The Essential Guide to Practical Applications

CASIO FZ-1 & FZ-10M

CASIO® FZ-1 & FZ-10M

Digital Sampling Synthesizer

Joe Scacciaferro
Steve DeFuria

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Casio FZ-1 & FZ-10M
Revisions & Corrections

Due to last minute changes in the numbering of the experiments that appear throughout this book, the following changes must be noted:

Page 48	Experiment # 3 • Step by Step - 1st & 2nd paragraphs - 'Experiment # 3' should read 'Experiment # 2'
Page 51	Experiment # 4 • Observations - 2nd paragraph 'Experiment # 6 ' should read 'Experiment # 5'
Page 54	First Paragraph • (see Experiment # 7) should read (see Experiment # 6)
Page 58	Experiment # 7 • Step by Step - 3rd paragraph 'Experiment # 9 ' should read 'Experiment # 10'
Page 61	Experiment # 8 • Step by Step - 2nd paragraph 'Experiment # 9' should read 'Experiment # 10'
Page 67	Fourth Paragraph • last sentence 'pages ?? - ??' should read 'page 124'
Page 69	Experiment # 9 • Step by Step - 1st paragraph should read '•Sample the phrase, I don't know where I am.' Leave a little space between each word as you speak.
Page 72	Experiment # 10 • Step by Step - 5th paragraph 'Experiment 8' should read 'Experiment 9'
Page 74	First Paragraph • 'Experiment # 14' should read 'Experiment # 13'
Page 81	1st Paragraph • 'Experiment # 15' should read 'Experiment #14'
Page 84	Experiment # 14 • Step by Step - 6th paragraph 'Figure 40' should read 'Figure 41'
Page 92	Experiment # 16 • Step by Step - 9th paragraph 'page ??' should read 'page 79'
Page 94	Experiment # 17 • Step by Step - 1st paragraph 'Experiment 16' should read 'Experiment 15'. Paragraph 3 - 'Experiment 20 should read 'Experiment 19'
Page 95	Experiment # 18 • Step by Step - 3rd paragraph 'Experiment 21' should read 'Experiment 20'
Page 117	9.4 Key Layering with Create Bank • 4th paragraph - (See Page *) should read (page 107)
Page 128	Experiment # 28 • Step by Step - last paragraph 'Experiment 28' should read 'Experiment 26'

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Part 1: SAMPLING BASICS

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1. Overview Of Sampling Process

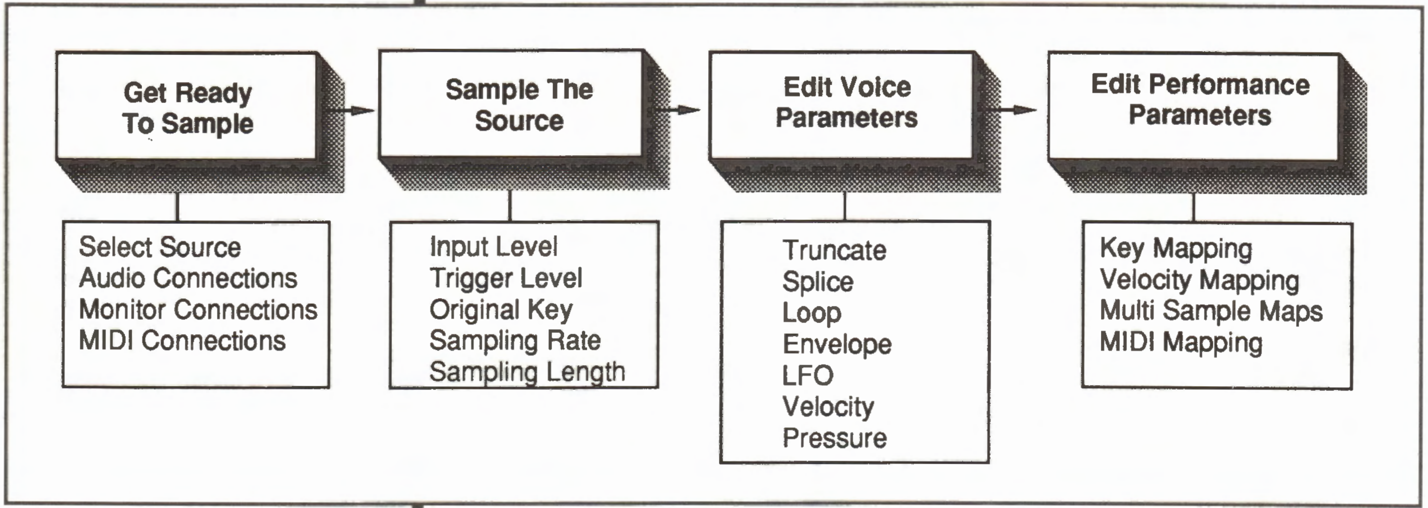

Figure 1: There are four basic steps to the overall sampling process. Associated with each step is a specific set of parameters to use and/or decisions you'll be faced with. We'll take a detailed look at each step.

Your FZ is an incredibly powerful sampling/synthesis instrument. If you're like most new FZ owner's, your primary interest is sampling. That's certainly understandable! There's something very appealing about the idea of being able to capture any sound, and then being able to play it back instantly from your Casio FZ-1 or FZ-10M (kind of a sonic snap shot). You don't need much to plunge right in (the FZ, a microphone, perhaps a disk or two). Once you've gotten your feet wet, you will undoubtedly want to know more about what's going on. Even more to the point, you'll want to know how to use the FZ more effectively and how to get the most out of all of its built-in features and functions.

We've designed this book to bring you quickly up to speed with this amazing instrument and sampling technology. As you can see from *Figure 1*, we've organized the sampling process into a series of steps. In the first part of the book, we will explore these steps as they apply to sampling technology in general. In the second part of the book, each chapter will take you on an in-depth guided tour of the major functions of your FZ-1 or FZ-10M. In fact, you can think of this book as your tour bus of the FZ. We'll go beyond sampling and check out the FZ's synthesis, performance, and MIDI features as well. You can sit back and enjoy the ride, and if you own an FZ, you can get off the bus and explore each of the major sites along the way on your own! You see, for every major FZ feature, we've provided a step-by-step, hands-on experiment for you to try.

No tour would be complete without an itinerary, so here's a list of the main attractions we'll be checking out along the way:

Table of Experiments

As you can see, we have quite a trip lined up for you, so get your FZ-1 or FZ-10M (don't forget the *Operations Manual*) and a mic, and let's go.

Experiment #1: What Is A Sample?

Focus: Sample **Edit** **Performance**

Key Settings:

- Sampling Rate: Default, Sampling Time: 1000 ms
- Original Key, High Key, Low Key: Default
- Input Level: 1/4 up
- Mic or Line: Mic

Operations Manual Page Reference:

- Sampling Operations: 24-30

Step by Step:

- Refer to pages 24 -29 in your *Operations Manual.*
- Plug a microphone into your FZ's sampling input. (Be sure this is the input for microphones and not line-level equipment).
- Turn on the FZ.
- Enter the Sampling: Length Set operation of the Source Select sub-mode. Set the "TIME" value to "001000 ms" (1 second).
- Raise the sampling input level about a quarter of the way up.
- Enter Manual Sampling and push **ENTER** and **YES** when you're ready to sample.
- Say the word "SAMPLE" into the mic the instant you begin sampling.
- Once the sampling process has ended, raise the listening volume of the FZ, and press the "original key" (C5) on the FZ's keyboard.
- Release the "original key" and press the "highest key."(C7)
- Release the "highest key" and now press the "lowest key."(C2)
- Play different notes within the key range.

Observations:

- Following the above process, you have just digitally recorded the word "SAMPLE" and loaded it into your FZ's computer memory.
- Internally, your FZ has taken the the analog signal ("SAMPLE") and converted it to its digital equivalent, which can be recalled and reconverted for playback instantly. Not only can this signal now be recalled exactly as it was originally recorded, but it can also be altered hundreds of ways.
- Every time you play the "original key," you will hear the word "SAMPLE" played back at the same pitch and timbre that was originally recorded.
- You should have also noticed that, when you played the "highest key," that the pitch (or frequency) of the word was shifted up by whatever the musical interval was between the"original key" and the "highest key" (and of course, the opposite shift takes place when you play the "lowest key.") The pitch isn't the only thing that shifts. Everything time-based shifts, too, (the length of the word, background noise, etc.).

1.1 What Is a Sample?

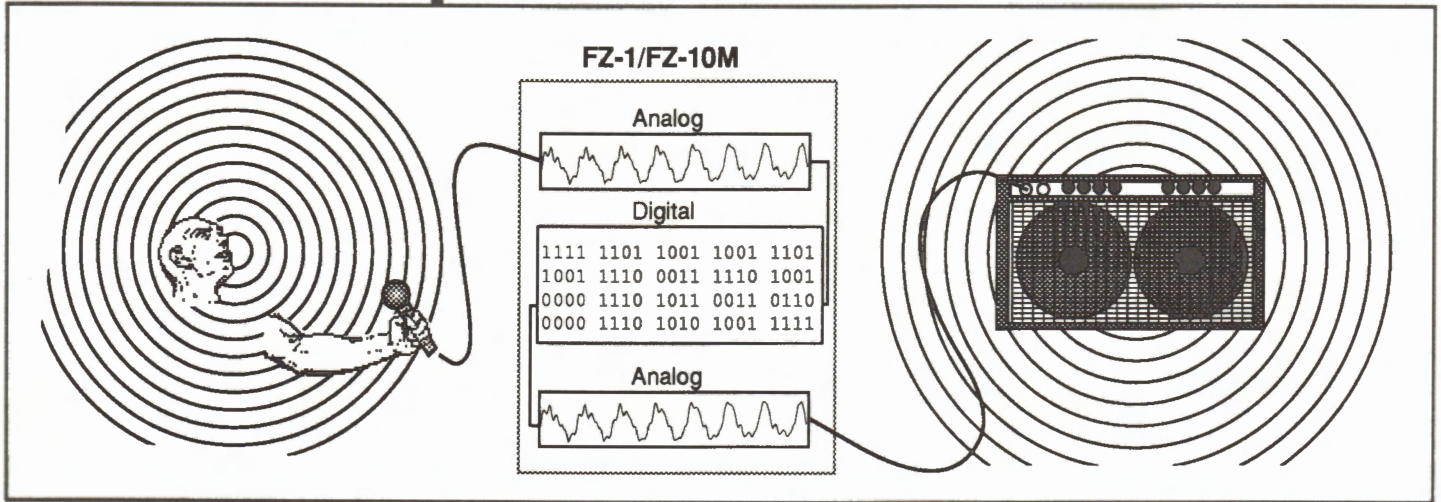

Figure 2: When you sample your voice, the sound is converted to an electronic signal by the microphone. Your FZ converts the electronic signal to digital data. When you play the sample, the conversion process is reversed.

We wanted to start right off with a simple hands-on experiment. Now that you've sampled and played back your voice, let's take a moment to look into what's happening. First of all, the sound you made with your voice has been converted into several different formats. Sound waves were converted into analog electronic signals by the microphone. Your FZ converted those analog signals into digital data. It is this data that is stored in your FZ's memory. When you play the sound back, the FZ converts the digital data back into analog signals again. These are in turn converted back into sound waves by the speakers in your amplification system. Whew! It seems like an awful lot of changes to put a sound through. It makes sense that sound waves have to be converted into analog signals (since they're what amplifiers, mixers, effects, etc. are already geared up to handle), but why change the sound into digital numbers?

One good reason is that once a sound is converted into numbers, those numbers can be stored and recalled very quickly and relatively cheaply. That's because once something exists in a numerical format (i.e., digital data), it becomes fair game for the technology marvel of our age, the micro computer. Yes, as you're no doubt aware, there's a 16-bit micro lurking in the depths of your FZ. The micro and its software manage the sampling process for you. Does this mean you have to be a computer star to work your FZ? Definitely not! Like so many other products, from microwaves to talking teddy bears, you need never think about that micro inside, but it's nice to know it's there

Another good reason why FZs use sound in a digital format is that it is possible to do some very interesting things to the data in between the time it is first stored and when it it recalled. For example, changing the speed at which the data is recalled will change the pitch of the sound. That's how you're able to play different pitches with the FZ. Each key you press changes the rate at which the data is read . If it is read at the same speed it was sampled, you hear the original pitch. If it is read at a higher speed, the pitch goes up. Lower speeds make the pitch go down.

The FZ will let you reverse the order in which the data is read, as well as the speed. Reversing the order makes the sound backwards. Of course, just changing pitch or playing sounds forwards and backwards is only the beginning of what your sampler can do. The FZ will let you play with the sound in a variety of interesting ways. It provides a host of digital and analog voice editing features. Some of the more exciting digital editing features include looping, splicing, reverse play, and truncate. Analog editing features include filters, amplifiers, envelopes, and more. *Bank Edit* and *Effects/MIDI* functions allow you to control your samples and synthesis voices with your performance, making possible such things as velocity and pressure dynamics,velocity cross-fades, multi-switching, and multi-channel MIDI operation. (All and all, more than 200 FZ features will be explored in detail in the upcoming pages.)

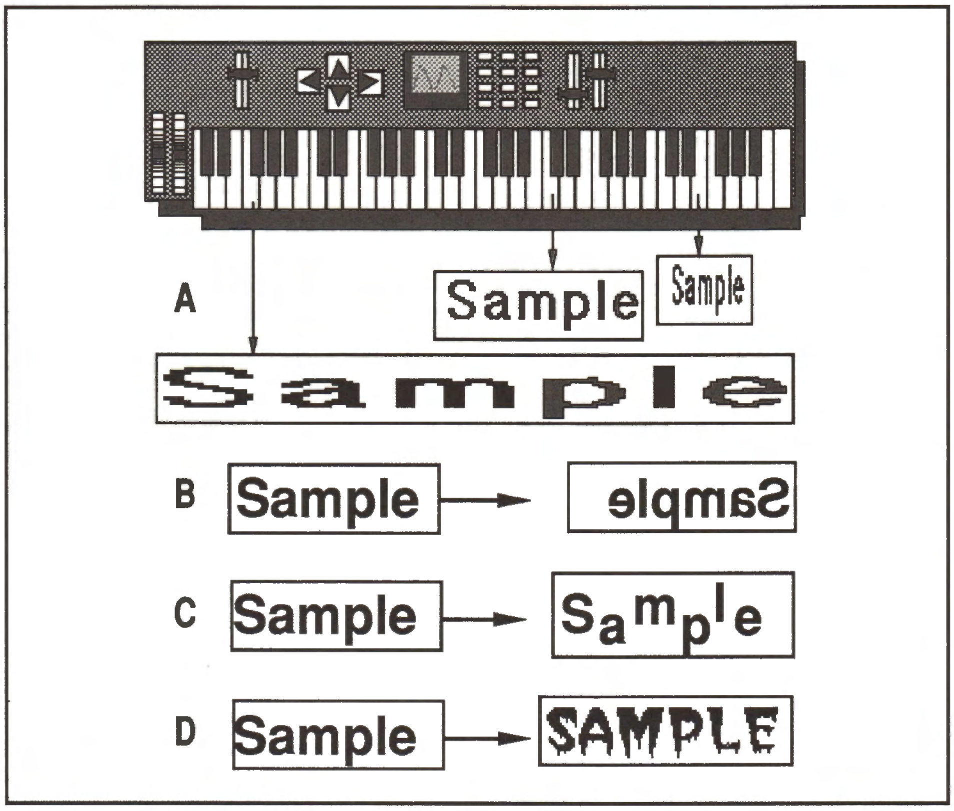

Figure 3: Typical ways of altering a sampled sound. A: Playing different keys speeds up or slows down the rate the sample is played back, changing its pitch and length. B: Samples can be played forwards or backwards. C: An LFO can also vary the sample's pitch. D: Synthesis functions can transform the sonic character of the sample.

1.2 What Are the Basic Steps Involved in Creating a Sample?

Creating samples is a process that involves several distinct steps. Often, at each one of these steps there are decisions that must be made. (See *Figure 1*.) You'll learn that often the decisions are based on trade offs. ("You can have more of this, but you'll have to give up some of that").

The basic steps for creating samples with any sampler are:

1. **Get Ready to Sample**
2. **Sample the Sound**
3. **Edit Voice Parameters**
4. **Edit Performance Parameters**

Each of these steps is distinct from the others and involves its own sets of tools and techniques. You don't necessarily have to learn them all to do great things with your FZ. You may find, for example, that your main interest lies in editing voice parameters, or sampling unusual sources. We present each of these areas separately so you can easily isolate and concentrate on what interests you most. Let's first get familiar with sampling in general. Then we'll go on to thê specifics of how to apply this to the FZ.

1.3 Sampling Buzzwords

The FZ, like all sampling instruments, is an offspring of computer, synthesis, and audio technologies. They have inherited several high tech terms to describe their features and functions. The decisions made at each step in the sampling process are each associated with a key sampling concept. Each of these concepts is described by a sampling buzzword. You will see them again and again throughout the book, as well as in the *Operations Manual.*

Some of these terms are used almost universally by the various manufacturers of sampling instruments, while others are not. In presenting FZ sampling techniques to you, we have tried to stick with the terms as they are presented in your *Operations Manual.* There are some commonly used terms, like *velocity cross-fade* or *splicing,* which you won't see in the *Operations Manual* . The FZ does indeed have velocity cross-fade and splicing features; they're just not labelled as such. Throughout the book, we have tried to use the most commonly used terms to describe any given feature. When FZ and common usage terms differ, we use both. By the way, one reason why the FZ's terminology is somewhat different from some common buzzwords is that the FZ's version of the function generally has more features than associated with the "normal" term. A good example of this is the FZ's *Mix Write* and *X-Mix Write* functions. Yes, they are used for splicing, but as you'll soon see, they do a lot more than a simple splice.

Too often, imposing terminology will stop us dead in our tracks when we're learning something new. If we are comfortable with the terms, then we generally have no trouble mastering the subject at hand. The only way to get comfortable is by defining the buzzwords. Definitions may themselves be confusing when taken out of context, so we have opted not to provide a quick and dirty glossary of terms here. Instead, we've used the buzzwords as chapter and topic headlines where appropriate. You'll find them easily by simply scanning the Table of Contents or just flipping through the book. You'll get a much better grasp on the significance of these terms if you see them within the overall context of how they are applied.

2. Get Ready To Sample

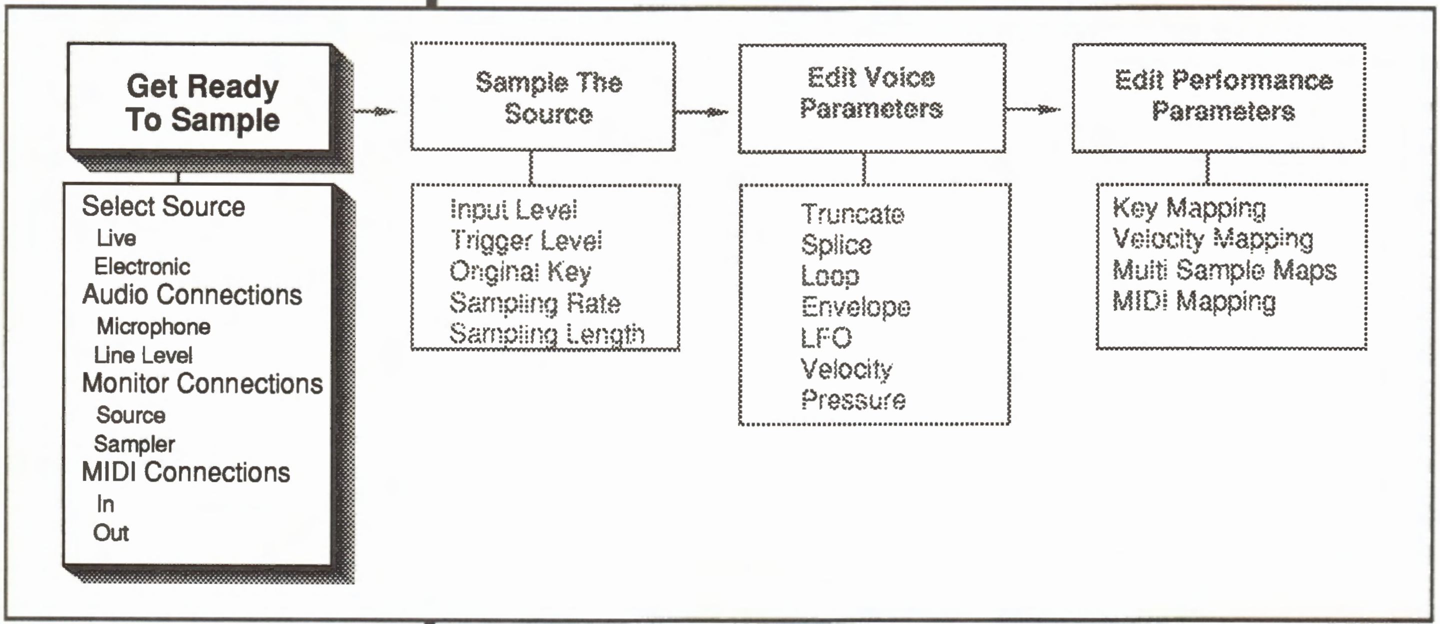

Figure 4: The first step in the sampling process involves making decisions about what to sample and how your FZ and other equipment will be inter-connected.

Before you start sampling, you must take care of details, such as deciding what it is you want to sample and setting up your equipment. If, for example, you want to sample your speaking voice, this can be as simple as plugging a microphone into the **Mic Input**, and its **Mix** or **Line Outputs** to an amplifier/speaker.

However, if your FZ is just one component of your own music studio, you may want to take full advantage of the other equipment you have. Even with a modest collection of equipment, such as tape recorders, CD players, and mixers, your setup can become considerably more sophisticated.

Let's look at each of the decisions to be considered in this first step of the sampling process.

- **What do you want to sample?**
- **How will you get the sound from the source into the FZ?**
- **How can you listen to the sound and the samples you'll make?**
- **Do you need to make any MIDI connections? This might be necessary if you're using the FZ-10M (the rack-mount version of the FZ).**

Figure 4 outlines the typical options you have for each choice.

2.1 Selecting a Source

Live Sounds

There are two general sources of sound to load into your sampler. One is acoustic sound, i.e. live sound that you can capture with a microphone. This would include voices, acoustic musical instruments, ambient sounds: traffic, crowd noises, airplanes, etc., as well as specific sound events: a door slam, breaking glass, a gunshot. In short, virtually anything you hear can be picked up by a microphone. If the mic is connected properly to the instrument, then you can sample just about anything.

Line Level Audio

The other source of sounds for your FZ is line level audio. By this, we mean the electronic signals generated by such devices as tape recorders, CD players, radios, electronic instruments (like electric guitars, synthesizers, and even samplers). The **Line Level Input** on the FZ can be connected to just about any line level source, allowing you to sample directly from recordings, synthesizers, and other electronic instruments, mixers, etc.

Sampling Live Sounds vs. Sampling Recorded Sounds

When you're first learning your way around, or if you want to work fast, you may find it most convenient to simply plug a mic into the **mic input** and then sample live sounds. This is a perfectly good way to learn the basics. If you're careful and precise, there's no reason why you can't create the professional quality samples this way.

However, as your sampling skills increase and you begin to create more sophisticated samples, you'll probably find that you don't want to connect the source directly to the instrument. For instance, it may be impractical to bring the FZ to the source. Suppose you want to sample the sound of a jet taking off, or the crash of the surf onto a rocky coast? You wouldn't want to bring your FZ-1 or FZ-10M to the airport or the beach. In situations such as these, you can collect your source material with a microphone and a good quality portable tape recorder. Then you can sample the sounds directly from the tape recorder in a more convenient setting, whether it's your living room or your own personal-use music studio.

Another reason not to sample direct is insurance. Suppose you want to sample a very special sound, say the noise a fine crystal vase makes upon colliding with a concrete driveway at high speed. Unless you have an unlimited supply of crystal vases, you'll probably have to get this right the first time. If the FZ isn't set up properly or the sampling begins a little late or the sound takes longer than you thought, you'll end up with an incomplete and possibly unusable sample. So how do you get insurance? Once again, the best approach to take is to record the sound with a tape recorder. In general, a tape recorder is easier to set up, and there's so much recording time on a reel of tape that you don't have to worry about how long the sound will last, or about starting the recording precisely in sync with the event. Sample directly from the tape. Since the recorder will play back the original sound exactly the same way each time, you'll be able to adjust the FZ with great care and precision. If you don't get it right the first time, no problem; just rewind the tape and start over.

Although the examples given above are extreme, they're not all that unusual. Many sampling artists go to great lengths to build up collections of "found" sounds. You may find that such sonic material is a rich source of creative ideas. If you don't have the facilities, time, or dedication to create such a collection, Casio and others have been busy doing it for you. Casio has a complete presampled library of FZ sounds (FL disks), and an additional set of FZ sounds is offered by Soundwaves as well (FLS disks).

Other companies also supply sampling libraries in FZ format or as high quality, ready-to-sample audio recordings. For example, Sound Ideas offers a sampling library of over 3,000 musical sounds and sound effects recorded on CDs. For more information on what sounds are available and how you can acquire them, just contact your local Casio dealer.

2.2 Connecting the Source to the FZ

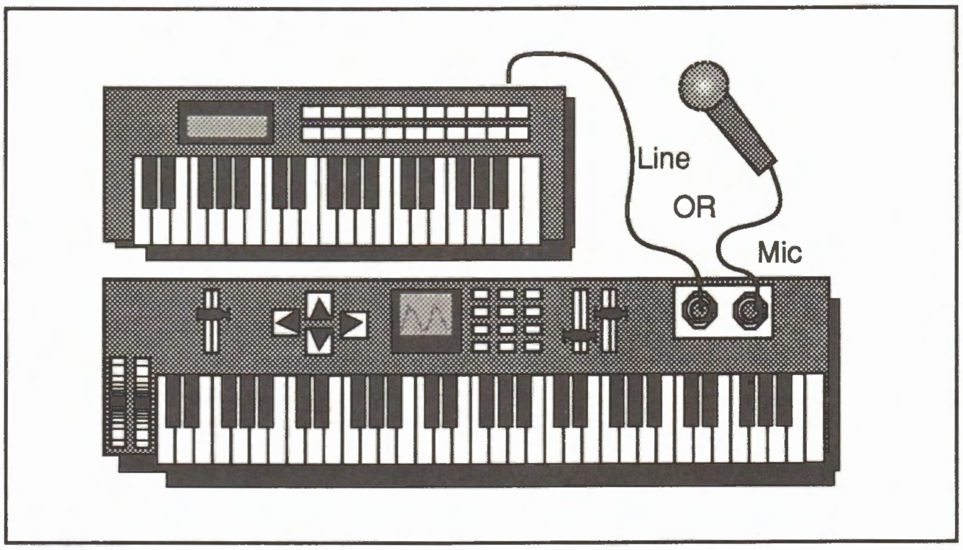

Figure 5: Your FZ has two separate input jacks on the rear panel–one for mic, another for line.

Microphone Inputs

The FZ-1 provides separate inputs for mic and line levels; any high impedance mic can be connected to a jack labelled **Mic Input**. It is a standard 1/4" female phone plug. This is the standard connection for high impedance (Hi-Z) microphones. The FZ-10M also offers a female 3-pin "Amphenol" connector. This connector (also called a "Cannon" plug) is the standard connection for low impedance (Lo-Z) microphones.

Make sure that the mic you are using is the proper impedance for your sampler. Chances are, if the connectors match, so does the impedance. If you're not sure, check the manual that came with the mic. If the impedances/connectors don't match, you will need a transformer/adaptor to change the mic's impedance to match the sampler's.

Line Level Inputs

The FZ-1 provides a 1/4" high impedance jack for its line level input. Although this connection is standard for most line level devices, you may come across some equipment that does not follow this standard. Once again, we recommend you check with the *Operations Manual* of each device you intend to use. And use adaptors or transformers when necessary.

Experiment #2: Mic Or Line Level

Focus: Sample Edit Performance

Key Settings:

- Sampling Rate: Default, Sampling Time: 1000 ms.
- Original Key, Highest Key, Lowest Key: Default
- Input Level: 1/4 up
- Mic or Line selector: refer to **Step by Step** below

Operations Manual Page Reference:

- Sampling Operations: 24-30

Step by Step:

- Plug a microphone into your FZ's sampling input (be sure this is the input for microphones and not line level equipment).
- Turn on the FZ.
- Enter the Sampling: Length Set operation of the Source Select sub-mode. Set the "TIME" value to "001000 ms" (1 second).
- Raise the sampling input level about a quarter of the way up.
- Enter Manual Sampling and push **ENTER** and **YES** when you're ready to sample.
- Say the word " MIC" into the mic the instant you begin sampling.
- Before making the next sample, enter Define Voice and set the value of "VOICE No." to "02."
- Remove the microphone from the "mic" input, and place it into the "line" input.
- Without making any changes in the sampling parameters, sample the word "LINE."
- Push **PLAY** and use the **DOWN** button to select "VOICE No."
- Play and listen to the two samples you've made. Use the **YES/NO** buttons to select between voice number 1 and voice number 2.

Observations:

- You will notice that the major difference between the sample made at the Mic input and the sample made at the Line input is that the line seems to be lower in volume. This is because the mic input is much more sensitive than the line input. Since a microphone produces such a low signal, it needs to go through an input that will boost a signal before it is recorded.
- Another point to mention is that, when comparing the two signals, notice the noise level of both signals. You will find that, when you attempted to record the weak signal through the line input, not only did you lose signal but you gained noise.
- Try the same experiment, but this time, instead of sampling your voice, try using the output of a tape recorder, synthesizer, or other line-level device.
- Be sure to learn how tc set the best levels, regardless of whether you're sampling a mic or line-level source. (That's covered in **Experiment #3**.)

2.3 Listening to Your Work

Of course you want to hear your FZ. You will also want to listen to your source as well. This is particularly important if you are sampling from a line level source. You should provide a way to monitor the output of the source so that you can hear what's being sent into the FZ.

The key to setting up your monitoring scheme is clean and accurate sound quality. At the initial stages of the sampling process, you will want to compare the sound of the original source to your sample. (This is called an "A/B " test.) For most situations you will want the sample to sound as close to the original sound as possible. In order for your ears to fairly judge the accuracy of your sample, the amplifier/speaker combination you use should be of the highest possible fidelity. Both the source and the FZ should be played through the same system to ensure that any coloration introduced by the amp and/or speakers is applied equally to both.

Audio processors and effects like reverberation, chorus, EQ, compression, etc., are all very useful, but be sure that you can take them out of the system when you don't need them. For now, we'll assume you are sampling without any effects.

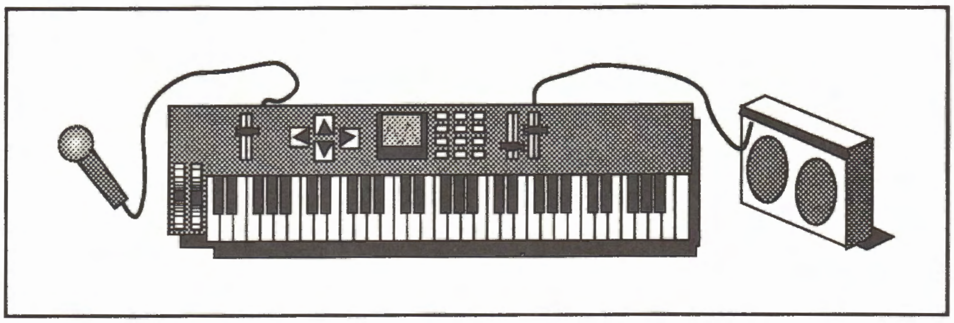

Figure 6: For most of the experiments in this book, all you need is your FZ, a mic, and an amp (or headphones). A tape recorder will also come in handy if you have one.

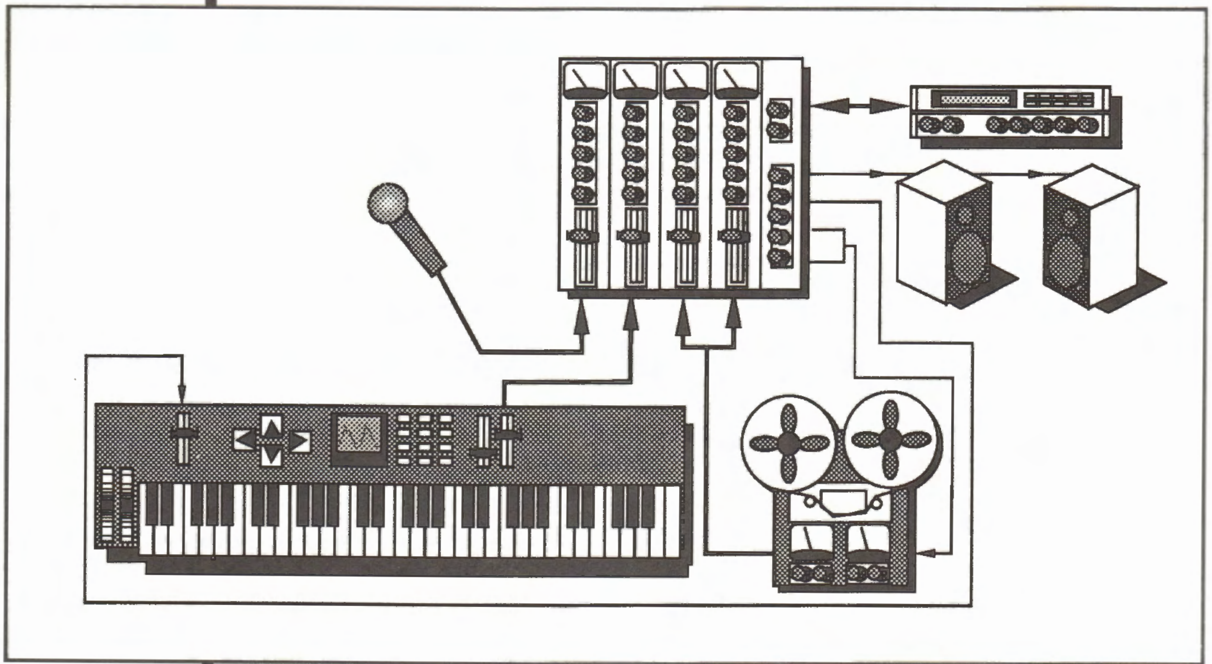

Figure 7: A more sophisticated FZ sampling setup would include a mixer, tape deck, outboard effects, and a pair of monitor speakers.

2.4 MIDI Connections

On the back of your instrument you will find three 5-pin connectors–MIDI IN, OUT, and THRU. These three ports provide you with the ability to communicate not only with other FZ's but with any instrument or computer that incorporates MIDI. You won't need to use any MIDI functions when you are simply creating and listening to samples, unless you are using the FZ-10M (this is the rack-mounted version of the FZ and does not have a keyboard). The *MIDI Implementation Chart* that came with your FZ lists all of its MIDI functions.

Simply put, MIDI is the means that allows your FZ-10M (or FZ-1 for that matter) to be played from another keyboard. If you want to be able to play voices on the FZ-10M with different pitches or in chords, then you will need to control it from a MIDI controller of some type. For our examples here, we are using a keyboard, but it could just as easily be a MIDI wind controller, guitar, percussion controller, or a sequencer.

In order to connect your FZ to a MIDI controller:

- Connect a MIDI cable from the MIDI OUT port of the controller to the MIDI IN port of the FZ.
- Set the FZ's *Receive* parameter to "BASIC" and set the *Basic CH* parameter to the same MIDI channel that the controller is transmitting on.

OR

- Set the FZ's Receive parameter to "AREA " and set the *Areas* in the *Bank* you wish to use to the same MIDI channel(s) that the controller is transmitting on.

If you're not sure how to do this, consult pages 97-99 and 84-85 in the *Operations Manual* for the details. If you're not sure why this is necessary, when you get a chance, take the time to learn more about MIDI.

> If you are unfamiliar or uncomfortable with MIDI, we strongly recommend that you get hold of our book, ***The MIDI Book***, for a complete and unintimidating look at all things MIDI. If you want to know even more about MIDI from a technical point of view, then by all means check out our **MIDI Reference Series**.

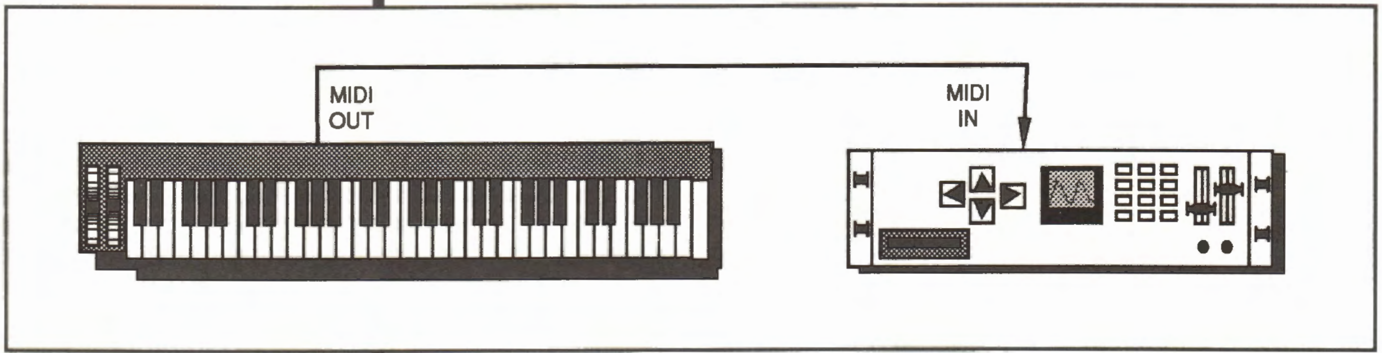

Figure 8: Here's the basic connection to make to control an FZ-10M from an external MIDI controller. You only need to connect the controller's MIDI OUT to the FZ's MIDI IN.

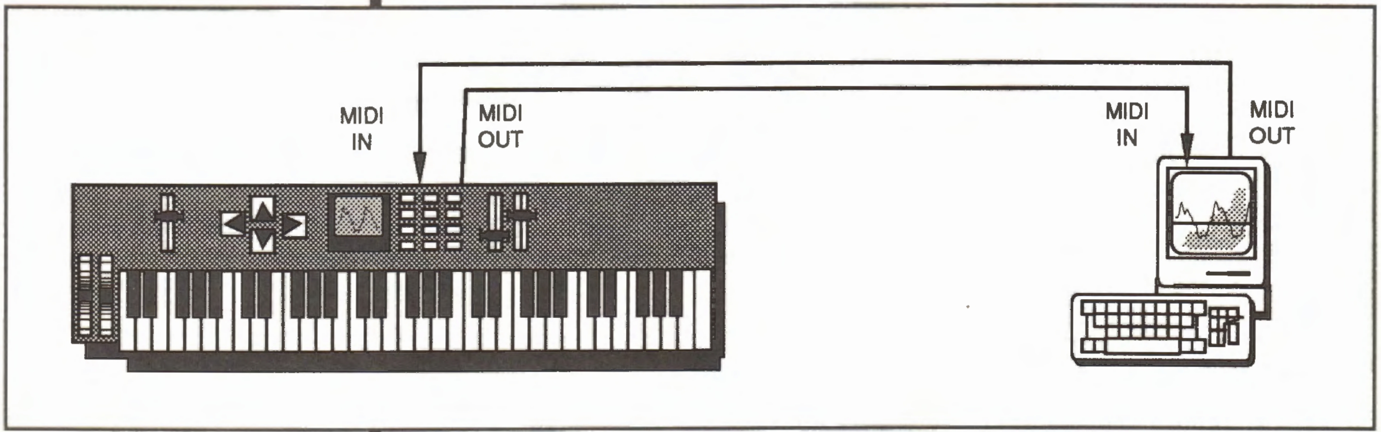

Figure 9: When you use a MIDI/computer system with the FZ, you'll need to connect both the MIDI IN and MIDI OUT of the FZ to the computer's MIDI interface, since both the computer and the FZ will exchange data with each other.

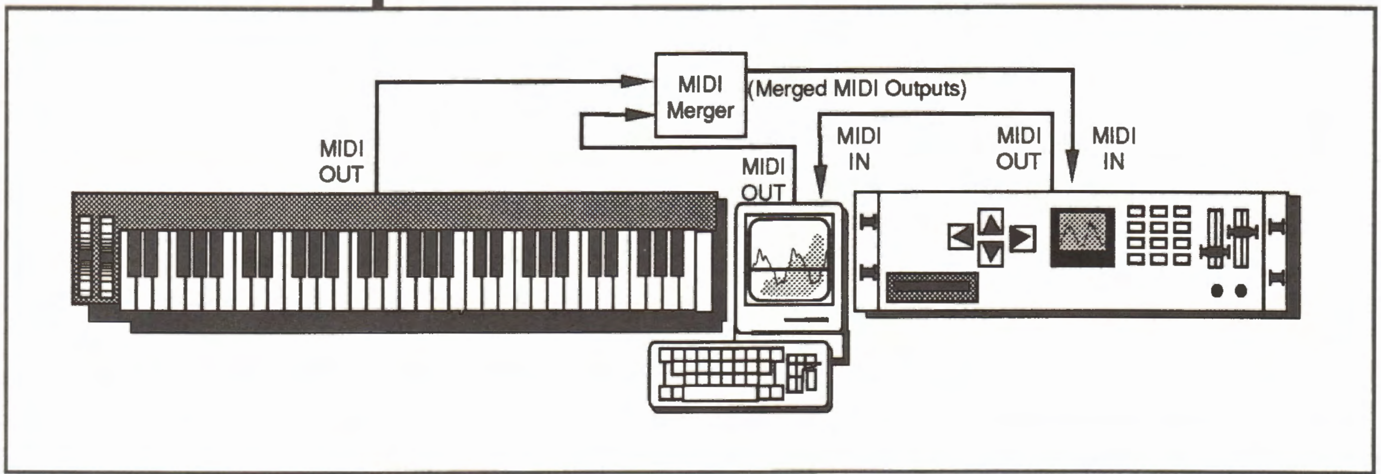

Figure 10: If you want to play a rack mount FZ from a keyboard *and* use the computer at the same time (for sequencing, etc.), you may have to merge together the MIDI OUT of both the computer and the controller. Send the merged signal to the FZ's MIDI IN.

3. Let's Get Technical

When you really want to get creative with a musical instrument, it helps to understand as much as possible about how it works. Learning how something does what it does isn't necessary for you to be able to play it or use it. After all, you don't have to know what makes a jet go if you want to fly, and you don't have to know how an electric guitar works if you want to play rock 'n' roll. But if you want to push that jet through high-G loops or make that guitar scream, the more you know about the technology you're using, the more creative you can get with your toys.

With that in mind, we feel that it's important to provide you with the details of how and why your FZ does what it does. (Remember, all samplers use the same technology, so the principles we're explaining here apply to all samplers.) Once you understand these basic fundamentals, you'll have a better appreciation for how your FZ works. We feel that understanding is an essential key to unlocking many creative options.

> If you're in a hurry to get on with our tour of FZ features, then by all means jump ahead to **Part 2**. When you're ready to get technical, come on back!

By now you might have some questions about all of this, such as:

- How is sound converted into an electronic signal?
- How is the electronic signal converted into a series of numbers?
- The sound is converted from acoustic to electronic, to digital, to electronic, and finally to acoustic again before we ever hear it. Doesn't something get lost in all that translation?
- Why does the sampling rate effect the fidelity of my samples and the amount of sampling time I have?
- Why does the number of bits a sampler uses effect its dynamic range?

All of these questions (and many others that will occur to you along the way) can be answered if we take a look at what is really going on inside the FZ. Earlier, we introduced you to the basic sampling process (*Figure 2*). Let's review the steps involved in turning a sound into a sample.

1. The sound, which is an acoustic waveform, must first be converted into an electronic representation of the sound wave by a microphone.
2. The electronic waveform (analog audio) is sent to the FZ's input, where it is converted into a digital representation of the wave.
3. The digital representation of the sound, a series of numbers, is stored in the FZ's memory.
4. To play the sample, the digitizalized representation of the sound must be converted back to an electronic waveform.
5. The electronic waveform is sent from the FZ's output to an amplifier and speaker(s).
6. The speakers convert the electronic waveform back into an acoustic waveform.
7. We hear the acoustic waveform as sound.

