All this happens digitally in bits and bytes, incredibly fast and absolutely silently.

The MIDI interface is serial - all the information goes down the same line. All the instruments receive all the information simultaneously. But how does a receiving synthesizer know which events to respond to, and which events to ignore ? For this a system of 16 MIDI channels was set up. Every piece of MIDI information that comes down the line carries a little "flag", an identifying number tag that says what MIDI channel it is meant for. Each instrument can be switched to respond to only one of the 16 channels, and to ignore events that are labeled for a different channel.

To standardize the ways in which MIDI instruments respond to incoming messages, FOUR MIDI MODES were set up.

MIDI MODE 1: OMNI ON/POLY. Instruments respond to any and all channels simultaneously ("omni" is Latin for "all"). Furthermore, they play polyphonically - chords are possible. ("Poly" is Green for "many".) This MODE is useful when all the receiving instruments are supposed to play all the same transmitted notes and other events.

MIDI MODE 2: OMNI ON/MONO. Instruments respond to any and all channels simultaneously and play monophonic (single notes only - chords are not possible). You'll probably never use this MODE.

MIDI MODE 3: OMNI OFF/POLY. Instruments respond only to one specified channel and play polyphonic - chords are possible. This is the most common MODE for a system where different synthesizers are playing different musical parts, usually from a sequencer or from a split-keyboard controller.

MIDI MODE 4: OMNI OFF/MONO. Instruments respond only to one specified channel and play monophonic (single notes only - chords are not possible). This mode is used for certain multi-umbral instruments that can play different monophonic musical parts, using different memory PROGRAM sounds simultaneously, one for each part.

So, the receiving MIDI instruments select the data of all channels or of just one channel. Do they have to respond to all the events ? No, not as a rule. Certain types of events can be filtered out, and other types of events can't be responded to in the first place. Example: if you don't wish for a receiving instrument to reset its PROGRAM numbers when such a command comes down the line - set it to ignore that particular type of event. Or maybe you can program the transmitting instrument to not transmit certain types of events in the first place.

As a rule of thumb, MIDI can't cause an instrument to perform functions that the instrument couldn't do by itself.

Example: If you transmit VELOCITY or AFTER TOUCH data from your DSS-1 to a keyboard that wasn't built to be touch sensitive, you'll hear no VELOCITY or AFTER TOUCH response from the receiving instrument – it simply ignores those types of commands. But this rule of thumb has recently become a little blurred. Several instruments, notably from ROLAND and YAMAHA, don't have VELOCITY or AFTER TOUCH sensors built into their keyboard mechanisms. Their brains, however, can respond to those commands as long as they arrive from an external MIDI controller.

There are many more areas of incompatibility - too many to list here. That's why most manufacturers spell out the MIDI capabilities of their instruments in great detail. The user can look up the tables and figure things out for him/herself. As long as the user knows what to look for...

As a basic reference I recommend that you get David Crigger's *Making MIDI Work* and the *Murphy's Law MIDI Book* by Jeff Burger. These books from Alexander Publishing quite nicely give you the MIDI ins and outs.

Chapter 49
MIDI ON THE DSS-1

F1 CHANNEL SELECT

When first powered up, the DSS-1 is always set to transmit and receive MIDI data on channel 1. Use the CURSOR and the DATA ENTRY A to set other channel numbers for transmission and reception of MIDI data.

F2 FUNCTION SELECT

Use the CURSOR and the DATA ENTRY A to select and change the following:

PROGRAM CHANGE MODES

Incoming PROGRAM CHANGE messages can be responded to in several ways, or not at all.

OFF means that incoming PROGRAM CHANGE messages are ignored -- the DSS-1 keeps playing the currently selected PROGRAM.

MODE 1 means that the following incoming PROGRAM CHANGE numbers result in the selection of these DSS-1 PROGRAMS:

Incoming	DSS-1 plays
00 - 31	SYS A PRG 01 - 32
32 - 63	SYS B PRG 01 - 32
64 - 95	SYS C PRG 01 - 32
96 -127	SYS D PRG 01 - 32

Remember that the DSS-1 can have only one SYSTEM in RAM at a time. When you send it a PROGRAM CHANGE number that is asking for a PROGRAM in SYSTEM not currently in RAM, the DSS-1 starts to load that new SYSTEM. You can't play the DSS-1, nor can it send or receive MIDI data, while the new SYSTEM is being loaded.