Ultimately we need to know what is going on inside your FZ, where sound is represented digitally (as a series of numbers). However, when you look at the steps involved in turning a sound into a sample, we see this isn't where the process begins. Before the sound can be digitalized, it must first be converted into an electronic signal. Before that ever happens, something must produce a sound. So where do we begin? At the beginning, of course.

3.1 What Is Sound?

Not surprisingly, everything revolves around sound. You're turning sounds into samples and then turning the samples back into sounds again. Any changes you make along the way (whether they're planned or accidental) will ultimately effect the sound of the sample when you play it back. So let's begin here by defining what sound is.

From a subjective point of view, sound is anything we are capable of hearing, but what stimulates our sense of hearing in the first place? Our ears respond to very small changes in air pressure. These pressure changes are called sound waves.

Sound waves are disturbances caused by something vibrating in a medium (like air or water). When that tree falls in the woods (whether there is anyone around to hear it or not), a series of sound waves are generated as surely as a series of water waves would be generated if it fell into a pond (whether or not there was anyone there to see the ripples in the water).

Sound has three unique subjective properties: pitch (high-low), timbre (bright-dark), and loudness (loud-soft). These three attributes of sound are mutually independent, but must each be present to some degree in order for us to actually hear anything. They correspond to three physical properties: frequency (pitch), waveshape (timbre), and amplitude (loudness).

Sound waves can be depicted graphically in a variety of ways. The most common method (and often the most useful) of drawing a sound wave is to show amplitude as it changes over the passage of time. Vertical distance on such a graph (called an oscillogram) shows a wave's amplitude. Note that amplitude is shown above *and* below the the zero line. If we are talking about acoustic waves, amplitude represents loudness. It represents voltage for electronic waveforms and a numeric value for digital waveforms. In any case, the graph of the waveform will look the same regardless of whether it is representing acoustic, electronic, or digital information. Horizontal distance shows the passage of time. Waveforms from pitched sources of sound show up as repeating patterns. The number of patterns in a given space is frequency, and the shape of the individual pattern is waveshape (which we hear timbre). *Figure 11* illustrates how these different sound parameters look graphically. The FZ's display also shows a representation of sound waves using this format.

> This isn't the time or place to go into a complete discussion all the subtleties and nuances of what makes up sound. There's so much to explore we could write a book about it. In fact, we did! If you'd like to know more about the properties of sound, and in particular, how they apply to music and synthesis, we strongly recommend ***Secrets of Analog and Digital Synthesis*** by **Ferro Technologies**, which is available as both a book and a two-hour video course.

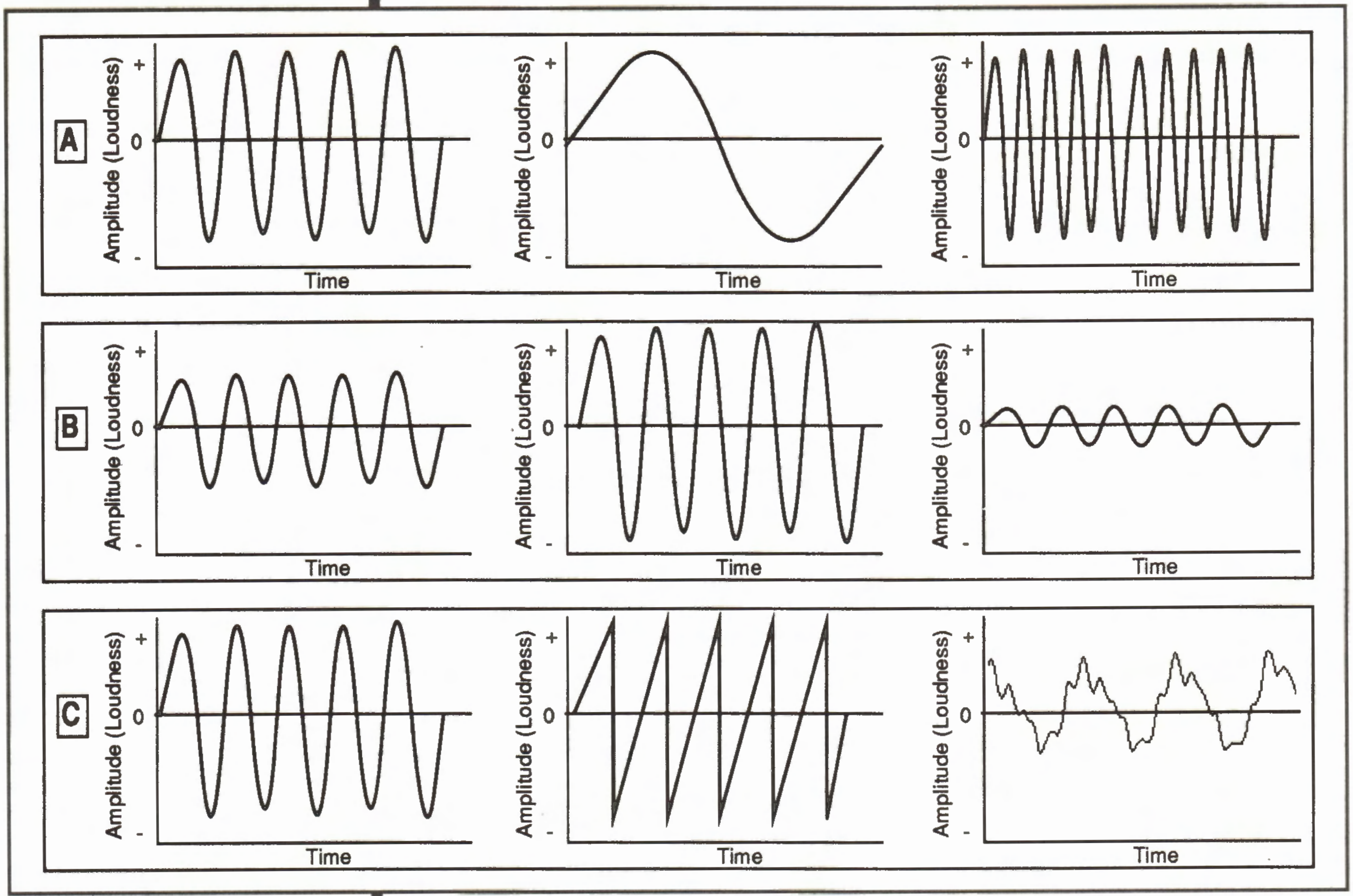

Figure 11: Waveform parameters. The three waves in row A differ only in frequency. They have equal amplitude and are the same shape (sine wave). The waves in row B are identical except for amplitude. The first two waves in row C differ only in waveshape. The final wave in row B differs in all characteristics from the other two. Not only is it a different shape, its frequency and amplitude are different as well.

3.2 Changing Sound into Electronic Signals

A microphone is a device made to convert small changes in air pressure (sound) into small changes in electrical pressure (voltage). Essentially, it is a very simple machine. A very thin membrane, called a diaphragm, can move in and out in response to changing air pressure. When there is no sound present (and therefore the ambient air pressure is constant), the diaphragm remains stationary. When the pressure becomes higher than normal, the membrane is pushed inwards. When the pressure becomes lower than normal, the membrane is pulled outwards.

Microphones are designed so that whenever the diaphragm moves from its normal position, an electrical voltage is produced. (There are various methods for doing this, thus the various types of microphones: carbon, dynamic, condenser, electret condenser, etc.) The farther the mic's diaphragm is displaced (in either direction), the higher the voltage produced. When the diaphragm is pushed inwards, the voltage is positive. When it is pulled outwards, the voltage is negative.

That's how a microphone converts sound, a continuously changing pressure wave, into audio, a continuously changing electrical wave; pressure changes are converted directly into voltage changes. The term *analog* is often used to describe this kind of smoothly changing electrical signal. If we were to graph both the sound wave and the electrical output of the microphone, we'd see that both graphs were identical (*Figure 12*). (Given a perfect microphone, of course)

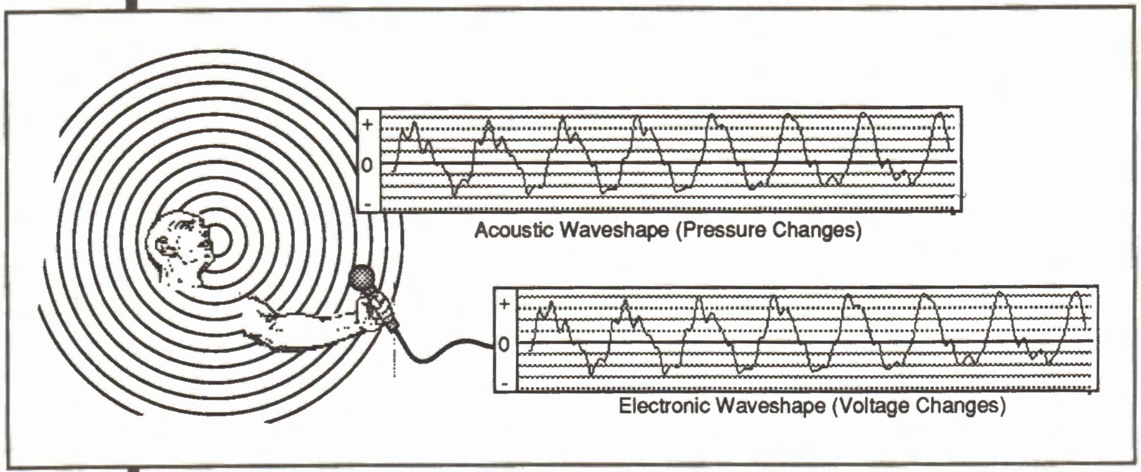

Figure 12: A microphone converts changes in air pressure (sound) into changes in voltage levels (analog audio). The pressure and voltage changes correspond to each other in a simple, one-to-one manner. Graphing either the acoustic or electronic signal will produce the same image.

The outputs of the other potential sampling sources we've mentioned, such as tape recorders, CD players, electronic musical instruments, etc., all produce the same kind of continuously changing analog electronic signals. You should be aware that these electronic signals, i.e., analog audio, are what your FZ is actually sampling.

3.3 Changing Electronic Signals into Digital Samples

So, as we've already mentioned, the sampler actually converts an analog electronic signal into a digital signal. How exactly is that done? Even though we use the FZ to record and playback sounds, in some ways it has more in common with a motion picture camera than it does with a tape recorder. So much so in fact, that it's worth taking a brief look at how movies are made to find out how samples are made.

In a sense, the the camera is a sampling system, and the film is its memory. Instead of sampling sound, it samples images. Its internal mechanism periodically "looks," via a system of lenses, at whatever it happens to be pointed at. Each of these glimpses lasts for only a brief instant. That single instant is captured on a single frame of film. After a short period of time, the camera advances to the next frame, looks again, and this new moment is recorded there. The exposed film becomes a series of these frozen moments–each one showing a sample of what was going on the instant it was exposed. The process continues until you stop the camera or run out of film.

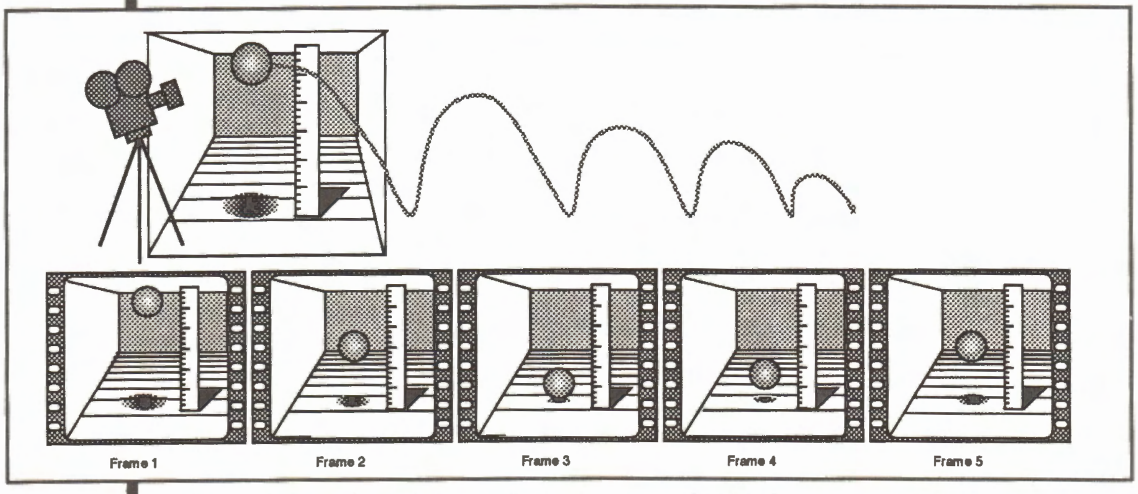

Figure 13: A movie camera periodically "looks" at what it is focused on and records continuous action as still pictures. The movie consists of a series of frames. Each frame is analogous to a single sample.

What's interesting about this process is that it creates a partial record of what occurred in the real world. Although we call them motion pictures, the motion isn't recorded on the film. Each frame is an individual still picture. We only perceive motion when the frames are displayed very rapidly, one after the other.

Suppose, for example, that you made a movie of a ball bouncing alongside a ruler (*Figure 13*). As you watch the scene live, you see the ball move up and down in one continuous, uninterrupted motion. If you examine the frames of the movie, however, you 'll find that each one is just a still picture of the ball and the ruler. If you look closely, you'll see that the ball's vertical position changes slightly from frame to frame. When you compare the ball's position against the ruler, you'll see that there are indeed gaps between its location from frame to frame. So how come when the movie is shown, you see motion?

You've just hit upon the fundamental premise of sampling systems. *It is not necessary to capture motion in one continuous (i.e.,. analog) stream.* As long as enough individual samples are taken in a given amount of time, it is possible to recreate the original motion by rapidly playing back the samples in their original order. If they are played back at the same speed they were taken, the motion will appear to be the same as it originally occurred. If they are played back slower, the motion will appear to be slowed down, and if they are played back faster, the motion will appear to be speeded up. You can also change the order in which they are played–for which your movie may earn critical acclaim and an Oscar for editing.)

Let's stay with the movie analogy of sampling a bit longer and see if we can come up with the critical factors that effect how well (technically, not artistically) a given movie will turn out.

1. **Sampling Rate** (the rate at which progressive frames of film are exposed): The higher this rate, the more accurate the representation of action will be. Actions will seem more fluid as more frames are exposed in a given period of time. Actions will seem jerky as fewer frames are exposed in the same period of time.
2. **Sampling Resolution** (the resolving power of the film): In other words, how precise will the images recorded on each frame be? With motion pictures, resolution is measured by the frame size. 70 mm has higher resolution, and potentially sharper images, than 8 mm. We emphasize the word potentially because frame size alone does not determine the quality of the imagery. Other factors, such as the optical system, will also effect overall image quality.
3. **Sampling Memory** (the amount of film available): This will ultimately determine the length of the movie you can take.

Nothing very hard to understand here, is there? It all makes a lot of sense. If you've ever seen any super slow-motion photography or those high-speed silent skits on "The Benny Hill Show," you've seen how the rate of exposure effects the way that action is reproduced (sampling rate). All of us have seen the difference in clarity between CinemaScope and Dad's home movies (sampling resolution). You also know that the length of a movie is determined by how much film is used (sampling memory). Guess what? If you've stayed with us so far, then you already understand the fundamental principles that all samplers (including your FZ) use.

Think of the FZ as an electronic movie camera. Unlike the movie camera, however, it has no lenses. Instead, the FZ has an audio input jack. Rather than capturing samples of actions that occur in front of its lens, it captures samples of the analog audio signal being fed to its input. (*Figure 14*)

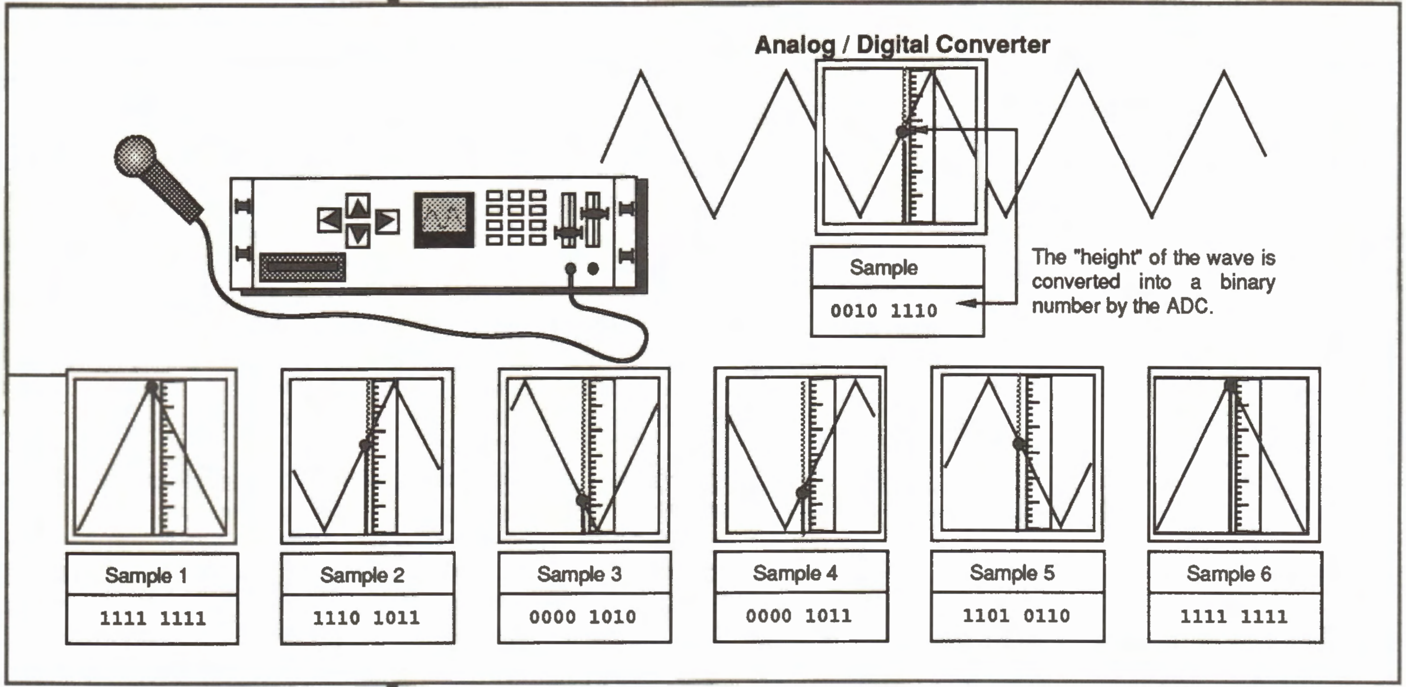

Figure 14: Inside your FZ, the *Analog to Digital Converter (ADC)* periodically "looks" at the input waveform and measures its height (amplitude) with its own built in ruler. A binary number representing the sampled voltage level is stored in a single word of sampling memory. Each of these single samples is analogous to a single movie frame.

Its internal mechanism periodically looks at whatever signal that happens to be connected to its input via an electronic circuit called an *Analog To Digital Converter (ADC).* Each glimpse lasts for only a brief instant.

What is sees is a voltage level moving up-and-down (just like the bouncing ball). This up and down motion is the amplitude changes in the electronic waveform. When it takes a sample, the ADC measures the height of the wave (it compares the level to its own ruler). The measurement is, simply enough, a whole number–big numbers for large amplitudes, small numbers for small amplitudes. Plus (+) and minus (-) signs represent positive and negative voltage. (Samplers use the binary version–ones and zeros–of numbers instead of the decimal version we're accustomed to working with. A number is a number is a number, whether you call it "seven," "7," or "0111".) The number is put into a single word of sampling memory. (In this example, a word of memory is to a sampler as a frame of film is to a movie camera.) After a brief period of time, the sampler looks again, measures the input level, and puts the new measurement into the next word of memory. The memory contains a series of these values. Each one holds the level of the input signal at the instant the sample was taken. The process goes on until you stop the FZ or run out of memory.

Sound familiar? As long as enough samples are taken in a given amount of time, it is possible to recreate the original waveform by converting the digital values back into voltage levels rapidly and in the same order that they were sampled. The conversion of digital numbers to voltage levels is done by a *Digital to Analog Converter* , or *DAC* (what else?). If the samples are played back at the same speed they were taken, the sound will have the same pitch (and last as long) as the original. If the playback rate is slower, the pitch will be lower (and the sound will last longer). Increasing the playback rate will raise the sound's pitch (and make it shorter). You see, the camera and the sampler really do work in a very similar way. You can even change the order in which the samples are played. (No, they don't give out Oscars for sample editing; how about a Grammy?)

Samplers and movie cameras are so similar that the three critical factors for taking good movies apply to making good samples as well. All we have to do is substitute some of the film terminology with sampling terminology.

Critical Factors for Making Good Samples

1. **Sampling Rate:** (the rate at which progressive words of memory are written): The higher the sampling rate, the more accurate the representation of the original signal will be. Higher sampling rates will produce samples with better frequency response (fidelity) than lower sampling rates.

2. **Sampling Resolution** (the resolving power of the ADC): In other words, how precise will the value written to each word of memory be? With samplers, resolution is measured by the word size of the *analog to digital converter*, 16 bits has higher resolution, and potentially cleaner audio, than 12 or 8 bits. We emphasize the word potentially because word size alone does not determine the quality of the audio. Other factors, such as the analog input and output circuits, will also effect the overall audio quality.

3. **Sampling Memory** (the amount of memory available): The number of digital words, one word for each sample, will determine the ultimate length of the sound(s) that can be recorded.

Sampling Rate

Sampling rate has a direct effect on the fidelity of a sample. The higher the rate, the better the fidelity. By fidelity, we mean frequency response. Faster sampling rates can capture higher frequencies than slower rates can. This means that higher sampling rates will yield samples with more accurate top end. For any sampling rate there is a best case limit of the highest frequency it can record. This is called the *Nyquist limit*, and it is about one half the sampling rate. Samplers can't record frequencies above the Nyquist limit accurately. They will produce low frequency tones in the sample that were never in the original signal. This undesirable effect is called aliasing . In order to prevent it, the FZ has special anti-aliasing filters built into its input circuitry to remove frequencies above the Nyquist limit.

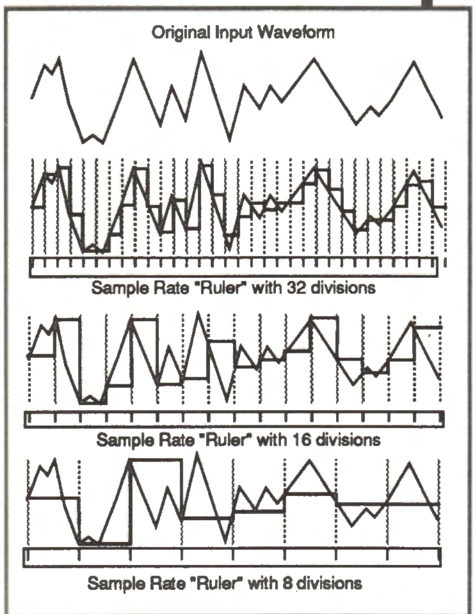

Figure 15: Sampling Rate

Samplers and movie cameras really do have a lot in common. Aliasing can be a problem in motion pictures too. Instead of hearing something that wasn't part of the original sound, you'll see something that wasn't part of the original image. Next time you're watching an old western on the "So Late It's Early Show," watch those stage coach wheels. During the big chase (you know, when it careens out of control through the gulch), you may notice that the wheels look like they're going backwards. You've probably seen this before. If you've ever wondered about it, now you know that it is the visual equivalent to aliasing. It is caused by the wheels turning faster the the movie camera's version of the Nyquist limit.

You should keep in mind that the Nyquist limit is an ideal figure. For instance, given a sampling rate of 36 kHz, the best possible frequency response is about 18kHz (see *Figure 24*). However, in the real world there are other factors that effect frequency response (like the design of the audio input circuitry, etc.). You can use the Nyquist limit as a convenient reference ("Hmm, let's see... I'm sampling at 18 kHz, so I should be able to catch highs up to about 9kHz"), but be aware that the actual top-end limit is probably somewhat less than the ideal figure.

As we learned above, sampling can be looked at as a matter of measurements. The sampler looks at, and measures, the input waveform. The measurements are recorded, and then the original is reconstructed by using the recorded measurements. Obviously, the authenticity of the reconstruction will be limited by the accuracy of the rulers used to take the measurements. *Figure 15* shows the same original waveform sampled at three different rates. The rates are shown as rulers. The higher the sampling rate, the more divisions on the ruler. You can see that with higher sampling rates, you can measure (record) finer horizontal details in a waveform. Horizontal details translate into frequency components of a waveform. The finer the detail, the more high frequencies (harmonics, etc.) in a sound. Now you know why changing the sampling rate changes your FZ's frequency response.

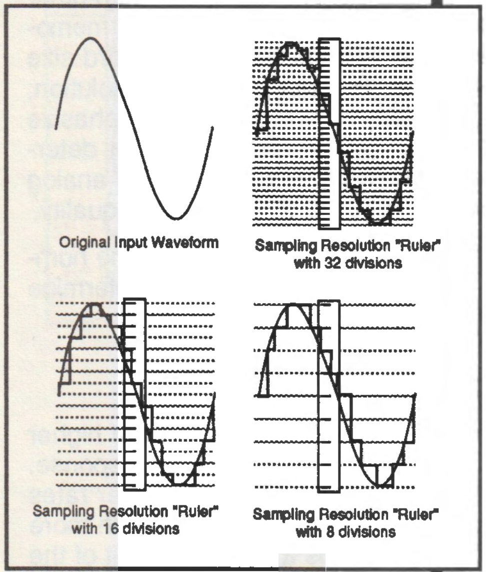

Figure 16: Sampling Resolution

Sampling Resolution

The FZ must also make measurements on a vertical scale as well. Vertical details on a waveform graph translate to loudness changes in the original waveform. So the more divisions on a sampler's vertical ruler, the more detailed loudness changes the sampler can measure and record. This will be the determining factor in establishing the dynamic range of the sampler, and is referred to as sampling resolution. Sampling resolution sets the size of the sampling "window" (see *Figure 22*). The higher the resolution, the wider range of dynamics you can sample before they become distorted or noisy.

The number of divisions on this vertical ruler is determined by the the number of bits used by the analog to digital converter. The common sizes used by sampling instruments are 8 bit, 12 bit, and 16 bit. These translate to 256, 4096, and 65,536 divisions respectively. Therefore, a 16-bit sampler should have a bigger dynamic range than an 8-bit sampler, and so on. In theory, each bit means a difference of 6 dB in the dynamic range. So an 8-bit system will have a 48 dB range, 12-bit would be 72 dB, and a16-bit works out to 96 dB. However, just like the Nyquist limit, these distinctions are based on ideal, theoretical limits. In the real world there are other factors besides word size that will affect dynamic range. *Figure 16* illustrates three different sampling resolutions against the same input waveform. Note that higher resolution allows you to make finer measurements on the vertical scale.

Sampling Memory

$$\boxed{\text{Length In Words}} = \boxed{\text{Sampling Rate}} \times \boxed{\text{Length In Seconds}}$$

$$\boxed{\text{Length In Seconds}} = \boxed{\text{Length In Words}} \,/\, \boxed{\text{Sampling Rate}}$$

Figure 17: Sampling Memory Calculations

Unlike sampling rate and sampling resolution, sampling memory doesn't effect the quality of a sample, but rather the quantity. The amount of sampling memory determines the maximum length of sampling time available to you. The length will, vary with the sampling rate. You will often need to convert either memory size into a time value ("how many seconds can I sample with 128k of memory?") or time into memory size ("how much memory do I need to sample 3.5 seconds?"). Figure 17 shows how to do the math for these all important sampling questions. The FZ-1 with 1 meg of memory uses 522 k for sampling data (the rest is used for system software, voice and Area definitions, etc.). An FZ-1 with 2 meg, or an FZ-10M, uses 1044 k for sampling data.

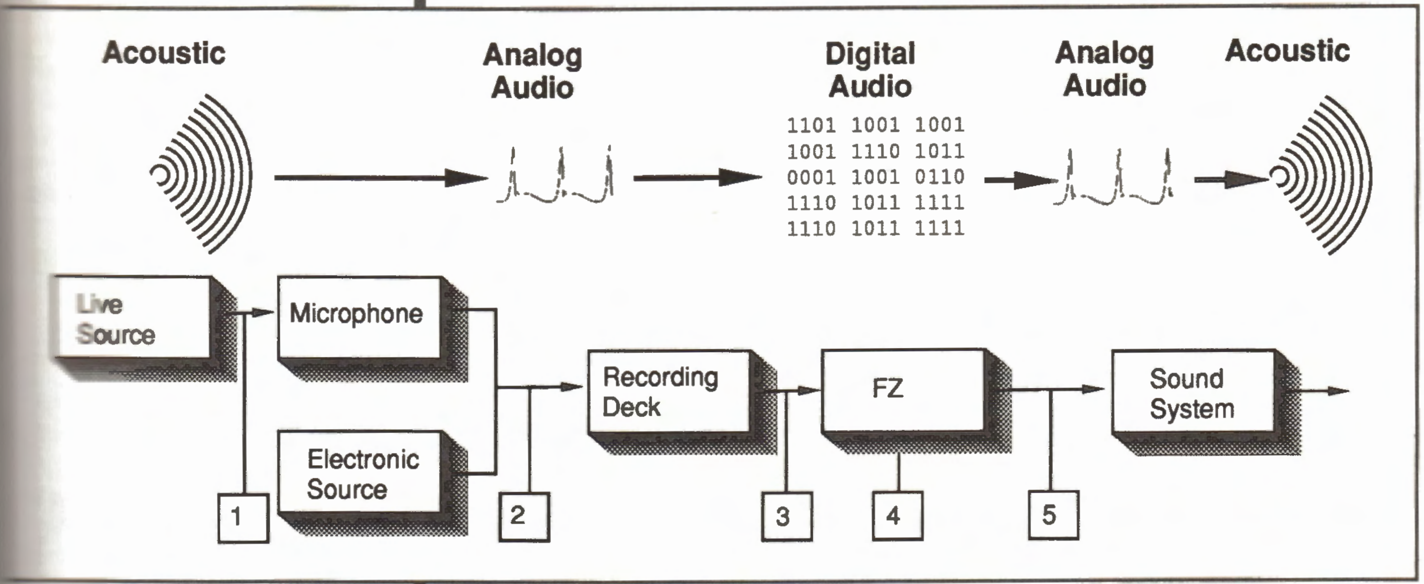

Figure 18: This diagram shows where a sound is converted from one form to another when it is sampled and played back. You have the option of modifying the signal at the points where it travels between two places. At point 1, your options include choosing a particular microphone and the environment where the recording takes place. At points 2, 3, and 5, you can use any number of audio processors, such as mixers, reverbs, equalizers, and compressors. At point 4, you can modify the original sound with the FZ's Voice Editing functions.

3.4 What's the Point?

Why is it important to know all this stuff? Well, there are three basic "facts of life" that will have a direct effect on your success (or lack of it) with sampling.

1. Whenever something gets converted from one format to another (in this case, acoustic to analog audio to digital audio and back again), there is always a chance that some changes will be introduced by the conversion process. In fact, you can count on it.

2. Another important factor is that there are usually some overall limitations to the conversion process. If you try to convert things beyond the limits, the results of the conversion will be distorted.

3. Finally, whenever you convert from one format to another, you have a chance to alter or enhance the original, so the converted result is no longer a "clone" of the original.

What it all boils down to is this. A lot can and/or will happen to a sound between when it is generated acoustically and when it is played back by your sampler. Once the conversion is completed, be it acoustic to analog or analog to digital, there is no going back. For the most part, changes to your original sound (intentional or not) will be there to stay for the life of that particular sample.

Once you understand these basic facts, you can apply them in a variety of ways. Much of what was once mystifying or intimidating becomes common place. As we mentioned above, your understanding of theory will open up a lot of creative options for you, but there are some very practical benefits to this understanding as well. If you haven't taken the plunge yet and bought yourself one of these remarkable machines, you'll find that the "facts of life" can help you make intelligent and informed decisions on how to spend your hard-earned cash.

Part 2: GETTING THE MOST FROM YOUR FZ

Part 2: Getting The Most From Your FZ

Now that we've gone over the basics of sampling, let's start our in-depth tour of the FZ. The tour is organized in the same general manner as the menus on the FZ. We've designed this part of the book to enhance and augment your *Operations Manual*, not to replace it, so be sure to keep it handy as you read on.

Speaking of the *Operations Manual,* we mentioned in *1.3 Sampling Buzzwords* that there are a lot of terms used by different instrument makers. Some have been more or less adopted by the music community at large, and others are unique to particular companies or instruments. Terms such as *splice, cross-fade splice, velocity switch, velocity cross-fade, key mapping, MIDI mapping,* and *controller mapping* are frequently used to describe certain instrument features by some manufacturers. You won't see these actual terms in the FZ's *Operations Manual,* but you should know that your FZ can do all of these and a great deal more.

Of course, we point out and demonstrate how to get these effects, and many others, with your FZ. You'll find that we've given you a lot of hands-on experiments and examples to try out. Be sure you do!

4. Getting Around On The FZ

The FZ is a sophisticated sampling/synthesis system. As you have undoubtedly already discovered, it offers a wide variety of features and functions. There are well over two hundred different parameters on the FZ. Rather than provide you with a confusing array of a hundred or more individual sliders, switches, and knobs, Casio has organized the FZ 's features neatly into a system of *modes, sub-modes, functions, and operations.* This method keeps related functions and operations grouped together in a logical manner while at the same time allowing you to use one common set of buttons and sliders for all 200+ parameters. The groups are displayed on the LCD display as a *Menu.* Menus are accessed with the cursor keys (**UP, DOWN, LEFT, RIGHT**), **ENTER** key, and **ESCAPE** key. We will refer to these as a group as *Menu Keys* (see *Figure 19*).

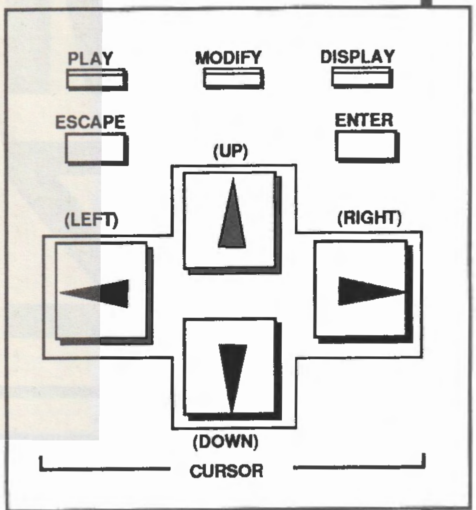

Figure 19: The Menu Keys

The key to mastering the FZ is learning how to use the Menu Keys to move quickly from mode to mode, sub-mode to sub-mode, and within each sub-mode–how to move between functions and operations. You get around on the FZ by manipulating the Menu Keys. The LCD display will show you where you are at any given time.

4.1 Overview of FZ Modes

The FZ has two modes which are activated by pushing the appropriate button on the front panel. The **PLAY** button puts the FZ into its *PLAY MODE.* The structure or the *PLAY MODE* is very simple. It has three operations: *Bank Play* lets you select and play any one of up to eight Banks (complete keyboard setups); *Voice Play* lets you select and play any one of up to sixty-four voices (individual samples); *Load Exec* loads all data (voices, Banks, effects, etc.) from the disk currently inserted in the disk drive.

Pushing the **MODIFY** button activates the *MODIFY MODE.* From this mode you have access to all of the FZ's sampling, synthesis, voice, Bank, Efects/MIDI, and Data Dump functions. Since there are so many things you can do in the *MODIFY MODE,* the different options are organized into a series of six sub-modes: Source Select, Voice Edit, Bank

Edit, Effects/MIDI, Data Dump, and OPT Software. (Each sub-mode has a corresponding menu display.) Each of these sub-modes is further organized into functions and operations (which also have their own menu displays).

To help you find your way around, we've given you a map (*Figure 20*).

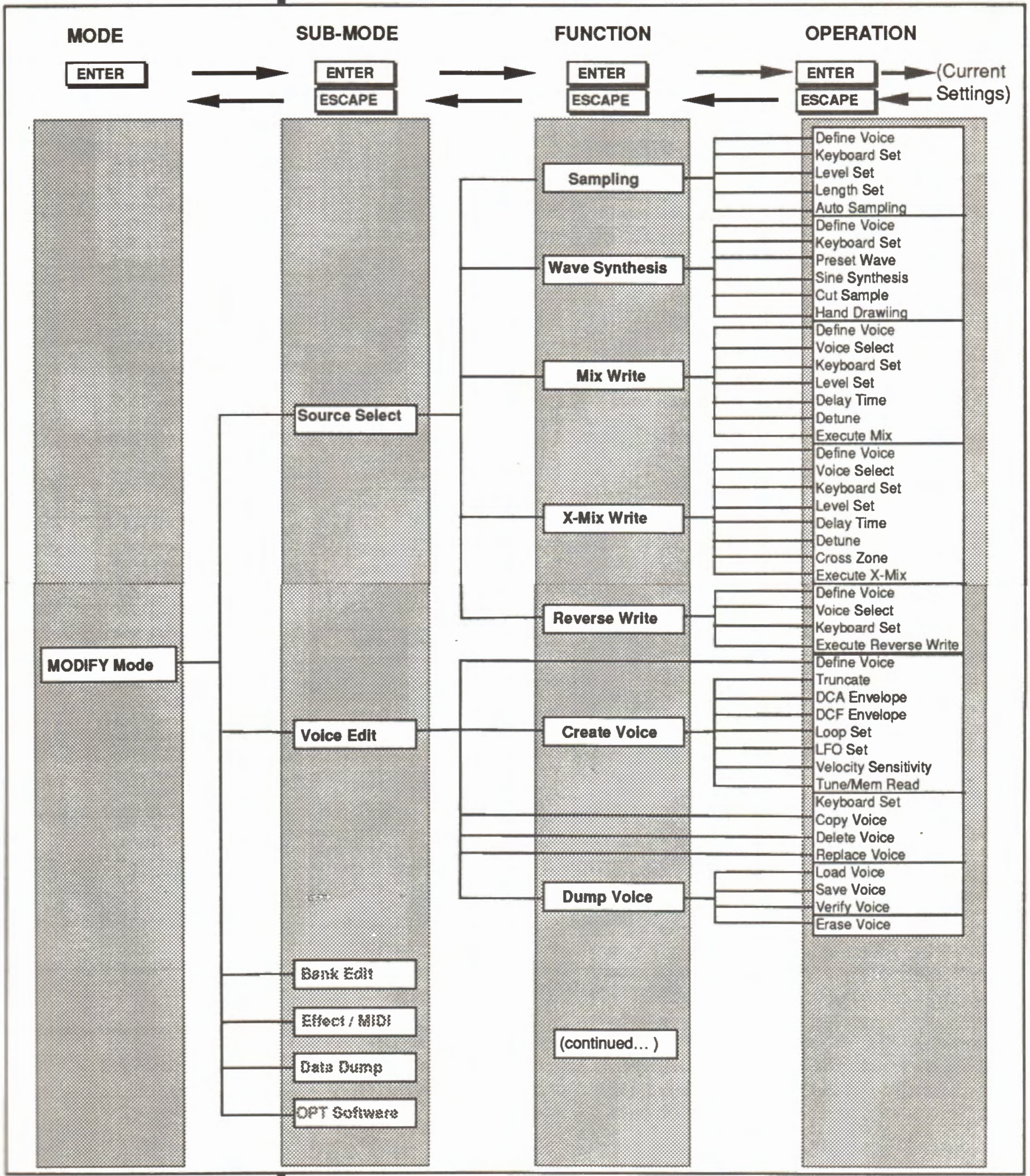

Figure 20: FZ Menu Map

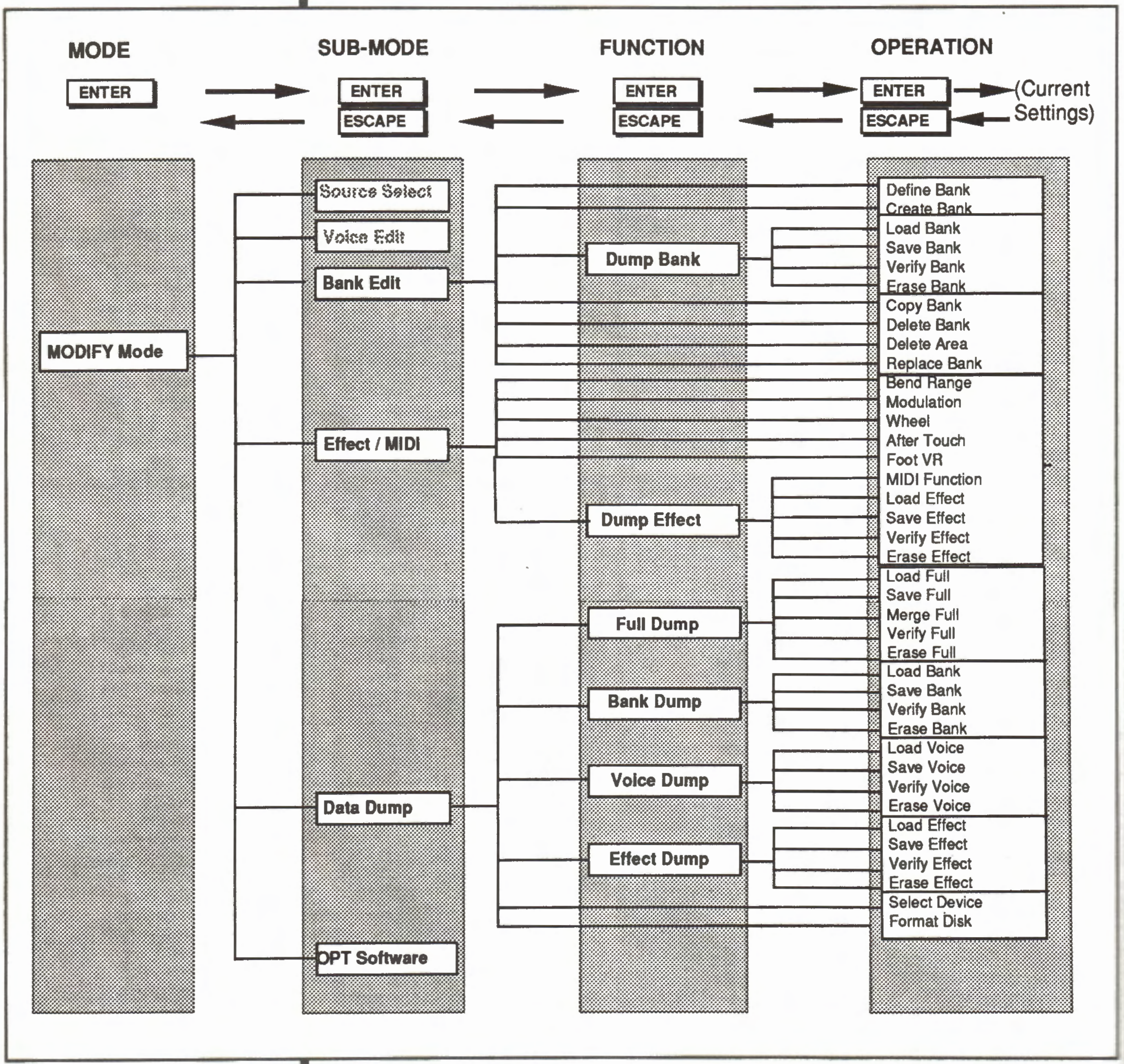

Figure 20 continued.

4.2 MODIFY MODE Menu Map

As you can see from the illustration, the menus in the *MODIFY MODE* are interconnected somewhat like the branches of a tree. (In fact, this kind of menu structure is indeed called a menu tree.) When you enter the *MODIFY MODE* by pushing the **MODIFY** button, the *Main* menu is displayed. You can branch out from there into any one of the six sub-modes listed in the menu:

MAIN MENU

[Source Select]
[Voice Edit]
[Bank Edit]
[Effect/MIDI]
[Data Dump]
[OPT Software]

From each of these sub-modes, you can branch out still further. For example, if you enter the *Source Select* sub-mode, you can branch out into any of the five functions displayed in the Source Select menu:

SOURCE SELECT

[Sampling]
[Wave Synthesis]
[Mix Write]
[X-Mix Write]
[Reverse Write]

From each of these functions, you can again branch out into another set of menus displaying operations. From each operation you can reach the final limbs in the menu tree. These are the *Parameter Settings* menus for the selected operation. Every operation on the FZ is associated with one or more parameter value. As we mentioned earlier, there are more than two hundred Parameter Settings active on the FZ. You can imagine how difficult it might be to try to locate one setting out of two hundred if they weren't organized efficiently. The beauty of the FZ's menu structure is that it lets you approach its features by moving from the general (sub-modes) to the specific (operations). Each level of menus is more focused on a specific task than the previous level. You'll find that this approach is quite natural and intuitive. At first, you may find it helpful to keep our menu map handy, but you'll soon be able to find your way around without it.