MODE 2 means that the following incoming PROGRAM CHANGE numbers result in the selection of these DSS-1 PROGRAMS:

Incoming	DSS-1 plays
00 - 31	SYS C PRG 01 - 32
32 - 63	SYS D PRG 01 - 32
64 - 95	SYS A PRG 01 - 32
96 - 127	SYS B PRG 01 - 32

Remember that the DSS-1 can have only one SYSTEM in RAM at a time. When you send it a PROGRAM CHANGE number that is asking for a PROGRAM in SYSTEM not currently in RAM, the DSS-1 starts to load that new SYSTEM. You can't play the DSS-1, nor can it send or receive MIDI data, while the new SYSTEM is being loaded.

MODE 3 means that the DSS-1 keeps playing PROGRAMS from within the currently selected SYSTEM. The incoming PROGRAM CHANGE numbers wrap around every 32 numbers.

Incoming	DSS-1
00 - 31	SYS PRG 01 - 32
32 - 63	SYS PRG 01 - 32
64 - 95	SYS PRG 01 - 32
96 - 127	SYS PRG 01 - 32

MODULATION

MOD ON/OFF selects between responding to or ignoring incoming MODULATION messages. This includes PITCH BEND messages. Note that, when set to ON, the DSS-1 responds according to its own settings for the parameters affected by MODULATION. Example: If the DSS-1 is set to **F61 PITCHBEND RANGE 02**, an incoming PITCH BEND message from a controller that is set to an interval larger than 02 is still only going to cause a PITCH BEND of 02 on the DSS-1.

AFTER TOUCH

AFT. ON/OFF selects between responding to or ignoring incoming AFTER TOUCH messages. Note that, when set to ON, the DSS-1 responds according to its own settings for the parameters affected by AFTER TOUCH.

F3 OMNI MODE

Use the DATA ENTRY A to select OMNI MODE to be either ON or OFF. The DSS-1 is always responding to POLY data. If you wish it to play in monophonic mode, set **F63 KEY ASSIGN MODE** to UNISON.

F4 LOCAL ON/OFF

This stands for LOCAL CONTROL ON/OFF. The normal mode is ON, meaning that the "brain" of the DSS-1 listens and responds to DSS-1's keyboard. When LOCAL CONTROL is OFF, the DSS-1 doesn't respond to its own keyboard. Use the DATA ENTRY A to change this value.

Use this feature when a) the DSS-1 is being played via MIDI from an external controller or sequencer, and b) at the same time you wish to transmit MIDI data from the DSS-1 while playing its keyboard.

F5 SAVE MIDI PARAMETERS

Use this feature (press YES) if you wish to SAVE the current MIDI set-up (F1-F4) to become a permanent part of the SYSTEM currently in RAM. Your selected MIDI values will overwrite the previous MIDI set-up for that SYSTEM.

SECTION 9
THE REFERENCE TABLES

HOW TO INTERPRET TABLES

These tables list all the PROGRAMS on Factory Disks KSD-001 and KSD-002, with the MULTI SOUNDS as they are assigned to OSCILLATORS 1 & 2. The MIX RATIO and the OCTAVE RANGES apply to each OSCILLATOR. The DETUNE and INTERVAL values only apply to OSCILLATOR 2.

If you wish to establish your own tables for other DISKS, use the blank form provided (make copies of the blank form). After loading a SYSTEM, enter PLAY mode (cancel the SYSTEM mode tab), and call up one PROGRAM after the other. For each PROGRAM, enter the PROGRAM PARAMETER mode, select the FUNCTIONS as I list them below, and write the values in the appropriate columns. Before calling up the next PROGRAM, re-enter the PLAY mode (cancel the PROGRAM PARAMETER mode).

The first column shows the PROGRAM NUMBER.

The second column shows the PROGRAM NAME.

The third column shows the name of the MULTI SOUND assigned to OSCILLATOR 1. (PROGRAM PARAMETER mode, **F12 OSCI MULTI SOUND**).

The fourth column shows the name of the MULTI SOUND assigned to OSCILLATOR 2. (PROGRAM PARAMETER mode, **F13 OSC2 MULTI SOUND**).

The fifth column shows the MIX RATIO. To the left of the / is the percentage value for OSC1, to the right of / is the percentage value for OSC2. (PROGRAM PARAMETER mode, **F14 MIX RATIO**).

The sixth column shows the OCTAVE range settings for the OSCILLATORS. To the left of the / is the OCTAVE range for OSC1, to the right of / is the OCTAVE range for OSC2. (PROGRAM PARAMETER mode, **F11 OSC OCT**).

The seventh and eighth columns show the DETUNE and INTERVAL values for OSCILLATOR 2. (PROGRAM PARAMETER mode, **F15 OSC2 DETUNE & INTERVAL**).