4.3 Moving from Menu to Menu

So how do you go from one place to another in the menu tree? You climb it, of course. To climb to the next level of menus, push the **ENTER** button. To climb to the previous level, push the **ESCAPE** button. You select the branch you wish to climb with the cursor **UP** and **DOWN** buttons. Move the cursor (the little arrow to the left of the list displayed in the menu) next to the sub mode, function, operation, or parameter you want and then push **ENTER.** Once you're done with the items of a particular menu, push **ESCAPE** to return to the previous menu. That's all there is to it. (See page 12 in the *Operations Manual.*)

There are even some short cuts you can take. No matter where you are in the menu tree, pushing the **MODIFY** button will always return you to the Main menu. This can save you some steps. If you wanted to go from the Sampling Parameter Settings menu to the Voice Edit menu, for example, the **MODIFY** button would let you jump directly to the Main menu in one button push, as opposed to the three required if you use the **ESCAPE** button.

Another very useful short cut provided by Casio is the **CALL/SET MENU** button. Use this to jump instantly from the *PLAY MODE* to any of the menus in the *MODIFY MODE*. It's easy to use. Before you push the **PLAY** button, push the **CALL/SET MENU** button. (Its red indicator light will go on.) The FZ will memorize where you are in the menu tree. Now push the **PLAY** button. When you want to leave the *PLAY MODE* and return to the memorized menu, just hit the **CALL/SET MENU** button. You'll jump instantly to the same menu that was displayed when you first pushed the **CALL/SET MENU** button.

4.4 Menu Overviews

The FZ's menu system will let you cover a lot of territory. To make things easier for you, we've provided an overview for each sub-mode, function, and operation menu used in the FZ. (These are similar to the *Mode Transition Diagrams* like those found on pages 10 and 47 in Casio's *Operations Manual*.) Each overview shows which buttons to push and which menu items to select to get to any desired FZ menu. The complete path is shown so that you can see how to get there even if you just plugged your FZ in for the first time.

The first column shows which of the two modes to select. The sub-mode, functions, operations, and parameter settings columns show you what items will appear in the LCD display. The second column shows which of the six sub-modes to select with the **UP/DOWN** buttons before pushing **ENTER.** (We've indicated the selected item in bold-face type.) The next column shows which function to select with the **UP/DOWN** buttons before pushing **ENTER.** The next column shows all operations that will be displayed for the selected sub-mode or function. The last column lists the parameter settings that will be displayed when any of the operations are selected and entered. Above the display columns we've indicated the buttons to push to climb to the next (**ENTER–>**) or previous (**<–ESCAPE**) level of menus. You'll notice that not all operations are nested inside of functions. Many are accessed directly from the sub-mode menu. We've indicated this with a double arrow (**<-------->**).

PLAY MODE: Menu Overview

MODE PLAY ->	OPERATIONS			
MODIFY **PLAY**	Play Bank Play Voice Load Exec			

Sampling Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER->	ENTER -> <--ESCAPE	ENTER -> <--ESCAPE	<--ESCAPE
MODIFY PLAY	**Source Select** Voice Edit Bank Edit Effect MIDI Data Dump OPT Software	**Sampling** Wave Synthesis Mix Write X-Mix Write Reverse Write	Define Voice	Voice No. Voice Name
			Keyboard Set	Original Key Highest key Lowest Key
			Level Set	Record Level Trigger Level Sampling Level
			Length Set	Time Rate
			Auto Sampling	Start Stop
			Manual Sampling	Start Stop

Wave Synthesis Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <--ESCAPE	ENTER -> <--ESCAPE	<--ESCAPE
MODIFY PLAY	**Source Select** Voice Edit Bank Edit Effect/MIDI Data Dump OPT Software	Sampling **Wave Synthesis** Mix Write X-Mix Write Reverse Write	Define Voice	Voice No. Voice Name
			Keyboard Set	Original Key Highest Key Lowest Key
			Preset Wave	Saw Tooth Square Pulse Double Sine Saw Pulse Random
			Sine Synthesis	Harmonic Level Execute
			Cut Sample	Source No. Start End
			Hand Drawing	Position Level

Mix Write Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER ->**	**ENTER -> <-ESCAPE**	**ENTER -> <-ESCAPE**	**<-ESCAPE**
MODIFY	**Source Select**	Sampling	Define Voice	Voice No.
PLAY	Voice Edit	Wave Synthesis		Voice Name
	Bank Edit	**Mix Write**		
	Effect/MIDI	X-Mix Write	Voice Select	1st Voice No.
	Data Dump	Reverse Write		2nd Voice No.
	OPT Software			
			Keyboard Set	Original Key
				Highest Key
				Lowest Key
			Level Set	1st Level
				2nd Level
			Delay Time	2nd Start:Coarse
				2nd Start:Fine
			Detune	1st Tune
				2nd Tune
			Execute Mix	Yes/No

X-Mix Write Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER ->**	**ENTER -> <-ESCAPE**	**ENTER -> <-ESCAPE**	**<-ESCAPE**
MODIFY	**Source Select**	Sampling	Define Voice	Voice No.
PLAY	Voice Edit	Wave Synthesis		Voice Name
	Bank Edit	Mix Write		
	Effect/MIDI	**X-Mix Write**	Voice Select	1st Voice No.
	Data Dump	Reverse Write		2nd Voice No.
	OPT Software			
			Keyboard Set	Original key
				Highest Key
				Lowest Key
			Level Set	1st Level
				2nd Level
			Delay Time	2nd Start:Coarse
				2nd Start:Fine
			Detune	1st Tune
				2nd Tune
			Cross Zone	Start:Coarse
				Start:Fine
				End:Coarse
				End:Fine
			Execute X-Mix	Yes/No

Reverse Write Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER ->**	**ENTER ->** **<-ESCAPE**	**ENTER ->** **<-ESCAPE**	**<-ESCAPE**
MODIFY	**Source Select**	Sampling	Define Voice	Voice No.
PLAY	Voice Edit	Wave Synthesis		Voice Name
	Bank Edit	Mix Write		
	Effect/MIDI	X-Mix Write	Voice Select	1st Voice No.
	Data Dump	**Reverse Write**		2nd Voice No.
	OPT Software			
			Keyboard Set	Original key
				Highest Key
				Lowest Key
			Execute Reverse	Yes/No

Voice Edit Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER ->**	**ENTER ->** **<-ESCAPE**	**ENTER ->** **<-ESCAPE**	**<-ESCAPE**
MODIFY	Source Select	**<---------------->**	Define Voice	Voice No.
PLAY	**Voice Edit**			Voice Name
	Bank Edit			
	Effect/MIDI	Create Voice	Truncate	Start:End
	Data Dump			
	OPT Software		DCA Envelope	Rate KF
				Level KF
				Step,Rate,Level
				Copy From DCF
			DCF Envelope	Cutoff Frequency
				Resonance
				Rate KF
				Level KF
				Step, Rate, Level
				Copy From DCA
			Loop Set	Start: End
				Loop Time
				Cross Time
				Next
			LFO Set	Wave
				Sync
				Delay
				Rate
				OSC Depth
				DCA Depth
				DCF Depth
			Velocity Sens	DCA Level
				DCA Rate
				DCF Rate
				Resonance
			Tune/Mem Read	Tune/ Mem Read

Voice Edit Menu Overview (continued)

		<----------------->	Keyboard Set	Original Key Highest Key Lowest Key
		<----------------->	Copy Voice	Voice Name Destination Execute: Yes/No
		<----------------->	Delete Voice	All Part Unused Part
		<----------------->	Replace Voice	Voice Name Destination
		Dump Voice	Load Voice	Voice Name Execute: Yes/No
			Save Voice	Voice Name Execute: Yes/No
			Verify Voice	Voice Name Execute: Yes/No
			Erase Voice	Voice Name Execute: Yes/No

Bank Edit Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER ->**	**ENTER -> <--ESCAPE**	**ENTER -> <--ESCAPE**	**<--ESCAPE**
MODIFY PLAY	Source Select Voice Edit **Bank Edit** Effect/MIDI Data Dump OPT Software	<----------------->	Define Bank	Bank No. Bank Name
		<----------------->	Create Bank	Voice No. Original Key Highest Key Lowest Key Max Touch Min Touch Area Level MIDI Channel Output
		<----------------->	Copy Bank	Source No. Destination No. Execute: Yes/No
		<----------------->	Delete Bank	Bank Only Bank & Voice
		<----------------->	Delete Area	Area No. Execute: Yes/No
		<----------------->	Replace Bank	Source No. Destination No. Execute: Yes/No
		Dump Bank	Load Bank	Bank Name Execute: Yes/No
			Save Bank	Bank Name Execute: Yes/No
			Verify Bank	Bank Name Execute: Yes/No
			Erase Bank	Bank Name Execute: Yes/No

Effect / MIDI Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER ->**	**ENTER ->** **<--ESCAPE**	**ENTER ->** **<--ESCAPE**	**<--ESCAPE**
MODIFY PLAY	Source Select Voice Edit Bank Edit **Effect/MIDI** Data Dump OPT Software	<---------------->	Bend Range	Bend Range
		<---------------->	Mod Wheel	LFO OSC LFO DCA LFO DCF DCA Level DCF Level
		<---------------->	After Touch	LFO OSC LFO DCA LFO DCF DCA Level DCF Level
		<---------------->	Foot VR	LFO OSC LFO DCA LFO DCF DCA Level DCF Level
		<---------------->	MIDI Function	Basic Channel Receive Basic/Area Control ENA/DIS Program ENA/DIS
		Dump Effect	Load Effect	Effect Name Execute: Yes/No
			Save Effect	Effect Name Execute: Yes/No
			Verify Effect	Effect Name Execute: Yes/No
			Erase Effect	Effect Name Execute: Yes/No

Data Dump Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER ->**	**ENTER ->** **<-ESCAPE**	**ENTER ->** **<-ESCAPE**	**<-ESCAPE**
MODIFY PLAY	Source Select Voice Edit Bank Edit Effect/MIDI **DATA DUMP** OPT Software	**Full Dump** Bank Dump Voice Dump Effect Dump	Load Full Save Full Merge Full Verify Full Erase Full	Execute: Yes/No Execute: Yes/No Execute: Yes/No Execute: Yes/No Execute: Yes/No
		Full Dump **Bank Dump** Voice Dump Effect Dump	Load Bank Save Bank Merge Bank Verify Bank Erase Bank	Bank Name Execute: Yes/No Bank Name Execute: Yes/No Bank Name Execute: Yes/No Bank Name Execute: Yes/No Bank Name Execute: Yes/No
		Full Dump Bank Dump **Voice Dump** Effect Dump	Load Voice Save Voice Verify Voice Erase Voice	Voice Name Execute: Yes/No Voice Name Execute: Yes/No Voice Name Execute: Yes/No Voice Name Execute: Yes/No
		Full Dump Bank Dump Voice Dump **Effect Dump**	Load Effect Save Effect Verify Effect Erase Effect	Effect Name Execute: Yes/No Effect Name Execute: Yes/No Effect Name Execute: Yes/No Effect Name Execute: Yes/No
		<----------------> <---------------->	Select Device Format Disk	Device: Disk, Pat, MIDI Port, MIDI Disk Name Execute: Yes/No

4.5 Adjusting the LCD Display

Certainly, one of the most powerful features of the FZ is its high resolution LCD display. Not only is the display your window into the FZ's menu structure, but it also allows you to view and edit waveforms and envelopes visually. You may have noticed that the display can be hard to read from certain viewing angles (a common problem with almost all LCD displays). Casio has provided a means of adjusting the display contrast so that you can make the display easy to read from virtually any angle. Unfortunately, the method for how to do this isn't explained in the *Operations Manual.* Here's how it's done:

- Enter the *MODIFY MODE* by pressing the **MODIFY** button.
- Hold down the **DISPLAY** button and move the **VALUE** slider.
- Move the **VALUE** slider up and down through its complete range until you find the spot where the display looks best for your current viewing angle. That's all there is to it. (Remember to hold the **DISPLAY** button while you're moving the slider.)

4.6 Common FZ Operations

If you examine the Menu Map (*Figure 20*) and the Menu Overviews, you will see that most of the sub-modes and functions have three or more operations in common: *Define Voice, Keyboard Set,* and *Voice Select,* as well as *Load..., Save..., Verify...,* and *Erase....* Since they will appear in so many different menus, but are always used in the same way, we felt it would be a good idea to go over them now, before we look at FZ functions in-depth.

Define Voice

Define Voice appears in all five of the Source Select menus and also in the Voice Edit menu as well. When you enter this operation from any of these menus you will see a Parameter Settings display showing you the *Voice No.* and *Voice Name* of the currently selected voice. (If you've just turned the FZ on, this will default to Voice No.1)

Define Voice is used to select any one of the sixty-four FZ voices to work with. If you are working in one of the Source Select menus: Sampling, Wave Synthesis, Mix Write, X-Mix Write, Reverse Write, you will want to use this operation to define a new voice. If you are working in the Voice Edit sub-mode, you'll want to use Define Voice to select a voice previously created in the Source Select sub-mode. You can scan through the sixty-four voices by moving the cursor to Voice No. and then using the **YES/NO** buttons to scroll through the list. If you are creating a new voice, select any voice number that shows the words "NO SOUND" in the display. Voices previously created with Source Select functions will show either "Recorded" or "Synthesized." Voices supplied by Casio (or other commercial libraries) will also show a name, like "Piano C3." Of course, voices you create yourself can display their names as well, but you'll have to enter the names as you create the voices. That is the other function of the Define Voice operation. (See page 13 in the *Operations Manual.*)

The important thing to remember about Define Voice is to be sure to select a new voice ("NO SOUND") before you execute any of the Source Select functions. Otherwise, you may unintentionally delete a previously created voice. If you have selected an existing voice, the FZ will ask you if you really want to continue. You'll see "VOICE EXISTS, DELETE? YES/NO" in the display. If you press **YES** the voice will be deleted, so you can put a new one in its place. If you press **NO**, you will *escape* to the previous menu. From there, you can enter Define Voice to select a new voice.

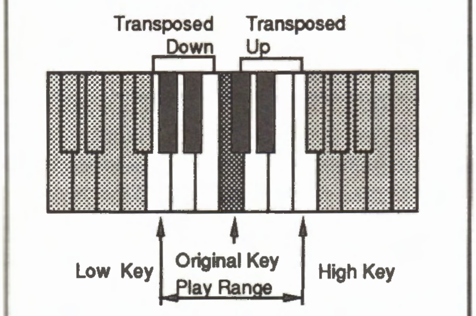

Figure 21: Original Key

Keyboard Set

The Keyboard Set operation is used to select the range of keys which will play the selected voice. The *Original Key* is the key that will play back a sampled voice with the same pitch it was sampled at. The *Highest Key* is the upper key that will sound with this voice, and the *Lowest Key* is the Lowest Key that will sound the voice. The operation is described on page 25 of the *Operations Manual*. Be aware that when the parameters for this operation are displayed, you will not hear any sounds from the FZ when you play on the keyboard. Instead, every key you press updates the values for the Original, Highest, and Lowest Keys. The cursor will move automatically between these three keys. (You don't have to use the **UP/DOWN** buttons.) Once you've set up the desired range, be sure to return to the previous menu (with the **ESCAPE** button) before you play any other notes on the keyboard.

Voice Select

The Mix Write, X-Mix Write, and *Reverse Write* functions each create a new voice by altering a previously created (sampled) voice. The Voice Select operation is used to select which previously sampled voice(s) will be used as source material for the new voice. (See pages 41, 48, and 58 in the *Operations Manual*.) Make sure the the Voice No. for the function is not the same as the 1st Voice No, 2nd Voice No, or Source No., or you may inadvertently delete a voice. Use the **VALUE** slider if you want to set a range of keys that extends above and/or below the limits of the keyboard. You can set any limits within a ten-octave range.(This can sometimes happen if you execute the function without ever entering the Define Voice operation.)

Load, Save, Verify, Erase

Memory operations that deal with the different FZ data types can be accessed from within the appropriate sub-modes and also from the *Data Dump* sub- mode. This is a great convenience, since you don't have to leave a sub-mode to load, save, verify, or erase Banks, voices, or effects. The operations, as they appear in the sub-modes, look and work the same as they do in the Data Dump sub-mode. (See pages 100 through 122 in the *Operations Manual*.)

Understanding the different data formats and how to use them is an important part of getting the most from your FZ. Be sure to read about ***Memory Management*** in Chapter 13.

4.7 OPT Software

This sixth sub-mode is reserved for soon-to-be-released expansion software for the FZ. The computer hardware in your FZ can be made to do any number of things as long as it is driven with the appropriate software. Casio and others will be releasing optional software packages that will add even more power to the FZ. As this is written, the first set of optional software from Casio is about to be released. The package contains enhanced looping tools, audio effects, and note and pattern sequencers for the FZ. Now let's move on and look into how to create samples with the FZ.

Experiment #3: Original Key/Key Range

Focus: Sample Edit Performance

Key Settings:

- Sampling Rate : Default, Sampling Time: 00100 ms
- Input Level: 1/4 up
- Mic or Line: Line
- Original Key, Highest Key, Lowest Key: refer to **Step by Step** below

Operations Manual Page Reference:

- Sampling Operations: Keyboard Set Pg. 25

Step by Step:

- Record the word "PITCH" onto your tape deck. (This time, try singing the word instead of just speaking it.). Remember to take all precautions outlined in **Experiment #3.**
- Set the FZ's input level and the sample length as described in **Experiment #3**.
- Enter the Sampling: Keyboard Set
- Set the FZ's "Original" to "C04" (either simply press middle C or use the Value slider).
- Using the cursor Down select "Highest."
- Set "Highest" to "C05" (by either selection method).
- Set "Lowest" to "C05". **ESCAPE.**
- Enter Manual Sampling and press **YES** after your tape has been cued and is running.
- When sampling is complete, press and release middle C,"Original," then C one octave above middle C, "Highest," and then C one octave below middle C.
- Repeat the entire above procedures altering only the key assignment. Assign Original Key to C02 (the lowest C), Highest Key to C03 (the second C), and the Lowest Key to C05 (the third C).

Observations:

- Two points should be immediately apparent. One, Key assignment sets the maximum limits the sample will be transposed. Two, placement of the sample on the keyboard has no relation to that keys normal musical pitch, i.e., placing a sample on middle A does not mean that the sample will be A440 or concert A.
- The relationship of Highest Key and Lowest Key to Original Key is the musical intervals between high and low to original. For example, if high key is one octave above original key, then high key will be exactly one octave higher in pitch than any pitch assigned to original key, or exactly doubled in frequency. The same relationship is true for low key.
- You should have also noticed that, when you attempted to assign low key higher than the original key, the the FZ will adjust lowest and highest till the relationship is correct. Low key must be equal to or lower than original key, and consequently, high key must be equal to or higher than original key.
- Experiment with placing different sounds on the keyboard, and listen for the limit of the transposition of each sound.

5. *Creating Samples*

On the FZ, all of the operations that have to do with sampling sounds, in other words, converting analog source material into digital data, are grouped together in the Sampling function of the Source Select menu. They are detailed in pages 22 through 30 of the *Operations Manual*. Here is the Menu Overview for this function.

Sampling Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER->	ENTER -> <--ESCAPE	ENTER -> <--ESCAPE	<--ESCAPE
MODIFY	**Source Select**	**Sampling**	Define Voice	Voice No.
PLAY	Voice Edit	Wave Synthesis		Voice Name
	Bank Edit	Mix Write		
	Effect MIDI	X-Mix Write	Keyboard Set	Original Key
	Data Dump	Reverse Write		Highest key
	OPT Software			Lowest Key
			Level Set	Record Level
				Trigger Level
				Sampling Level
			Length Set	Time
				Rate
			Auto Sampling	Start
				Stop
			Manual Sampling	Start
				Stop

5.1 *Setting Input Levels*

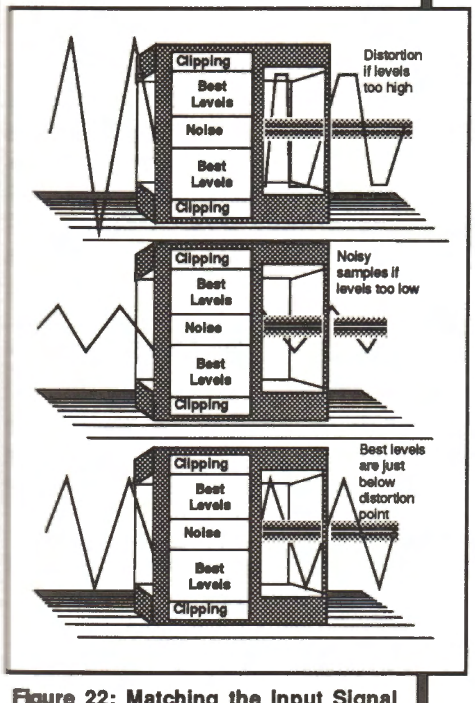

Figure 22: Matching the Input Signal Level to the FZ's Sampling Window

Input Level, more than any other parameter, has an enormous effect on the overall sound quality of your samples. If the level is set too high, a very harsh (and particularly unpleasant) distortion will be introduced into your sound. If the level is too low, your samples will be noisier than they could be.

If you were to sample a continuous tone like an organ note, the ideal level setting would be at the point just below where the distortion occurs. This would produce the cleanest and most quiet sample on your instrument. Most of the sounds you'll be sampling won't be continuous tones, however. One of the things that makes a sound interesting is changes in its loudness. Piano tones, for example, start loud and gradually fade away. When you set the level to sample the beginning of the tone without distortion, the level of "tail," or the decay portion of the tone, will eventually become so low that the noise of the sampler is louder than the piano sound.

If your sound is flawed because of improper level settings, there is nothing you can do to fix it later on. Your only choice is to sample the sound again. Now you can see why we recommended that you first record the sounds you want to sample on a good quality analog or digital tape (or hi-fi video) recorder. You can replay the sound over and over until you get the levels set just where you want them.

Figure 22 shows the relationship between input levels and the resulting sampled sound. Your FZ (indeed, all audio devices) has a "window" that defines where the best levels should be. Input signals above the window will be clipped (distorted). While distortion has becomes a sought after effect in certain situations, for most sampling applications you will find distortion to be extremely unpleasant. The FZ (like any audio device) will also add a small amount of noise to the signal it processes. If the input level is *below* the sampling window, the added noise will be loud enough (relative to the sampled sound) to be quite audible in the final sample.

Technically, the sampling window is referred to as dynamic range. The spec. sheets of most audio gear will list dynamic range as signal to noise ratio or S/N. The overall dynamic range of any sampler is determined, for the most part, by the number of bits used for each sample. In general, the more bits used, the wider the dynamic range. The reasons for this were explored previously in *Changing Electronic Signals into Digital Samples.*

5.2 Setting Length and Rate

Length Set

All samplers use RAM (random access memory) to store digitalized sounds. The total amount of memory available for storage is fixed. Memory size is usually given as the number of kiloBytes, as in "128 kBytes" or "128K" (which means 128,000 bytes). The FZ-1 comes from the factory with 1 mBytes of memory (that's 1 million bytes). This can be expanded to 2 mBytes with the MB-10 option. The FZ-10M comes standard with 2 mBytes of internal memory. Out of that 1 mByte, 522 kBytes are used for sampling memory (1044 kBytes for the 2 mByte machines). Although the total size of sampling memory is fixed, the length of time for each individual sample can usually be varied. For example, the normal FZ-1 has 14.5 seconds of sampling memory (at a sampling rate of 36K). You may divide up those 14.5 seconds any way you wish. For example, you could use all 14.5 seconds for one sample, or create 14 samples of one second each and still have room for a 15th 0.5 second sample. You can divide that memory any number of ways into as many as 64 samples. Not only can they be of different lengths, but you can also use different sampling rates as well. *Figure 23* shows examples of different divisions of a 522 kByte memory block.

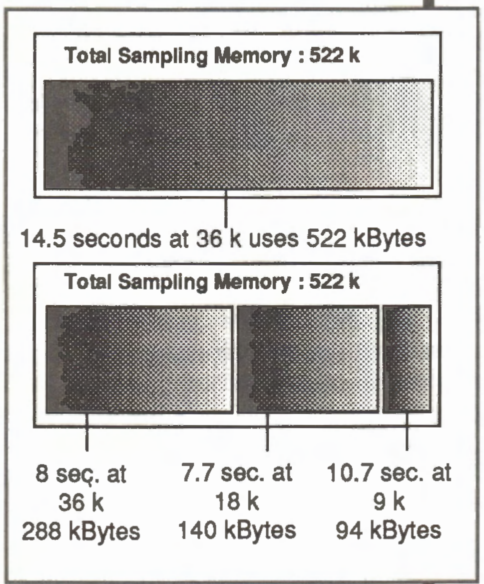

Figure 23: Allocating Sampling Memory

This is another one of those decisions that often involves a trade-off. As mentioned above, the sampling rate will effect how much memory is used to sample a certain period of time. As the sample rate is increased, more memory is used to sample the same amount of time. You can, of course, lower the sampling rate to stretch the sampling time, but we're back to another one of those trade-offs. Remember? Lowering the sampling rate will give you more time, but it also degrades the fidelity of the sample.

Here are some tips on how to get the most out of your sampling memory:

- Try to start recording the sample as close to the start of the sound you're sampling as possible. The *Auto Trigger* operation may be a great help in this regard (more about Auto Trigger coming up).
- When possible, use loops to recreate the sustained portions of continuous sounds. (We'll be looking at loops in just a bit.)
- Remove any dead space from the front and/or back of the sample with Truncate and Delete Voice: Unused Part.
- Make the sound you are sampling as short as you can without spoiling it.

Experiment #4 : Sampling Rate

Focus: Sample Edit Performance

Key Settings:

- Sampling Rate: refer to **Step by Step** below
- Sampling Time: 002000 ms
- Input Level: 1/4 up
- Mic or Line : Line

Operations Manual Page Reference:

- Sampling Operations: Length Set Pg. 27 (block # 3)

Step by Step:

- Use your cassette recorder or drum machine to record a long cymbal crash. If you can't get a recording of a cymbal, record another sound that has a lot of "top end."
- Refer to page 27
- Enter Sampling: Length and set the "Rate" to 36 kHZ and the "Time" to 002000 ms.
- Using the "Manual Sample" fountain sample the cymbal sound.
- Enter Define Voice and set "Voice No." to "02".
- Enter Length and set the "Rate" to 18 kHZ and clear and reset the "Time" to "002000 ms."
- Using the Manual Sample fountain sample the cymbal sound.
- Enter Define Voice and set "Voice No." to "03".
- Enter Length and set the "Rate" to 9 kHZ and clear and reset the "Time" to "002000 ms."
- Using the Manual Sample function, sample the cymbal sound.
- Push **PLAY**and use the **DOWN** button to select "Voice No."
- Play and listen to the three samples you've made. Use the **YES** and **NO** buttons to select between voices.

Observations:

- One of the differences that should have been noted was the quality, or lack of quality, of the sample at the various sampling rates. The higher the sampling rate the brighter and clearer it sounded. We highlighted this by recording a sound with a lot of high frequencies. If for example, we had sampled a bass drum, the difference in sampling quality would have been less noticeable
- Another point you should have recalled from experiment #6 "Sampling Length" (which is the reason for not using the highest sampling rate constantly) is the increase in memory used at the higher sampling rates. The higher the rate, the more memory needed to record the same length sample. Since some sounds don't need the broad of a bandwidth that the higher sampling rates yield, you can conserve memory by using a lower sampling rate (and still get a good quality sound). Test this theory by redoing the experiment, only this time use a bass drum. When you're done, compare both the fidelity and the memory consumption.

Sampling Rate

Sampling Rate has a direct effect on the sound fidelity of your samples. We went over the reasons for this earlier, but the basic relationship between Sample Rate and sound quality is this: the higher the Sampling Rate, the better the fidelity. By fidelity, we mean frequency response, which translates to how much top end, or treble, will be captured in your sample. In short, samples at high rates will sound brighter than the same sound sampled at lower rates. *Figure 24* shows the basic relationship between sampling rate and frequency response. *Figure 25* shows the amount of memory required to sample one second of sound at the FZ's three sampling rates.

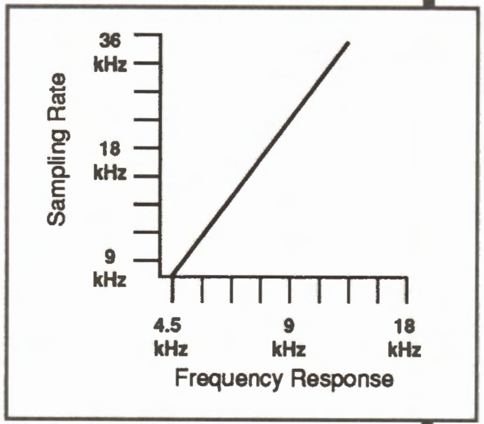

Figure 24: Sampling Rate vs. Frequency Response

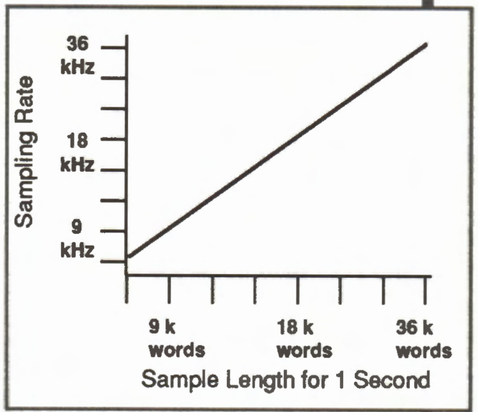

Figure 25: Sampling Rate vs. Sampling Time

So why do you have to make a choice? Why not sample at the highest possible rate all of the time? Like so many things, sampling is a series of trade-offs. It's like the old cliché, "I've got good news, and bad news...." The good news is that as the sampling rate goes up, the fidelity goes up too. The bad news is that as the sampling rate goes up, the maximum amount of sampling time available to you goes down. As a matter of fact, the relationship between these two factors is inversely proportional. If you double the sampling rate, you halve the time available. Conversely, if you halve the sampling rate, you double the time available.

The FZ gives you a choice of three sampling rates, so you can select the optimum setting for your needs. If fidelity is your highest priority for a given sound, then you may want to use the highest sampling rate, 36k. In doing so, bear in mind that you are also limiting the total amount of time available to sample to 14.5 seconds (with 1 mByte FZs).

If, on the other hand, your highest priority is to capture an entire sonic event (for example, you might want to sample a complete phrase or sentence), then you'll want to be able to set the sample rate to 18k (29 seconds), or maybe even 9k (58 seconds), to give you enough time to catch the whole event. A good rule of thumb is to sample at the highest rate that will still give you enough time to catch the whole event.

There may be times when no matter how you set the sampling rate, you cannot achieve the results you want. There are many tricks you can use to extend the apparent length of a sample and/or enhance its fidelity. The most important of these is looping. As you'll learn in Chapter 7, loops and envelopes can be used to extend even a very short sample to an infinite length. This means that you can sample short sounds at high rates for the best fidelity, and use loops and envelope settings to synthesize the desired length of the sound.

5.3 Auto Sampling

We mentioned above that one of the ways of making the most of your available memory was to start recording the sample as close to the beginning of the sound as possible. That sounds easier than it is. Sometimes it can be pretty difficult to hit the FZ's **ENTER** button just as the sound starts. You may not know (or be able to control) when the sound will begin. If you're playing the sound, it may be awkward to play a note and hit the button simultaneously.

This is why your FZ has special operation called Auto Trigger. It will start the FZ automatically for you. Basically, it works like this: the FZ "watches" the signal coming from its input jack, and when it senses the beginning of a sound, it automatically triggers (starts) the sampling process for you.

Experiment #5: Sampling Length

Focus: Sample Edit Performance

Key Settings:

- Input Level: 1/4 up
- Mic or Line: Line
- Sampling Rate: refer to **Step by Step** below
- Sampling Time: refer to **Step by Step** below

Operations Manual Page Reference:

- Sampling Operations: Length Set Pg. 27

Step by Step:

- Using your tape deck, record this sentence. "How long will this sample run, 1,2,3,4,5...."
- Enter the Sampling: Length Set operation of the Source Select sub-mode. Move the cursor to "Time=".
- Move the value slider to maximum. Note the maximum sampling time available at the rate of 36 kHZ is "14560 ms" (14.56 seconds).
- Move the cursor **DOWN** to "Rate=". Adjust the value slider until the "Rate = 18 kHZ"
- Move the cursor **UP** to "Time =," and bring the value slider all the way down, then all the way back up to maximum. (In order to get an accurate reading of "time" you must first clear the old setting by bringing the slider down then back up to your new setting.)
- Note that the maximum sampling "Time" at 18 kHZ is "29120 ms" (29.120 seconds).
- Adjust the Rate to the lowest sampling rate of 9 kHZ.
- Readjust the time to maximum. Your reading at 9 kHz should be "Time = 58250 ms" (58.250 seconds).
- Using either the value slider or the number buttons set the "Time for 001000 ms" and leave the rate at 9 kHZ.
- **ESCAPE** and enter Manual Sample
- Sample the sentence you have recorded on your deck.
- **ESCAPE** and set "Define Voice" to "Voice No. 02"
- Enter "Length" to verify that your display reads "Voice No. 02" "No Sound".
- Using the value slider, clear and reset the "TIME " to maximum. The display should show "57250 ms" @ 9 kHZ.
- Set the "Rate" to 18 kHZ. Using the value slider, clear and readjust the "TIME" to maximum. The display should read "TIME = 028620" at a "rate = 18kHZ".
- Reset the "Rate" to 36 kHZ. Using the value slider, clear and readjust the "TIME" to maximum. The display should read "TIME = 014310" at a "rate = 36kHZ".

Observations:

- It is obvious to that the longer you set the sampling time, the remainder of available sampling time will be the maximum time - the time used to sample. This formula is only true when you keep the rate constant. In the example above we sampled 1 second at 9 kHZ, and from a maximum available time of 58250 we were left with 57250, a difference of 1000 (1 second). But when we check the other rates, we found that at 18 kHZ only 500 ms were used, and at 36 kHZ only 250 ms were used.
- Remember that sample time is directly related to memory space. Spend some time considering alternatives to long samples or high sampling rates. For example, looping a short sample, or remembering to lower the sampling rate when an increased band width is not necessary. There are always alternatives to wasting memory. Read on to learn how!

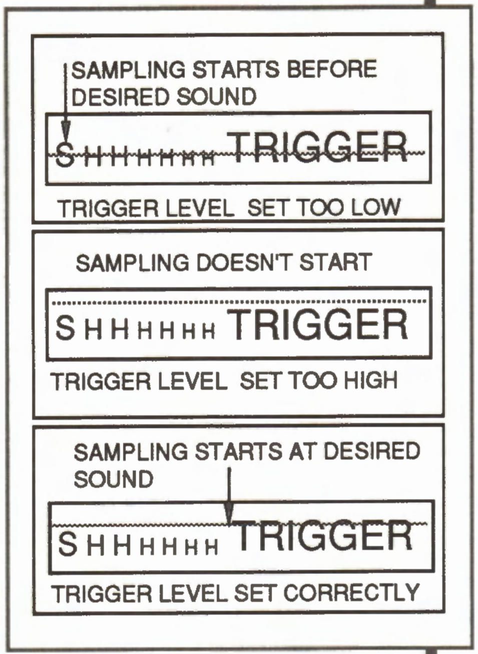

Figure 26: Setting the Auto Trigger Level Properly

So how does the FZ know when a new sound begins? When you select and enter Auto Trigger, the FZ will not start recording until the input signal is greater than "TRIGGER LEV" value you set in the *Level Set* operation (see **Experiment #7**). This way you can enter Auto Trigger, select and execute *Start*, walk slowly across the room, contemplate the meaning of life, the universe, and all that, and then–when you're good and ready–whack a four-foot Chinese gong with an old army boot. Even though many minutes (hours? days?) passed between the time you started the sampling process and when you hit the gong, the FZ didn't start recording until you whacked the gong. (Of course, the"TRIGGER LEV" had to be set properly or your effort would have been spoiled by premature digitalization.)

At the lowest "TRIGGER LEV" settings, the FZ will trigger (begin recording) when even the slightest sound comes its way (via microphone, tape recorder, or whatever your sampling from). Very low trigger settings are most desirable, since they will ensure that you catch the sound from very beginning. If there is some low-level background noise coming from the source (that you can't eliminate any other way), set the trigger level just beyond the point where the noise sets it off. This will start the FZ when your sound begins, and you won't be recording the noise up front.

If there are no noise problems, you can almost always use the trigger level set to its lowest level. Now the Auto Trigger operation acts as a smart record feature. It will hold the FZ until your sound gets there. Since the level is set for zero, sampling will start at the very beginning of the source sound.

Be aware that if the trigger level is set above the very lowest settings, two things are almost bound to happen. The very beginning of the source sound will cut off, and there will be an audible click at the start of the sample. You can use voice editing functions to clean up the sample. Move the sample start point to the nearest zero crossing (see Truncate). Another way would be to adjust the DCA envelope (see *DCA envelope*) to produce a very slight attack. This make the start of the sample sound less clipped, and at the same time it will hide the click.

Be careful with the trigger setting. If it's too low and you're sampling a noisy source, you'll waste valuable memory to record useless noise. If the level is too high, you'll cut off the beginning of the sound you're trying to sample. The best way to avoid trouble is to always sample in a very quiet environment.

Experiment #6: Auto Trigger

Focus: Sample Edit Performance

Key Settings:

- Sampling Rate: Default, Sampling Time: 1500 ms
- Mic or Line input: Line
- Input Level: 1/4 up
- Original Key, Highest Key, Lowest Key: Default

Operations Manual Page Reference:

- Sampling Operations: Auto Sampling Pg. 28

Step by Step:

- Using your cassette recorder and a microphone,we need to make a cue that has a decreasing level of noise that then ends in the word "TRIGGER" (See *Figure 26.*)
- Referring to page 26 of your owner's manual.
- Enter the *S*ampling: Length Set operation of the Source Select submode. Set the "time" value to "001000 ms" (1 second). **ESCAPE**
- Set the sampling input level to about a quarter of the way up.
- Enter the Level Set operation. Move the curse to *Trigger Level =.*
- Using the value slider set the "Trigger level" to equal 000. **ESCAPE**
- Enter Auto Sample and press YES, and start the tape deck.
- Once the sampling process is complete, **ESCAPE** and go back to Define Voice, and set value of "Voice No." to "02".
- Repeat the above process starting from the step relating to Level Set. This time set the *Trigger Level* to equal "064" (which is mid point).
- When the sample is complete **ESCAPE** and Enter Define Voice, and set "Voice No." to "03".
- Repeat the above process starting from the step relating to Level Set. This time sets the Trigger Level equal to "100" (which is about the maximum auto trigger level we can achieve if we wish to keep our input levels at their original levels).
- Finally, push **PLAY** and use the **DOWN** button to select "Voice No."
- Play and listen to the three samples you've made. Use the **YES/NO** buttons to select between voice number 1, 2 or 3.

Observations:

- One of the obvious critical points about using auto triggering is the more accurately the triggering threshold is set, the more accurately we can determine where our sample will begin.
- As we have demonstrated the auto trigger can be a very useful tool in sampling only the true signal and eliminating unwanted noise.
- It is also very useful when you are attempting to sample a sound that you have little or no control over when it is going to be heard.
- Sampling sounds without noise or long silent gaps in the front helps conserve memory space and saves editing time, too!

6. Resampling Functions

The FZ has three functions in the Source Select sub-mode that generate a new sampling voice by *resampling* one or more previously sampled voices.

- Mix Write is use to splice, layer, or overlap two previously sampled voices. Detuning and level adjustment for each voice are possible.
- X-Mix Write is similar to Mix Write, but it also allows you to set up a cross-fade between both voices.
- Reverse Write creates a new version of a sampled voice with the sound data in reverse order.

6.1 Mix Write

The Mix Write function is detailed in pages 39 through 46 of the *Operations Manual.* Below is the Menu Overview for the complete function.

Mix Write Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <--ESCAPE	ENTER -> <--ESCAPE	<--ESCAPE
MODIFY PLAY	**Source Select** Voice Edit Bank Edit Effect/MIDI Data Dump OPT Software	Sampling Wave Synthesis **Mix Write** X-Mix Write Reverse Write	Define Voice	Voice No. Voice Name
			Voice Select	1st Voice No. 2nd Voice No.
			Keyboard Set	Original key Highest Key Lowest Key
			Level Set	1st Level 2nd Level
			Delay Time	2nd Start:Coarse 2nd Start:Fine
			Detune	1st Tune 2nd Tune
			Execute Mix	Yes/No

Splicing With Mix Write

Combining two or more samples into one is often referred to as *splicing*. The Mix Write and X-Mix Write functions of your FZ go well beyond the basic splicing operations found on other sampling instruments. The samples you combine can be completely different sounds, say a tuba and a hand bell, or pieces extracted from samples like the attack of a kick drum and the ring or a bass guitar note (you extract them with the Truncate operation). Once you've competed a splice, you have a new sample, and that new sample can in turn be spliced to any other sample.

Butt Splicing

The two most common types of splices are the butt splice and the cross-fade splice . The butt splice is so named because the two sounds butt up against each other. The transition from one sound to another is very abrupt (*Figure 27*).

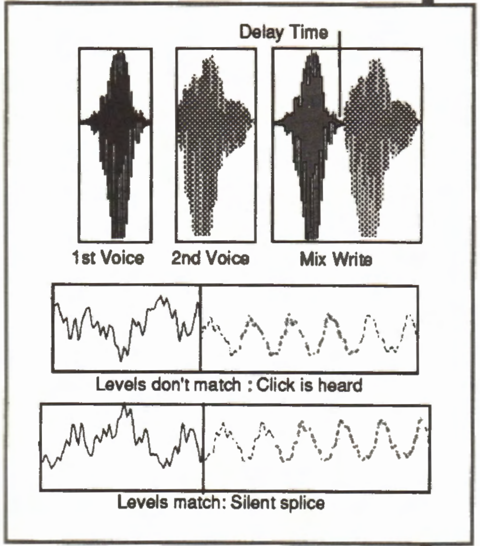

Figure 27: 1st Voice and 2nd Voice are spliced together to create a new FZ voice with Mix Write. The end point of 1st Voice and the start point of 2nd Voice must be carefully matched to produce a silent splice. If the levels are different, a click will be heard at the splice point. The Delay Time determines when the 2nd Voice will start.

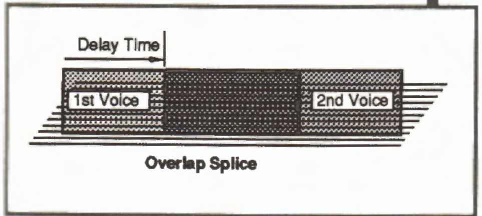

Figure 28: This illustrates an overlap splice created with Mix Write. The Delay Time is set so 2nd Voice starts before 1st Voice ends.

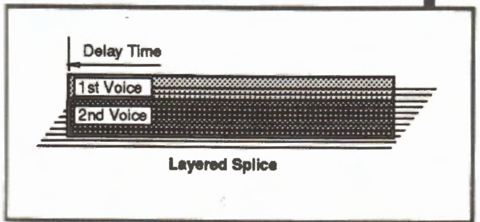

Figure 29: To create a layered splice with Mix Write, the Delay Time is set at zero. Both 1st Voice and 2nd Voice start together.

When you create a butt splice, you have to be careful or you will hear a click or pop at the splice point. This will occur if the end point of the first sample and the start point of the second sample are at different levels. The FZ's graphic display will help you locate the best possible start and end points. Use the Truncate operation (read more about how to truncate in Editing Voices in this book and page 63 in the *Operations Manual*) in the *DISPLAY* mode to set the start and end points to a zero crossing. A zero crossing is a sampling point that sits right on the zero line (the horizontal line that runs through the middle of the graphics display).