TABLE 1: FACTORY DISK KSD-001 MULTI SOUNDS AS IN ALL FOUR SYSTEMS A-D

MS#	MULTISOUND	LENGTH	SYSTEM:A	SYSTEM:B	SYSTEM:C	SYSTEM:D
(01)	A.PF	254,790	1. 254,790	not used	not used	not used
(02)	HDSAWO	1,020	2. 1,020	1. 1,020	1. 1,020	1. 1,020
(03)	HDPW50	1,020	3. 1,020	2. 1,020	2. 1,020	2. 1,020
(04)	HDPW12	1,020	4. 1,020	4. 1,020	4. 1,020	4. 1,020
(05)	ADCOMBO	1,020	5. 1,020	not used	not used	not used
(06)	ADOrgan	1,020	6. 1,020	6. 1,020	6. 1,020	6. 1,020
(07)	AD8ORHO2	1,020	7. 1,020	not used	9. 1,020	9. 1,020
(08)	HD pf	1,020	8. 1,020	10. 1,020	not used	not used
(09)	HDPW25	1,020	not used	3. 1,020	3. 1,020	3. 1,020
(10)	HDCOMBO	1,020	not used	5. 1,020	5. 1,020	5. 1,020
(11)	ADMetal	1,020	not used	7. 1,020	7. 1,020	7. 1,020
(12)	HDSINO	1,020	not used	8. 1,020	8. 1,020	8. 1,020
(13)	HD8ORHO2	1,020	not used	9. 1,020	not used	not used
(14)	CLAV	30,001	not used	11. 30,001	not used	not used
(15)	Harpsi	30,001	not used	12. 30,001	not used	not used
(16)	MB	44,001	not used	13. 44,001	not used	not used
(17)	HARP	64,001	not used	14. 64,001	not used	not used
(18)	EP	30,001	not used	15. 30,001	not used	not used
(19)	DWO3-A.P	1,020	not used	not used	10. 1,020	10. 1,020
(20)	HDO-PW50	1,020	not used	not used	11. 1,020	not used
(21)	HDO-SWXO	1,020	not used	not used	12. 1,020	11. 1,020
(22)	HDO-PW25	1,020	not used	not used	13. 1,020	not used
(23)	HDYAJIO	1,020	not used	not used	14. 1,020	12. 1,020
(24)	HDO-PWXO	1,020	not used	not used	15. 1,020	13. 1,020
(25)	HDTRGO	1,020	not used	not used	16. 1,020	14. 1,020
(26)	ADVIBE	1,020	not used	not used	not used	15. 1,020
(27)	ADVOIC.N	1,020	not used	not used	not used	16. 1,020
	Total	474,215	261,930	208,205	16,320	16,320
	===============	Free	214+	53,939+	245,824+	245,824+
		Maximum RAM	262,144	262,144	262,144	262,144

TABLE 2: FACTORY DISK KSD-002 MULTI SOUNDS AS IN ALL FOUR SYSTEMS A-D

MS#	MULTISOUND	LENGTH	SYSTEM:A		SYSTEM:B		SYSTEM:C		SYSTEM:D	
(01)	HDSAW0	1,020	1.	1,020	1.	1,020	1.	1,020	1.	1,020
(02)	HDPW50	1,020	2.	1,020	2.	1,020	2.	1,020	2.	1,020
(03)	HDPW25	1,020	3.	1,020	3.	1,020	3.	1,020	3.	1,020
(04)	HDPW12	1,020	4.	1,020	4.	1,020	4.	1,020	4.	1,020
(05)	ADCOMBO	1,020	5.	1,020	5.	1,020	5.	1,020	5.	1,020
(06)	ADOrgan	1,020	6.	1,020	6.	1,020	6.	1,020	6.	1,020
(07)	ADMetal2	1,020	7.	1,020	7.	1,020	7.	1,020	7.	1,020
(08)	ADSINO	1,020	8.	1,020	8.	1,020	8.	1,020	8.	1,020
(09)	ADBORHO2	1,020	9.	1,020	9.	1,020	9.	1,020	9.	1,020
(10)	HD-pf	1,020	10.	1,020	10.	1,020	10.	1,020	10.	1,020
(11)	A.SAX	125,001	11.	125,001		not used		not used		not used
(12)	T.SAX	125,001	12.	125,001		not used		not used		not used
(13)	TP	125,001		not used	11.	125,001		not used		not used
(14)	BrassEns	114,466		not used	12.	114,466		not used		not used
(15)	HDO-SAW0	1,020		not used		not used	11.	1,020		not used
(16)	HDO-SAW3	1,020		not used		not used	12.	1,020		not used
(17)	ADVOIC.L	1,016		not used		not used	13.	1,016		not used
(18)	HDO-XXX1	1,020		not used		not used	14.	1,020		not used
(19)	HDO3-A.P	1,020		not used		not used	15.	1,020		not used
(20)	HDO-SWX2	1,020		not used		not used		not used	11.	1,02[illegible]
(21)	HDO-PW10	1,020		not used		not used		not used	12.	1,02[illegible]
(22)	HDSAW1	1,020		not used		not used		not used	13.	1,02[illegible]
(23)	HDSAQW3	1,020		not used		not used		not used	14.	1,0[illegible]
(24)	HDTRG0	1,020		not used		not used		not used	15.	1,0[illegible]
(25)	HDPWMT1	1,020		not used		not used		not used	16.	1,0[illegible]
	Total	510,885		260,202		249,667		15,296		16,3[illegible]
	===============		Free	1,942+		12,477+		246,848+		245,[illegible]
		Maximum	RAM	262,144		262,144		262,144		262,1[illegible]