You can also match the level by ear. When you truncate the two pieces in preparation for the splice, listen very carefully. Adjust the end point of the first piece so that the sound stops cleanly with no click. Adjust the starting point of the second piece in the same way.

NOTE: When you adjust the start and end points with the Truncate operation of your FZ, the unused portion of the sample is not deleted unless you subsequently enter the Delete Voice operation and select, enter, and execute the "UNUSED PART" parameter. When preparing samples for Mix Write operations, be sure to use Delete Voice to remove the unused parts of the sample. If you don't, they will be copied into the Mix Write splice, regardless of where the start and end points are set. Since you may want to truncate a sample in a variety of different ways for different kinds of splices, it's always a good idea to make a copy of the sample first. Use the Save Voice operation to make a copy of samples, not Copy Voice (see *8.11 Copy, Delete, and Replace*). Then perform the Truncate and Delete Voice operations on the copy. Use the copies for the Mix Write splices. This way, your original sample remains unchanged.

To create the splice, enter the Mix Write function from the Source Select sub- mode. Follow the basic procedure for using Mix Write as outlined in the *Operations Manual*. The sample that you want to hear first is selected with the "1ST VOICE NO." parameter. The second sample in the splice is selected with the "2ND VOICE NO." parameter. In the *Delay Time* operation, use the graphic display to set the "2ND START" at the last sample point in the first sample. This will be the zero crossing you selected with the Truncate operation.

Overlap Splice With Mix Write

You can do a lot more than just a simple butt splice with Mix Write. You can, for example, create an overlap splice by adjusting the "2ND START" parameter so that it occurs before the end of the first sample. This way, you will hear the second sample in the splice before the first sample stops sounding.

Layered Splice With Mix Write

Another very handy use of Mix Write is to use it to create a layered splice . For this type of splice, set the "2ND START" parameter to the start point of the first sample. The resulting sound will be a mixture of the two samples. This generally works best when both samples are of about the same length. One of the strengths of this technique is that it allows you to layer two sounds together without using up two voices to do it. Once the splice is completed (when you hit **EXECUTE**), the levels between the two sounds will be fixed, so don't forget to use the Level Set operation to balance the loudness of your two samples first.

Doubling Effects With Mix Write

You can get doubling effects like those produced with chorus and digital delay devices with the Mix Write operation too. Use the Voice Select operation to select the same sample for "1ST VOICE NO." and "2ND VOICE NO." To create a slap echo (a single repeat echo), use the Level Set operation to lower the "2ND LEVEL" (try it at about 75 or so). Now enter the Delay Time operation and set "2ND START" to produce the desired echo time.

For chorus-like effects, set up a layered splice as described above and then enter the *Detune* operation and alter the "1ST TUNE" and/or "2ND TUNE" parameters to create the chorus. Slight changes in tuning will produce subtle chorusing. Drastic changes will produce honky-tonk type effects.

Experiment #7: Mix Write (Butt Splice)

Focus: Sampling **Editing** Performance

Key Settings:

- 1st Voice, 2nd Voice, Level, Delay , Detune: refer to **Step by Step** below

Operations Manual Page Reference:

- Truncate: 63-65, Delete Voice:79-80, Mix Write 39-46

Step by Step:

- Create two sample voices: Voice 1 = "Sample one." and Voice 2 = "Sample two."
- Truncate "Sample one" to the word "one." Leave some silence at the end of the word. Make sure the new sample ends without a click or pop.
- Enter the Voice Edit: Delete Voice operation and select and execute "UNUSED PART." (See **Experiment #9**.)
- Truncate "Sample two" to the word "sample." Set the start point right at the beginning of the word.
- Enter the Voice Edit: Delete Voice operation and select and execute "UNUSED PART".
- Enter Source Select: Mix Write. Use Define Voice to select "Voice No. 3" (or any "NO SOUND" voice).
- Set "1st VOICE No." to "1" and "2nd VOICE No." to "2".
- Using the display, set the "DELAY TIME" to the end of the 1st Voice (the word "One"). Now when you strike a key you hear "One sample."
- Experiment with the "LEVEL" and "DETUNE" settings.
- Enter and execute Execute Mix.

Observations:

- A butt splice combines two samples by starting the second one immediately after the first one. This makes butt splices very effective with speech samples.
- Sometimes you may need a longer pause between the words of a spliced phrase. You can sample *silence* (sample half a second or so with the input level all the way down), and splice truncated portions of it in between the words in the spliced phrase.
- You can alter the level and tuning of each voice before completing the splice.
- *Mix Write* can do layered and overlapped splices, as well as butt splices. It depends on how the *Delay Time* parameter is set. Setting it to the very start of the 1st voice will produce a layered splice. Setting it in the middle of the 1st voice will create an overlapped splice.

6.2 X-Mix Write

The X-Mix Write function is detailed in pages 47 through 55 of the *Operations Manual.* Below is the Menu Overview for the complete function.

X-Mix Write Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <--ESCAPE	ENTER -> <--ESCAPE	<--ESCAPE
MODIFY PLAY	**Source Select** Voice Edit Bank Edit Effect/MIDI Data Dump OPT Software	Sampling Wave Synthesis Mix Write **X-Mix Write** Reverse Write	Define Voice	Voice No. Voice Name
			Voice Select	1st Voice No. 2nd Voice No.
			Keyboard Set	Original key Highest Key Lowest Key
			Level Set	1st Level 2nd Level
			Delay Time	2nd Start:Coarse 2nd Start: Fine
			Detune	1st Tune 2nd Tune
			Cross Zone	Start: Coarse Start: Fine End: Coarse End: Fine
			Execute X-Mix	Yes/No

Cross-Fade Splicing With X-Mix Write

The X-Mix Write function is identical to Mix Write with the addition of one operation, *Cross Zone*. The Cross Zone operation fades out the first sample while it simultaneously fades in the second sample. This makes it possible for you to create cross-fade splices with your FZ. (You can also do each of the splicing variations described above in Mix-Write with the added twist of having a variable cross zone.)

In a cross-fade splice, the two sounds are overlapped (one fades out as the other fades in). This produces a smooth transition from one sound to the other (*Figure 30*).

The first step in creating a cross-fade splice is to decide what two samples you want to splice together. You are not limited to using entire samples for this function. You can splice together pieces of samples. (Use the Truncate function to extract the portion of a sample you want to use in a splice.)

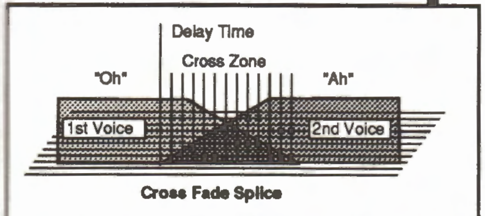

Figure 30: X-Mix Write can create a cross-fade splice. As 1st Voice fades out, 2nd Voice fades in. Delay Time determines where the splice begins. The length of the cross-fade is set with the Cross Zone parameter.

NOTE: When you adjust the start and end points with the Truncate operation of your FZ, the unused portion of the sample is not deleted unless you subsequently enter the Delete Voice operation and select, enter, and execute the "UNUSED PART" parameter. When preparing samples for X-Mix Write operations, be sure to use Delete Voice to remove the unused parts of the sample. If you don't, they will be copied into the X-Mix Write splice, regardless of where the start and end points are set. Since you may want to truncate a sample in a variety of different ways for different kinds of splices, it's always a good idea to make a copy of the sample first. Use the Save Voice operation to make a copy of the samples, not Copy Voice (see *8.11 Copy, Delete, and Replace*). Then perform the Truncate and Delete Voice operations on the copy. Use the copies for the X-Mix Write splices. This way, your original sample remains unchanged.

Once you have the pieces, you're ready to splice them together. Of course, you have to decide which one comes first. You may want to adjust the loudness of the two pieces to get a good balance between the sounds. This is done with the Level Set operation in the same manner as the Mix Write function. Also, you can use the Detune operation (the same as in Mix Write) to alter the tuning of either or both of the samples.

The "2ND START" parameter of the Delay Time operation determines where the cross-fade will begin. The "CROSS START/CROSS END" parameters of the Cross Zone operation set the length of the cross-fade. In general, cross-fades with longer lengths will be smoother than cross-fades with shorter lengths.

6.3 Reverse Write

The Reverse Write function is detailed in pages 56 through 60 of the *Operations Manual.* Below is the Menu Overview for the complete function.

Reverse Write Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <-ESCAPE	ENTER -> <-ESCAPE	<-ESCAPE
MODIFY PLAY	Source Select Voice Edit Bank Edit Effect/MIDI Data Dump OPT Software	Sampling Wave Synthesis Mix Write X-Mix Write **Reverse Write**	Define Voice Voice Select Keyboard Set	Voice No. Voice Name 1st Voice No. 2nd Voice No. Original Key Highest Key Lowest Key Execute Reverse Yes/No

The Reverse Write function does exactly what its name implies. It resamples a sound in reverse. The result is an exact copy of the original sample except it sounds backwards. If you want to create a backwards version of a truncated sample, be sure to use the Delete Voice operation to remove the "UNUSED PART" of the sample before you do the Reverse Write. If you don't, the resulting sample will be a backwards copy of the entire source sample, not just the truncated part.

Experiment #8: X-Mix Write (Cross-Fade Splice)

Focus: Sample **Edit** Performance

Key Settings:

- 1st Voice, 2nd Voice, Level, Delay, Detune: refer to **Step by Step** below

Operations Manual Page Reference:

- Truncate : 63-65, Delete Voice : 79-80, X-Mix Write 47-56

Step by Step:

- Create two one second (1000 ms.) samples. Sing a C with "Ah" (Voice 1) for one, and sing an E with "Oh" (Voice 2) for the other.
- If you need to, use Truncate and Delete Voice to remove any dead spots before or after the sung notes. (See **Experiment** 9.)
- Enter Source Select: X-Mix Write . Use Define Voice to select any "NO SOUND" voice.
- Set "1st VOICE No." to "1" and "2nd VOICE No." to "2".
- Using the display, set the "DELAY TIME" to the minimum value. Set "CROSS START" to the minimum value and "CROSS END" to the maximum value. This sets up the longest possible cross fade splice for the two samples.
- Experiment with different Delay Time and Cross Zone settings.
- Experiment with the "LEVEL" and "DETUNE" settings.
- Enter and execute Execute Mix.

Observations:

- A cross-fade splice combines two samples by overlapping the end of the first and the start of the second. The Cross Zone Time determines how much they overlap. Cross-fade splices are very effective when you want to create a new sound that is merger of two different sounds.
- To create an effective cross-fade splice, your samples shouldn't have any dead spots where they overlap. Use Truncate and Delete Voice to remove silent areas before you make the splice.
- You won't have to worry about level matching the splice point of a cross-fade splice. The function adjusts the start and end point levels of the two samples automatically. This will give you a clean transition from sound to sound with no clicks or pops.

7. *Digital Synthesis*

Although the FZ made its name as a powerful sampling instrument, it has a complete set of analog and digital *synthesis* features as well. You can use your FZ as a full function hybrid (analog/digital) synthesizer as well as a sampler. In fact, it can play synthesized and sampled voices at the same time. The *Wave Synthesis* operation provides four different ways to digitally synthesize various waveshapes.

- The *Preset Wave* operation gives you a choice of six different waveshapes similar to those found on most traditional analog synthesizers.
- The *Sine Synthesis* operation lets you create unique waveshapes via additive synthesis. You have precise control over the level of up to forty-eight harmonics in your custom waveshape.
- The *Cut Sample* operation is a unique method for creating hybrid waveshapes using sampled sounds as source material.
- The FZ's *Hand Drawing* operation lets you modify waveshapes created by any of the above operations by drawing the modifications using the graphic display.

Once you've created a waveshape, you can use the FZ's Voice Edit operations, like DCF envelope, DCA envelope, and LFO, to synthesize any number of unique sounds and effects. (We'll look into voice editing features and techniques in Chapter 7.) Be aware that voices created with Wave Synthesis functions are not samples. Therefore, certain sampling related operations, such as Truncate, Loop, Memory Read, Mix Write, X-Mix Write, and Reverse Write, will not work with synthesized voices.

Voices using digital waveforms don't suffer from the pitch shift problems related to all sampling voices. You can assign a voice to the entire five-octave keyboard. This means synthesized voices work very well in layered keyboard setups. In fact, with the multi-switching technique we describe in Chapter 9, you'll find you can easily do wave table synthesis with the FZ. By wave table synthesis, we mean the ability to select a source waveshape from a list of pre-set waveforms with key velocity. The FZ can switch between as many as sixty-four different waveshapes with velocity.

Synthesis is an art unto itself. If you want to learn the secrets of making music with your FZ's synthesis function, we recommend our course, **Secrets of Analog and Digital Synthesis**.

The Wave Synthesis operations are detailed in pages 31 through 38 of the *Operations Manual.* Here is a Menu Overview of these operations.

Wave Synthesis Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <-ESCAPE	ENTER -> <-ESCAPE	<-ESCAPE
MODIFY PLAY	**Source Select** Voice Edit Bank Edit Effect/MIDI Data Dump OPT Software	Sampling **Wave Synthesis** Mix Write X-Mix Write Reverse Write	Define Voice	Voice No. Voice Name
			Keyboard Set	Original Key Highest Key Lowest Key
			Preset Wave	Saw Tooth Square Pulse Double Sine Saw Pulse Random
			Sine Synthesis	Harmonic Level Execute
			Cut Sample	Source No. Start End
			Hand Drawing	Position Level

7.1 *Preset Wave*

The Preset Wave operation is detailed on pages 33 and 34 of the *Operations Manual.*

The *Pulse, Square,* and *Sawtooth* waveshapes are common to virtually any analog-type synthesizer. The *Double Sine* and *Saw Pulse* shapes are similar to waveshapes that can be created with Casio's CZ series synthesizers. Be aware that the *Random* waveshape will not produce noise effects that you might have expected from its name. It generates a pitched sound with a very bright timbre.

Using the Preset Wave operation, create a voice for each of these preset waves. Name each one with the name of the Preset Wave. For example, "VOICE NO. 13 {SAWTOOTH}." Be sure to save them too (see Chapter 12). They'll serve as the foundation for synthesizer voices created with your FZ. When you want to synthesize a sound from scratch, enter the Voice Editing sub-mode and call up the voice with the waveshape you want to work with. Make a copy with a new name, for instance "VOICE NO. 14 {SYN BRASS}." Do your editing on the renamed copy. (Don't forget to save it when you're all done.)

7.2 *Sine (Additive) Synthesis*

The Sine Synthesis operation is detailed on pages 34 and 35 of the *Operations Manual.*

The Sine Synthesis operation is an additive synthesis waveform generator. Unique waveshapes can be created by dialing in the value of each of up to forty-eight harmonics in the wave. Be aware that you will not hear the wave you've created until you select and enter "EXECUTE" on the Sine Synthesis display. (If you've entered Sine Synthesis directly from any of the other Wave Synthesis operations without first selecting a new voice with the Define Voice operation, you will hear the waveshape you were last working on. The new waveform won't be heard until you select and enter "EXECUTE" in the Sine Synthesis display.)

This is one of the operations where the FZ's graphic display is a real plus. Activate the graphics display by pressing the **DISPLAY** button. You will see the spectrum of the waveform displayed as a bar graph (forty-two bars, one for each harmonic). The height of the bar corresponds to the level of the harmonic in the synthesized wave. Although it isn't mentioned in the *Operations Manual*, you can see the sound display as a waveshape too. Here's how:

- Push **DISPLAY** and adjust the level(s) of the desired harmonic(s) using the **LEFT/RIGHT** buttons and **VALUE** slider.
- Push the **DOWN** button. The display will change from a bar-graph (spectrum) display to a waveform graph. Below the graph, "EXECUTE" will be selected.
- Push **ENTER** and then **YES** to create the waveform. The display will change to show you the shape of the wave you've created.
- Push the **DOWN** button to return to the spectrum display if you want to change the sound further.
- Repeat the second and third steps to see and hear the changes you have made.

7.3 Cut Sample

The Cut Sample operation is detailed on pages 35 through 37 of the *Operations Manual.*

With the Cut Sample operation, you can remove a slice of a sampled or preset waveform to create a new hybrid waveform. The wave defined by the *start* and *end* points you select with this operation is compressed or expanded to fill a ninety-six-point wave table (which can be further modified with Hand Drawing—see below).

The resulting sound is similar to the effect of a very short loop. You can create many interesting waveforms by cutting out various pieces of a sample. Surprisingly, this also works well with pieces of the preset waveshapes too. If you cut from a Preset Wave, set the start and end points to define an incomplete number of cycles. This is a snap to do if you use the display to set the points. (If you set the points on a complete number of cycles, the resulting sound will be no different than the sound you can obtain with the original preset waveshape.)

7.4 Hand Drawing

The Hand Drawing operation is detailed on pages 35 through 39 of the *Operations Manual.*

The waveshapes created with Preset Wave, Sine Synthesis, or Cut Sample are made of ninety-six points. Each point can have any one of two hundred fifty-five levels. The Hand Drawing operation allows you to change the level of any (or all) of the points in the waveshape. You can also start with a new voice and draw an original waveshape from scratch.

8. Voice Editing

Your FZ can do more than just sample and play back sounds. It has several Voice Editing functions and operations that allow you to manipulate and fine tune your basic sample. You can remove parts of the sample you don't want, splice together pieces of different samples, and loop as many as eight separate sections of the sample to repeat in a variety of ways. When you are done editing the sample's data, you can edit its actual sound. You have access to many of the same powerful sound design features found on the most sophisticated synthesizers. And don't forget, you can use these same sound editing features with voices created via the Wave Synthesis sub-mode as well. You can use Casio's unique eight-stage envelopes, LFO, DCF, DCA, and other functions to mold the sample's loudness, tone color, and pitch in any number of interesting ways. The final result will be a unique sampling voice of your own design. When you're done with the editing process, the voice may sound completely different from the original sample or digital waveform. That is up to you.

Voice editing is independent of the sampling process itself. You will be editing sounds that have already been sampled, but the editing functions are quite separate from the actual sampling functions. Even if you never plan on actually sampling sounds yourself, and instead work with factory supplied samples or commercial sample libraries, you'll find that proficiency with voice editing techniques on the FZ is extremely useful. As with the initial sampling process, voice editing involves a series of related functions and decisions.

There are two basic types of voicing parameters available on your FZ. *Data* parameters alter the actual sample data. Sound parameters alter the three aspects of sound, pitch timbre, and loudness of sample or digital synthesis voices. In the pages below, we'll examine the Voice Edit functions and operations of the FZ. We've organized them into Data parameters and Sound parameters for you. The Voice Edit sub-mode is detailed in pages 61 through 81 of the *Operations Manual.*

Below is the Menu Overview for the complete sub-mode.

Voice Edit Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER ->**	**ENTER ->** **<-ESCAPE**	**ENTER ->** **<-ESCAPE**	**<-ESCAPE**
MODIFY PLAY	Source Select **Voice Edit** Bank Edit Effect/MIDI Data Dump OPT Software	<---------------->	Define Voice	Voice No. Voice Name
		Create Voice	Truncate	Start:End
			DCA Envelope	Rate KF Level KF Step,Rate,Level Copy From DCF
			DCF Envelope	Cutoff Frequency Resonance Rate KF Level KF Step,Rate,Level Copy From DCA
			Loop Set	Start:End Loop Time Cross Time Next
			LFO Set	Wave Sync Delay Rate OSC Depth DCA Depth DCF Depth
			Velocity Sens	DCA Level DCA Rate DCF Rate Resonance
			Tune/Mem Read	Tune Mem Read
		<---------------->	Keyboard Set	Original Key Highest Key Lowest Key
		<---------------->	Copy Voice	Voice Name Destination Execute: Yes/No
		<---------------->	Delete Voice	All Part Unused Part
		<---------------->	Replace Voice	Voice Name Destination
		Dump Voice	Load Voice	Voice Name Execute: Yes/No
			Save Voice	Voice Name Execute: Yes/No
			Verify Voice	Voice Name Execute: Yes/No
			Erase Voice	Voice Name Execute: Yes/No

What is an FZ Voice?

The term *voice* has come to mean many things depending on the particular instrument it refers to. Casio has taken a different approach to what makes up a voice on FZ instruments. This unique approach gives the FZ more performance oriented features than even the most expensive sampling instruments available today.

On most sampling instruments, a voice is made up of the sample's data and all parameters settings related to it. Such things as loop points, tuning, envelope, filter and amp settings are linked to the actual sample data as part of the voice. If you want another version of the sound with a different loop, LFO wave, etc., you must copy the voice and change the parameters on the copy. Copying the voice, however, also copies the sample data as well. If the original sample was three seconds long, the copy will take up another three seconds of sampling memory. After you've made a few copies, you'll start to run low on memory. The standard way around this has been to make some (not all) parameters variable with velocity. This is a good compromise as long as the parameters you want to change are among the ones that can be varied.

Casio has designed the FZ so that a voice is independent of its sampling (or digital waveform) data. What does this mean to you?

Simply put, every single *Create Voice* parameter can be made velocity sensitive for any sample on the FZ. Furthermore, you have complete freedom in setting any combinations of these parameters to interact with any velocity. (Be sure to read *9.8 Multi Switching: Expanding the FZ's Velocity Capabilities* on pages ??-??.)

Separating the voice parameters from the the sample data is what makes this possible. When you copy a sampled voice on the FZ, only the Create Voice parameters are copied, not the sample itself. You can make up to sixty-three copies of a single sample, and it won't use a single bit more of sampling memory than the original sample. For every copy of the voice, you can select an entirely different set of start and end points, loops, tuning, envelope, forward or reverse play, LFO wave rates and routings, whatever. Then, using the FZ's powerful Bank and Area features, you can use velocity to switch between the copies. The end result will be sampling and synthesis voices with expressive capabilities that go beyond those offered by virtually any other sampling system. We call the technique multi-switching, and in the next chapter we'll show you how to set it up.

This formidable capability isn't detailed in the *Operations Manual*. In effect, it adds more than one hundred additional velocity parameters to each FZ voice. (Aren't you glad you bought this book?) Keep in mind, this is above and beyond the *Velocity Sensitivity* parameters of the Create Voice function. So, as you read on and experiment with Create Voice parameters, remember that different values for each setting can be switched with velocity.

Define Voice

This operation is used to select the voice to be edited. You may edit voices created with any of the Source Select functions. Sampled or resampled voices will show the word "RECORDED" under the selected voice number. Voices created with the Wave Synthesis function will have the word "SYNTHESIZED" in the display. (Look over *6.6 Common FZ Operations* for additional details.) Within the Create Voice operation, there are seven separate parameters, each with its own set of Current Settings. Truncate, Loop Set, and Memory Read are data parameters. They can only be used with "RECORDED" voices. (In other words, samples.) The remaining Create Voice parameters–DCA envelope, DCF envelope, LFO set, Velocity Sensitivity, and Tune–are sound parameters. They can be used with both "SYNTHESIZED" and "RECORDED" voices.

8.1 Create Voice: Data Parameters

We learned in the previous section that the data of your sample is a series of numbers representing voltage values. Even a very short sample will be made up of thousands (or tens of thousands) of these points. Loop Set and Truncate will require you to select specific points from within this large collection of data.

Graphic Display Mode

The FZ can represent these values as points on a graph with its *DISPLAY* mode. The resulting picture is an excellent representation of the original sampled sound wave. The basics for using the display with sample data are covered on page 65 of the *Operations Manual.* However, a very important feature of the display system isn't explained in the manual, so let's go over it here.

When you are viewing a sample in the Truncate, Loop Set, or X-Mix Write: Cross Zone displays, you can switch back and forth between the "START" and "END" points very easily. Once you are looking at the waveform in the display, simply push the **ENTER** button to toggle from "START" to "END." This makes it very convenient to compare levels, etc. (more on that below).

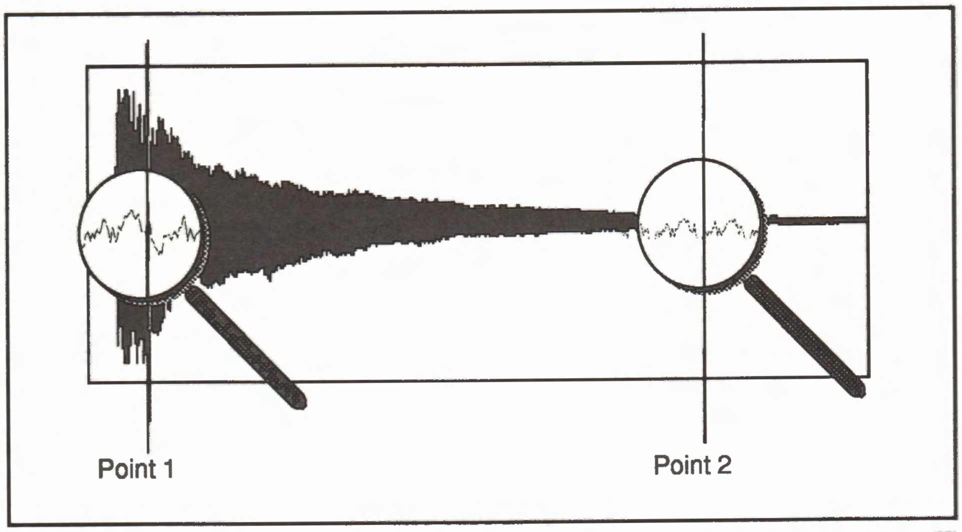

Figure 31: A sample is actually made up of thousands of individual sampling points. FZ operations, such as Loop, Truncate, Mix Write, and X-Mix Write, will often require you to select two specific points from within the entire sample.

8.2 Truncate

Truncate is detailed in pages 63 through 65 of the *Operations Manual.* Below is the Menu Overview showing each of the Truncate parameters.

Truncate Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <--ESCAPE	ENTER -> <--ESCAPE	<--ESCAPE
MODIFY PLAY	Source Select **Voice Edit** Bank Edit Effect/MIDI Data Dump OPT Software	Define Voice **Create Voice** Keyboard Set Copy Voice Delete Voice Replace Voice Dump Voice	**Truncate** DCA Envelope DCF Envelope Loop Set LFO Set Velocity Sens Tune/Mem Read	Start: Coarse Start: Fine End: Coarse End: Fine

Experiment #9 : Sample Start/End Points

Focus: Sample **Edit** Performance

Key Settings:

- Sample Start Point, Sample End Point: refer to **Step by Step** below
- Sample Length: 2000 ms, Sample Rate: 36 kHz

Operations Manual Page Reference:

- Truncate:63-65

Step by Step:

Leave a little space between each word as you speak.

- Enter the Create Voice: Truncate operation of the Voice Edit sub-mode.
- Push **DISPLAY** to see the sample in the FZ's display screen.
- While watching the display, move the **VALUE** slider. You'll see the cursor move indicating the current location of the start point. Under the waveform display the sample number will be displayed. Set the sample start point to the start of the first "I." Write down the sample number. (For example: " 'I' start point = 0000:0374.") (See page 65 of the *Operation Manual* for using the display with truncate.)
- Now move the sample start point to the beginning of the next word, "don't." Write down the sample number. Do this for each word in the phrase. You'll find the display makes it easy to locate the general area where each word begins.
- Reset the start point to the first "I" again. (A cinch to do because you have the number already written down.)
- Now we'll set the sample end point to the end of this word. Push **ENTER**. The display will change from "START" to "END." While looking at the display move the **VALUE** slider and set the end point to the end of the word "I." Write down the sample number. (For example, " 'I' " end point = 0005:0130.") Do this for each word in the phrase. You'll find the display makes it easy to locate the general area where each word ends.
- Now that you know where everything is, practice moving around in the sample. Set the start and end points so you hear only one word at a time, "I," "don't," etc. Try listening to combinations of words, "don't know," "know where," etc.

Observations:

- When you are looking for loop, splice (Mix write and X-Mix Write), and truncate points, you'll need to be able to isolate to small parts of the sample. The FZ's graphic display helps you to find major sections of a sample visually.
- Read more about this technique in Splicing and Loops.

The Truncate operation is used to remove unwanted spots from the beginning and end of a sample. Usually, this means taking out sections of silence. You are not limited to using truncate for removing silences however. You may, for example, wish to remove all but one word or syllable from a phrase. You could also use truncate to remove a small portion of a sample, like the snap of a bass note, for Mix Write or X-Mix Write functions. With the FZ's multi-switching abilities, you can even control the location of the truncate start and end points in a sample with your keyboard dynamics. For example, you can sample a very hard hit on a snare drum and use velocity to switch the start point. High velocities start the sample at the very beginning, producing the hard hit. Lesser velocities would start further into the sample (you'll hear less stick), producing a softer hit.

When you sample a sound (especially if you're not using the auto-trigger operation), very often you will catch a moment or two of silence before the sound begins. If the sound fades away before the sampling process stops, there will be a silent spot at the end of your sample as well. These dead-spots can cause some performance problems, and they waste sampling memory.

If you have a dead-spot at the start of a sample, your FZ will have to play through it every time you play a note. This means that whenever you hit a key, there will be a slight pause between when you hit a key and when you hear your sample.

By now, you should be aware that sampling memory is a very valuable commodity. You only have so much of it, so you must use it wisely. Using up memory to sample silence is very wasteful. Since it is almost impossible to tell exactly when a sound will start and stop, it is almost impossible to create a sample that doesn't have some wasted memory at the beginning or end.

The Truncate operation lets you remove sections from the beginning and end of a sample (*Figure 32*). This makes it a perfect tool for cleaning up your samples. Whenever you create a sample you want to keep, make it a practice to truncate it so it has no dead spots. The sample start point and sample end point should be the first and last samples of your sound, not silence.

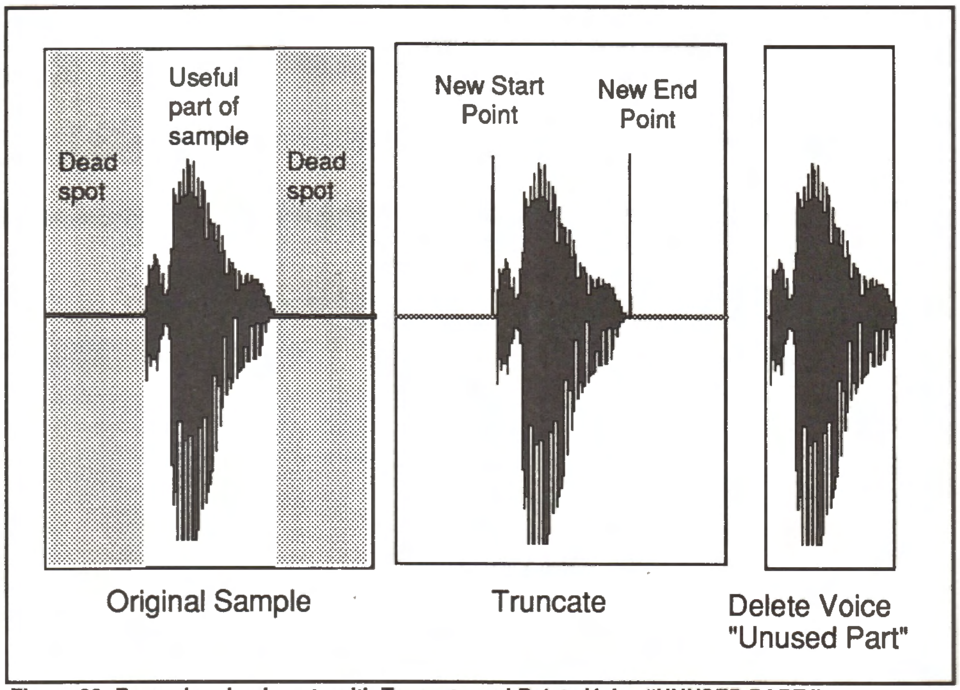

Figure 32: Removing dead spots with Truncate and Delete Voice "UNUSED PART."

NOTE: On the FZ, the unused portion of the sample is not thrown away automatically by setting the Truncate "START" and "END" points. If you want to save memory with Truncate, you must enter the Delete Voice operation and select, enter, and execute "UNUSED PART." This will permanently remove the unwanted portion(s) of the sample. If you are truncating a section of a sound for a resampling function (Mix Write, X-Mix Write, Reverse Write) and want to keep your original sample intact, make of copy of the sample(s) first. Use the Save Voice operation to make a copy of samples, not Copy Voice (see *8.11 Copy , Delete, and Replace*). Truncate and delete the "UNUSED PART" from the copies and use them for the resampling functions.

Don't forget, as we mentioned earlier, you can use Truncate to change where in a sample a voice starts or stops playing. If you make copies of a sampled voice with the Copy Voice function, each copy can have a different set of start and end points. Since more than one FZ voice can share a sample this way, copied voices will not show "UNUSED PART" in the Delete Voice display (since each copy could have a different unused part).

8.3 Loop Set

Loop Set is detailed in pages 69 through 71 of the *Operations Manual.* Below is the Menu Overview showing each of the Loop Set parameters.

Loop Set Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <-ESCAPE	ENTER -> <-ESCAPE	<-ESCAPE
MODIFY PLAY	Source Select **Voice Edit** Bank Edit Effect/MIDI Data Dump OPT Software	Define Voice **Create Voice** Keyboard Set Copy Voice Delete Voice Replace Voice Dump Voice	Truncate DCA Envelope DCF Envelope **Loop Set** LFO Set Velocity Sens Tune/Mem Read	Start: Coarse Start: Fine Start: Ex Fine End: Coarse End: Fine Loop Time Cross Time Next

Perhaps the most powerful sampling operation is looping. Loops can extend the duration of a sample for as long as you want, regardless or how long the original sample was. They can be super memory savers since you may not need to sample a very long sound to be able to play long notes. In fact, once you've looped a sample, you may want to re-truncate the sample again, this time from the start of the sample to the end of the loop. You can discard the part of the sample remaining after an "END" loop (or the last loop if *Next* is set to "SKIP"—more below), since it won't be heard.

Loop Modes: Sustain Loop, End Loop, Timed Loop

The FZ offers a unique set of looping features found on no other sampling instrument. While most samplers allow you to place one or two loops within a sample, the FZ lets you specify up to eight completely independent loops. Copies of a sampled voice made with Copy Voice can each have a different set of loop parameters. A loop is a section within a sample that will be played repeatedly when the sample is heard. Each loop can be placed anywhere within the start and end points of the entire sample. Depending on the loop mode, a loop can occur in differ-

Experiment # 10: Truncate

Focus: Sampling **Editing** Performance

Key Settings:

- Manual Sampling (not Auto Sampling), Sample Length: 2000 ms, Sample Rate : 36 kHz

Operations Manual Page Reference:

- Manual Sampl: 29-30,Truncate: 63-65, Delete Voice:79-80

Step by Step

- Enter the Sampling sub-mode and manually sample the phrase "Rock and Roll." (Start the sampler before you start talking.)
- Listen to the sample to make sure you haven't cut off the beginning of the word "rock."
- Enter the Create Voice: Truncate operation of the Voice Edit sub-mode. Note the values of the end point parameters: "COARSE = 0070" and "FINE = 0312." This is the length of a 2000 ms sample in the FZ'scoarse and fine units.
- Play some rapid notes on the keyboard. Notice that when you play quickly from key to key that you don't hear your voice. (That's because you've sampled some silence before you started to speak.)
- Set the sample start point at the very beginning of the word "rock." Be sure to locate the point so you don't hear a click at the beginning of the word. (Refer to **Experiment 8** for how to locate start and end points.)
- Set the sample end point at the very end of the word "roll." Be sure to locate the point so you don't hear a click at the end of the word.
- Play some rapid notes. You should hear the sampled phrase begin as soon as you strike a key.
- Enter the Voice Edit: Delete Voice operation and select and execute "UNUSED PART".
- Enter the Create Voice: Truncate operation of the Voice Edit sub-mode. Note the values of the end point parameters. These show the new, shortened length of the truncated sample. The "dead spots" have been permanently removed from the sample.
- Reset the start and end points of the truncated version to the start and end of the word "and." Enter the Voice Edit : Delete Voice operation and select and execute "UNUSED PART".
- Enter the Create Voice : Truncate operation of the Voice Edit sub-mode. Note the values of the end point parameters. Compare the values of the end point parameters with the previous two versions. The words "Rock" and the word "Roll" have been permanently removed from the sample.

Observations:

- Truncate and Delete Voice can be used together to permanently to remove dead spots from the beginning of a sample. The truncated sample will be heard as soon as a key is pressed.
- Removing unwanted portions of a sample also saves memory. The truncated version will use less memory than the original. Giving you more memory for new samples.
- You can use truncate to isolate small parts of a sample for splicing, etc.
- On your FZ, the sample is not permanently altered until you execute Delete Voice. Since Delete Voice will alter the original, its a good idea to make a copy of the sample with Save Voice first. Work on the copy. If you make a mistake you can always make another copy and try again.
- **Note:** Copies of a voice made with the Copy Voice operation may each have a different set of start and end points. However, once a voice is copied with this operation, the "UNUSED PART" option is not active with Delete Voice.

ent places, relative to how you play a note. The FZ offers you three different loop modes:

- *Sustain loop* – Any one of the eight loops can be set as a Sustain loop. A sustain loop will continue to repeat as long as you are holding down a key. There can be only one Sustain loop per voice.
- *End loop* – Any one of the eight loops can be set as an End loop. An end loop (often called a *release loop* on other samplers) will continue to repeat even after the key is released. Any part of the sample after the end point of an End loop will not be heard. There can be only one End loop per voice.
- *Timed loop* – This third loop mode allows you to set a specific time value for the loop. A Timed loop will repeat for the specified amount of time regardless of how long you hold down a key.

There can be no more than one Sustain loop and/or one End loop assigned to a voice. There can be more than one Timed loop (up to eight, if there is no Sustain Loop or End Loop assigned).

The Loop Cycle

Since there can be so many loops in a voice, it's important to understand the order in which the loops are played (*Figure 33*). They occur in the same sequence as the FZ's *envelope steps.* Loops will always occur in numerical order: Loop 1, Loop 2, Loop 3, Loop 5, ... Loop 8.

- The voice cycles through the first loop as soon as you hold down a key. If it is not a Sustain or End loop, it repeats for the selected Loop Time and then plays through the sample until it reaches the start of Loop 2.
- If Loop 2 isn't a Sustain or End loop, it repeats Loop 2 for the selected Loop Time and then plays through to the start of the next loop.
- This process continues until all of the loops have been cycled through in order or a Sustain loop is encountered.
- The Sustain Loop is repeated as long as you continue to hold down a key or the sustain pedal. When the key or pedal is let up, the voice plays through the sample until it reaches the next loop start point.
- If this next loop isn't an End loop, it repeats the loop for the selected Loop Time and then plays through the sample until it reaches the next loop start point. This process continues until all eight loops have been played or an End loop is reached.
- If an End loop has been set, then the voice will repeat this loop indefinitely as long as the voice is still audible (even if long envelope release times are used for the DCA and DCF).

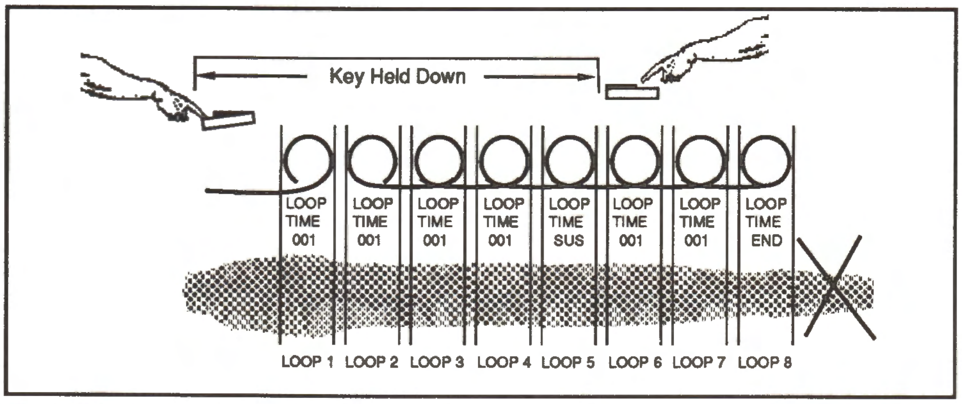

Figure 33: Here's what you would hear with the Loop Time settings shown in this example. When a key is pressed, Loops 1 through 4 will each repeat three times. Next, Loop 5 (the Sustain Loop) will repeat for as long as the key is held. After the key is released, Loops 6 and 7 will each repeat three times. Finally, Loop 8 (the End Loop) will repeat continuously until the sound fades away. The portion of the sample after the End Loop will not be heard. (A Loop Time value of one causes a loop to repeat three times.)

Next Loop: Trace and Skip

Another unique (and very useful) FZ loop feature is *Next* . This feature allows you to skip unlooped portions of a sample. Next affects unlooped portions of the sample in between the current loop and the next loop in the cycle. When this parameter is set to "TRACE," the sample will play normally. When it is set to "SKIP," the voice will jump directly to the start of the next loop, without playing any part of the sample between the previous loop's end point and the next loop's start point (*Figure 34*). Be sure to try **Experiment #14: Next (Trace/Skip)** to get a hands-on demonstration of this powerful feature.

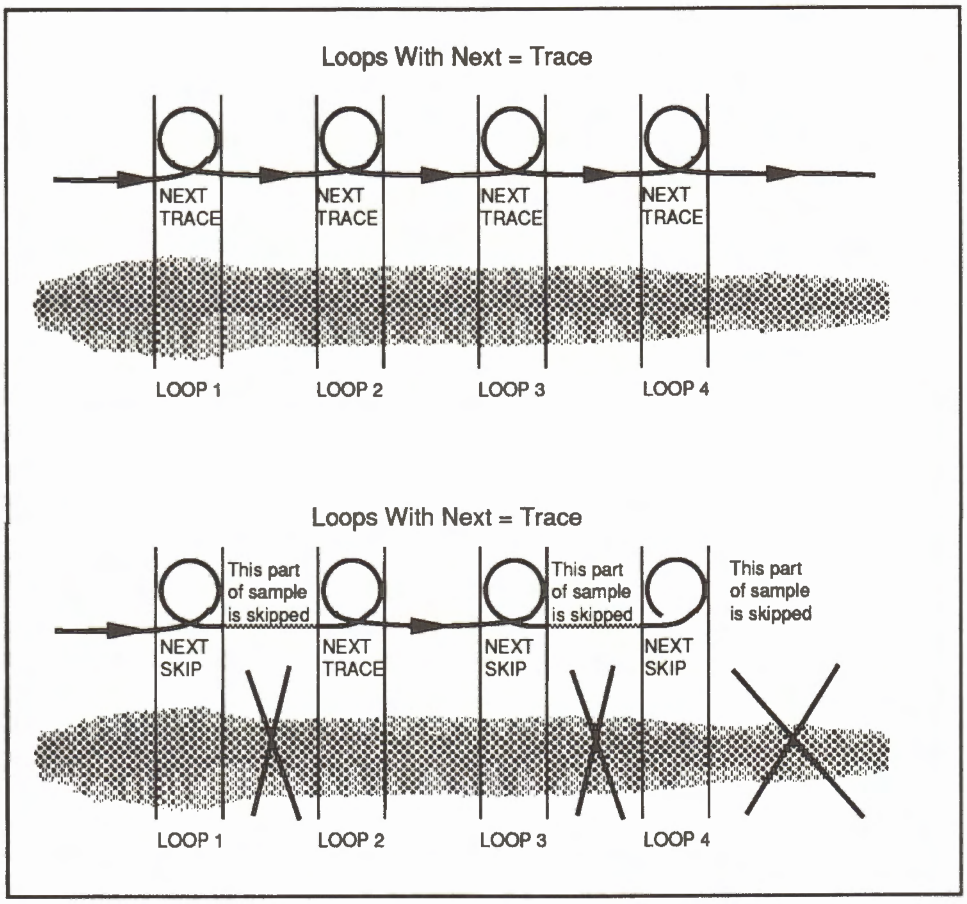

Figure 34: When a loop's Next parameter is set to Trace, the unlooped section of the sample before the next loop is played normally. If it is set for Skip, the sound jumps immediately from the end of the loop to the start of the next loop. The unlooped section is not heard.

Regardless of where in a sample it occurs, all loops are defined by a start point and an end point. These two points can be located anywhere within the sample, from the first sample to the last. Where they are placed will determine what part(s) of the entire sample will be repeated. It is very important to understand that the portion of the sample following an "END" loop (or the last loop if its Next parameter is set to "SKIP") will not be heard. This is the key big memory savings. If you know you'll always be using the sample with the loop active, you can truncate the sample to the loop end point and throw away the unused portion of the sample, using the Delete Voice ("UNUSED PART") operation. This will give you more memory for other samples (*Figure 35*).

Experiment #11: Loop Modes (Sustain, Release)

Focus: Sample **Edit** Performance

Key Settings:

- DCF Cutoff Frequency: 127, DCA Rate 2: 0
- Loop Time: refer to **Step by Step** below
- Loop Start Point, Loop End Point: refer to **Step by Step** below

Operations Manual Page Reference:

- Sampling: 22-29, DCA Envelope: 66-67, DCF Envelope:67-68,
  Loop Set: 69-70

Step by Step:

- Sample the phrase, "Endlessly looping."
- Enter Voice Edit: Create Voice and set the DCF and DCA parameters to the values listed in **Key Settings** above.
- Enter Loop Set and set the Loop 1 start point immediately before "endlessly."
- Set the Loop 1 end point immediately after "endlessly."
- Play the sample at the original pitch. Notice that "endlessly" repeats several times, then you hear "looping." The length of time a key is held doesn't effect the sound.
- Set the "LOOP TIME" value for Loop 1 to "SUS." Now when you will hear "endlessly" as long as you continue to hold down a key. After the key is released, you hear "looping."
- Set the "LOOP TIME" value for Loop 1 to "END." You will hear "endlessly" repeated for as long as you hold down a key. It will continue after you release the key as well. The word "looping" is never heard.
- Reset the "LOOP TIME" for Loop 1 to "SUS" again.
- Use the **RIGHT** button to select Loop 2.
- Set the Loop 2 start point immediately before "looping."
- Set the Loop 2 end point immediately after "looping."
- Set the "LOOP TIME" value for Loop 2 to "END." Now when you play, you'll hear "endlessly" as long as a key is held, and "looping" after the key is released.
- Try playing different pitches and chords.

Observations:

- A sustain loop can make even a short sample last indefinitely by repeating a portion of it while a key is held down.
- The portion of the sample after the end loop will not be heard . This means that you can delete the portion of the sample after the loop end point if you want to save memory. (Remember, save the original first and work on a copy.)
- The portion of the sample after a sustain loop will not be heard until after the key is released.
- In order to hear anything after a key is released, you must adjust the DCF and DCA so they remain open for some amount of time after the key is let up. (Experiments with these settings are coming up.)

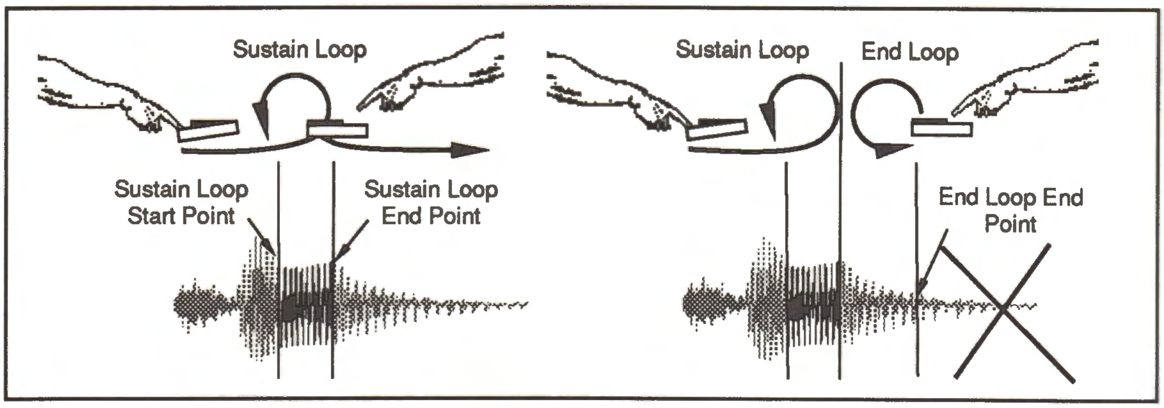

Figure 35: Sustain and End Loops

Experiment #12 : Multi Loops

Focus: Sample **Edit** Performance

Key Settings:

- DCF Cutoff Frequency: 127, DCA Rate 2: 25
- Loop Time: refer to **Step by Step** below
- Loop Start Point, Loop End Point: refer to **Step by Step** below

Operations Manual Page Reference:

- Sampling: 22-29, DCA Envelope: 66-67, DCF Envelope:67-68,
  Loop Set: 69-70

Step by Step:

- Sample the phrase, "One, two , three, four, five, six, seven, eight."
- Enter Voice Edit: Create Voice and set the DCF and DCA parameters to the values listed in **Key Settings** above.
- Enter Loop Set and set the "LOOP TIME" value for all eight loops to "0001." Use the **RIGHT/LEFT** buttons to move from loop to loop.
- Set Loop 1 to start and end on the word "One." You'll find this very easy to do if you use the graphic display to set your loop points. (Don't forget, you can use the **ENTER** button to toggle the display between "START" and "END.")
- Play the sample at the original pitch (keep the key down). Notice that "One" repeats a few times, then you hear "two, three, four, five, six, seven, eight."
- Now set Loop 2 to start and end on the word "two." Play and listen.
- Set the Loop 3 to start and end on the word "three." Do the same with the remaining loops: Loop 4 = "four," Loop 5 = "five," etc. Play and listen.
- You now have eight separate loops in the sample! Each one looping a different word. Experiment with different "LOOP TIME" values
- Try playing different pitches and chords.

Observations:

- You have total freedom as to where you can place each loop's start and end points. For instance, make a loop that cycles the words in reverse order: "One, two, three, four, five, six, seven, eight-eight-eight, seven-seven-seven, six-six-six, (etc.) one-one-one."
- Try looping different combinations of words: Loop 1 = "One two three,"
  Loop 2 = "seven eight," Loop 3 = "five six seven," etc.

Experiment #13: Next (Trace, Skip)

Focus: Sample **Edit** Performance

Key Settings:

- DCF Cutoff Frequency: 127, DCA Rate 2: 25
- Loop Time: refer to **Step by Step** below
- Loop Start Point, Loop End Point: refer to **Step by Step** below

Operations Manual Page Reference:

- Sampling: 22-29, DCA Envelope: 66-67, DCF Envelope: 67-68, Loop Set: 69-70

Step by Step:

- Sample the phrase, "One, two , three, four."
- Enter Voice Edit: Create Voice and set the DCF and DCA parameters to the values listed in **Key Settings** above.
- Enter Loop Set and set the "LOOP TIME" value for Loops 1 and 2 to "0001." Use the **RIGHT/LEFT** buttons to move from loop to loop.
- Set Loop 1 to start and end on the word "One." You'll find this very easy to do if you use the graphic display to set your loop points. (Don't forget, you can use the **ENTER** button to toggle the display between "START" and "END.")
- Now set Loop 2 to start and end on the word "three."
- Play and listen. You'll hear "One-one-one, two, three-three-three, four."
- Set the "NEXT" value for Loop 1 to "SKIP."
- Play and listen. This time you'll hear "One-one-one,three-three-three, four. (The word "two" is gone.)
- Set the "NEXT" value for Loop 1 to "SKIP."
- Play and listen. This time you'll hear "One-one-one,three-three-three, four. (The word "two" is gone.)
- Set the "NEXT"value to Loop 2 to "SKIP."
- Play and listen. Now you'll hear "One-one-one, three-three-three." (The word "four" is gone.)

Observations:

- When a loop's "NEXT" value is set to "SKIP," the FZ will skip any unlooped portion of the sample between the loop's end point and the next loop's start point.
- If the last loop in a sample is set to "SKIP," any portion of the sample after the loop will not be heard.

Cross Time

You may select two different loop types on your FZ, depending on how you set the *Cross Time* parameter (*Figure 36*):

- Setting the Cross Time value to zero creates a *forward loop.* This is the normal loop type for most samplers. It plays from the start point to the end point, then it goes back to the start point and plays to the end point again. This pattern repeats as long as the loop is active.
- Setting the Cross Time to a value greater than zero creates a *cross-fade loop.* A cross-fade loop plays from the start point to the end point over and over again (like a forward loop), but the start and end of the loop overlap. During the overlap, the start and end sections of the loop are automatically balanced against each other. One grows softer as the other grows louder. The Cross Time value adjusts the length of this overlap. The Cross Time parameter functions in essentially the same way as the X-Mix Time parameter (see pages 53 and 54 in your *Operations Manual*).

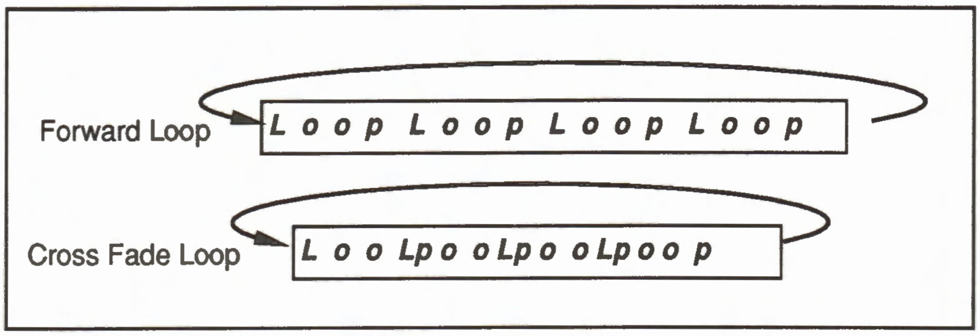

Figure 36: FZ Loop Types

8.4 The Art of Looping

As we mentioned above, looping is one of the most important sampling operations. It is also one of the trickiest to set up correctly. Why are loops so important? Well, for one thing, looping provides you with a method for extending the duration of a sample without using additional memory. You could have a sample of a trumpet that is only one-half second long, but with a well placed loop, you could play the trumpet note for as long as you wanted. Also, the sound of loops has become a very popular effect (á la M-M-M-Max Head-roo-roo-roo-room.)

Why are loops so tricky? Improper placement of the the start and end points can introduce glitches into your sample. If the levels are not properly matched, you will hear repeated clicks or pops during the loop. If the loop is in the wrong spot in the sound, a repetitious breathing may be heard instead of a smooth sustained note. If the start and end points are not set at a proper distance from each other, strange buzzing sounds may be produced during the loop. A lot depends on which two points you select out of the several thousand (or tens of thousands) points that make up your sample. Fortunately, the FZ provides you with special loop types, modes, and functions that will help simplify the looping process. You'll find, for example, that the FZ's graphic display is a great help in setting up loops, since it lets you select loop points visually. Let's take a look at what causes the different kinds of looping glitches and how to avoid them.

Level and Pitch Matching

A loop is very similar to a splice. A forward loop is like a butt splice. The point in the splice where one sound stops and the other starts is just like the point where the loop changes from the end point back to the start point again. The start and end points should have the same level. The waveform pattern of both points should also be very similar, if not identical. If the levels and patterns don't match, you'll hear a pop when the loop jumps from the end point to the start point. If the pitch of the sound drifts at all during the looped section of the sample, a vibrato or other bump in the pitch will result (*Figure 37*). If you want to produce a steady pitched sound with a loop, it's extremely helpful if the source has a steady pitch. If not, you can use a short loop. More on that in a bit.

Level Matching

LOoooPLOoooPLOoooPLOoooP

Loudness changes in a Forward Loop will cause a "glitch

LOoooPqoooO⅃LOoooPqoooO⅃

A Reversing Loop may help compensate for this ...

L O O P L O O P L O O P L O O P

Even loudness producess the best loops

Pitch Matching

Loop drifts flat Loop drifts flat

Any pitch change in the loop will also cause a "glitch"

one vibrato cycle one vibrato cycle

To loop a sound with vibrato, loop one complete vibrato cycle

Figure 37: Level and Pitch Mapping in a Loop

Looking For The Perfect Match

As we mentioned earlier, the graphic display is extremely helpful in finding two points with matching levels and patterns. Here's the basic procedure for setting matching points with the display:

- After entering *Loop Set* from the Create Voice menu, use the **LEFT/RIGHT** buttons to select the loop number you want to work with.
- Select "START: COARSE" with the **UP/DOWN** buttons and push **DISPLAY.** The entire sample will be shown on the display screen. The word "START" will appear under the sampled waveform (as well as the value of the start point).
- Use the **VALUE** slider to move the start point to the general spot you want the loop to begin at. Restrike the key after each move so that you can hear where the new location is.
- Once you've gotten the start point in the ball park, push the **ENTER** button. Now you will see the word "END" under the waveform. Use the **VALUE** slider to place the end point using the same procedure you used for the start point.

- To fine tune the loop points, push the **DOWN** button two or three times until the pattern of the wave is displayed as a curving line.
- If you are still viewing the end point in the display, push **ENTER** to display the location of the loop start point. Use the **VALUE** slider to move the start point to a zero crossing.
- Now, repeatedly pushing the **ENTER** button a few times will overlay the two patterns in the display. This is the key to finding matching points with the FZ's display system. Both patterns will be centered around the cursor in the display. You'll find it makes comparing waveshapes quite simple. If you have a good match, the display will hardly seem to change at all when you switch from start point to end point with the **ENTER** button.
- If the two patterns are very different, adjust the end point location as follows:

  Push **ENTER** to show the end point. Use the **LEFT** button to move the end point location. Continue to push the **LEFT** button until you find a pattern similar to the start point pattern. Don't worry about remembering what the start point looked like, just push **ENTER** to re-display it. Push **ENTER** again to show the end point.
- When the patterns are similar, use the **VALUE** slider to move the loop end point to a zero crossing that matches the setting of the loop start point. There should be no pop or click when you listen to the loop.

> If you have trouble getting a perfect match, you might want to try using a Cross-fade loop here. Cross-fade loops may automatically eliminate clicks at the loop point since they effectively set the loops start and end points to zero. Raising the "CROSS TIME" parameter above zero activates the cross-fade.

Setting Loop Lengths

Since there are so many points within a single sample, where's the best place to put a loop, and how long should it be? There is no single answer to this all- important question. The best locations and lengths of loops will vary with the type of sound and the effect you want to achieve. There are, however, some general guidelines we can pass on to you.

Loops for Repeating Effects

- If you are looping to create repeated speech effects, your ears will help you find the portion of the sample to loop. Don't forget to include some silence at the end of the loop, or the word (or words) that are looped will run together. If you loop the word "loop" for example, and don't leave a pause in the loop, you'll think you're hearing "plooplooploo" not "loop loop loop."
- To create specific looping rhythms, set the loop length to equal the time of the desired note. In other words, if you want to loop an eighth-note rhythm at a tempo of 120 beats per minute (quarter note = 1 beat), make the loop length .25 seconds. Then adjust the start or end points to get a click-free loop. (See below for how the calculations are done.)
- If you need to insert a pause into a loop, sample silence (*Manual Sample* with nothing connected to the input) and splice a piece of it into the sample to be looped.

How To Convert Loop Sizes To FZ Coarse and Fine Values

To convert a musical value to a loop length, multiply the sampling rate times the length of the rhythm unit. The length of the rhythm unit is equal to 60 divided by the number of notes per minute. For example, here's how we got the value for the quarter notes in **Experiment #15.** At a tempo of 120 beats per minute, there are 120 quarter notes every minute. So 60 divided by 120 equals .5. If the sampling rate is 36,000 (36k), then the number of samples in a quarter-note loop is 36,000 multiplied by .5 (18,000).

On the FZ, you can't just set the loop length to 18,000 samples. You must first translate the loop length into the appropriate number of "COARSE" and "FINE" units. Each coarse unit equals 1024 fine units. Each fine unit equals one sample. The highest fine unit value you can set is 1023. To find the right settings for an 18,000 sample loop, you have to do some more figuring.

- For the "COARSE" value, **divide** the loop size by **1024** (18,000/1024 = 17.6) and throw away the remainder. The result,**17**, is the "COARSE" value.
- For the "FINE" value, **multiply** 1024 by the "COARSE" value you just figured out (1024 X 17 =.17408). Subtract this number from the loop size. The result is the "FINE" value (18,000 - 17498 = **592**).
- Set the loop start point in the normal manner. For this example, let's say the "COARSE" value is 5 and the "FINE" value is 312.
- The end point "COARSE" value will be the start "COARSE" value added to the value obtained in the first step above (5 + 17 = **22**).
- The end point "FINE" value will be the start "FINE" value added to the value obtained in the second step above (312 + 592 = **904**).
- The loop is now the correct length for the rhythm: START COARSE = 5, START FINE = 312; END COARSE = 22, END FINE = 904. If the loop has a click, zoom in on the display with the **DOWN** button and move the end point to the nearest zero crossing.

You may occasionally run into a calculation which produces a value greater than 1023 for the "FINE" value. If so, there's an additional step to go through. Suppose that the start point's original "FINE" value was 500 in the example above. The addition would produce 1092 (500 + 592 = 1092). Simply subtract 1024 from this number and use the result as your END FINE value. Then add 1 to the "COARSE" value:

1092 - 1024 = **68**   22 + 1 = **23**

END COARSE = **23**, END FINE = **68**

We've included a table that has the "COARSE" and "FINE" values for a variety of note values and tempos at each of the FZ's sampling rates (*Figure 38*).

Loop Sizes For Common Rhythmic Values

To find the loop size (in sampling points) for any rhythmic value, multiply the **sampling rate** times the length (in seconds) of the desired **rhythm unit.** (Throw away the remainder.) To find the length of the rhythm, divide **60** (seconds) by the number of the desired **rhythm units** per minute. To convert the number of sampling points in the loop to the FZ's COARSE and FINE Values, divide the points by 1024 and throw away the remainder. That's the COARSE value. Then, multiply that number by 1024 and subtract the result from the number of points. That's the FINE value. For example, what loop size will produce eighth note triplets if the tempo is 120 beats=per=minute and the sampling rate used was 18 k?

- **60** seconds/**360** 8th note triplets per minute = **1.66667**
- **18,000** sampling rate X **1.66667** = **3,000.60**
- after throwing away the remainder, the number of sampling points in the loop is 3000
- **3000/1024** = **2.93**
- after throwing away the remainder, the FZ COARSE value is **2**
- **2** X **1024** = **2048** and **3000** - **2048** = **952**
- the FZ FINE value is **952**

Tempo (BPM) Quarter Note =120

		9K Loop Size			18K Loop Size			36K Loop Size		
Rhythm Unit	SECONDS	Points	COARSE	FINE	Points	COARSE	FINE	Points	COARSE	FINE
Whole Note	2	18000	17	592	36000	35	160	72000	70	320
Half Note	1	9000	8	808	18000	17	592	36000	35	160
Quarter Note	0.5	4500	4	404	9000	8	808	18000	17	592
Eighth Note	0.25	2250	2	202	4500	4	404	9000	8	808
Sixteenth Note	0.125	1125	1	101	2250	2	202	4500	4	404
Whole Note Triplet	1.333333	12000	11	736	24000	23	448	48000	46	896
Half Note Triplet	0.666667	6000	5	880	12000	11	736	24000	23	448
Quarter Note Triplet	0.333333	3000	2	952	6000	5	880	12000	11	736
Eighth Note Triplet	0.166667	1500	1	476	3000	2	952	6000	5	880
16th Note Triplet	0.083333	750	0	750	1500	1	476	3000	2	952

Tempo (BPM) Quarter Note = 100

		9K Loop Size			18K Loop Size			36K Loop Size		
Rhythm Unit	SECONDS	Points	COARSE	FINE	Points	COARSE	FINE	Points	COARSE	FINE
Whole Note	2.4	21600	21	96	43200	42	192	86400	84	384
Half Note	1.2	10800	10	560	21600	21	96	43200	42	192
Quarter Note	0.6	5400	5	280	10800	10	560	21600	21	96
Eighth Note	0.3	2700	2	652	5400	5	280	10800	10	560
Sixteenth Note	0.15	1350	1	326	2700	2	652	5400	5	280
Whole Note Triplet	1.6	14400	14	64	28800	28	128	57600	56	256
Half Note Triplet	0.8	7200	7	32	14400	14	64	28800	28	128
Quarter Note Triplet	0.4	3600	3	528	7200	7	32	14400	14	64
Eighth Note Triplet	0.2	1800	1	776	3600	3	528	7200	7	32
16th Note Triplet	0.1	900	0	900	1800	1	776	3600	3	528

Figure 38: Looping Rhythms with the FZ **(continued on next page)**

Tempo (BPM) Quarter Note = 80

		9K Loop Size			18K Loop Size			36K Loop Size		
Rhythm Unit	SECONDS	Points	COARSE	FINE	Points	COARSE	FINE	Points	COARSE	FINE
Whole Note	3	27000	26	376	54000	52	752	108000	105	480
Half Note	1.5	13500	13	188	27000	26	376	54000	52	752
Quarter Note	0.75	6750	6	606	13500	13	188	27000	26	376
Eighth Note	0.375	3375	3	303	6750	6	606	13500	13	188
Sixteenth Note	0.1875	1688	1	664	3375	3	303	6750	6	606
Whole Note Triplet	2	18000	17	592	36000	35	160	72000	70	320
Half Note Triplet	1	9000	8	808	18000	17	592	36000	35	160
Quarter Note Triplet	0.5	4500	4	404	9000	8	808	18000	17	592
Eighth Note Triplet	0.25	2250	2	202	4500	4	404	9000	8	808
16th Note Triplet	0.125	1125	1	101	2250	2	202	4500	4	404

Figure 38 continued.

Inaudible Loops

Loops are most frequently used to extend the duration of a sample without requiring that any additional memory be used. A properly set sustain or end loop can extend even a very short sample to any musical duration you want. In order for the looped sound to retain its natural quality, the loop must be inaudible. In other words, you don't want to hear the looping effect, just a smooth continuous sound. Here are some things to keep in mind when setting up these kinds of loops.

- In general, sounds with noticeable changes in pitch, timbre, or loudness during the course of a note are trickier to loop than sounds that are relatively stable. For this reason, it is usually good to avoid sampling sounds with vibrato, tremolo, or other modulation effects (like chorus, or flanging) if you intend to loop them. The same holds true with many percussion sounds, since their loudness changes radically over a very short time.
- In order to create an undetectable loop, it is important that the pitch and general waveshape in the Area of the loop start and loop end points match as closely as possible.
- Many sound sources drift in pitch over the course of a note. Sometimes this can be avoided by careful performance. At other times, a certain amount of pitch shift may be unavoidable. If a pitch change occurs inside of a loop, you will hear what sounds like vibrato during the loop. As you play higher keys, the vibrato speed will increase, and when you play lower keys, the vibrato speed will slow down.
- If the amplitude or shape of the waveform is dramatically different at the loop points, you will hear repeating tremolo, or breathing, during the loop. The speed of this effect will also increase or decrease as you play higher and lower keys.

As you can see, there are a lot of factors that can effect the sound of a loop. Finding the right starting point and determining the proper length for a loop will often be an exercise in patience, using a trial-and-error approach. However, the art of looping is not as mysterious as many seem to think. There are two basic kinds of loops, *short loops* and *long loops*. Once you understand what they are, how to set them up, and the effects they produce, looping goes from esoteric to common sense.

A short loop is one that a happens so quickly that we are not able to hear it repeat. Instead, we will hear an actual buzz or pitch. This will occur with loop lengths smaller than about 50 ms. (that's 900 samples at a 18K sampling rate). When using a short loop, it is important to match the pitch of the loop with the pitch of the original sample. Care should be taken in the setting of the sampling rate and the tuning of the instrument

Experiment #14 : Tuning Loop Rhythms

Focus: Sample **Edit** Performance

Key Settings:

- Sampling Rate 36 kHz, Sampling Length 1000 ms.
- Loop Set parameters: refer to **Step by Step** below
- For this experiment you'll need to play your FZ and listen to a sequence, drum machine or metronome playing a steady tempo at the same time. You can listen to a repeating sequence of music, or just a steady beat. Set the tempo for 120 beats (quarter notes) per minute.

Operations Manual Page Reference:

- Sampling: 22-29, Loop Set: 69-70

Step by Step:

- Sample the sound of a single finger snap. (Be sure to sample an entire second. You'll need at least a half second of silence following the snap. If you have to, adjust the start point with Truncate so that you hear the snap as soon as you touch a key.)
- Enter Create Voice: Loop Set of the Voice Edit sub-mode.
- Set the "LOOP TIME" value of Loop 1 to "SUS."
- Set the loop start point at the beginning of the snap. Write down this number. (For example, Loop start point = 0002:0069.)
- Set the "COARSE" loop end value to the start "COARSE" value plus **17**. (For example, 0002 + 17 = 0019.)
- Set the "FINE" loop end value to the start "FINE" value plus **592**. (For example, 0069 + 0592 = 661.) Be sure to read, *How To Convert Loop Sizes To FZ Coarse And Fine Values* for how the calculations were done. (*Figure 40* contains a table with the correct values for most of the pitches in the A440 equal tempered scale.)
- Start the music or metronome. Play the original key (C5) on a down beat. You'll hear finger snapping in perfect quarter note rhythm.
- Play the C above, and then the C below (also on down beats.) You'll hear eighth notes and then half notes. Now try it with the F# above, and below the original key. You'll hear quarter note triplets, and half note triplets.
- Try playing two note chords, octave C's, tri-tone C - F#, etc.

Observations:

- If you know the tempo and the sampling rate, you can create loops that will be in perfect time. The arithmetic to figure out the loop lengths is very simple. *Figure 34* shows the formula and has a chart with timing values for many tempos, and the FZ's three sampling rates are already figured out for you.
- Transposing from the original pitch will of course alter the rhythm. Higher pitches will be shorter, lower pitches longer. Octave shifts will double (or halve) the rhythmic value. Tri-tone shifts (a "flat five") will produce a triplet effect (3 against 2).

being sampled. We've given you a chart (*Figure 41*) to help you pick the best settings. It is important to realize that the pitch of a tuned loop is determined by the size of the loop, not the pitch of the sampled sound. Since these loops are so short, many of the problems mentioned above will not be heard. However, these loops are very dry sounding and generally don't sound natural with samples of sounds with effects (like phase shifters or delays) or with samples of groups of instruments playing at once.

Long loops (more than 50 ms.) won't be heard as a pitch, but rather as a repeating section of sound within a sample. A long loop is generally best when you want to loop a sample of multiple instruments (like a string section), or a sound with chorus, or some other repeating type of effect. The trick to setting long loops is to find a section of the sound that can repeat without sounding unnatural. Regardless of whether you use a long loop or a short loop, the first step is to define a loop start point.

Finding the Loop Start Point

Most sounds will change radically during the attack portion of a note. The best place to start a loop is often right after the attack when things settle down. (In a sample of the word "loop," this would be during the "oo" portion of the sound.)This generally holds true whether you are going to set up a long or short loop. Changes in pitch and loudness during the attack of a note are by no means undesirable. Much of what identifies an instrument are changes that occur as the instrument begins to "speak." In most musical situations, once the attack has been completed, the player generally will try to keep the pitch and loudness steady for the held part of the tone. (We're not worrying about effects like vibrato or tremolo here.) You can accomplish this with even a short sample by putting the loop right after the attack. With the help of the FZ's graphic display, it is usually quite easy to see this area and pick a spot to place the loop start point(*Figure 39*).

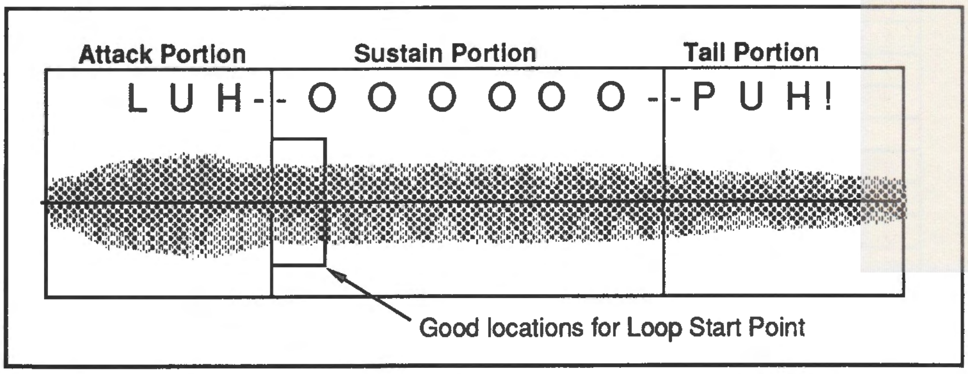

Figure 39: Loop start points generally work best when they are placed in the sustain portion of a sample. The best spot is often right after the attack portion settles down. This illustrates the placement of the start point in a sample of the word "loop" as described in our looping experiments.

You can also locate the start point by ear. Move the sample start point (using the Truncate operation), by small amounts, from the start of the sample. Continue to readjust the start point until you no longer hear any of the attack sound of the sample. Once you've gotten past the attack, adjust the start point in single sample increments until you can start the modified sample without a click. Write down this location and put the sample start point back at the first sample again. Now set the loop start point to the location you wrote down.

Finding the Loop End Point: Short Loops

Ideally, the loop length should be equal to some whole number multiple of the frequency of the sound you are looping. With the graphic display, you merely have to look for the pattern of the waveform and place the end point on a matching point in the pattern some place beyond the start point (*Figure 40*). Use the **ENTER** button to switch the display between the start point and the end point; this will make it easy to find matching spots. If the sampled sound was out of tune, your loop will have to be tuned up if you are going to use it with other, in-tune samples. Use the Create Voice "TUNE" parameter to retune the sample to true pitch.

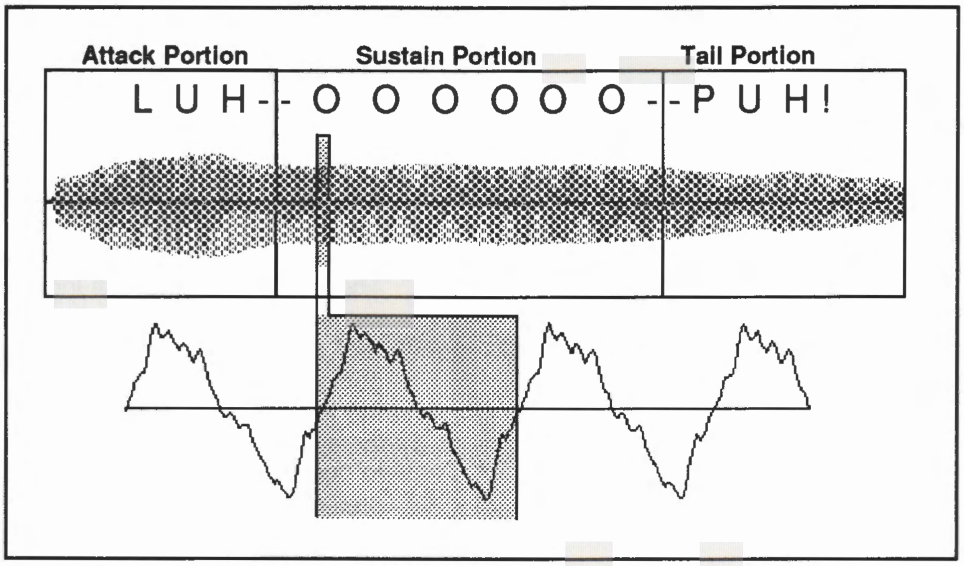

Figure 40: The best short loops are just one complete cycle long.

If you don't use the display, you can locate the end point by the numbers. As we mentioned above, the tuning of the loop is determined by the number of samples in the loop. It doesn't matter what the pitch (if any) of the sample is. Of course, most of the time you will want the loop and sample pitches to match. Always be sure to check the tuning of the instruments as you sample them. To find the loop size, divide the sampling rate by the frequency of the pitch you've sampled. That is the number of samples in the loop.

Converting Tuned Loop Sizes to FZ Coarse and Fine Values

Your FZ lets you specify samples in both "COARSE" and "FINE" values. The "FINE" values are equal to single samples. Each "COARSE" value is equal to 1024 samples. When setting up tuned loops, the loop size will almost always be less than 1024 samples. Since it isn't necessary to start tuned loops on zero crossings, you can usually get great results by setting the start point "FINE" value to zero. (This can eliminate that extra step we used to deal with "FINE" values greater than 1024.) Here's how we recommend setting up start and end point values for tuned loops on the FZ:

- Set the start point with the "COARSE" parameter.
- Set the start point "FINE" value to "0000."
- Set the end point "COARSE" parameter to the same value as the start point.
- Set the end point "FINE" value to the number of samples in the loop.

Frequency And Loop Size For Equal Tempered Scale

To find the loop size for any pitch, divide the **sampling rate** used by the **frequency** of the pitch you want to loop. Round off your answer to a whole number. That will be the correct loop size in your FZ's FINE units. For example, what's the loop size for **F1** at a sampling rate of **36 kHz**?

- **36,000/43.65 = 824.74**
- after rounding off the result, the loop size is **825**

FZ Key Number	MIDI Key Number	Frequency Hertz	9k Loop Size FINE Units	18k Loop Size FINE Units	36k Loop Size FINE Units
A 1	21	27.50	327	655	285 (COARSE+1)
A# 1	22	29.14	309	618	211 (COARSE+1)
B 1	23	30.87	292	583	142 (COARSE+1)
C 1	24	32.70	275	550	77 (COARSE+1)
C# 1	25	34.65	260	519	15 (COARSE+1)
D 1	26	36.71	245	490	981
D# 1	27	38.91	231	463	925
E 1	28	41.20	218	437	874
F 1	29	43.65	206	412	825
F# 1	30	46.25	195	389	778
G 1	31	49.00	184	367	735
G# 1	32	51.91	173	347	694
A 2	33	55.00	164	327	655
A# 2	34	58.28	154	309	618
B 2	35	61.74	146	292	583
C 2	36	65.40	138	275	550
C# 2	37	69.30	130	260	519
D 2	38	73.42	123	245	490
D# 2	39	77.82	116	231	463
E 2	40	82.40	109	218	437
F 2	41	87.30	103	206	412
F# 2	42	92.50	97	195	389
G 2	43	98.00	92	184	367
G# 2	44	103.82	87	173	347
A 3	45	110.00	82	164	327
A# 3	46	116.56	77	154	309
B 3	47	123.48	73	146	292
C 3	48	130.80	69	138	275
C# 3	49	138.60	65	130	260
D 3	50	146.84	61	123	245
D# 3	51	155.64	58	116	231
E 3	52	164.80	55	109	218
F 3	53	174.60	52	103	206
F# 3	54	185.00	49	97	195
G 3	55	196.00	46	92	184
G# 3	56	207.64	43	87	173

Figure 41: FZ Loop Sizes for Tuned Loops **(continued on next page)**

FZ Key Number	MIDI Key Number	Frequency Hertz	9k Loop Size FINE Units	18k Loop Size FINE Units	36k Loop Size FINE Units
A 4	57	220.00	41	82	164
A# 4	58	233.12	39	77	154
B 4	59	246.96	36	73	146
C 4	60	261.60	34	69	138
C# 4	61	277.20	32	65	130
D 4	62	293.68	31	61	123
D# 4	63	311.28	29	58	116
E 4	64	329.60	27	55	109
F 4	65	349.20	26	52	103
F# 4	66	370.00	24	49	97
G 4	67	392.00	23	46	92
G# 4	68	415.28	22	43	87
A 5	69	440.00	20	41	82
A# 5	70	466.24	19	39	77
B 5	71	493.92	18	36	73
C 5	72	523.20	17	34	69
C# 5	73	554.40	16	32	65
D 5	74	587.36	15	31	61
D# 5	75	622.56	14	29	58
E 5	76	659.20	14	27	55
F 5	77	698.40	13	26	52
F# 5	78	740.00	12	24	49
G 5	79	784.00	11	23	46
G# 5	80	830.56	11	22	43
A 6	81	880.00	10	20	41
A# 6	82	932.48	10	19	39
B 6	83	987.84	9	18	36
C 6	84	1046.40	9	17	34
C# 6	85	1108.80	8	16	32
D 6	86	1174.72	8	15	31
D# 6	87	1245.12	7	14	29
E 6	88	1318.40	7	14	27
F 6	89	1396.80	6	13	26
F# 6	90	1480.00	6	12	24
G 6	91	1568.00	6	11	23
G# 6	92	1661.12	5	11	22
A 7	93	1760.00	5	10	20
A# 7	94	1864.96	5	10	19
B 7	95	1975.68	5	9	18
C 7	96	2092.80	4	9	17

Figure 41 continued.

Experiment #15: Tuning Short Loops

Focus: Sample **Edit** Performance

Key Settings:

- Sampling Rate 36 kHz, Sampling Length 1000 ms.
- Loop Set parameters: refer to **Step by Step** below

Operations Manual Page Reference:

- Sampling: 22-29, Loop Set: 69-70

Step by Step:

- We're going to sample the word "Loop" again, but this time sing it at the pitch of "middle C." Try to hold the pitch as steady as you can.
- Enter Create Voice: Loop Set of the Voice Edit sub-mode.
- Set the "LOOP TIME" value of Loop 1 to "SUS."
- Use the display to set the loop start point in the beginning of the "oo" sound, after the "luh" part of the word.
- Leave the display mode by pushing **ENTER.** Reset the start point "FINE" value to "0000."
- To find the loop length divide the sampling rate by 261.6, (the frequency of the pitch "middle C"). For example, 36,000 / 261.6 = 137.9. Round off your answer to a whole number to get the loop length for "middle C" (138 samples).
- Set the loop end "COARSE" value to the same value as the loop start "COARSE" value. Set the loop end "FINE" value to 138.
- Play the original key. You will hear your voice singing "Looooooo." The sustained portion will be tuned to "middle C."
- Enter the display mode and push the **DOWN** button until the pattern of the wave can be seen. Compare the pattern of the start and end points by pushing **ENTER.** If your singing was in tune, the patterns should be identical.
- Try setting up the loop using the display to locate start and end points instead of the math. You'll find that it is very easy to do as long as you can see at least one cycle of the wave in the display.

Observations:

- In a short loop, the number of samples determines the pitch of the loop, regardless of what pitch you sampled.
- When you use the display to set loop points for short loops, your loop will be in tune with the sample. If the sample is out of tune (with A440) the loop will be too. If you set the loop points mathematically, the loop pitch will be in tune (with A440) even if the sample is not.
- You don't have to worry about clicks and pops with short loops.
- The sound of a short loop is very steady. No pitch or timbre changes will occur during the loop (unless you create them with the LFO, amp, or filter functions).
- Short loops work great for single instrument sounds like voice, sax, trumpet, etc. The trick is to put the tuned loop right after the natural "attack" sound (in this case, the "luh" part of the sample.)
- Notice that only a very small portion of the sample is actually heard. You can save a lot of memory by truncating the sample after the loop end point if you don't want to hear that part of the sample.

If you set the loop size on your FZ this way, you will hear the correct pitch. That's all there is to it. We've given you a chart that shows the loop lengths for different frequencies and at the FZ's three sampling rates (*Figure 41*). Sample a pitched sound and set the start point as described above. Use the chart to pick a location for the end point and play a note. The loop will definitely be in tune. The question at this point is, does the loop tuning match the rest of the sample's tuning? Since most sounds fluctuate in pitch by at least some amount, you may still have a few adjustments to make. You can try placing the loop in a different spot, or resample the source (this time check the tuning as you record it).

We go over some additional techniques for tuning up out of tune loops (as well as a lot of other advanced sampling techniques) in our book, ***The Sampling Book***.

Another possibility is to fine tune the loop's pitch to the sample's pitch by ear. If the loop is sharp compared to the sample, make the loop length longer, one sample at a time. (Remember, use the "FINE" parameter to change loop points in single sample increments.) The loop's pitch will drop to match the sample's. If the loop is flat, shorten the loop length, one sample at a time. The loop will rise to match the sample's. When you're done, you'll have a loop that's in tune with the sample, but the whole thing will be out of tune with the rest of the A=440 world (which is why the by the numbers technique didn't work right off the bat). Use the Create Voice "TUNE" parameter get the sample in tune with the rest of your sounds (and next time, check your tuning before you sample).

You may find that you can't match the pitches exactly, one end point and the loop pitch is flat, the next and the loop pitch is sharp. (This can sometimes happen when the frequency of the source can't be divided evenly by the sampling rate.) Here's a trick to try in this situation: multiply the loop length by two and set the end point for the new length. Listen to the new loop pitch and try retuning again. If the pitch difference is still unacceptable, multiply the original loop length by three and try again. Continue until the pitches match. (Each time you increase the loop length in this manner, you will decrease the pitch difference between the loop pitch and the sample pitch.)

If you can't get a perfect match, you may want to try resampling the sound again. This time, adjust its pitch (or the sampling rate) so that the values match those on the chart.

Finding the Loop End Point: Long Loops

As we mentioned above, longer loops are often best with samples of long notes played by multiple sound sources. Good examples are string sections, vocal choirs, multiple oscillator synthesizer patches. Long loops also work well with samples of sounds that have been played through effects devices like chorus, flange, or delay units. The richness that characterizes these sounds is caused by continual changes in their pitch, timbre, and loudness. Sometimes, as in the case of a sound processed with a chorus effect, the rate of change is regular and predictable. More often, as in the case of a string or brass choir, the rate of change varies and seems to be random. Using a short loop on sounds such as these would freeze the change for the duration of the sustain, and that rich quality would be lost. Long loops also work well with mechanical sounds, such as machinery and engines, or any sound with a repetitious nature (like a ticking clock or a telephone's busy signal).

The key for setting a long loop for sounds with regular cycles of change is try to identify one cycle of change and set the loop length to match that cycle. For example, if you wanted to create a sustain loop of a guitar note being played through a phase shifter, you should set the loop to be equal to one complete sweep of the phase shift effect. In the case of a string section, where the change is not regular, it is usually best to make the loop as long as possible (as long as starting and ending Areas of the loop match in pitch, timbre, and loudness). This way, within the overall loop, you have retained as many of the irregularities that occurred during the course of a note as possible, while creating an inaudible loop.

The graphic display will help you identify these patterns of change. Look for repeating patterns in the overall shape of the sample. Set the loop start and end points to match the start and end of one pattern (*Figure 42*). If you can see no regular pattern, look for the longest sustained portion of the sound. Set the loop start and end points to the edges of this area. Now zoom in, using the **DOWN** button so that you can see the exact samples you've set your loop points to and adjust them so that the levels match. (This eliminates clicks in the loop.)

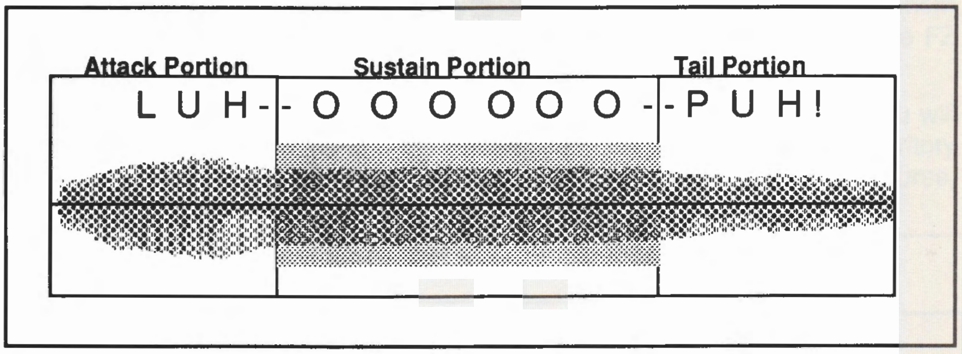

Figure 42: The longest loop that retains the natural sound is your objective when setting up a long loop.

Play a sustained note and listen. If you are unhappy with the loop, reset it on a different pattern.

If you can't identify patterns on the display, you can try finding them by ear. It takes some practice at first, but you can do it. To begin, set the loop start point using the method we outlined above (see *Finding the Loop Start Point).* Next, find the spot where the sample starts to decay (you'll want to put your loop end point before this spot). In our sample of the word "loop," this spot would be right before the "puh" sound at the end of the word. This is the point where the original sound stops holding and begins to fade away. Usually this is easy to hear and you can quickly move the loop end point to a spot right before the decay begins.