TABLE 3: FACTORY DISK KSD-003 MULTI SOUNDS AS IN ALL FOUR SYSTEMS A-D

MS#	MULTISOUND	LENGTH	SYSTEM:A		SYSTEM:B		SYSTEM:C		SYSTEM:D	
(01)	HDSAWO	1,020	1.	1,020	6.	1,020	1.	1,020	1.	1,020
(02)	ADCOMBO	1,020	2.	1,020	not used		2.	1,020	2.	1,020
(03)	ADOrgan	1,020	3.	1,020	not used		3.	1,020	3.	1,020
(04)	ADMetal	1,020	4.	1,020	not used		4.	1,020	4.	1,020
(05)	ADBORHO2	1,020	5.	1,020	7.	1,020	5.	1,020	5.	1,020
(06)	VL	100,001	6.	100,001	not used		not used		not used	
(07)	Strings	128,001	7.	128,001	not used		not used		not used	
(08)	K+Choir2	252,000	not used		1.	252,000	not used		not used	
(09)	FV1	1,911	not used		2.	1,911	not used		not used	
(10)	FV2	1,911	not used		3.	1,911	not used		not used	
(11)	MV1	1,911	not used		4.	1,911	not used		not used	
(12)	MV2	1,911	not used		5.	1,911	not used		not used	
(13)	HDO-SWX2	1,020	not used		not used		6.	1,020	not used	
(14)	HDPWMT1	1,020	not used		not used		7.	1,020	6.	1,020
(15)	HDO-PWXO	1,020	not used		not used		8.	1,020	not used	
(16)	HDYAJIO3	1,020	not used		not used		9.	1,020	7.	1,020
(17)	HDO-SAWO	1,020	not used		not used		10.	1,020	not used	
(18)	ADVOIC.L	1,016	not used		not used		not used		8.	1,016
(19)	ADVOIC.N	1,020	not used		not used		not used		9.	1,020
(20)	HDSAW4	1,020	not used		not used		not used		10.	1,020
(21)	HDSAWMT1	1,020	not used		not used		not used		11.	1,020
	Total	501,922		233,102		261,684		10,200		11,216
	================		Free	29,042+		.460+		251,944+		250,928+
		Maximum	RAM	262,144		262,144		262,144		262,144

TABLE 4: FACTORY DISK KSD-004 MULTI SOUNDS AS IN ALL FOUR SYSTEMS A-D

MS#	MULTISOUND	LENGTH	SYSTEM:A	SYSTEM:B	SYSTEM:C	SYSTEM:D
(01)	ALVAREZ2	262,112	1. 262,112	not used	1.262,112	not used
(02)	PULSE	32	2. 32	3. 32	2. 32	1. 32
(03)	SoftBass	131,056	not used	1.131,056	not used	not used
(04)	SlapBass	131,056	not used	2.131,056	not used	not used
	Total	524,256	262,144	262,144	262,144	32
	================	Free	0+	0+	0+	262,112+
		Maximum RAM	262,144	262,144	262,144	262,144

TABLE 6: FACTORY DISK KSD-001 SYSTEM:B PROGRAMS, OSC. AND MULTI SOUNDS

PROG#	PROG.NAME	M.S.OSC1	M.S.OSC2	MIX %	OCTAVES	DETUNE	INTERVAL
1	E.Piano1	EP	EP	100/0	16/16	0	0
2	E.Piano2	EP	EP	100/0	16/16	0	0
3	E.Piano3	EP	EP	100/0	8/8	0	0
4	E.Piano4	EP	EP	100/0	8/8	0	0
5	Clav 1	CLAV	Harpsi	75/25	8/16	0	0
6	Clav 2	CLAV	CLAV	100/0	8/8	0	0
7	Clav 3	CLAV	Harpsi	60/40	8/16	0	0
8	Clav 4	CLAV	MB	68/32	4/16	0	0
9	Harpsi1	Harpsi	Harpsi	100/0	16/16	1	0
10	Harpsi2	Harpsi	CLAV	61/39	16/8	3	0
11	Harpsi3	Harpsi	Harpsi	70/30	8/4	3	0
12	Harpsi4	Harpsi	CLAV	68/32	16/4	2	0
13	Marimb1	MB	MB	100/0	16/16	2	0
14	Marimb2	MB	HDSAWO	70/30	16/8	3	0
15	Harp 1	HARP	HARP	100/0	16/16	2	0
16	Mix	HARP	Harpsi	60/40	16/8	0	0
17	BR3/Bell	HDSAWO	ADMetal	0/100	16/16	5	0
18	BR3&Bell	HDSAWO	ADMetal	60/40	16/4	1	0
19	E.PIANO1	HD8ORHO2	ADMetal	92/8	16/4	1	7
20	E.PIANO2	HD8ORHO2	HD8ORHO2	72/28	8/8	5	0
21	STRINGS1	HDSAWO	HDSAWO	55/45	8/8	4	0
22	STRINGS2	HDSAWO	EP	55/45	16/16	2	0
23	BRASS1	HDSAWO	HDSAWO	52/48	16/16	4	0
24	BRASS2	HDSAWO	HDSAWO	49/51	16/8	3	0
25	ORGAN 1	ADOrgan	ADOrgan	60/40	16/4	0	0
26	ORGAN 2	ADOrgan	ADOrgan	52/48	16/4	0	7
27	PanFlute	MB	HARP	45/55	16/16	5	0
28	[%(YY)%]	EP	HARP	40/60	16/16	2	0
29	Synth 1	HDSAWO	HDCOMBO	34/66	16/16	0	11
30	Synth 2	HDPW12	HDPW25	48/52	4/4	0	0
31	Synth 3	HDSAWO	HDSAWO	56/44	16/8	0	0
32	Synth 4	HD pf	HD pf	100/0	16/8	1	0

TABLE 7: FACTORY DISK KSD-001 SYSTEM:C PROGRAMS, OSC. AND MULTI SOUNDS

PROG#	PROG.NAME	M.S.OSC1	M.S.OSC2	MIX %	OCTAVES	DETUNE	INTERVAL
1	E.Piano	AD8ORHO2	AD8ORHO2	53/47	16/16	9	0
2	ToyPIANO	HDSINO	HDPW50	70/30	4/4	18	0
3	H.CORD	HDO-SWXO	HDO-PW25	51/49	8/4	0	0
4	HARP	HDO-PWXO	HDO-PW50	72/28	8/8	8	0
5	COLLAGE1	HDO-PWXO	HDSINO	58/42	4/8	8	5
6	COLLAGE2	HDO-PWXO	HDSINO	56/44	4/4	10	7
7	COLLAGE3	HDO-PWXO	HDO-PWXO	50/50	4/4	10	8
8	COLLAGE4	HDTRGO	HDTRGO	62/38	4/4	27	8
9	USER	HDO-SWXO	HDO-PW25	100/0	8/8	0	0
10	USER	HDTRGO	HDYAJIO	100/0	8/8	0	0
11	USER	HDO-PWXO	HDO-PW50	100/0	8/8	0	0
12	USER	HDO-PW50	HDO-PWXO	100/0	8/8	0	0
13	USER	HDO-PWXO	HDSINO	100/0	8/8	0	0
14	USER	HDO-PWXO	HDSINO	100/0	8/8	0	0
15	USER	HDO-PWXO	HDO-PWXO	100/0	8/8	0	0
16	USER	HDTRGO	HDTRGO	100/0	8/8	0	0
17	PIANO 1	DWO3-A.P	DWO3-A.P	70/30	8/16	2	0
18	PIANO 2	HDCOMBO	DWO3-A.P	54/46	4/8	1	0
19	E.PIANO1	AD8ORHO2	AD8ORHO2	95/5	16/4	1	0
20	E.PIANO2	AD8ORHO2	AD8ORHO2	72/28	8/8	5	0
21	STRINGS1	HDSAWO	HDSAWO	55/45	8/8	6	0
22	STRINGS2	HDSAWO	HDSAWO	55/45	16/16	4	0
23	BRASS1	HDSAWO	HDSAWO	52/48	8/8	6	0
24	BRASS2	HDSAWO	HDSAWO	49/51	16/8	3	0
25	ORGAN 1	ADOrgan	ADOrgan	60/40	16/4	5	0
26	ORGAN 2	ADOrgan	HDSINO	52/48	16/4	1	7
27	SynBass1	HDPW12	HDPW12	0/100	16/16	1	0
28	SynBass2	HDPW25	HDPW12	51/49	16/16	2	0
29	Synth 1	HDSAWO	HDCOMBO	34/66	16/16	0	11
30	Synth 2	HDPW25	HDPW50	48/52	4/4	0	0
31	Synth 3	HDSAWO	HDSAWO	56/44	16/8	0	0
32	Synth 4	ADCOMBO	HDSAWO	0/100	16/4	1	0