You can use the Truncate operation to zero in on this area if you're having trouble hearing it. Move the start point all the way to the end of the sample. (You'll hear nothing when you play a note.) Now start moving it back towards the front of the sample. (Use the "COARSE" parameter.) At first you will hear just a soft click, but soon you will recognize the sound of the end of your sample. It will be relatively soft and percussive (i.e., it will start to fading out the instant you hit a key). As you keep moving the start point, this percussive sound will grow louder and louder, and the fade-out will become longer and longer. Eventually, you will hear the level hold steady for an instant before the sound starts to fade away. This is the point you are looking for. Write down the location. Move the sample start point back to the first sample in the sound and set the loop end point to the location you wrote down.

Your loop is now set to the maximum size for this particular sample. It will probably be too long a loop for most sounds, but once you've set these limits, you can narrow it down from here. Hold down a key and listen. If you feel you can get a better loop, begin by either moving the loop start point forward or the loop end point backward. After you've done this a few times, you'll get a feel for which one you want to try first.

In either case, move the selected loop point a few coarse values at a time. Continue to listen with smaller slices of the sound in the loop each time until you find the one that works best. When you hit a setting that sounds close, don't forget to try the different loop types before moving on. (This is the kind of situation where cross-fade loops may work best.) Write down the locations that sound good so that you can return to them.

If you still haven't found the perfect loop after this process, reset the loop points to the settings that sounded the best and try fine tuning them by moving the loop points in smaller amounts with the fine parameter.

Experiment #16 : Long Loops

Focus: Sample **Edit** Performance

Key Settings:

- Sample Length 2000 ms.
- Loop Set parameters: refer to **Step by Step** below

Operations Manual Page Reference:

- Sampling: 22-29, Loop Set: 69-70

Step by Step:

- Sample the word "Loop." Sing it at "middle C." Draw out the "oo" part of the word ("Loooop").
- Enter Create Voice: Loop Set of the Voice Edit sub mode.
- Set the "LOOP TIME" value of Loop 1 to "SUS."
- Set the Loop 1 start point to just after the "luh" part of the word. Use the graphic display to locate the spot.
- Set the Loop 1 end point to just before the "puh" part of the word using the graphic display to locate the spot.
- Play and listen, the loop probably has an audible click or "thump" in it.
- To fine tune the loop points, push the **DOWN** button two or three times until the pattern of the wave is displayed as a curving line.
- If you are still viewing the end point in the display, push **ENTER** to display the location of the loop start point. Use the **VALUE** slider to move the start point to a zero crossing.
- Use the **ENTER** button to compare the start and end points in the display. If the two patterns are very different, use the technique we describe on page ?? to find a matching pattern.
- Play and listen. The click in the loop should be gone from the sound. (The more closely the two patterns match, the less click or thump will be heard. A perfect match generally yields no click.
- Listen to and compare the same sample with different "CROSS TIME" settings.

Observations:

- In a long loop, the pitch is determined by the sound you sampled, not the size of the loop. Changes in pitch (as well as timbre and loudness) are natural to many sounds. These changes can make finding good loop spots quite a challenge.
- With long loops, you must be careful to avoid clicks caused by mismatched start and end point levels. For the best loops, it's necessary to match both the level, and the overall pattern of the start and end point. The FZ's graphic display will help you find matching spots quickly.
- Even when the patterns match, there may still be "glitches" in the loop if there are loudness, pitch, or timbre changes within the loop. Sometimes, the FZ's Cross Time parameter can help to smooth out this kind of problem.

8.5 Create Voice: Sound Parameters

The parameters discussed above provide you with ways to actually manipulate the data in your samples. As you have seen, sampling data can be moved, re-ordered, and recombined in a variety of musically useful ways. The FZ also provides methods for manipulating your samples even further. Once you have finished editing the data parameters, you can edit sound parameters to alter a sample's pitch, timbre, and loudness. These parameters will effect voices created with Wave Synthesis operations as well.

> The sound functions of your FZ are set up and behave in the same way as most analog synthesizers. In other words, filters, amplifiers, envelope generators, and LFOs do precisely the same things whether you find them on a synthesizer or a sampler. The only difference between them is that synthesizers use oscillators also as a sound source, and the FZ uses digital samples.
>
> If you are already comfortable programming synthesizers, then you will find these functions and their uses quite familiar. If this is new territory for you, once again we would like to recommend our synthesizer course, ***Secrets of Analog and Digital Synthesis.***

The Audio Path

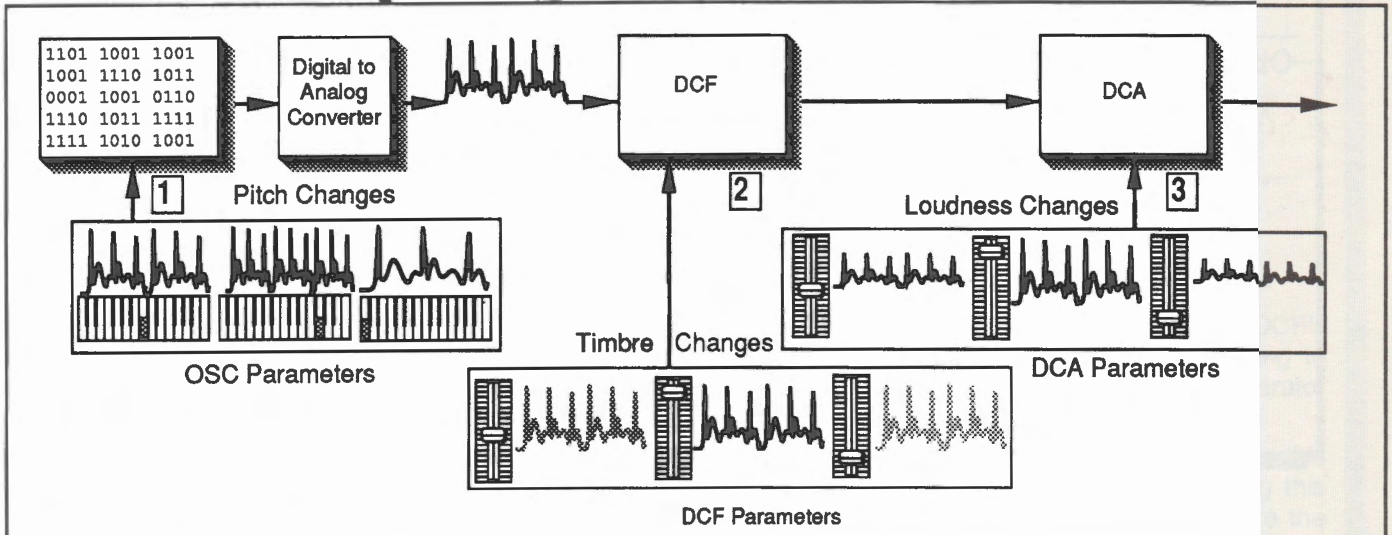

Figure 43: This diagram shows the audio path of an FZ. The sample is converted to an analog signal, and that signal runs through a synthesizer-type DCF and DCA before you hear it. The three main points in the path are where you can change the pitch (1), timbre (2), and loudness (3) of your sampled and synthesized voices. Pitch changes are made by changing the rate the digital data are converted into analog signals. Changing the DCF's cutoff frequency will alter the timbre of the sample. Changing the DCA's output level will alter the loudness of the sample.

When you play a key on your FZ, the sample passes through a synthesizer-type DCF (Digitally Controlled Filter) and DCA (Digitally Controlled Amplifier), before it gets to the output jack on the back of the instrument (*Figure 43*). The DCF and DCA can be controlled by the FZ's envelope generators, keyboard, low frequency oscillators, and performance controls, such as velocity, mod wheel, and pressure.

What Does the DCA Do?

After going through the DCF, your sample is routed through a DCA. The DCA is a sophisticated loudness control. It determines the loudness of your samples and synthesized voices when you play them back. It is important to realize that the DCA is the last stop in the audio path. If the level of the DCA is set to zero (by an envelope or other controller), you will hear no sound from your FZ, even if you're holding down a key.

Experiment #17: What Does The DCA Do?

Focus: Sample **Edit** Performance

Key Settings:

- DCA Level 1: refer to **Step by Step** below

Operations Manual Page Reference:

- DCA Envelope, 66-67

Step by Step:

- Try this with the tuned loop version of "Loop" from **Experiment 16**.
- Enter Create Voice: DCA Envelope.
- For this experiment, all we want to do is listen to the effect that the DCA has on FZ voices. To do this we'll change the value of "LEVEL 1." For now, don't worry about what the envelope generator does. We'll look at it fully in **Experiment 20.**
- When you play a key you should hear " Looo...." Continue as long as a key is held down (like an organ note). While playing, adjust "LEVEL 1" parameter. Play a note, alter the value, replay the same note, alter the value, etc. Notice only the loudness of the voice changes.

Observations:

- The function of the DCA is simple, yet very important. It determines how loud the sound coming from your FZ will be.
- It is important to realize that, when you control the DCA from another function, like the DCA envelope generator, LFO, or velocity, it will do the same thing you did in this experiment – change the loudness of the sample. That's all that happens!
- The DCA must be "opened" for you to hear anything. The FZ defaults to an envelope that opens the DCA fully as soon as you press a key. When you release the key, the DCA quickly shuts down.
- As you'll learn in upcoming experiments, you can really change the character of the original sound by controlling the DCA with other functions.

What Does the DCF Do?

The DCF is a sophisticated tone control (a low-pass filter) that can effect (dramatically!) the timbre or tone color of your voices. The *Level* parameters associated with the DCF alter the *cutoff frequency* of the filter. Changing the DCF's level changes timbre by removing frequencies above its cutoff point from any sound that is passed through it. If the cutoff point is set to its highest value, then no frequencies are removed from the sound (and the DCF is said to be "open"). If the cutoff point is set to its lowest value, then all frequencies are removed from a sound (and the DCF is said to be closed). When the DCF is open, your voice will have its original timbre. As the DCF is closed, the voice will sound progressively darker and darker until, at some point, you hear no sound at all.

Experiment #18: What Does The DCF Do?

Focus: Sample **Edit** Performance

Key Settings:

- DCF Level 1: refer to **Step by Step** below

Operations Manual Page Reference:

- DCF envelope: 67-68

Step by Step:

- Use the same sample you used in the DCA experiment.
- Enter Create Voice: DCF envelope.
- For this experiment, all we want to do is listen to the effect that the DCF has on FZ voices. The DCF's cutoff frequency is controlled by the levels of the envelope. For this experiment, we're going to change the value of "LEVEL 1" only. For now, don't worry about the what the envelope generator does. We'll look at it fully in **Experiment 21.**
- The DCF envelope sustain parameter controls the DCF's cut off point. The FZ automatically sets this to the maximum value when you first create a voice. As you play some notes, try changing this parameter to different values from maximum all the way to minimum and back again. Listen to the effect this has on your sound.
- Notice that at some point near the lowest setting the sound will completely disappear.

Observations:

- The DCF alters the timbre (brightness) of your sample by removing high frequencies from it. The lower the setting, the fewer high frequencies in your sound. It is possible to set the value so low that *no* frequencies (and therefore no sound!) will pass through the DCF. Be aware that, if the DCF is "closed," you will hear nothing, even if the sample is a sustain loop and the DCA is wide open.
- When the DCF is opened all the way, you hear your sample as it was originally recorded.
- It is important to realize that when you control the DCF from another function, like an envelope generator, LFO, pressure or velocity, it will do the same thing you did in this experiment –change the brightness of the sample. That's all that happens!
- As you'll learn in upcoming experiments, you can really change the character of the original sound by controlling the DCF with other functions.

Remote Control

What makes the DCF and DCA such powerful features is that their settings can be changed automatically while you're playing a note, by other functions called envelope generators and LFOs. Equally important, they can also be controlled by your own performance, such as how quickly you strike a key or how much pressure you apply to the key when you hold it down (*Figure 44*).

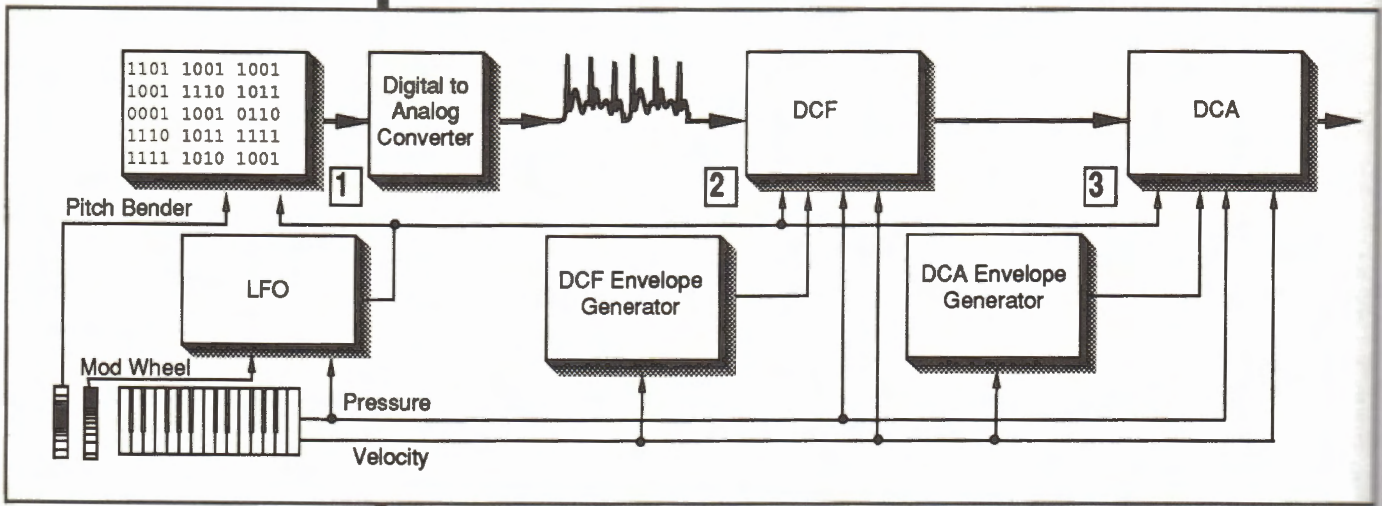

Figure 44: The FZ allows you to control the three main points in the audio path in a variety of ways. This diagram shows the overall controller configuration.

Eight-Stage Envelope Generators

The FZ uses the same powerful eight-stage envelope generators found on Casio's CZ series of digital synthesizers. These envelope generators (EGs) offer considerably more flexibility than the traditional ADSR-type envelope generators. Their basic operation is simple enough:

- Each envelope can have up to eight steps. Each step has an associated rate and level. Higher rate settings produce faster rates; 99 is the fastest. Higher level settings produce greater levels; 99 is the greatest.
- The envelope cycles through the steps in numerical order (similar to the way loops are cycled).
- Any one of the steps can be made the *sustain step.*
- Any one of the steps can be made the *end step.*

The Envelope Step Cycle

The power of your FZ's envelope generator is its flexibility. You can create simple or complex envelope shapes, depending on how many steps you use and where you set the rates and levels. You can even set the EGs to behave exactly like a traditional ADSR envelope. In order to get the most from your EGs, it's important to understand the way they cycle through the various steps.

You can think of the EG as moving the DCF and DCA levels up and down when you play a key. When no keys are down, the DCF level returns to the current setting of the "CUTOFF FREQ" parameter. The DCA level eventually returns to zero (no sound) when no keys are held down. (The speed at which these changes occur is determined by the rate of the EG's last step).

Both the DCF EG and the DCA EG operate in the same way. Here's what happens when you play on the keyboard:

- When you push down a key, the EG's "LEVEL 1" goes from zero to whatever level is set for step 1.
- If step 1 is not set to "SUS," then the EG's level immediately begins to change to whatever step 2's "LEVEL 2" is set for.
- If step 2 is not set to "SUS" or "END," the EG's level immediately begins to change to whatever step 3's "LEVEL 3" is set for.
- This process continues until the "SUS" or "END" step is encountered.
- When the EG hits the "SUS" step, the Level Set for that step is held for as long as a key or the sustain pedal is held down.
- When the key or pedal is released, the EG's level goes to the Level Set for the next step. This continues until the "END" step is encountered.
- When the EG hits the "END" step, its "LEVEL" goes to zero, even if you are still holding down a key or the sustain pedal.

You control how *quickly* the EG changes from level to level between each step with the "RATE" settings. "RATE 1" determines how quickly the EG level changes from zero to "LEVEL 1." "RATE 2" determines how quickly the EG changes from "LEVEL 1" to "LEVEL 2," and so on.

Your envelopes don't have to use all eight steps. When you set a step to "END," it becomes the last step in the envelope cycle. Its "LEVEL" will automatically be set to zero by the FZ.

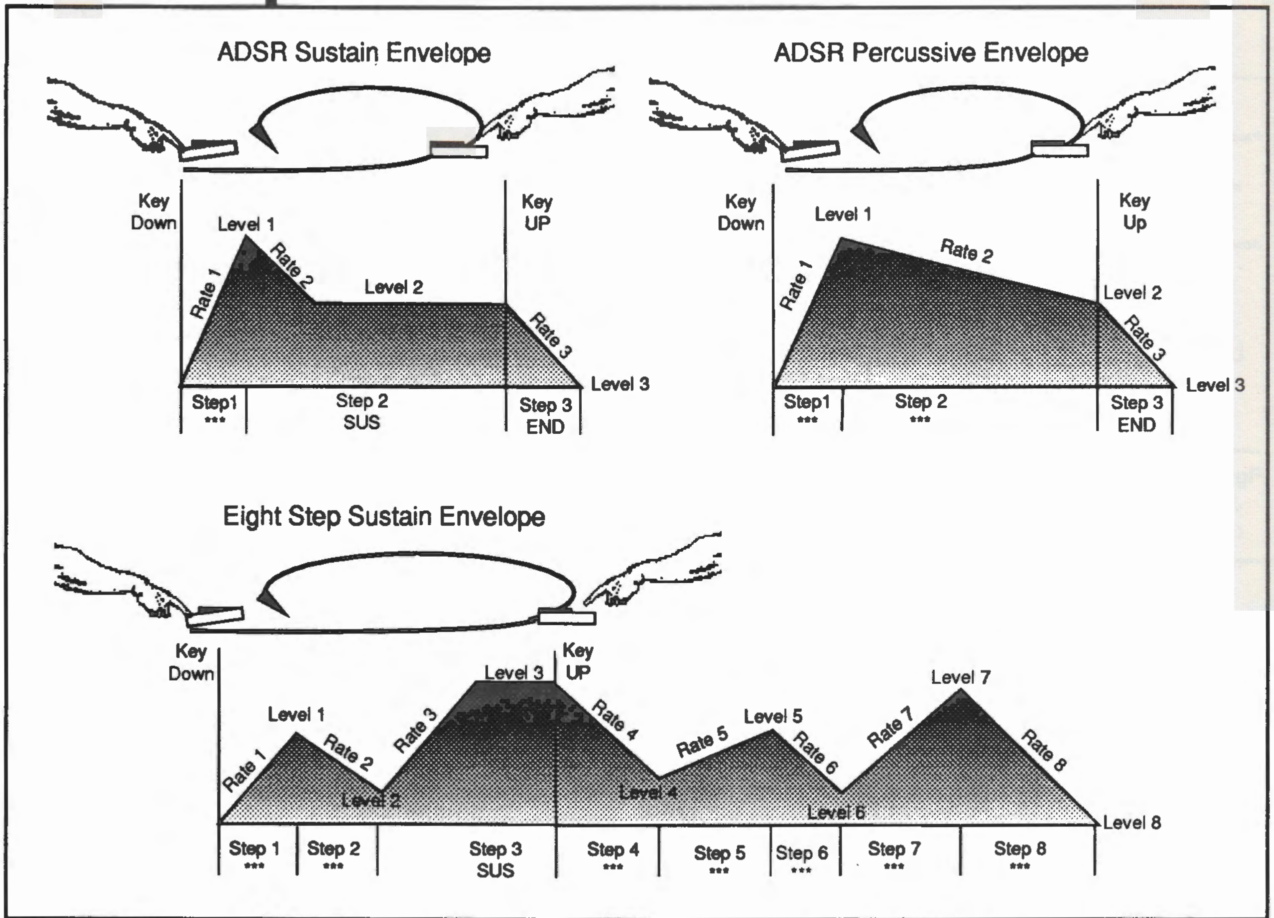

Figure 45: Envelopes with Sustain step level settings greater than 0 will keep the amp or filter open as long as a key is held down. To create a sound that will continue as long as you want, use this kind of envelope with a sample with a Sustain and/or End Loop. Envelopes with no Sustain step fill eventually fade away to nothing, no matter how long a key is held. In this kind of percussive envelope, the rate settings determine how long the fade-out will be. The FZ's envelopes can have up to eight steps.

This is another feature where the FZ's graphic display really shines. You can display your envelope shapes as graphs on the LCD (see page 67 of the *Operations Manual*). You'll find that designing envelopes graphically will help you develop an intuitive "feel" for adjusting EG parameters quickly. The graphs are very easy to understand. If your EG shape has a "SUS" step, it will show as a *horizontal line.*. Rates are shown as *vertical lines.* (Straight up and down is the quickest rate, 99). *Figure 45* shows examples of some useful envelope shapes for the CZ.

8.6 DCA Envelope

DCA envelope is detailed in pages 66 and 67 of the *Operations Manual.* Below is the expanded Menu Overview showing each of the DCA envelope parameters.

DCA Envelope Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <-ESCAPE	ENTER -> <-ESCAPE	<-ESCAPE
MODIFY PLAY	Source Select **Voice Edit** Bank Edit Effect/MIDI Data Dump OPT Software Dump Voice	Define Voice **Create Voice** Keyboard Set Copy Voice Delete Voice Replace Voice	Truncate **DCA Envelope** DCF Envelope Loop Set LFO Set Velocity Sens Tune/Mem Read	Rate KF Level KF Step 1-8 Rate 1-8 Level 1-8 Copy From DCF

Rate KF

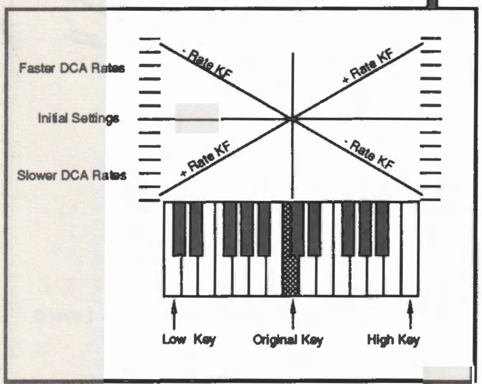

Figure 46: DCA Rate KF

This parameter is used to scale the rates of your DCA envelope within the range of keys set with the Keyboard Set operation. The resulting effect will be changes in the attack(s) and decay(s) of your sound's loudness based on where you play on the keyboard.

If the value for this parameter is positive ("+01" to "+15"), rates will get faster as you play above the Original Key and get slower as you play below the Original Key. If the value for this parameter is negative ("-01" to "-15"), levels will get slower as you play above the Original Key and get faster as you play below the Original Key. Be aware that this parameter interacts with the current "RATE" values. For instance, if one or more rates is already set for 99, *Rate KF* can't make them change any faster.

Level KF

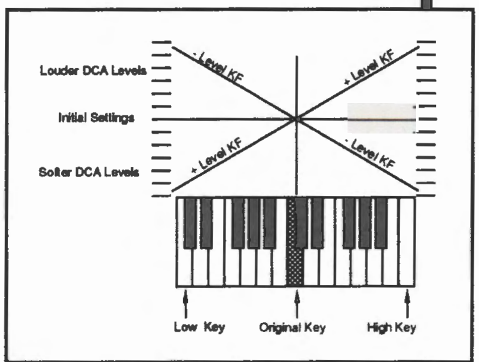

Figure 47: DCA Level KF

This parameter is used to scale the levels of your DCA envelope within the range of keys set with the Keyboard Set operation. The resulting effect will be changes in your sound's loudness based on where you play on the keyboard. If the value for this parameter is positive ("+01" to "+15") levels will increase as you play above the Original Key and decrease as you play below the Original Key. If the value for this parameter is negative ("-01" to "-15"), levels will decrease as you play above the Original Key and increase as you play below the Original Key. Be aware that this parameter interacts with the current "LEVEL" values. For instance, if one or more levels is already set for 99, *Level KF* can't make them get any louder.

NOTE: *DCA Level KF* is the key to creating up-key layer cross-fades when designing Banks and Areas on the FZ. You can overlap the Areas of two voices in a Bank and use the *Level KF* parameters for the voices to create a cross-fade. Be sure to read about Keyboard Splits in *Editing Banks.*

Step, Rate, and Level

These parameters are used to set up the actual envelope shape as described above in *Eight-Stage Envelopes.* Use the **UP/DOWN** buttons to select the desired EG parameter and the **LEFT/RIGHT** buttons to select the desired step.

Copy From DCF

This convenient feature lets you copy the current "STEP," "RATE," and "LEVEL" values from the DCF envelope to the DCA envelope.

Experiment #19: DCA Envelopes

Focus: Sample **Edit** Performance

Key Settings:

- DCA Envelope parameters:refer to **Step by Step** below

Operations Manual Page Reference:

- DCF envelope: 67-68, DCA envelope: 67-68

Step by Step:

- Use any voice with an "END" loop.
- Enter Create Voice: DCF envelope and set the value of "CUTOFF FREQ" to "127."
- Enter Create Voice: DCA envelope and set the envelope parameters as follows:

DCA	1	2	3	4	5	6	7	8
STEP	***	***	***	***	***	***	SUS	END
RATE	99	75	99	75	99	75	99	92
LEVEL	99	0	99	0	99	0	99	0

- Push **DISPLAY** to display the envelope graphically. You will see an envelope shape with four peaks.
- Play a note. You will hear three quick attacks and then a held tone while the key is down.
- Reset "STEP 1" to "SUS" and play a key. Now you will hear a held tone while the key is down, and three short attacks after the key is lifted.
- Listen to the difference in the envelope when you change each step to "SUS."
- Listen to the envelope with none of the steps set to "SUS."
- Replay a note while adjusting the value of "RATE 1"parameter. Change the value and play a note. You will hear the note fade in (quickly at first, more slowly as you increase the values). Be sure to wait to hear the complete event (they'll get pretty long!) before you stop the note and reset the value. When you've listened to the complete range of values, reset the value to its original setting.
- Repeat the above with each of the other rates.
- When you are done with the rates, try the same procedure with the level values.

Observations:

- The DCA envelope generator can dramatically alter a sample's loudness shape. In order to hear the original loudness dynamics of a sample, set the envelope to the default values. Changing from the default settings will create new loudness shapes for the sample.
- Envelopes with a sustain step will continue as long as a key is held down. Envelopes with no sustain step will fade away eventually, even if you continue to hold down a key.
- If you want to hear DCA release effects, make sure that the filter envelope is open, or set for the same, or even longer, release time as the DCA envelope.

8.7 DCF Envelope

DCA envelope is detailed in pages 66 and 67 of the *Operations Manual.* Below is the expanded Menu Overview showing each of the DCA envelope parameters.

DCF Envelope Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <-ESCAPE	ENTER -> <-ESCAPE	<-ESCAPE
MODIFY PLAY	Source Select **Voice Edit** Bank Edit Effect/MIDI Data Dump OPT Software	Define Voice **Create Voice** Keyboard Set Copy Voice Delete Voice Replace Voice Dump Voice	Truncate DCA Envelope **DCF Envelope** Loop Set LFO Set Velocity Sens Tune/Mem Read	Cutoff Freq Resonance Rate KF Level KF Step 1-8 Rate 1-8 Level 1-8 Copy From DCA

Cutoff Frequency

This parameter sets the initial brightness of the DCF. The changes in timbre caused by the DCF envelope will start at and return to this setting. The default setting for this parameter is "000." Be aware that cutoff frequency settings and DCF envelope level settings interact. For example, if the cutoff frequency is set to the maximum amount (127), the EG level settings will have no further effect on the brightness of your sound. Why? Once the filter is "wide open" (by setting the Cutoff to maximum) it simply can't be opened any further.

You can use this interaction to your advantage when you make sounds with the FZ. The cutoff parameter can be used to "scale" all of the levels in a multi-step envelope at once. Here's how :

- Temporarily, set one of the steps of the DCF EG to "SUS". For now, set the level of that step to zero.
- While holding down a key, increase the Cutoff Frequency setting until you can just barely hear your sound. (Use a sound with a Sustain loop.)
- Reset your DCF EG steps, rates, and levels for whatever effect you want.
- Now, the Cutoff Frequency parameter can scale all of the DCF EG levels at once. Increasing its value will make the overall envelope brighter, and decreasing it will make the overall envelope darker. Unless you need to make a drastic change in the envelope, you'll find this is a good way to "fine tune" your sounds.

Resonance

Resonance is used to emphasize frequencies at or near the filter's cutoff frequency. This can produce a "nasal" characteristic in a sound's timbre if the DCF envelope level settings don't change much from step to step. If the level settings do change between steps, a "wah" quality (characteristic of analog synthesizers) will be produced.

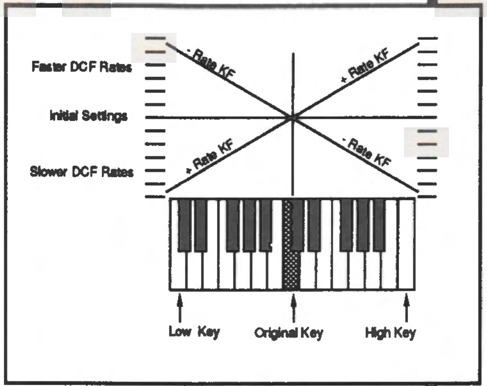

Figure 48: DCF Rate KF

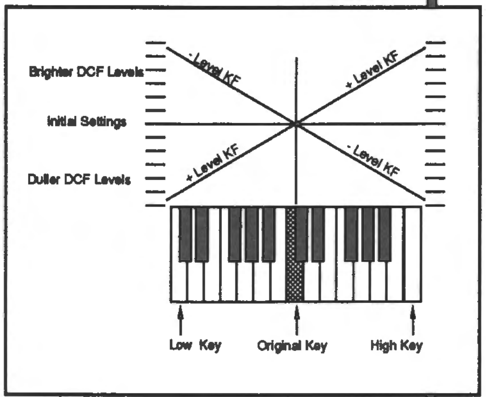

Figure 49: DCF Level KF

Rate KF

This parameter is used to scale the rates of your DCF envelope within the range of keys set with the Keyboard Set operation. The resulting effect will be changes in the attack(s) and decay(s) of your sound's timbre based on where you play on the keyboard. If the value for this parameter is positive ("+01" to "+15"), rates will get faster as you play above the Original Key and get slower as you play below the Original Key. If the value for this parameter is negative ("-01" to "-15"), levels will get slower as you play above the Original Key and get faster as you play below the Original Key. Be aware that this parameter interacts with the current "RATE" values. For instance, if one or more rates is already set for 99, *Rate KF* can't make them change any faster.

Level KF

This parameter is used to scale the levels of your DCF envelope within the range of keys set with the Keyboard Set operation. The resulting effect will be changes in your sound's timbre based on where you play on the keyboard. If the value for this parameter is positive ("+01" to "+15"), timbre will get brighter as you play above the Original Key and get duller as you play below the Original Key. If the value for this parameter is negative ("-01" to "-15"), timbre will get duller as you play above the Original Key and brighter as you play below the Original Key. Be aware that this parameter interacts with the current "LEVEL" values. For instance, if one or more levels is already set for 99, *Level KF* can't make them get any brighter.

Step, Rate, and Level

These parameters are used to set up the actual envelope shape as described above in Eight-Stage envelopes. Use the **UP/DOWN** buttons to select the desired EG parameter and the **LEFT/RIGHT** buttons to select the desired step.

Copy From DCA

This convenient feature lets you copy the current "STEP," "RATE," and "LEVEL" values from the DCA envelope to the DCF envelope.

Experiment #20: DCF Envelopes

Focus: Sample **Edit** Performance

Key Settings:

- DCF and DCA Envelope parameters:refer to **Step by Step** below

Operations Manual Page Reference:

- DCF envelope: 67-68, DCA envelope: 66-67

Step by Step:

- Use any voice with an "END" loop.
- Enter Create Voice : DCA envelope and set the envelope parameters as follows:

DCA	1	2
STEP	SUS	END
RATE	99	30
LEVEL	99	0

- Enter Create Voice: DCF envelope and set the "CUTOFF FREQ" value to "000." Set the envelope parameters as follows:

DCF	1	2	3	4	5	6	7	8
STEP	***	***	***	***	***	***	SUS	END
RATE	99	75	99	75	99	75	99	92
LEVEL	99	0	99	0	99	0	99	0

- Push **DISPLAY** to display the envelope graphically. You will see an envelope shape with four peaks.
- Play a note. You will hear three quick attacks and then a held tone while the key is down.
- Reset "STEP 1" to "SUS" and play a key. Now you will hear a held tone while the key is down, and three short attacks after the key is lifted.
- Listen to the difference in the envelope when you changing each step to "SUS."
- Listen to the envelope with none of the steps set to "SUS."
- Replay a note while adjusting the value of "RATE 1"parameter. Change the value and play a note. You will hear the note fade in (quickly at first, more slowly as you increase the values). Be sure to wait to hear the complete event (they'll get pretty long!) before you stop the note and reset the value. When you've listened to the complete range of values, reset the value to its original setting.
- Repeat the above with each of the other rates.
- When you are done with the rates, try the same procedure with the level values.

Observations:

- The DCF envelope generator can dramatically alter a sample's timbre shape. To hear the timbre dynamics of the original sample, set the envelope to its default settings. Changing these settings will create new timbre shapes.
- If your DCF envelope closes the DCF (very low or zero sustain values) and nothing else keeps it open (like the LFO or pressure), you'll hear no sound regardless of how the amplifier envelope is set.
- If you want to hear DCF release effects, make sure that the amplifier envelope release is set to the same, or even longer, release time as the DCF envelope.

Experiment #21: Synthesizing ADSR Sustain Envelopes

Focus: Sample **Edit** Performance

Key Settings:

- Master Volume: optimum setting
- DCA Envelope Sustain Level: refer to **Step by Step** below

Operations Manual Page Reference:

- DCF envelope: 67-68, DCA envelope: 66-67

Step by Step

- Use the same voice as with the other envelope experiments.
- If necessary, set the DCF envelopes to its default settings.
- Enter Create Voice: DCA envelope and set the envelope parameters as follows:

DCA	1	2	3
STEP	***	SUS	END
RATE	99	99	50
LEVEL	99	99	0

- The EG settings given above create an ADSR type envelope with your FZ. Rate 1 = Attack, Rate 2 = Decay, Rate 3 = Release, and Level 2 = Sustain.
- Lower the DCA envelope sustain level ("LEVEL 2" in this example) to the point where the sound is barely audible. Write down the value. Return the level to its maximum value.
- From now on, when setting up sustain envelopes for this sample, never set the sustain level below this point. This will give you the cleanest signals possible for the sample.
- If you want to hear the sample's natural loudness dynamics, set the envelope parameters to the default settings given above.
- If you want to synthesize new dynamics, set the sustain level parameter to a loudness level between 99 and the the level you wrote down. Exactly where you set it within these limits will depend on the effect you are trying to create. Adjust the rates to suite your taste.

Observations:

- Even a very short sample can have a natural sounding, unlimited sustain if the DCA envelope and sustain loop are set properly.
- If you sampled the sound with compression, this is the way to restore the original dynamics. The advantage of this technique is improved signal to noise, which means that your samples will be quieter! To restore the peak that gets squashed during compression, set the sustain to about 75. Now adjust Rate 2 (make it slower) until the peak sounds natural.
- You can use any number of steps in your envelope shapes. Any one of those steps can be the sustain step. For the quietist possible signals, don't set any of the levels below the limit you found in step four above.

Experiment #22: Synthesizing ADSR Percussive Envelopes

Focus: Sample **Edit** Performance

Key Settings:

- DCA and DCF Envelope parameters: refer to **Step by Step** below

Operations Manual Page Reference:

- DCF envelope: 67-68, DCA envelope: 66-67

Step by Step

- Use the same voice as with the other envelope experiments.
- If necessary, set the DCF envelopes to its default settings.
- Enter Create Voice: DCA envelope and set the envelope parameters as follows:

DCA	1	2	3
STEP	***	***	END
RATE	99	99	50
LEVEL	99	0	0

- The EG settings given above create an ADSR type envelope with your FZ. Rate 1 = Attack, Rate 2 = Decay, Rate 3 = Release, and Level 2 = Sustain. (Sustain is always 0 in a percussive envelope.)
- Adjust the decay rate until sample has the percussive shape you want.
- If you want to, adjust the release rate to simulate "ring." (You may have to alter the DCF's "CUT OFF FREQ" or the rate of the "END" step in order to hear the DCA release.)
- If you want to create an unnatural attack for the sample, adjust the attack time.

Observations:

- Even a very short sample can have a natural sounding, long decay if the DCA envelope and sustain/end loop are set properly.
- If you sampled the sound with compression, this is the way to restore the original dynamics. The advantage of this technique is improved signal to noise, which means that your samples will be quieter! To restore the original decay that was "squashed" during compression, adjust the decay rate (make it slower) for a natural sounding fade out. Try using this envelope shape with a piano sample to create plucked (shorten Rate 2) and bowed (lengthen Rate 1) piano sounds.
- You can use any number of steps in to make a percussive envelope shape as long as none are set for "SUS."

8.8 LFO Set

Like EGs, an LFO is a control function. It allows you to continuously vary pitch, timbre, and loudness with a variety of waveshapes. This type of control is called *modulation*, and the FZ has a special controller (called the *Mod Wheel*) that can be used in conjunction with the LFO (more on that in a while). LFO set is detailed in pages 66 and 67 of the *Operations Manual.* Below is the expanded Menu Overview showing each of the LFO set parameters.

LFO Set Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <-ESCAPE	ENTER -> <-ESCAPE	<-ESCAPE
MODIFY PLAY	Source Select **Voice Edit** Bank Edit Effect/MIDI Data Dump OPT Software	Define Voice **Create Voice** Keyboard Set Copy Voice Delete Voice Replace Voice Dump Voice	Truncate DCA Envelope DCF Envelope Loop Set **LFO Set** Velocity Sens Tune/Mem Read	Wave LFO Sync Delay Rate OSC Depth DCA Depth DCF Depth

Wave

You can select from the following six waveshapes for LFO effects on the FZ: random, square, triangle, saw down, saw up,and sine.

LFO Sync

This parameter allows you to control the synchronization of the LFO. You will only notice this effect when you play more than one note at the same time (and you must be modulating the OSC, DCF, or DCA with the LFO). When the sync is set to "ON," the modulation for all notes you play assigned to this voice will be "in time" with the first note held down. When the sync is set to "OFF," the modulation for each note will begin relative to when each key is struck. If you want to create ensemble effects (lots of instruments playing at once), you'll find the "OFF" setting works very well. If you want a uniform effect (like the tremolo of a vibraphone), the "ON" setting works best. Be aware that if you create a Bank with several different voices, you can synchronize the LFO for all of the voices by simply setting the same "RATE" for each voice and setting the sync parameter to "ON."

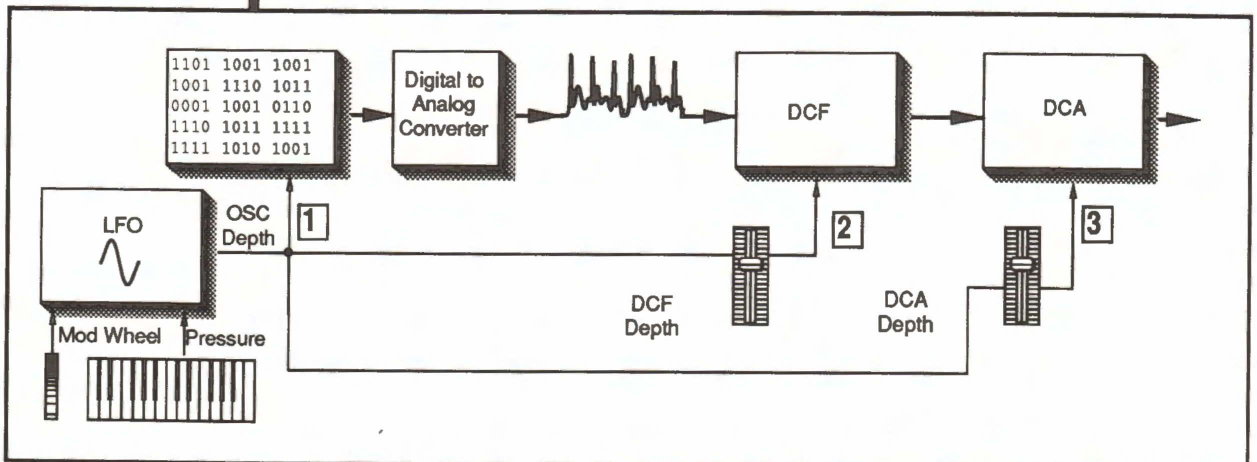

Figure 50: The LFO is routed to all three points in the audio path. Note that the amount of LFO signal sent can be controlled by pressure (After Touch) and the Mod Wheel, as well as the value of the LFO OSC Depth parameter.

Delay and Rate

The *Delay* parameter determines the rate that depth of your LFO effect will change from zero to the current depth settings. Higher delay settings cause longer LFO fade-ins. This produces a similar effect to fading in the LFO with the wheel, but it leaves both your hands free to play on the keyboard.

OSC, DCA, and DCF Depth

You can set the amount of LFO effect independently for the OSC (vibrato effects), DCA (loudness tremolo effects), and the DCF (timbre tremolo effects) with these three settings. Be aware that these settings are also independent of the Mod Wheel. If any of the depth settings is greater than zero, the LFO effect will be heard even if the Mod Wheel is all the way off.

Experiment #23: LFO

Focus: Sample **Edit** Performance

Key Settings:

- LFO parameters: see **Step by Step** below

Operations Manual Page Reference:

- LFO set: 72-73

Step by Step:

- Use the same sample you used in the envelope experiments. If necessary, return the envelope settings to their original default values.
- Enter Create Voice: LFO set.
- Set the "OSC DEPTH" value to "025." Play and listen. You will hear a slow, wide vibrato. Listen to the different LFO waveshapes by changing the "WAVE" parameter.
- Listen to the difference in the sound caused by changing the "LFO SYNC" from "ON" to "OFF." This will be most obvious when you hold down more than one key at a time. You'll notice that the pitch changes of all of the notes don't "line up" with each other. (Try this comparison with the "SQUARE" waveshape if you're having trouble hearing the effect.) Reset the "LFO SYNC" parameter to "ON".
- Listen to the effect produced by changing the "DELAY" value. (Restrike a key each time you reset the value.)
- Reset the "OSC DEPTH" value to "000" and repeat the above with tests with "DCA DEPTH" and then "DCF DEPTH."
- When you are comfortable with how the LFO can affect the voice's pitch, timbre and loudness , try the above different rate and depth settings.

Observations:

- The most common use of the LFO is for vibrato, sine, or triangle control of a sampler's pitch. Many interesting effects can be obtained by controlling the DCA (loudness changes) and the DCF (timbre changes) as well.
- The delay function can be used to fade the LFO effect in gradually. This is more natural sounding than leaving the effect on all of the time.
- The sync function can "randomize" LFO effects. This can be very effective when you want to create ensemble (many instrument) effects, like a string section or vocal choir.

8.9 Velocity Sensitivity

The FZ also lets you control the DCF and/or DCA levels and EG rates with velocity. Velocity is determined by how quickly you push down a key (*Figure 51*). To get the most dynamics from any velocity keyboard, the key should travel its full distance when it is struck. (Although it may seem you have to play harder to play louder, be aware that this is not the case. It's not the force you use, but the speed which the key travels, that determines the velocity value.) Controlling the DCA levels with velocity will allow you to create piano-like dynamics with any sound you sample. (Yes, even with piano samples too!) Controlling the DCF in the same way further enhances the dynamic expression available to you. Almost any sample will sound better with a small (or not so small) amount of velocity control on the DCA and DCF levels.

Control of DCA and DCF EG rates with velocity allows you to shape the attack(s) and decay(s) of your voice with your keyboard playing style. You change a sound from a bowed acoustic bass to a plucked one by simply changing how quickly you depress the keys. Be aware that velocity is affected by how fast you push the keys down, not how hard you strike them.