TABLE 8: FACTORY DISK KSD-001 SYSTEM:D PROGRAMS, OSC. AND MULTI SOUNDS

PROG#	PROG.NAME	M.S.OSC1	M.S.OSC2	MIX %	OCTAVES	DETUNE	INTERVAL
1	VIBE	ADVIBE	ADVIBE	86/14	8/8	9	0
2	XYLPHONE	ADMetal	ADMetal	93/7	4/4	63	6
3	MARIMBA	ADVIBE	ADMETAL	89/11	8/4	0	7
4	BELLS	ADMetal	ADMetal	65/35	16/16	5	0
5	SteelDR	ADMetal	HDCOMBO	61/39	16/4	17	0
6	COLLAGE5	ADVIBE	ADVIBE	56/44	8/16	4	5
7	COLLAGE6	ADVIBE	ADMetal	62/38	4/8	10	5
8	COLLAGE7	HDTRGO	ADMetal	62/38	8/16	4	5
9	USER	HDO-SWXO	HDO-SWXO	100/0	8/8	0	0
10	USER	HDTRGO	HDYAJIO	100/0	8/8	0	0
11	USER	HDO-PWXO	DWO3-A.P	100/0	8/8	0	0
12	USER	DWO3-A.P	HDO-PWXO	100/0	8/8	0	0
13	USER	HDO-PWXO	HDSINO	100/0	8/8	0	0
14	USER	HDO-PWXO	HDSINO	100/0	8/8	0	0
15	USER	HDO-PWXO	HDO-PWXO	100/0	8/8	0	0
16	USER	HDTRGO	HDTRGO	100/0	8/8	0	0
17	PIANO 1	DWO3-A.P	DWO3-A.P	70/30	8/16	2	0
18	PIANO 2	HDCOMBO	DWO3-A.P	54/46	4/8	1	0
19	E.PIANO1	HD8ORHO2	HD8ORHO2	95/5	16/4	1	0
20	E.PIANO2	HD8ORHO2	HD8ORHO2	72/28	8/8	5	0
21	STRINGS1	HDSAWO	HDSAWO	55/45	8/8	6	0
22	STRINGS2	HDSAWO	HDSAWO	55/45	16/16	4	0
23	BRASS1	HDSAWO	HDSAWO	52/48	8/8	6	0
24	BRASS2	HDSAWO	HDSAWO	49/51	16/8	3	0
25	ORGAN 1	ADOrgan	ADOrgan	60/40	16/4	5	0
26	ORGAN 2	ADOrgan	HDSINO	52/48	16/4	1	7
27	SynBass1	HDPW12	HDPW12	0/100	16/16	1	0
28	SynBass2	HDPW25	HDPW12	51/49	16/16	2	0
29	Synth 1	HDSAWO	HDCOMBO	34/66	16/16	0	11
30	Synth 2	HDPW25	HDPW50	48/52	4/4	0	0
31	Synth 3	HDSAWO	HDSAWO	56/44	16/8	0	0
32	Synth 4	HDCOMBO	HDSAWO	0/100	16/4	1	0

TABLE 9: FACTORY DISK KSD-002 SYSTEM:A PROGRAMS, OSC. AND MULTI SOUNDS

PROG#	PROG.NAME	M.S.OSC1	M.S.OSC2	MIX %	OCTAVES	DETUNE	INTERVAL
1	A.SAX 0	A.SAX	A.SAX	100/0	8/8	0	0
2	A.SAX 2	A.SAX	A.SAX	100/0	16/8	0	0
3	A.SAX 3	A.SAX	A.SAX	100/0	16/16	0	0
4	A.SAX 4	A.SAX	A.SAX	100/0	8/8	0	0
5	T.SAX 1	A.SAX	A.SAX	100/0	8/8	0	0
6	V.Switch	T.SAX	A.SAX	100/0	8/8	0	0
7	SAX 2	T.SAX	A.SAX	70/30	8/8	1	0
8	T.SAX 4	T.SAX	T.SAX	100/0	8/8	0	0
9	SAX Ens1	A.SAX	T.SAX	50/50	8/8	0	0
10	SAX Ens2	A.SAX	T.SAX	50/50	8/8	2	0
11	A.SAX 1	A.SAX	A.SAX	100/0	8/8	0	0
12	A.SAX 2	A.SAX	A.SAX	63/37	8/8	2	7
13	T.SAX 1	T.SAX	T.SAX	100/0	8/8	0	0
14	T.SAX 2	T.SAX	T.SAX	53/47	8/8	0	0
15	SAX Ens1	A.SAX	T.SAX	60/40	8/8	6	0
16	SAX Ens2	A.SAX	T.SAX	60/40	8/8	5	0
17	Organ[Y]	HDSAW0	HDSAW0	52/48	16/16	4	0
18	Sun Rise	ADCOMBO	ADMetal2	40/60	16/16	2	0
19	E.PIANO1	AD8ORHO2	ADMetal2	85/15	16/4	1	7
20	E.PIANO2	AD8ORHO2	AD8ORHO2	65/35	8/8	5	0
21	STRINGS1	HDSAW0	HDSAW0	55/45	8/8	4	0
22	STRINGS2	HDSAW0	HDSAW0	55/45	16/16	2	0
23	BRASS1	HDSAW0	HDSAW0	52/48	8/8	6	0
24	BRASS 0	HDSAW0	HDSAW0	100/0	16/16	3	0
25	ORGAN 1	ADOrgan	ADOrgan	60/40	16/4	0	0
26	ORGAN 2	ADOrgan	ADSIN0	52/48	16/4	1	7
27	BR3&Bell	HDSAW0	ADMetal2	60/40	16/4	1	0
28	SynBass2	HDPW25	ADOrgan	51/49	16/16	2	0
29	Synth 1	HDSAW0	ADCOMBO	34/66	16/16	0	11
30	Synth 2	HDSAW0	HDSAW0	48/52	4/4	0	0
31	Synth 3	HDSAW0	HDSAW0	56/44	16/8	0	0
32	Synth 4	HDSAW0	HD pf	65/35	16/16	1	0

TABLE 10: FACTORY DISK KSD-002 SYSTEM:B PROGRAMS, OSC. AND MULTI SOUNDS

PROG#	PROG.NAME	M.S.OSC1	M.S.OSC2	MIX %	OCTAVES	DETUNE	INTERVAL
1	TP 1	TP	TP	100/0	16/16	0	0
2	TP 2	TP	TP	100/0	8/8	0	0
3	TP 3	TP	TP	55/45	16/8	0	0
4	TP 4	TP	TP	100/0	4/4	0	0
5	Brass 1	BrassEns	ADCOMBO	83/17	8/16	0	0
6	Brass 2	BrassEns	TP	54/46	8/16	0	0
7	Brass 3	TP	BrassEns	52/48	8/8	2	0
8	Brass 4	TP	BrassEns	52/48	4/8	2	0
9	Sax1	ADCOMBO	TP	95/5	16/16	0	0
10	Brass	HDSAWO	BrassEns	30/70	16/8	0	0
11	TP +	TP	HDSAWO	85/15	16/16	0	0
12	TP #	ADCOMBO	TP	14/86	16/8	0	0
13	Brass 0	BrassEns	ADCOMBO	83/17	8/16	0	0
14	TP >>><<	TP	HDSAWO	82/18	8/8	0	0
15	Brass 1	BrassEns	BrassEns	100/0	8/8	0	0
16	ORGAN 2x	ADOrgan	ADSINO	68/32	16/4	63	11
17	Organ[Y]	HDSAWO	HDSAWO	52/48	16/16	4	0
18	Sun Rise	ADCOMBO	ADMetal2	40/60	16/16	2	0
19	E.PIANO1	AD8ORHO2	ADMetal2	85/15	16/4	1	7
20	E.PIANO2	AD8ORHO2	AD8ORHO2	65/35	8/8	5	0
21	STRINGS1	HDSAWO	HDSAWO	55/45	8/8	4	0
22	STRINGS2	HDSAWO	HDSAWO	55/45	16/16	2	0
23	BRASS1	HDSAWO	HDSAWO	52/48	16/16	6	0
24	BRASS 0	HDSAWO	HDSAWO	100/0	16/16	3	0
25	ORGAN 1	ADOrgan	ADOrgan	60/40	16/4	0	0
26	ORGAN 2	ADOrgan	ADSINO	52/48	16/4	1	7
27	BR3&Bell	HDSAWO	ADMetal2	60/40	16/4	1	0
28	SynBass2	HDPW25	HDPW12	51/49	16/16	2	0
29	Synth 1	HDSAWO	ADCOMBO	34/66	16/16	0	11
30	Synth 2	HDSAWO	HDSAWO	48/52	4/4	0	0
31	Synth 3	HDSAWO	HDSAWO	56/44	16/8	0	0
32	Synth 4	HDSAWO	HD pf	65/35	16/16	1	0