Velocity Sensitivity is detailed in pages 73 and 74 of the *Operations Manual*. Below is the expanded Menu Overview showing each of the Velocity Sensitivity parameters.

Velocity Sensitivity Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <-ESCAPE	ENTER -> <-ESCAPE	<-ESCAPE
MODIFY PLAY	Source Select **Voice Edit** Bank Edit Effect/MIDI Data Dump OPT Software	Define Voice **Create Voice** Keyboard Set Copy Voice Delete Voice Replace Voice Dump Voice	Truncate DCA Envelope DCF Envelope Loop Set LFO Set **Velocity Sens** Tune/Mem Read	DCA Level DCA Rate DCF Level DCF Rate Resonance

DCA Level

This parameter can be set with either positive ("+001" to "+127") or negative ("-001" to "-127") values. Positive values will make the voice get louder as velocity increases. Maximum velocity will produce the normal loudness of the voice; lesser velocities will produce softer volume levels. Negative values will produce the opposite effect. Minimum velocity will produce the normal loudness of the voice, and higher velocities will produce softer volume levels.

NOTE: This is how you can create velocity cross-fades on the FZ. Layer two voices over the same range of keys. Set one voice's Velocity Sensitivity "DCA LEVEL" to "+127" and the other's to "-127." Now when you play, the balance between the two voices will be controlled by how quickly you strike the keys. (Be sure to read more about velocity splits in Editing Banks.)

DCA Rate

This parameter can be set with either positive ("+001" to "+127") or negative ("-001" to "-127") values. Positive values will make the DCA EG's rates get quicker as velocity increases. Maximum velocity will produce the normal rates of the voice; lesser velocities will produce slower rates. Negative values will produce the opposite effect. Minimum velocity will produce the normal rates of the voice, and higher velocities will produce slower rates.

DCF Level

This parameter can be set with either positive ("+001" to "+127") or negative ("-001" to "-127") values. Positive values will make the voice get brighter as velocity increases. Maximum velocity will produce the normal brightness of the voice; lesser velocities will produce duller timbres. Negative values will produce the opposite effect. Minimum velocity will produce the normal brightness of the voice, and higher velocities will produce duller timbres.

DCF Rate

This parameter can be set with either positive ("+001" to "+127") or negative ("-001" to "-127") values. Positive values will make the DCF EG's rates get quicker as velocity increases. Maximum velocity will produce the normal rates of the voice, lesser velocities will produce slower rates. Negative values will produce the opposite effect. Minimum velocity will produce the normal rates of the voice and higher velocities will produce slower rates.

Resonance

You can also use velocity to control the amount of resonance on the DCF. This parameter is positive only ("000" to "127"). Higher velocities increase the amount of resonance.

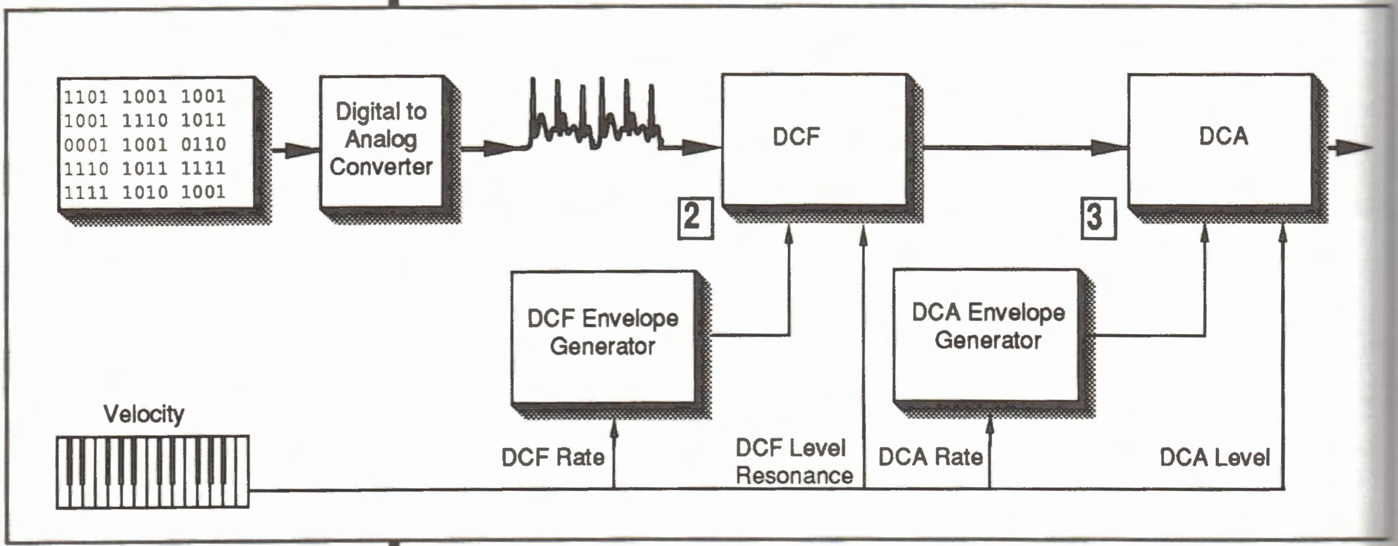

Figure 51: Velocity is routed to both the DCF level and resonance as well as DCA level. At each point, the sensitivity is adjustable. This will give you the most flexibility in performing timbre and loudness dynamics with your FZ.

Experiment #24: Velocity

Focus: Sample **Edit** Performance

Key Settings:

- Velocity parameters: see **Step by Step** below

Operations Manual Page Reference:

- Velocity Sensitivity: 73-74

Step by Step:

- Use the same sample as for the previous experiments
- Enter Create Voice: Velocity Sens.
- Set the "DCA LEVEL" parameter to the maximum value ("+127"). Play the keyboard with a variety of different velocities and note the effect on the sound. The quicker you strike keys, the louder the sample. Lower the sensitivity parameter and play some more. Continue in this manner until you have returned the level to "000." Now every note is the same loudness, no matter how quick or slow you strike the keys.
- Set the "DCA LEVEL"parameter to the minimum value ("-127"). Play the keyboard with a variety of different velocities and note the effect on the sound. The quicker you strike keys, the softer the sample. Raise the sensitivity parameter and play some more. Continue in this manner until you have returned the level to "000." Now every note is the same loudness, no matter how quick or slow you strike the keys.
- Set the "DCA RATE" parameter to the maximum value ("+127"). Play the keyboard with a variety of different velocities and note the effect on the sound. The quicker you strike keys, the faster the EG rates. Lower the sensitivity parameter and play some more. Continue in this manner until you have returned the level to "000." Now every note has the same rates, no matter how quick or slow you strike the keys.
- Set the "DCA RATE" parameter to the minimum value ("-127"). Play the keyboard with a variety of different velocities and note the effect on the sound. The quicker you strike keys, the slower the EG rates. Raise the sensitivity parameter and play some more. Continue in this manner until you have returned the level to "000." Now every note has the same rates, no matter how quick or slow you strike the keys.
- Repeat the above, but this time adjust the "DCF LEVEL" and "DCF RATE" velocity parameters. This time the timbre (brightness) of the the notes will change with different velocities. Make sure you can hear the difference between controlling the DCF with velocity and the DCA with velocity. If the distinction isn't clear, alternate between the two.
- When you are comfortable with what's going on, try changing *both* the DCA and the DCF at the same time. If you can do it, try using reverse sensitivity on just the DCF. (Then do it again with just the DCA reversed.)

Observations:

- Velocity sensitivity allows you to play the sample expressively, in a manner that is very similar to playing a piano (or for that matter, a dynamic synthesizer). The most common usage is to change the DCA level only, so that fast is louder and slow is softer.
- You can make many sounds more expressive by also changing the DCF level just a little along with the DCA. Now softer sounds will also be a little darker, and louder sounds will be a little brighter. Many acoustic instruments change timbre and loudness together in just this manner.
- Changing EG rates with velocity is also a very expressive technique. Particularly with brass and string sounds.
- Reversing the sensitivity (especially on only the DCA level or only the DCF level) can produce some novel effects with a single sample. As we'll see later on, it is the key to creating Velocity Cross-Fade effects.
- Velocity interacts with several operations on the FZ. Be sure to experiment with the velocity related parameters of *Create: Bank* (Min Touch, Max Touch) and *DCA / DCF Envelope* (Rate KF, Level KF) as well!

8.10 Tune/Memory Read

The "TUNE" parameter allows you to adjust the tuning of a voice sharp or flat up to one semitone. The values for this parameter are equal to cents. (One hundred divisions per semitone.) You can detune to voices in layered Areas to create chorus effects. Another way to use tuning is to *multi-switch* between voice copies that have been set with different tuning values. This allows your keyboard dynamics to control a voice's tuning. (Be sure to read about *multi-switching* in the next chapter.)

The "MEM READ" parameter gives you three options for how the data of "RECORDED" voices (samples) will be played by the FZ.

- "FWD" is the normal setting. The voice is heard as sampled, and pitch is controlled from the keyboard.
- "REV" plays the sample data in reverse order, creating a backward sound. The pitch is controlled from the keyboard.
- "CUE" is a unique FZ function. It allows you to play the sample from the *Pitch Bender.* Moving the bender forwards (away from you), plays the sample data as it was recorded. Moving the bender backwards (towards you), plays the sample data in reverse. How quickly you move the bender determines the pitch of the sample. The number of keys you hold down determines the number of voices that will be played by the bender. As noted in the *Operations Manual,* the resulting effect is indeed very similar to "scratch" effects created by manually moving a record back and forth on the turntable.

8.11 Copy, Delete, and Replace

Copy Voice

Copy Voice creates a duplicate version of a source voice's editing parameters in a different location (the *destination* voice). If the destination already contains voice data, you will be prompted with "VOICE DELETE?" Pushing the **YES** key in response to the prompt will erase the voice currently in the destination and put a new copy of the selected voice in its place. Note that the source voice is unaffected by the *Copy* operation.

As we mentioned in the beginning of this chapter, copying a voice does not make a copy of the sampled data. Only the Create Voice parameters are copied. Both the copy and the original use the same sample data. To create a duplicate of the sampling data as well as the voice data, use the *Save* operation.

Delete Voice

Delete Voice is slightly different from the Voice Dump: Erase Voice operation. It gives you a choice. You can delete the entire voice or just the "UNUSED PART" of a sampled voice. The unused part is the part of a sample before the "START" and after the "END" values set with the Truncate operation. This is a very useful feature. Use it to trim dead spots from a sample (to save memory) or to isolate sections to splice with the Mix Write and X-Mix Write functions. (Note that, contrary to the *Operations Manual*, Delete Voice on the current version of the FZ-1 and FZ-10M doesn't remove "skipped" portions within a sample.)

Use this to remove unused memory before you make any copies for velocity or other effects. If you have one or more copies of a voice active on the FZ at the same time, you won't be able to delete the "UNUSED PART." This allows each of the copies to use its own set of truncate start and stop points, etc.

Replace Voice

Replace Voice is essentially the same as Copy Voice, with one significant difference. The source voice is deleted by the operation. Use Replace Voice to move a voice from one location to another. Note that you can't replace a voice with itself (as you might want to in order to save editing changes). Use the Voice Dump: Save Voice operation to store updated versions of a voice.

9. Bank Editing

In *Creating Samples,* we showed you how to use the FZ's sampling features to digitalize sounds and make samples. In Voice Editing, we demonstrated the operations and techniques used to turn those samples into expressive musical sampling voices. Now it's time to explore ways of combining sampling voices to make a complete sampling instrument by editing the FZ's Create Bank parameters.

The emphasis here is on performance. Where on the keyboard do you want to locate the voice? Do you want to combine the voice with others, to make a composite sound? Do you want the entire range of the keyboard to be made up of one voice, or do you want to split it into several different voices? Do you want to hear different voices, depending on how or where you play the keys? If you are controlling your FZ with a MIDI controller or sequencer, you may want to assign different voices (or combinations of voices) to different MIDI channels.

As you can see, most of these questions have to do with where (or under what circumstances) FZ voices will appear on your keyboard. For this reason, they are generally referred to as *mapping* operations. In this context, mapping simply means to assign FZ voices to respond to particular sections of the keyboard and/or particular performance controllers or MIDI Channels.

It is important to note that these operations are used with voices that have already been created with Create Voice parameters. At this point, you no longer have to work with the sampling operations or voice editing operations. Even if you plan on never actually sampling any sounds yourself and intend to use only factory supplied or commercial sound libraries, you must become familiar with the FZ's mapping operations. At the performance level they are the most useful (not to mention fun) operations to master, even if you only have a modest library of sampling voices. You can use mapping operations to rearrange them into many powerful performance setups.

The mapping features handled by the FZ's Bank Edit: Create Voice operations fall into three basic categories.

- *Key Mapping* operations are used to assign FZ voices to individual keys or ranges of keys called Areas on the FZ.
- *Velocity Mapping* operations assign FZ voices and Areas to respond to keyboard dynamics. The FZ's Velocity Mapping operations are particularly powerful since multiple copies of a voice share the same sample data. We told you in the last chapter that each of the more than one hundred Create Voice parameters can be velocity sensitive. In this chapter, we'll show you how.
- *MIDI Mapping* assigns FZ Areas to MIDI channels. We'll show you how to use your FZ as a full function, multi timbral, polyphonic MIDI sound module.

The Bank Edit sub-mode is detailed in pages 82 through 90 of the *Operations Manual.* Below is the Menu Overview for the complete sub-mode.

Bank Edit Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER -->**	**ENTER -->** **<--ESCAPE**	**ENTER -->** **<--ESCAPE**	**<--ESCAPE**
MODIFY PLAY	Source Select Voice Edit **Bank Edit** Effect/MIDI Data Dump OPT Software	<---------------->	Define Bank	Bank No. Bank Name
		<---------------->	Create Bank	Voice No. Original Key Highest Key Lowest Key Max Touch Min Touch Area Level MIDI Channel Output
		<---------------->	Copy Bank	Source No. Destination No. Execute: Yes/No
		<---------------->	Delete Bank	Bank Only Bank & Voice
		<---------------->	Delete Area	Area No. Execute:Yes/No
		<---------------->	Replace Bank	Source No. Destination No. Execute: Yes/No
		Dump Bank	Load Bank	Bank Name Execute: Yes/No
			Save Bank	Bank Name Execute: Yes/No
			Verify Bank	Bank Name Execute: Yes/No
			Erase Bank	Bank Name Execute: Yes/No

Define Bank

This operation is very similar to the Define Voice operation we've seen before. It is used to select the Bank to be edited. The FZ can hold up to eight Banks in memory at a time. Each Bank is numbered (1-8). When a new Bank is created, you should give it a name (like "VIB / W BASS"). This will make it much easier to remember what the keyboard setup consists of. (All of the Banks supplied with Casio's sampling library and other commercial libraries for the FZs will come already labelled with names.)

9.1 Create Bank

Create Bank is detailed in pages 84 and 85 of the *Operations Manual.*

For maximum performance flexibility, your FZ organizes Areas into groupings called Banks. Each Bank can hold up to sixty-four Areas. The FZ can keep up to eight complete Banks "on-line" at a time. This means you can access up to eight complete keyboard setups made of up to sixty-four different sounds, keyboard maps, velocity maps, and MIDI maps at the push of a button–without having to reload any disks.

Understanding how to set up Area parameters is the key to such sampling techniques as *key splits, key layers, velocity cross-fades, cross-fade key layers, multi-sampling, multi-switching,* and more. All in all, there are nine critical parameters for each Area. Since each Bank holds up to sixty-four Areas, that means 576 critical parameters per Bank. Eight Banks hold 4608 Area parameters! To help you keep organized, we've included a blank chart that you can copy and fill in with the vital statistics of your Banks (*Figure 52*). We'll be using a simplified version of the chart in several of the examples below.

Let's take a closer look at the parameters that make up an Area

- **Voice Number**: Assigns any sound created by Sampling, Wave Synthesis, Mix Write, X-Mix Write, or Reverse Write functions to this Area.
- **Key Range** (Original, Highest, and Lowest): Assigns a range of keys to this Area. (The assigned voice will be heard only when these keys are played and the key velocity is within the assigned velocity range.)
- **Velocity Range** (Max Touch, Min Touch): Assigns a range of velocities to this Area. (The assigned voice will be heard only when the key velocity is within the assigned velocity range and the keys are in the assigned key range.)
- **Area Level**: An independent loudness setting for the FZ voice assigned to this Area. The maximum value, 127, equals the voice's normal loudness.
- **MIDI Channel**: Assigns a MIDI channel number (1-16) to control the voice assigned to this Area.
- **Output Channel:** Assigns the output of this voice to one or more of eight monophonic outputs in the FZ.

Defining New Areas

As we mentioned earlier, each Bank can contain up to sixty-four different Areas. The FZ will automatically give you new Areas as you need them. When you first define a new Bank, the display will show you "AREA No.1." All of the values will show asterisks ("*"), indicating that no voice has been assigned to this Area yet. Once you enter a "VOICE No." with the **VALUE** keys or slider, all of the parameters will be automatically filled in as follows:

- "ORIGINAL," "HIGHEST," and "LOWEST" will be set to the current settings for the selected voice with the Voice Edit: Keyboard Set operation for the selected voice.
- "MAX TOUCH" will be set to "127."
- "MIN TOUCH" will be set to "001."
- "AREA LEVEL" will be set to "127."
- "MIDI CH" will be set to "01."
- "OUTPUT" will be set to all eight outputs.

Bank Chart For FZ-1 and FZ-10M

			Key Range			Velocity Range				
Area	Voice Name	No.	Original	Highest	Lowest	Max Touch	Min Touch	Level	MIDI Ch.	Output
1										
2										
3										
4										
5										
6										
7										
8										
9										
10										
11										
12										
13										
14										
15										
16										
17										
18										
19										
20										
21										
22										
23										
24										
25										
26										
27										
28										
29										
30										
31										
32										

Continue Area Numbers Through 64.
For a complete reproducable chart, see page 143.

Figure 52 : FZ Bank Chart.

Of course, you can change these settings any way you wish. Once you've adjusted the settings to suit your needs, you can create another Area. Pushing the **RIGHT** button will advance the display to "AREA No. 2." You can assign a voice to this Area, adjust the parameters to your liking, and continue to create new Areas by simply pushing the **RIGHT** button. Use the **LEFT** button to go back to Areas that you've previously defined.

If you are editing a previously defined Bank, the display will show you the current settings of "AREA No. 1" when you first enter Create Bank. If you want to add a new Area to this Bank, use the **RIGHT** button to advance the display to an undefined Area. If you want to erase an Area, you can use the *Delete Area* operation.

Working With Areas

Many of the Create Bank parameters interact with where or how you play the keys of your FZ-1 (or MIDI controller if you have an FZ -10 M). Be sure that you are playing within its key range when you are editing an Area. The values displayed for "HIGHEST" and "LOWEST" will show you where to play. For reference, be aware that the lowest "C" on the keyboard corresponds to "C2" in the display, and the highest "C" corresponds to "C7" in the display. If you're working with the FZ-10M, "middle C" corresponds to "C4" on the FZ (on most MIDI controllers, "middle C" is C3).

9.2 Key Mapping with Create Bank

The FZ-1 has a sixty-one-note keyboard. The rack mount FZ-10M will respond to Note On/Off messages from a remote MIDI keyboard (or other controller). Playing a single voice across an entire keyboard is not particularly musical. As we have seen, different pitches are produced by speeding up or slowing down the rate at which the sampled voice is played back by the sampler. The original sound undergoes radical changes as its pitch is shifted beyond a small amount. The end result is that the useful keyboard range for most single samples is only a few semitones. (Synthesized voices don't have this problem.) That leaves you with an awful lot of empty keys and very few pitches with which to play melodies. Casio's system of Areas lets you assign several sampling voices across the keyboard so that you can take full advantage of all its available notes.

There are three basic types of key mapping. *Splitting* allows you to assign different sampling voices to different keys to create an instrument that will produce several different sounds, depending on where you play. *Layering* is used to assign more than one FZ voice to the same key (or keys), letting you play two or more voices with each key you depress. Multi-sampling and multi-switching are actually specialized uses of splitting, layering, and velocity mapping (more on velocity mapping in a while). Rather than dividing the keyboard between different voices made from samples of different instruments, the keyboard is divided between voices sampled from different pitches of the same instrument. Likewise, rather than layering different sampled voices, different samples (or copies of samples) of the same instrument with different dynamics are layered. The result is a realistic simulation of the sampled instrument spread over the entire keyboard and dynamic range. Casio's Area system makes it possible for you to set up all four types of key mapping, and several variations as well, with your FZ.

9.3 Keyboard Splits with Create Bank

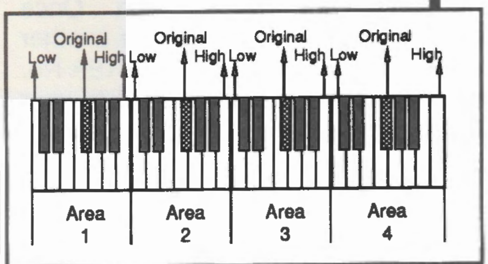

Figure 53: In this illustration, the keyboard is split into four one-octave Areas. Each Area can play a different sample.

Since the FZ lets you define up to sixty-four Areas, you can assign a different voice to each key and still have room for three additional voices. Most times, however, you'll want to assign a voice to an Area made up of several keys so that you can play the voice melodically. When you assign a key to a split, there are three parameters you will have to set. The Original Key determines the note that will sound the original sampled pitch. The *upper key* is the Highest Key the sample will be played from. The Lowest Key is the lowest note that will play the sample (*Figure 53*).

When defining a key range for an Area, the Original Key doesn't have to be inside the highest and Lowest Keys. Also, it doesn't have to be set to the true pitch of the sampled voice.

Suppose you want to put two Areas next to each other on the keyboard–let's say, for instance, Area 1 right below Area 2. Start by assigning a key range to Area 1. Then assign the "LOWEST" key of Area 2 to the value 1 semitone above the "HIGHEST" key of Area 1. Here's how the key split between the bass and vibes is set up in Bank 5 of your FL-B library disk. Note that the bass sound's Highest Key is C4 and the vibe's Lowest Key is C#4.

Area	Voice	Name	Original	Highest	Lowest	Level
3	16	W Bass D3	D04	**C04**	#F03	127
4	9	Vib A41	A04	#G04	**#C04**	127

9.4 Key Layering with Create Bank

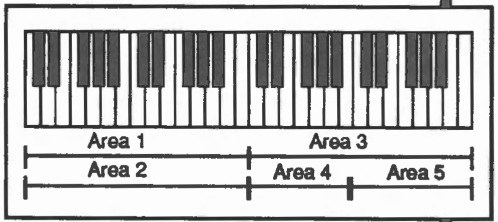

Figure 54: Layering allows two or more Areas to be played by the same keys.

Key layering allows you to create splits that overlap each other. With the FZ, you can overlap more than two Areas at once (stacking three or more sounds on the same range of keys, or producing split and layers – *Figure 54*).

You can also set up a *Cross-Fade Key Layer.* Here's how:

- Overlap the upper part of one Area with the lower part of another. Here's one possible layer:

Area	Voice	Name	Original	Highest	Lowest	Level
1	1	Lower Area	#F03	**C04**	C03	127
2	2	Higher Area	C04	#F04	**#F03**	127

- Set the *DCA Level KF* parameter of the voice assigned to the lower Area to a negative value (see page *). It will fade out as you play into the upper part of its key range.
- Set the *DCA Level KF* parameter of the voice assigned to the upper Area to a positive value. It will fade out as you play into the lower part of its key range.
- Since the two Areas overlap as you play from one to the other, you will hear a gradual transition from sound to sound when you play between F#3 and C4.

Experiment #25 : Key Split

Focus: Sample   Edit   **Performance**

Key Settings:

- Bank Edit: Create Bank

Operations Manual Page Reference:

- Bank Edit: Create Bank Pg. 84-86

Step by Step:

- Refer to page 84
- Sample and save the phrase SPLIT ONE as Voice No. 01.
- Sample and save the phrase SPLIT TWO as Voice No. 02.
- Sample and save the phase SPLIT THREE as Voice No. 03.
- Enter Bank Edit: Create Bank
- Your display should read "Area No. 01" "Voice No. = **". Set "Voice No." to "01" (the display will immediately set up the default key assign of C05-C07-C02.)
- Using either the keyboard or value slider set "Original = C#04, Highest = D04 and Lowest = C04"
- Using the **RIGHT** button select "Area 02".
- Your display should read "Area No. 02" "Voice No. = **". Set "Voice No." to "02" (the display will immediately set up the default key assign of C05-C07-C02.)
- Using either the keyboard or value slider set "Original = E04, Highest = F04 and Lowest = D#04"
- Using the **RIGHT** button select "Area 03".
- Your display should read "Area No. 03" "Voice No. = **". Set "Voice No." to "03" (the display will immediately set up the default key assign of C05-C07-C02.)
- Using either the keyboard or value slider set "Original = G04, Highest = G#04 and Lowest = F#04"
- Push the **PLAY** button and compare the three samples by playing each note from middle C to middle G#.

Observations:

- Play a chromatic scale up the middle octave starting at middle C. You should notice that the sampled phrase changes every three chromatic steps. (You will here each phrase at its original pitch and a pitch shifted up and down one semitone.)
- Playing outside the range of the assigned keys should not produce any sound. (Remember only voices assigned to specific areas with designated key ranges will sound in a bank set-up.)
- Try the same basic experiment. This time substitute different sounds for the sampled phrases. (For example, try assigning several drum sounds across an octave.) Key splitting allows you to play many different instrument sounds from a single keyboard. To get the largest possible group of instruments on the FZ at one time, only use the amount of keys needed for each sound. (See *9.5 Key Splits For Multi-Samples.)*

Experiment #26 : Key Layering

Focus: Sample Edit **Performance**

Key Settings:

- Bank Edit: Create Bank

Operations Manual Page Reference:

- Bank Edit: Create Bank Pg. 84-86

Step by Step:

- Refer to page 84
- Sample and save the phrase LAYER ONE as Voice No. 01.
- Sample and save the phrase LAYER TWO as Voice No. 02.
- Enter Bank Edit: Create Bank
- Your display should read "Area No. 01" "Voice No. = **". Set "Voice No." to "01" (the display will immediately set up the default key assign of C05-C07-C02.)
- Using the cursor **DOWN** button until the value in "Area Level" is flashing.
- Using the value slider set the "Area Level" to "060"
- Using the cursor **RIGHT** button select "Area 02".
- Your display should read "Area No. 02" "Voice No. = **". Set "Voice No." to "02" (the display will immediately set up the default key assign of C05-C07-C02.)
- Using the cursor **DOWN** button until the value in "Area Level" is flashing.
- Leave the "Area Level" at it's default value of "127".
- Push the **PLAY** button and listen to the balance of the two samples as you play the keyboard.

Observations:

- Playing any where on the keyboard, you will hear both samples at the same time. The only difference that should be noted is the level difference that was assigned during the experiment.

- You can obtain great effects by layering different musical sounds. Some of the combinations that work very well are: strings and acoustic guitar, male and female voices, and strings and piano.

9.5 Key Splits for Multi-Samples

Multi-sampling is a two-stage process. We'll look into the key mapping aspect of it here, but be aware that how the sounds are originally sampled and edited is crucial to the creation of a realistic multi-sample. If you plan on creating multi-samples from your own sampled sounds, be sure to read our book ***The Sampling Book***. We've packed it full of tips, techniques and secrets of how to record instruments specifically for multi-sampling.

As we mentioned above, multi-sampling is really just a specialized use of key splitting operations. The purpose is to reproduce the sound of one instrument (or groups of instruments) across a large pitch range. Many samplers come with at least one example of such a set up. (The most common are grand pianos and acoustic guitars.) A lot of thought and planning goes into creating a good multi-sample sound. Before the key mapping can be set, multiple samples of the source must be made and edited. Right now, we'd like to discuss how to set up key ranges for Areas when you use the FZ as a multi-sample instrument.

Picking Pitch Shift Intervals

Take for example, the FZ-1's sixty-one-note five-octave keyboard. Why not just make sixty-one separate splits for sixty-one separate sampled voices? Well, if you remember, we told you that many of the decisions you make with a sampler involve trade-offs. This is another example of one. If each sample is just one-half second, then you'll need a little more than thirty seconds of sampling time for just one multi sampled instrument. (The FZ will also have to hold sixty-one samples in memory at the same time as well!) So, even though the FZ can handle that many splits, you may still want to use less to create a multi-sample instrument. That way, you'll have room for additional sounds.

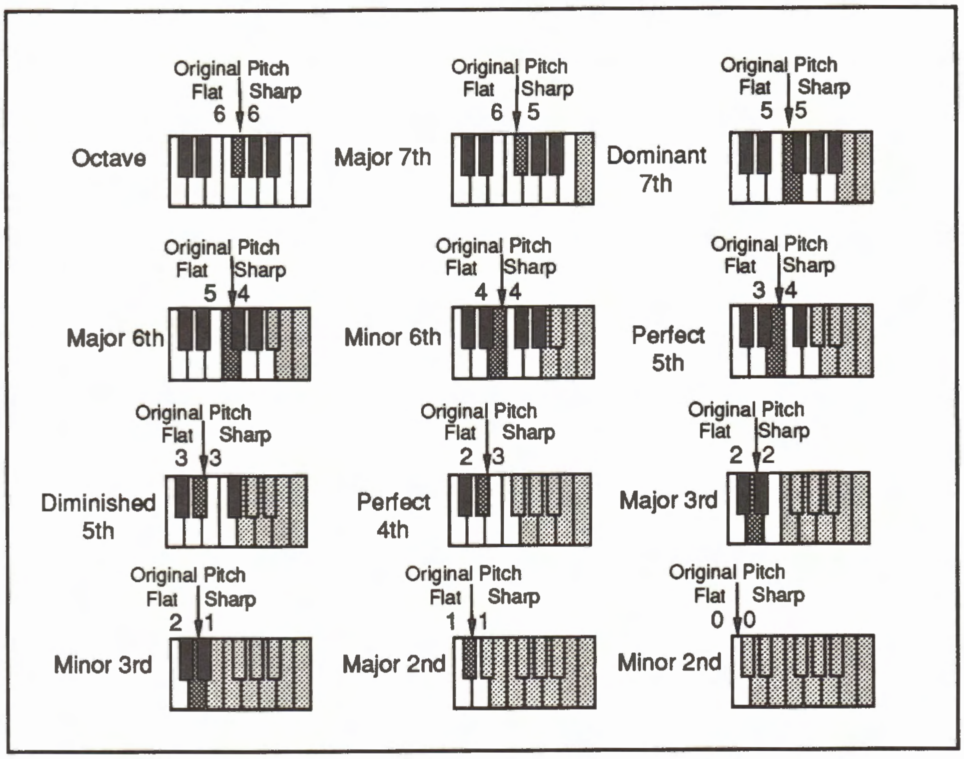

Figure 55: Useful Pitch Shift Intervals

Since most sounds can be shifted a few semitones sharp or flat before they start to sound unnatural, you'll find that you can use considerably less than sixty-one samples to create a five-octave instrument. A key question (pun intended) is how far apart in pitch should the samples be? For all practical purposes, no musical source will sound natural if you shift it more than a total of one octave. Most sounds will require considerably less pitch shifting to retain a natural quality.

Let's use an octave as the maximum limit for total pitch shift (plus or minus six semitones). If we split a keyboard by the different intervals within an octave, we can see how much pitch shift occurs for all the useful intervals. *Figure 55* shows the total pitch shift and the maximum sharp vs. maximum flat pitch shift for each interval. Note that some intervals shift equally in both directions, while others have a one-semitone difference between sharp and flat shift. (When the amount of shift is unequal, we put the "extra" semitone on the flat side of the Original Key. It could also have been shown on the sharp side. There really is no difference.)

When setting up a multi-sample instrument, intervals that create unequal shifts use your sampled voices more effectively than those that create equal shifts. (A more natural sound is obtained, since less pitch shifting accumulates over a given number of samples for the same approximate pitch range.) For this reason, the most efficient intervals to use for multi-sampling key splits are octave, flat seventh, minor sixth, flat fifth, major third. It can be somewhat awkward however, to set up a split keyboard in dominant 7ths, etc. The preferred method is to split the keyboard into intervals that divide an octave evenly. These intervals are easier to work with for most of us, since we are familiar with them from learning basic scale and chord patterns. Octaves are evenly divided by octave (of course!), flat five, major third, minor third, whole step and half step intervals.

Perhaps the most sensible way to assign splits to a multi-sample instrument is to divide the keyboard into intervals based on minor thirds, the diminished chord. This simple chord contains two of the best intervals defined above and also cleanly divides the keyboard into one, two, or four splits per octave. This reduces the number of splits needed to cover five octaves from sixty-one down to a number between five and twenty. The maximum number of semitones a sample will be shifted by is six (one split per octave), three (two splits per octave), or one (four splits per octave).

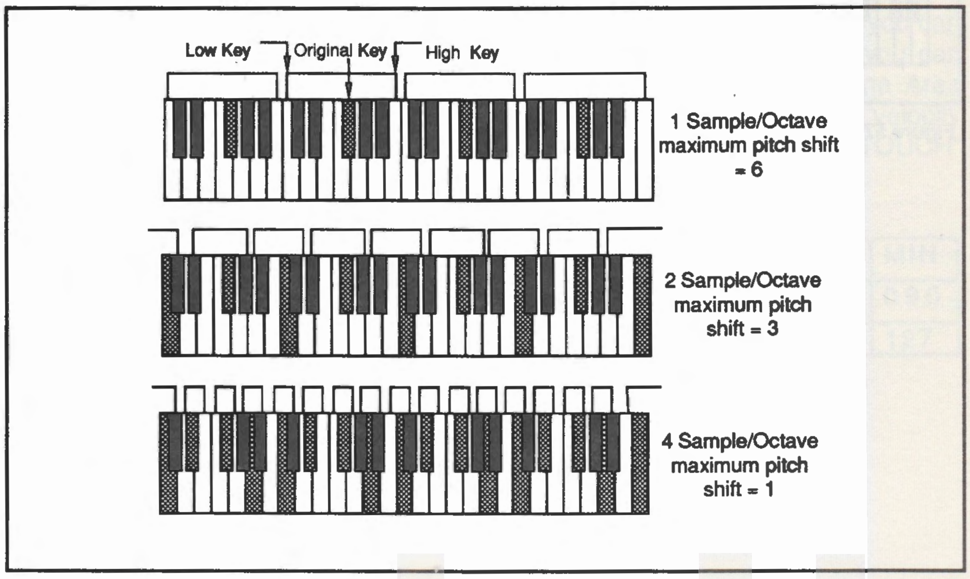

Figure 56: Interval Splits Based on a Diminished Scale

Figure 56 illustrates various ways of splitting a five-octave keyboard using one, two, or four splits to an octave. You'll find that these divisions work very well with samples of almost any instrument. (We show the Original Keys based on a "C" tuning. This will change, depending on the range and tuning of the instrument you sample.) Since they are all built on the interval of a minor third, you will find it quite easy to pick the notes to sample during your sampling sessions as well. All you have to do is sample the same four notes in each octave. Since there are only three unique spellings of a diminished chord, you'll find them easy and convenient to use (regardless of the playing range of the instrument being sampled). Once you've sampled the instrument in minor thirds, you can pick and choose between one, two, or four splits per octave as needed, without worrying about having samples of the right notes. Obviously, if you can afford the memory, the most desirable division is four to the octave. Each sample is only shifted by a single semitone sharp or flat.

You may find that you won't always need the same number of splits per octave. Depending on the instrument and where you are in its range, you may need three, two, or just one split for the right effect. If you've sampled the same notes in each octave, you can mix the above splits' sizes quite easily, without a lot of recalculation or resampling. *Figure 57* illustrates several variations.

Another approach is to space your splits at equal intervals: major thirds, perfect fourths, minor sixths, etc. This can work equally well, and since the splits occur at different places in each octave, it may help disguise the multi-sampling setup. It is not as simple to mix the splits' sizes, since octaves are not divided equally. You will also have to be careful to make sure you get the right notes when you sample the sound. Also be aware, as we mentioned above, that some intervals yield less pitch shifting than others. A good alternative is to set your splits in intervals based on the major thirds, i.e., the augmented triad. This tuning cleanly divides an octave into three equal divisions. The maximum pitch shift for each split is only two semitones, and there are only four unique spellings of the augmented triad. The same three notes are repeated in each octave. As with the diminished tuning, it is easy and convenient to use. *Figure 58* shows intervals that will yield splits with five, four, and two semitones of maximum pitch shift.

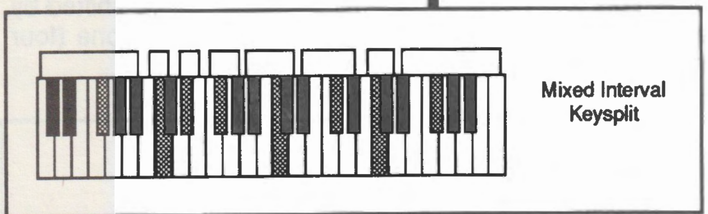

Figure 57 : Mixed Interval Splits

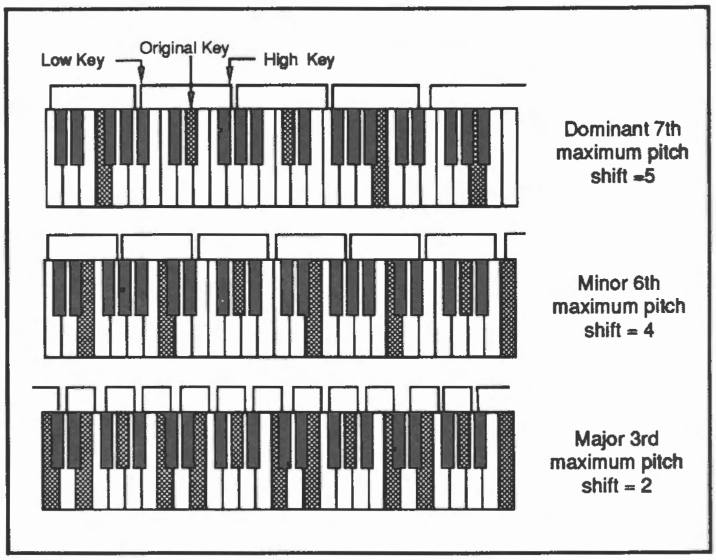

Figure 58: Equal Interval Splits

9.6 Velocity Mapping with Create Bank

The "MAX TOUCH" and "MIN TOUCH" parameters, in conjunction with the Create Voice: Velocity Sensitivity operation, allow you to create velocity mapping effects with your FZ. You can control these effects in real time with your performance on the FZ-1's built-in velocity sensitive keyboard. If you own an FZ-10M, then you must control it with a velocity sensitive MIDI controller in order to achieve velocity related effects. There are two general types of velocity mapping. Velocity Switching allows you to select between different Areas with your keyboard performance. Velocity Cross-Fade allows you to control the mixture of different Areas with your keyboard performance. Unlike many samplers that can only switch or cross-fade between two sampling voices, your FZ switches or fades between as many as sixty-four. This ability is the secret behind the multi-switching technique described below.

9.7 Velocity Switching with Create Bank

Velocity Switching works in conjunction with a layered keyboard. Two (or more) Areas are assigned to the same keyboard range. The sound you hear when you play a key will be determined by how quickly you depress the note. In other words, you can layer any two sounds, let's say a trumpet and a dog bark, and you'll hear one when you play loud, and the another when you play softly (*Figure 59*).

Here's how to set up a velocity switch between two sounds on the FZ:

- Assign two different sounding voices to two different Areas. (We'll use Area 1 and Area 2 for this example.) Assign both Areas to the same key range.
- For now, set the Create Voice: Velocity Sensitivity parameter for each voice to "000" (no velocity sensitivity).

Area	Voice	Name	Highest	Lowest	MAX	MIN
1	1	Slow	C04	C03	001	**064**
2	2	Fast	C04	C03	**065**	127

When you play a note in the key range, the voice you hear will be controlled by your keyboard velocity. Slower velocities will sound the voice assigned to Area 1. Quicker velocities will sound the voice assigned to Area 2.

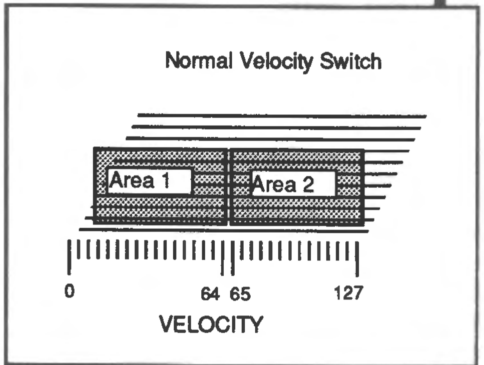

Figure 59: You can switch between two Areas with velocity by using the FZ's Min Touch and Max Touch parameters.

Since the FZ allows an Area to respond to a range of velocities, you can do more than just a simple two-sound switch. For example, you can overlap the ranges by setting the "MAX TOUCH" value of one Area greater than the "MIN TOUCH" value of another Area. Use the velocity switch example described above, but this time set the "MAX TOUCH" value for Area 1 to "090" instead of "064."

Area	Voice	Name	Highest	Lowest	MAX	MIN
1	1	Slow	C04	C03	001	**090**
2	2	Fast	C04	C03	**065**	127

Now when you play with your quickest touch, you'll hear only the voice assigned to Area 2, and your slowest touch will still play only the voice assigned to Area 1. When you play with a medium touch, however, you will hear both voices at the same time. This occurs because velocities between sixty-five and ninety will play both Areas.

To switch between three voices, simply assign each voice's Area to the same key range as before. Now set the velocity range for each Area to a different section of the total velocity range .

Area	Voice	Name	Highest	Lowest	MAX	MIN
1	1	Slow	C04	C03	001	050
2	2	Medium	C04	C03	051	080
3	3	Fast	C04	C03	081	127

You can overlap the velocity ranges too. Instead of switching from sound to sound, you'll hear different combinations of voices based on your keyboard dynamics. Here are two possibilities.

Area	Voice	Name	Highest	Lowest	MAX	MIN
1	1	Slow	C04	C03	001	066
2	2	Medium	C04	C03	051	080
3	3	Fast	C04	C03	065	127

Area	Voice	Name	Highest	Lowest	MAX	MIN
1	1	Slow	C04	C03	001	065
2	2	Always!	C04	C03	001	127
3	3	Fast	C04	C03	081	127

Figure 60 shows some examples of different velocity switching assignments.

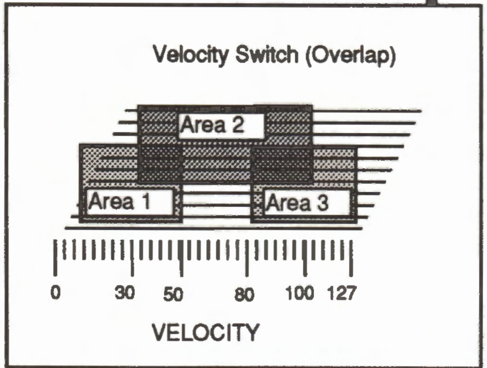

Figure 60: You can switch between as many as sixty-four Areas with velocity on your FZ. This example shows how to overlap three Areas.

Don't forget that each voice also has a set of velocity parameters that can be set with the Create Voice: Velocity Sensitivity operation. Be sure to try using these in conjunction with the velocity mapping techniques described here.

9.8 Multi-Switching: Expanding the FZ's Velocity Capabilities

Here's how to make all 100+ Create Voice parameters velocity sensitive on the FZ. We call it multi-switching. It's simple really. In the examples above, we showed you how to switch between two or more Areas with different voices. The secret to multi-switching is to switch between Areas with copies of the same voice. Since voice copies all share the original sample data, you can "stack" as many copies as you want (up to sixty-three) in the same Area. Doing this won't use any more sampling memory than you would for just the original sample. This means you can add multi-switching effects to many of your existing FZ Banks. For instance, consider a Bank that uses eight sampled voices assigned to eight Areas split across the keyboard. You can add seven copies of each voice to the Bank. This, in turn, gives you a total of eight sets of velocity switchable parameters for each voice.