TABLE 11: FACTORY DISK KSD-002 SYSTEM:C PROGRAMS, OSC. AND MULTI SOUNDS

PROG#	PROG.NAME	M.S.OSC1	M.S.OSC2	MIX %	OCTAVES	DETUNE	INTERVAL
1	BRASS 1	HDSAWO	HDSAWO	52/48	16/16	7	0
2	BRASS 2	HDO-SAWO	HDO-SAW3	71/29	16/16	7	0
3	BRASS 3	HDO-SAWO	HDO-SAWO	50/50	8/16	8	5
4	BRASS 4	HDSAWO	HDSAWO	52/48	8/8	7	0
5	BRASS 5	HDSAWO	HDSAWO	52/48	16/16	8	0
6	EXHAUST1	ADVOIC.L	ADVOIC.L	52/48	16/16	6	7
7	EXHAUST2	HDO-XXX1	ADVOIC.L	30/70	16/16	6	7
8	EXHAUST3	ADVOIC.L	HDO-XXX1	71/29	16/4	6	7
9	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
10	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
11	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
12	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
13	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
14	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
15	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
16	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
17	PIANO 1	HD pf	HD pf	70/30	8/16	2	0
18	PIANO 2	ADCOMBO	HD pf	54/46	4/8	1	0
19	E.PIANO1	AD8ORHO2	AD8ORHO2	95/5	16/4	1	0
20	E.PIANO2	AD8ORHO2	AD8ORHO2	72/28	8/8	5	0
21	STRINGS1	HDSAWO	HDSAWO	55/45	8/8	6	0
22	STRINGS2	HDSAWO	HDSAWO	55/45	16/16	4	0
23	BRASS1	HDSAWO	HDSAWO	52/48	8/8	6	0
24	BRASS2	HDSAWO	HDSAWO	49/51	16/8	3	0
25	ORGAN 1	ADOrgan	ADOrgan	60/40	16/4	5	0
26	ORGAN 2	ADOrgan	ADSINO	52/48	16/4	1	7
27	SynBass1	HDPW12	HDPW12	0/100	16/16	1	0
28	SynBass2	HDPW25	HDPW12	51/49	16/16	2	0
29	Synth 1	HDSAWO	ADCOMBO	34/66	16/16	0	11
30	Synth 2	HDPW25	HDPW50	48/52	4/4	0	0
31	Synth 3	HDSAWO	HDSAWO	56/44	16/8	0	0
32	Synth 4	ADCOMBO	HDSAWO	0/100	16/4	1	0

TABLE 12: FACTORY DISK KSD-002 SYSTEM:D PROGRAMS, OSC. AND MULTI SOUNDS

PROG#	PROG.NAME	M.S.OSC1	M.S.OSC2	MIX %	OCTAVES	DETUNE	INTERVAL
1	BASSOON	HDO-SWX2	HDO-SWX2	68/32	16/16	0	10
2	TROMBONE	HDSAW1	HDSAQW3	63/37	16/16	0	0
3	SYMPHORN	HDSAWO	HDSAQW3	52/48	16/16	4	0
4	FLUTE	HDTRGO	ADCOMBO	35/65	4/4	4	0
5	HAMONICA	HDPW50	HDPW25	52/48	4/4	8	0
6	5thSYN-1	HDO-SWX2	ADMetal2	54/46	8/8	0	7
7	5thSYN-2	HDPWMT1	HDPWMT1	54/46	16/16	0	7
8	5thSYN-3	AD8ORHO2	HDPWMT1	54/46	8/8	0	7
9	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
10	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
11	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
12	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
13	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
14	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
15	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
16	USER	HDSAWO	HDSAWO	56/44	16/8	0	0
17	PIANO 1	HD pf	HD pf	70/30	8/16	2	0
18	PIANO 2	ADCOMBO	HD pf	54/46	4/8	1	0
19	E.PIANO1	AD8ORHO2	AD8ORHO2	95/5	16/4	1	0
20	E.PIANO2	AD8ORHO2	AD8ORHO2	72/28	8/8	5	0
21	STRINGS1	HDSAWO	HDSAWO	55/45	8/8	6	0
22	STRINGS2	HDSAWO	HDSAWO	55/45	16/16	4	0
23	BRASS1	HDSAWO	HDSAWO	52/48	8/8	6	0
24	BRASS2	HDSAWO	HDSAWO	49/51	16/8	3	0
25	ORGAN 1	ADOrgan	ADOrgan	60/40	16/4	5	0
26	ORGAN 2	ADOrgan	ADSINO	52/48	16/4	1	7
27	SynBass1	HDPW12	HDPW12	0/100	16/16	1	0
28	SynBass2	HDPW25	HDPW12	51/49	16/16	2	0
29	Synth 1	HDSAWO	ADCOMBO	34/66	16/16	0	11
30	Synth 2	HDPW25	HDPW50	48/52	4/4	0	0
31	Synth 3	HDSAWO	HDSAWO	56/44	16/8	0	0
32	Synth 4	HDPW12	HDSAWO	10/90	16/8	0	0