Experiment #27 : Velocity Switching

Focus: Sample Edit **Performance**

Key Settings:

- Bank Edit: Create Bank

Operations Manual Page Reference:

- Bank Edit: Create Bank Pg. 84-86

Step by Step:

- Refer to page 84
- Sample and save the phrase QUICK HIT as Voice No. 01.
- Sample and save the phrase SLOW HIT as Voice No. 02.
- Enter Bank Edit: Create Bank
- Your display should read "Area No. 01" "Voice No. = **". Set "Voice No." to "01" (the display will immediately set up the default key assign of C05-C07-C02.)
- Using the cursor **DOWN** button select "Max Touch". Set it's value to 127.
- Next select "Min Touch". Set it to "65".
- Using the cursor **RIGHT** button select "Area 02".
- Your display should read "Area No. 02" "Voice No. = **". Set "Voice No." to "02" (the display will immediately set up the default key assign of C05-C07-C02.)
- Using the cursor **DOWN** button select "Max Touch". Set it's value to 64.
- Next select "Min Touch". Set it to "00".
- Push the **PLAY** button and compare the three samples by playing each note from middle C to middle G#

Observations:

- One of the first points you will notice in playing this set-up, is that when you strike a key quickly you will hear only, QUICK HIT and when you attack the key slowly you will only hear SLOW HIT.
- Experiment with this set-up until you can control which of the two samples will be heard in your playing style.
- Velocity switching can be a very powerful means of means of musical expression. Use it to switch between two different dynamics of the same sound.

Of course, each Area in the multi-switch plays the same sample, but how it is played is completely variable, since you have one hundred independent voice parameters for each Area. If you've set each Area to be played by a different velocity range (with "MIN TOUCH" and "MAX TOUCH"), then those one hundred parameters become velocity sensitive. We can't emphasize enough how incredibly powerful this ability is. Here's a list of just some of the performance effects this makes possible:

- Truncate "START" and "END" points can be shifted with velocity.
- The number of loops in a sample can change with velocity
- The locations of loops in a sample can change with velocity.
- *Trace* and *Skip* loops in a sample can change with velocity.
- *Mem Read* (forwards or reverse) in a sample can change with velocity
- *Loop Time* and Cross Time for each loop in a sample can change with velocity
- LFO *Wave, Rate, Delay,* and/or *Sync* in a sample or Wave Synthesis voice can change with velocity.
- Tuning in a sample or Wave Synthesis voice can change with velocity.
- The number of steps in an envelope in a sample or Wave Synthesis voice can change with velocity.
- The location of envelope sustain and/or end steps in a sample or wave synthesis voice can change with velocity.
- Envelope Rates and Levels can change in more complex ways than can be produced with Velocity Sensitivity settings.

Here's a quick example to introduce you to the technique:

- Enter Voice Edit and select any sample voice. (Pick one that has a key range of at least a fifth, so you'll have some room to play.)
- Enter Copy Voice and make seven copies (see page 77 in the *Operations Manual* for details).
- Set a different set of loop points in the original and each copy using the Loop Set operation.
- Enter Bank Edit: Create Bank and assign the voice and seven copies to Areas 1 through 8. They will automatically come up with the same key range. Adjust the "MIN TOUCH, MAX TOUCH" parameters as follows:

Area	Voice	Name	Highest	Lowest	MAX	MIN
1	10	Original	C04	C03	001	015
2	12	Copy 1	C04	C03	016	030
3	13	Copy 2	C04	C03	031	045
4	14	Copy 3	C04	C03	046	060
5	15	Copy 4	C04	C03	061	075
6	16	Copy 5	C04	C03	076	090
7	17	Copy 6	C04	C03	091	105
8	18	Copy 7	C04	C03	105	127

- Play the keyboard with various velocities. You'll be able to switch between eight different loop setups with your keyboard style.
- Return to Create Voice and remove the loops from each voice. Change another set of parameters in each voice. (Use the list we gave above as a guide.)

- Push **CALL/SET MENU** and then **PLAY.** (If you need to, select the Bank you've been working on.) Play and listen to the new changes.
- Push **CALL/SET MENU** to return to Create Voice. Reset the last set of parameters you made and alter some different ones.
- Push **CALL/SET MENU** and then **PLAY**. Play and listen to the new changes. You can repeat this process as long as you like. There are hundreds of parameters and thousands of combinations.

Depending on the number of different voices in Bank, your playing style, and the effects you want to produce, you'll want to try multi-switching with more or fewer copies. We've gotten some great effects with just two or three switches.

9.9 Velocity Cross-Fade with Create Bank

In a velocity cross-fade, the velocity determines the balance between two (or more) FZ voices. Here's how to set this up:

- Assign two different sounding voices to two different Areas. (We'll use Area 1 and Area 2 for this example.) Assign both Areas to the same key range and the maximum velocity range.

Area	Voice	Name	Highest	Lowest	MAX	MIN
1	1	Normal	C04	C03	001	127
2	2	Reversed	C04	C03	001	127

- Set the Create Voice: Velocity Sensitivity parameter for the voice assigned to Area 1 to "+127" (normal velocity sensitivity).
- Set the Create Voice: Velocity Sensitivity parameter for the voice assigned to Area 2 to "-127" (reversed velocity sensitivity).

The balance of the two voices will be controlled by your keyboard playing. At the quickest velocities, the voice assigned to Area 1 will be much louder than the voice assigned to Area 2. The reverse will be true at the slowest velocities. Both voices will have the same approximate loudness when you play with medium velocities.

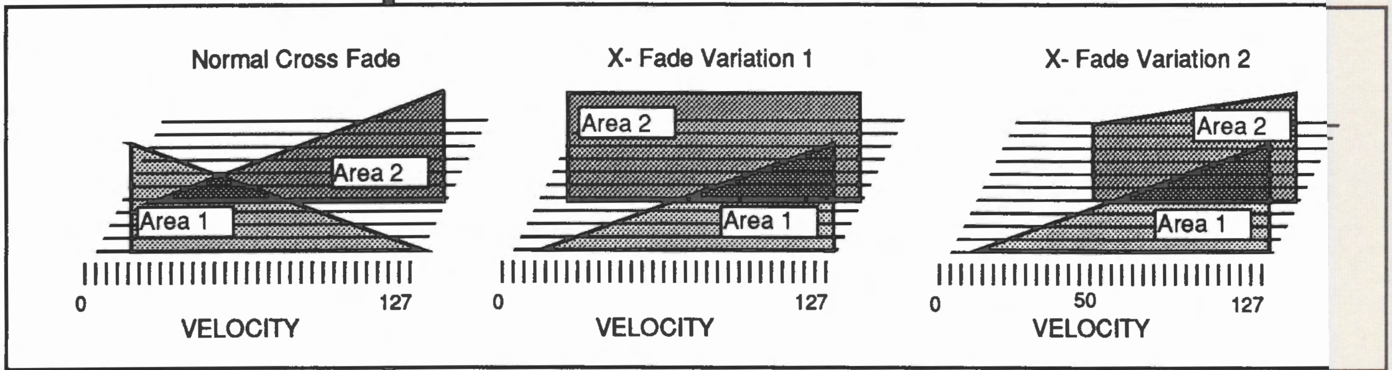

Figure 61: Here are three possible versions of a velocity cross-fade. In a normal cross-fade, keyboard velocity makes Area 1 get softer as Voice 2 gets louder. The first variation shows Area 1 changing loudness with velocity, while Area 2 does not. The second variation shows a combination cross-fade and switch. When you play very softly only Area 1 is heard. As you play louder, Area 2 will join Area 1 at the switch point. From there on, as you play louder, both 1 and 2 will increase in loudness.

There are some interesting variations on the standard velocity cross-fade. Layer a velocity sensitive voice with one that isn't to bring one voice in and out of the mix with your dynamics. Be sure to try it with normal and reversed velocity sensitivity. Try combining velocity switching with velocity cross-fading. For example, set one voice's velocity range so it switches on only for higher velocities, and the other for normal velocity sensitivity. Now you won't hear any cross-fade effect until you play louder than the switch point (*Figure 61*).

Experiment #28 : Velocity X-Fade

Focus: Sample Edit **Performance**

Key Settings:

- Bank Edit: Create Bank
- Velocity Sensitivity

Operations Manual Page Reference:

- Bank Edit: Create Bank Pg. 84-86

Step by Step:

- Refer to page 84 and 73
- Sample and save the phrase NORMAL VELOCITY as Voice No. 01.
- Sample and save the phrase REVERSED VELOCITY as Voice No. 02.
- Enter Voice Edit: Create Voice
- Enter "Velocity Sensitivity" function and set the "DCA Level" to "-127"
- Enter Bank Edit: Create Bank
- Your display should read "Area No. 01" "Voice No. = **". Set "Voice No." to "01" (the display will immediately set up the defaults. Change only the "Min Touch" to "025".)
- Using the cursor **RIGHT** select "Area 02".
- Your display should read "Area No. 02" "Voice No. = **". Set "Voice No." to "02". The display will immediately set up the defaults. Leave all of them set as is.
- Push the **PLAY** button and compare the two samples by playing each note from middle C to middle G#. (You may have to readjust the Area Levels to get a true equal balance. If so refer to experiment 28.)

Observations:

- One of the first points you will notice in playing this set-up, is that when you strike a key quickly you will hear only, NORMAL VELOCITY and when you attack the key slowly you will only hear REVERSE VELOCITY. At one point can you get both samples to sound at the same time on the same key.
- Experiment with this set-up until you can control which of the two samples will be heard by your playing style.
- You'll find that you can get some interesting musical results by setting up velocity cross-fades between contrasting sounds. Try cross-fading piano and strings, or forward and backwards cymbal crashes.

9.10 Area Level

Once you've assigned voices to Areas and you've assigned the Areas across the keyboard, you should balance their relative loudness. This will allow you to play across the keyboard, from Area to Area, without any voices sounding too loud or soft. This is very important, especially if you're recreating a single instrument from a number of different samples. Here's a method for balancing all of the voices in a given Bank:

- Temporarily defeat each voice's velocity sensitivity by setting the Create Voice: Velocity Sensitivity parameter for each voice to "000" (Write down the original settings so you can reset them later.)
- Enter Bank Edit: Create Voice and set the "AREA LEVEL" parameter for each Area to "127."
- Play a slow chromatic scale up and down the keyboard. All of the notes should have the same loudness. If not, determine which Area sounds the softest.
- Adjust the "AREA LEVEL" parameters of the other Areas to match the loudness of the softest Area.
- Reset the original Create Voice: Velocity Sensitivity parameters for each voice.

This method insures that you will have the greatest overall dynamic range for a given set of sounds in a Bank. As you can see, the overall loudness of a Bank is limited to the softest voice (*Figure 62*). This is one of the reasons why you should always try to create samples with the maximum loudness without distortion.

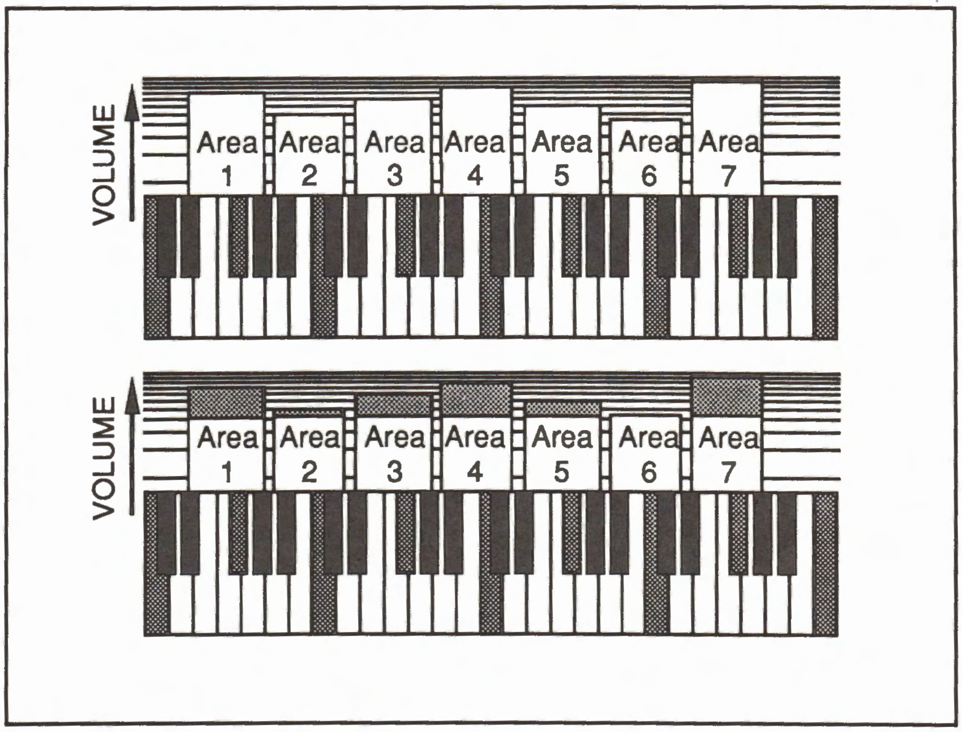

Figure 62: The FZ's Area level parameter lets you adjust the loudness of each Area so that you can balance the sound of the entire keyboard. When adjusting Area levels, the louder Areas will have to be made softer to match the most quiet Area. In this example, Area F is much softer than the others, so they all have to be reduced in volume.

9.11 MIDI Mapping with Create Bank

The FZ gives you the option of controlling all of the Areas in a Bank with one MIDI channel or controlling each separately from one of sixteen MIDI channels. The "MIDI CH" parameter is used to assign an Area to a MIDI channel. Be aware that this setting interacts with the "RECEIVE" parameter of the Effects/MIDI: MIDI function operation. (In order to control Areas with separate MIDI channels, the "RECEIVE" parameters must be set to "AREA.") We've outlined some of the most important features of the FZ's Bank-related operations below:

- Each Area behaves like an independent MIDI module. Making the FZ a polyphonic, multi-timbral MIDI sound source.
- Voices assigned to each Area can be played polyphonically. (Keep in mind that the FZ can never play more than eight notes at a time.)
- When the FZ-1's MIDI "RECEIVE" parameter is set to "AREA," the built in keyboard will play only those voices whose Areas are assigned to the FZ's "BASIC CHANNEL."
- Each Area will respond independently to pitch bend, modulation, velocity, after touch, sustain pedal, and volume pedal messages. (Response to these messages is effected by the settings of the *Effect /MIDI* submode.)

Take a look at the abbreviated Bank Chart below. It shows the Voice, MIDI Channel, and Key assignments for a Bank made of fourteen Areas. It demonstrates some of the power behind the FZ's MIDI mapping capabilities. If the FZ's Basic Channel is set to 1, you can play left-hand bass and right-hand vibes on the keyboard. (If you have an FZ-10M, the controller must be set to transmit on channel 1.) Notice that the key ranges for the bass and vibes sounds overlap the drum, brass, and guitar key ranges. Since each of those Areas is set to a different MIDI channel, those sounds won't be played by the keyboard. Instead, they can be played by a multi-track MIDI sequencer–while you're playing along on the bass and vibes. This is just one example of how MIDI mapping could be put to use.

Area	MIDI CH	Voice No.	Name	Highest	Lowest	Output
1	1	1	Bass F#2	B2	C2	All
2	1	2	Bass F#3	B3	C3	All
3	1	3	Bass F#4	B4	C4	All
4	1	9	Vibes F#5	B5	C5	All
5	1	10	Vibes F# 6	B6	C6	All
6	1	11	Vibes F#7	B7	C7	All
7	2	20	Kick	C2	C2	All
8	2	21	Snare	D2	D2	All
9	2	22	Hi Hat Open	E2	E2	All
10	2	23	Hi Hat Closed	F2	F2	All
11	2	24	Tom Tom	G3	G2	All
12	2	25	Crash Cymbal	A3	A3	All
13	3	30	Brass Hit	F#3	C3	All
14	4	40	Guitar A#3	D#4	E3	All

9.12 Output Channel

Each Area can also be assigned to any combination of the eight monophonic outputs on the back of your FZ. The normal mode of operation is to assign an Area to all eight. This produces up to eight-note polyphony for the voice. For special applications, you may want to change this assignment. For example, you may wish to let up your FZ as a drum machine. You could assign each drum voice to a different Area and each Area to a different output. This would make it possible to route each drum sound to a separate input channel on your mixer so that you could set different EQ, effects, etc., for each sound.

Here is an example of how you could set up an FZ Bank to send drum sounds to different outputs:

Area	MIDI CH	Voice No.	Name	Highest	Lowest	Output
1	3	20	Kick	C2	C2	1
2	3	21	Snare	D2	D2	2
3	3	22	Hi Hat Open	E2	E2	3
4	3	23	Hi Hat Closed	F2	F2	4
5	3	24	Tom Tom	C3	G2	5
6	3	25	Crash Cymbal	D3	D3	6
7	3	26	Ride Cymbal	E3	E3	7
8	3	27	Shaker	F3	F3	8

9.13 Copy, Delete, and Replace

Copy Bank

This is similar to the Copy Voice operation. Copy Bank creates a duplicate version of a Source Bank in a different Bank location (the *destination* Bank). Only Create Bank parameters are copied, not the voices. If the destination already contains Bank data, you will be prompted with "BANK DELETE?" Pushing the **YES** key in response to the prompt will erase the Bank currently in the destination and put a new copy of the selected Bank in its place. Note that the source Bank is unaffected by the Copy operation.

Delete Bank

Delete Bank is slightly different from the Bank Dump: Erase Bank function. It gives you a choice. You can delete the entire Bank and its voices, or just the Bank data (Areas) without deleting the voices.

Delete Area

Delete Area lets you remove individual Areas from a within a Bank.

Replace Bank

Replace Bank is essentially the same as Copy Bank with one significant difference. The source Bank is deleted by the operation. Use Replace Bank to move a Bank from one location to another. Note that you can't replace a Bank with itself (as you might want to in order to save editing changes). Use the Bank Dump/Save Bank operation to store updated versions of a Bank.

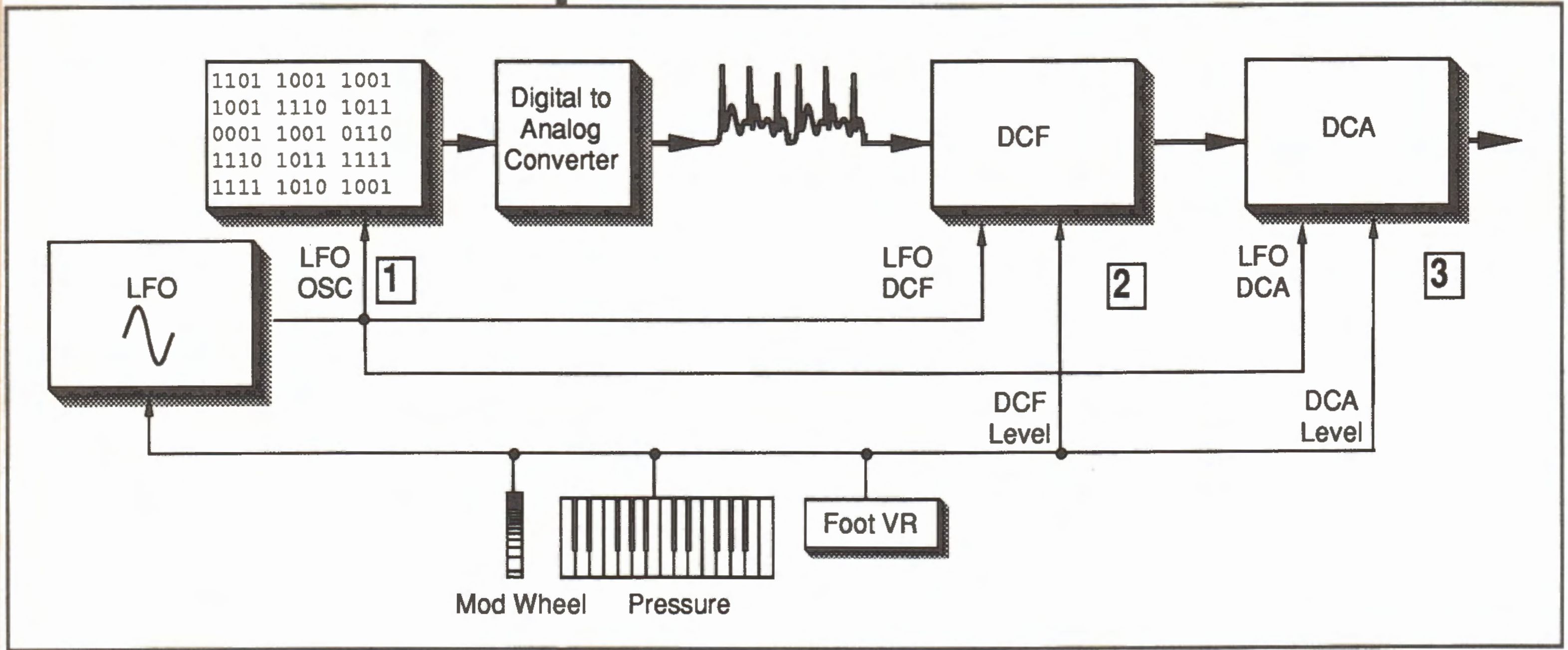

Figure 63: The Mod Wheel, After Touch, and Foot VR can each independently control LFO routing, DCF cutoff, and DCA level.

The FZ-1 offers a number of options for expressive control of its voices. As we've already seen, the FZ's keyboard is velocity sensitive. It is also pressure sensitive as well. The FZ-1 also provides a pitch bender, modulation wheel, sustain pedal, and foot pedal. If you own the rack mount FZ-10M, then you don't have a built-in keyboard, pitch bender, or mod wheel. However, if your MIDI controller has those features, the FZ-10M will respond to them the same way an FZ-1 responds to its built-in controllers. The *Effect/MIDI* sub-mode of the FZ provides four operations for using performance controllers. It is detailed on pages 92 through 97 of the *Operations Manual*. A Menu Overview of the Effect operations is given below.

Effect Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER –>**	**ENTER –> <–ESCAPE**	**ENTER –> <–ESCAPE**	**<–ESCAPE**
MODIFY PLAY	Source Select Voice Edit	<---------------->	Bend Range	Bend Range
	Bank Edit **Effect/MIDI** Data Dump OPT Software	<---------------->	Mod Wheel	LFO OSC LFO DCA LFO DCF DCA Level DCF Level
		<---------------->	After Touch	LFO OSC LFO DCA LFO DCF DCA Level DCF Level
		<---------------->	Foot VR	LFO OSC LFO DCA LFO DCF DCA Level DCF Level
		Dump Effect	Load Effect	Effect Name Execute: Yes/No
			Save Effect	Effect Name Execute: Yes/No
			Verify Effect	Effect Name Execute: Yes/No
			Erase Effect	Effect Name Execute: Yes/No

10.1 Bender

The maximum pitch bend interval is set with this operation. The range is adjustable in semitones from up and down a half step to up and down an octave.

> **Note:** When the FZ's *MIDI* function's "RECEIVE" parameter is set to "AREA," each Area will respond to pitch bending only on its assigned MIDI channel. For instance, you can bend the pitch of the voice assigned to Area 1 without bending pitches of any other voices. Although each Area responds only to its assigned MIDI channel, the "BEND RANGE" value will be the same for all Areas.

10.2 Mod Wheel, After Touch, and Foot VR

Each of these controllers has the same set of parameters, so we can consider them as a group. The three LFO parameters allow you to alter the amount of LFO signal sent to the OSC, DCF, or DCA with the mod wheel, after touch, or foot pedal. This is equivalent to having real time control over the LFO set depth parameters (see *10.8 LFO Set*). Use LFO OSC for vibrato and pitch effects, LFO DCA for loudness tremolo effects, and LFO DCF for timbre tremolo ("wah wah") effects. A very common technique is to use after touch to control LFO effects. Try it– you'll find that it's a very expressive way to control vibrato or tremolo.

DCA level allows you to control the loudness of a voice (or voices) with the mod wheel, after touch, or foot pedal. DCF level allows you to control brightness with any of the three performance controllers as well. DCA and/or DCF levels are often controlled with the foot pedal or after touch to produce crescendo and decrescendo effects. It can be particularly useful with brass and string sounds.

Note: When the FZ 's MIDI function's "RECEIVE" parameter is set to "AREA," each Area will respond to mod wheel, after touch, and foot pedal changes only on its assigned MIDI channel. For example, to use the mod wheel to add vibrato to your string sound without adding vibrato to voices assigned to Areas with different MIDI channel assignments. However, the settings of Effects parameters are used by all Areas. So if you have assigned after touch to control DCA levels, all Areas will change DCA levels when they receive after touch messages on their assigned MIDI channels.

11. MIDI

The FZ has a very comprehensive MIDI implementation. Its *MIDI* function operation part of the *Effect/MIDI* sub-mode. It is detailed on pages 97 through 99 of the *Operations Manual* . A Menu Overview of the *MIDI* function operation is provided below.

MIDI Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	ENTER ->	ENTER -> <--ESCAPE	ENTER -> <--ESCAPE	<--ESCAPE
MODIFY PLAY	Source Select Voice Edit Bank Edit **Effect/MIDI** Data Dump OPT Software	<----------------->	Bend Range Mod Wheel After Touch Foot Vr **MIDI Function** Dump Effect	Basic Channel Receive Basic/Area Control ENA/DIS Program ENA/DIS

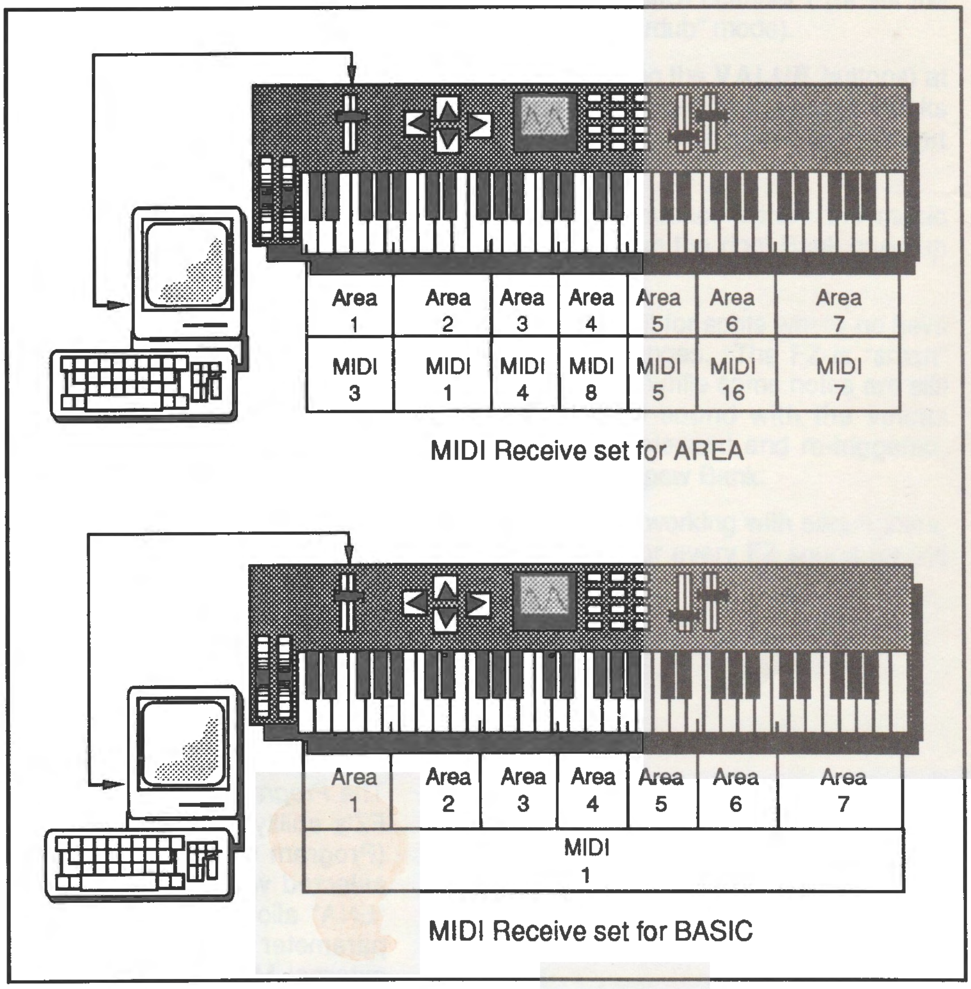

Figure 64: When the FZ's Receive parameter is set to "AREA", each Area in a Bank can be controlled from a separate MIDI channel. When it is set to "BASIC", all of the Areas are controlled by whichever MIDI channel has been selected with the FZ's Basic Channel parameter.

11.1 Basic Channel, Receive Mode

Your FZ will respond to MIDI messages in one of two ways, depending on how its "RECEIVE" parameter is set. When the parameter is set to "BASIC," all Areas in a Bank will be controlled by the same MIDI Channel. That is the channel selected with the "BASIC CH" parameter and is referred to as the FZ's Basic Channel. The basic channel is the only channel that the FZ uses to send/receive Program Change and System Exclusive Data Dump messages.

When the "RECEIVE" parameter is set to "AREA," each Area in a Bank will receive messages only on the channel selected with the Create Bank: MIDI CH parameter (see *11.10 MIDI Mapping With Create Bank*). Here is a list of the messages that can be received independently on each Area:

- Note On/Off (with Note On Velocity)
- Channel Pressure (After Touch)
- Pitch Bend
- Modulation (MIDI Controller 1)
- Sustain (MIDI Controller 64)
- Master Volume (MIDI Controller 7)
- Foot Controller (MIDI Controller 4)

Remember the effect created with the mod wheel, after touch, and foot pedal will be the same for all Areas. They will respond in accordance with the values you set in the Mod Wheel, After Touch, and Foot VR operations.

11.2 Controller Messages

The *Control* parameter lets you turn on (enable) or turn off (disable) the FZ's ability to receive and transmit MIDI Controller messages. Setting this parameter to "ENA" means that the FZ will respond normally to Modulation, Sustain, Master Volume, and Foot Controller messages from an external MIDI controller (in accordance with the setting of the "RECEIVE" parameter). If you set this parameter to "DIS," the FZ will not respond to any Modulation, Sustain, Master Volume, or Foot Controller changes.

The FZ will always respond to velocity and after touch (as long as the Create Voice: Velocity Sensitivity, Create Bank: Max/Min Touch, and Effects/MIDI: After Touch parameters have been set appropriately).

If you want to disable the FZ's response to Pitch Bend changes, simply set the *Bend Range* parameter to "00."

11.3 Program Change Messages

The *Program* parameter lets you turn on (enable) or turn off (disable) the FZ's ability to receive and transmit MIDI Program Change messages. (Program Change messages are only sent/received on the channel selected with the "BASIC CH" parameter.) Setting "PROGRAM" to "ENA" allows the FZ to send/receive Program Changes. When this parameter is set to "DIS," the FZ ignores Program changes from an external MIDI controller and will not transmit any Program Changes when you select Banks or Voices.

When "PROGRAM" is set to "ENA," Program Change messages are only sent/received when the FZ is in the *PLAY MODE* . If "BANK No." is selected in the *Play* display, the FZ will send/receive only program numbers 1-8. When "VOICE No." is selected, the FZ will send/receive only program numbers 1-64. When receiving program numbers, the FZ will recognize 1-127. Since the FZ has only eight Banks and sixty-four voices, numbers greater than eight or sixty-four will be automatically converted to an appropriate Bank or voice number.

Since Program Change messages are only recognized on the FZ's Basic Channel, you can use that channel to change Banks during a multi-track sequence. This will allow you to make complete changes of keyboard setups "on the fly" while the sequence is playing. Let's assume that you already have a multi-track sequence recorded and now you want to record a series of Bank changes as part of the sequence:

- Set the FZ's "RECEIVE" parameter to "AREA."
- Set the "BASIC CH" parameter to "01."
- Set "PROGRAM" to "ENA."
- Load the Banks you need for your sequence into the FZ (up to eight).
- Assign the desired MIDI channels to each Area of each Bank.
- Put the FZ in the *PLAY MODE*.
- Set your sequencer to record a new track on MIDI channel 1 (or set the previously recorded channel 1 track in the "overdub" mode).
- Run the sequence and change the Banks (using the **VALUE** buttons) at the appropriate spots. (If the music is too fast for you to change Banks in real time, use the sequencers "step edit" or "punch in" modes to insert the Program Changes in the right spot.)
- Insert the first Program Change into the sequence before the music starts. That way, the FZ will automatically have the right Bank cued up when the tune starts.
- Unlike other instruments, you don't have to wait for spots where no keys are held down when you make the Bank changes. The FZ is "smart" about how it handles Bank changes that occur while some notes are still being played. Held notes will continue to sound with the voices assigned to the old Bank. When they are released and re-triggered, they will sound with the voices assigned to the new Bank.

You'll find this technique very helpful if you're working with sequencers, since you won't have to use separate tracks for every FZ sound (or set of sounds).

12. Memory Management

Think for a minute about how much information your FZ can hold in its internal memory. The sampling memory can hold 14.5 seconds worth of sounds sampled at 36k (twice that if you have an FZ-10M or the MB-10 memory expansion for the FZ-1). The sampling memory can be split up into as many as sixty-four FZ voices, each voice having over one hundred unique parameters. Voices can be organized into Areas, which have an additional nine parameters each. Areas can can be organized into as many eight Banks, and each Bank can hold up to sixty-four unique Areas. Let's not forget the FZ's Effects settings either. All of that adds up to some twelve thousand or so parameters that your FZ holds in its internal memory for you!

That's an enormous amount of data to keep track of. If you're like most FZ users you'll quickly find that you want to be able to use more Banks, Areas, and Voices than the FZ can possibly hold internally. No problem. That's what the disk is for. You can store any kind of FZ data, voices, Banks, effects settings, even the entire contents of the instrument as files on micro floppy disks. Since there are so many things you can store, and several ways to store them, we thought we should spend a little time looking into how to use and organize the FZ's disks.

12.1 About Disks

The FZ uses *Double Sided-High Density* micro floppy diskettes. We refer to them simply as disks, but be aware that you must use *High Density (HD)* disks. There are three types of micro floppies commonly in use today: Single Sided- Double Density(SS/DD), Double Sided-Double Density (DS/DD) and Double Sided High Density (DD/HD). Physically, each of these disks appears exactly the same, but there is a difference. While the difference between disk types isn't visible to you, it is to your FZ.

Disk manufacturers certify the storage capacities of their disks. Each disk type is made to hold a different amount of data. SS/DD disks are certified to hold 400 kBytes of data. DS/DD are certified to hold 800 kBytes, and DS/HD disks are certified to hold 1000 kBytes of data. 1000 kBytes are the same as 1 mByte (or 1 million bytes). What does "certified" mean? It means that a disk is guaranteed to hold at least the specified amount of data. Just as important, it means the disk is not guaranteed to hold more than the specified amount of data. In other words, if you try to put more data on a disk than it is certified for, you may lose the data. Usually the contents of the entire disk become unreadable when this happens.

Your FZ was designed to use only DS/HD disks since the minimum memory size for the FZ-1 is 1 mByte. This insures that a *Full Dump* from an FZ-1 would fit onto a single DS/HD disk. Even if you are not storing a Full Dump, the FZ will let you pack up to 1 mByte of data (Banks, voices, etc.) on a disk. When you format a blank disk with the FZ, it expects it to be a DS/HD. Some FZs may be able to run the format routine with a DS/DD or even an SS/DD disk. (This isn't all that unusual in the computer world. Our trusty Macintosh, for example, can be "fooled" into formatting an SS/DD as though it were a DS/DD.)

Since DS/DDs are a little less expensive, you may find yourself tempted to use them in place of DS/HDs. Don't! Remember, those disks are not certified to hold more than 800 kBytes of data. Your FZ will put more than that onto the disk. You can bet that if you try to use DS/DD disks with your FZ, that one day the disk will fail. Murphy's Law guarantees (in this case maybe *certifies* is a better word) that the disk you lose will be the only one with the voice/Bank/sample you need. Murphy's Law also states that the disk will fail precisely when you need it the most, i.e., right in the middle of a gig or session. Believe us, it's not worth the pain and frustration. Be sure your disks are HDs.

Formatting New Disks

You'll undoubtedly acquire quite a collection of disks for your FZ. You'll find that you can get a considerable saving on the price of disks it you buy them in bulk from a mail-order supply house. Bulk disks are the same high quality certified disks you get at the computer boutique, but without the fancy (expensive) packaging. No, you won't have to buy five hundred or more at a shot. Most supply houses let you purchase bulk disks in lots of ten or twenty.

As you probably already know, you can't just take a new disk out of the box and save FZ data on it. You must first format the disk by entering and executing the *Format Disk* operation from the Data Dump sub-mode. When you get a new box of disks, it's generally a good idea to sit down and format several of them (or even the whole box). This way, you won't have to take the time later when you're in the middle of a hot project. Should you be in the middle of something and need to format a disk, don't worry. You can format disks without losing anything that you're currently working on.

12.2 Data Dumps

Your FZ saves and loads data to and from disks as files. There are four types of FZ files:

- Full Dump files contain all Bank, Area, Voice, and Effect settings active in the FZ as a single block.
- Bank Dump files contain all of the Areas and Voices that make up a single Bank in one file.
- Voice Dump files contain a single FZ voice in one file.
- Effect Dump files contain Bender, After Touch, and Foot VR settings as a single file. (Note that this does not include the *MIDI* function settings.)

The Data Dump sub-mode has four functions used to transfer these file types to and from disks. There is a separate function for each file type. Data Dump functions are detailed on pages 100-123 in the *Operations Manual*.

Below is a Menu Overview of the complete sub-mode.

Data Dump Menu Overview

MODE	SUB-MODE	FUNCTIONS	OPERATIONS	PARAMETER SETTINGS
MODIFY->	**ENTER ->**	**ENTER -> <--ESCAPE**	**ENTER -> <--ESCAPE**	**<--ESCAPE**
MODIFY PLAY	Source Select Voice Edit Bank Edit Effect/MIDI **DATA DUMP** OPT Software	**Full Dump** Bank Dump Voice Dump Effect Dump	Load Full Save Full Merge Full Verify Full Erase Full	Execute: Yes/No Execute: Yes/No Execute :Yes/No Execute: Yes/No Execute: Yes/No
		Full Dump **Bank Dump** Voice Dump Effect Dump	Load Bank Save Bank Merge Bank Verify Bank Erase Bank	Bank Name Execute: Yes/No Bank Name Execute: Yes/No Bank Name Execute: Yes/No Bank Name Execute: Yes/No Bank Name Execute: Yes/No
		Full Dump Bank Dump **Voice Dump** Effect Dump	Load Voice Save Voice Verify Voice Erase Voice	Voice Name Execute: Yes/No Voice Name Execute: Yes/No Voice Name Execute: Yes/No Voice Name Execute: Yes/No
		Full Dump Bank Dump Voice Dump **Effect Dump**	Load Effect Save Effect Verify Effect Erase Effect	Effect Name Execute: Yes/No Effect Name Execute: Yes/No Effect Name Execute: Yes/No Effect Name Execute: Yes/No
		<----------------> **<---------------->**	Select Device Port, MIDI Format Disk	Device: Disk, Disk Name Execute: Yes/No

Load File

When you load files to add additional Voices or Banks, be sure to choose an unused location, since loading a file *replaces* the contents of the Voice or Bank location with the new file.

You can only load files that have been saved in the same format as the loading function. In other words, Load Full can only load Full Dumps. Load Bank can only load Bank Dump , and so on. You can not, for example, load a voice file directly from a disk that contained only Bank Dumps. Instead, you would load the Bank Dump file that had the voice you were looking for assigned to one of its Areas.

Loading files replaces the current file of the same type. Load Full will replace all Bank, Voice, Area, and Effect parameters with the data from the new file. Load Bank replaces the selected Bank number with the Bank loaded from the disk. Load Voice replaces the selected voice number with the voice loaded from the disk. Load Effect replaces the current Effects setting with the Effect settings stored on disk.

Save File

These operations are used to save an active set of Bank, Voice, Area, or Effect parameters as a file on an FZ disk.

Merge File

Merge allows you to add new Banks, Voices, and Areas to the FZ without any of the current Banks, Voices, or Areas. Merge Full will fill empty Banks, Areas, and Voices currently in the FZ from a Full Dump file on disk. Note that all of the current Effects settings will be replaced with the Effects settings saved on disk. Merge Bank will fill empty Areas and Voices in the selected Bank with Areas and Voices in the Bank Dump file saved on disk.

Verify File

Verify compares the active Full, Bank, Voice, or Effect parameters with those of a file saved on disk. This comes in handy if you want to be sure that a file was saved correctly. Also, when you are doing a lot of editing, you can use this function to check to see if you've updated a file since your last save or load.

Erase File

Erase is used to remove a file from a disk.

12.3 Organizing Your Disks

By letting you store FZ data in a variety of file formats, Casio has provided you with a good deal of flexibility for organizing your data onto disks. The Full Dump format is perhaps the most convenient to use for live performances. You can load a Full Dump directly from the *PLAY MODE* with very little button pushing (all other file types most be loaded from within the *MODIFY MODE* and require considerable button pushing). Also, a Full Load completely re-configures the FZ, eight new Banks, sixty-four new sounds, etc. All of the disks in Casio's FZ sound library are stored in the Full Dump format.

When you're not on stage, but working at home or in the studio, you may find that working with Bank Dump files is more convenient. Depending on the sampling time used by the voices and the numbers of Areas in a Bank, a disk can hold well over eight different Banks. You'll also find it more convenient to work with Bank Dump files when you're designing your own custom Full Dump disks for live performance.

The procedure for transferring Bank Dumps from Full Dumps is outlined below:

- Have a couple of blank, formatted FZ disks handy. (Remember to use DS/HD disks.)
- Insert the Full Dump disk that contains the Bank (or Banks) you want and execute *Load* from the *PLAY MODE*. (All Casio FZ sound library disks are Full Dump disks.)
- Eject the Full Dump disk and insert a blank, formatted FZ disk. This will be your Bank Dump disk.
- Enter the Bank Dump function of the Data Dump sub-mode and use the Save Bank operation to save any of the current Banks you want onto your Bank Dump disk.
- If you want to save more Banks on the disk, eject the disk and insert another Full Dump disk. Enter the PLAY MODE and execute Load. Repeat the previous two steps until you've saved all the Banks you need. If your Bank Dump disk fills up, just start a new one.

Here's how to create your own custom library disks (Full Dump disks).

- Have a blank, formatted FZ disk and the Bank Dump disks with the Banks you want handy. (Use the procedure described above to create the Bank Dump disks.)
- Clear the FZ's memory by executing Load Exec from the *PLAY MODE* with no disk in the drive. (You'll see "DISK NOT READY" in the display.)
- Insert a Bank Dump disk in the drive.
- Enter the Bank Dump function of the Data Dump sub-mode and use the Load Bank operation to load any of the Banks you want from the disk into the FZ.
- If you want to add more Banks, and you still have room in the FZ, eject the disk and insert another Bank Dump disk. Use the Load Bank operation to load the Bank(s) you want from this disk.
- Repeat the previous step until you have all the Banks you want (or the FZ's memory is full).
- Insert the blank, formatted FZ disk.
- Enter the Full Dump function and use the *Save Full* operation to store your custom setup on the disk.

12.4 Select Device

The FZ can transfer files to and from other devices (like another FZ or a computer) via its MIDI ports the 25-pin *External Port* (used for direct connection with other FZ's). All of the file operations except *Erase* can be used with the MIDI or external ports. Be sure to read pages 125 and 126 of the *Operations Manual* for the details. Use the *Select Device* function to select where a file will be transferred to or from: *Disk, MIDI, PORT*.

Note: When you use MIDI to transfer FZ files, the MIDI IN and the MIDI OUT ports of both units most be connected.

Bank Chart For FZ-1 and FZ-10M

			Key Range			Velocity Range				
Area	Voice Name	No.	Original	Highest	Lowest	Max Touch	Min Touch	Level	MIDI Ch.	Output
1										
2										
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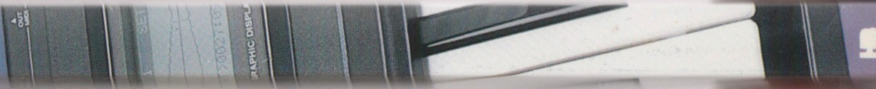
CASIO FZ-1 & FZ-10M The Essential Guide to Practical Applications



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