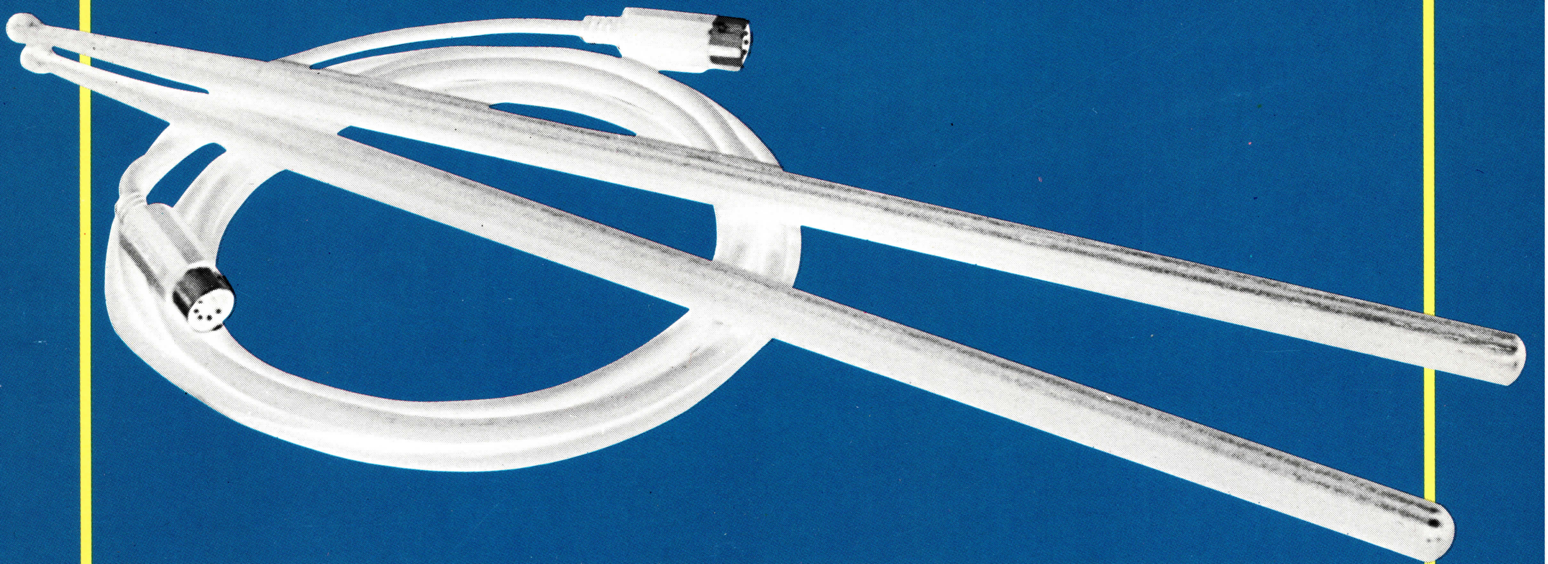
THE ELECTRONIC DRUMMER

BY NORMAN WEINBERG

Basic sound reinforcement of
acoustic drums to complete
electronic/MIDI setups for
today's drummer.



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PREFACE

Today's drummer is on the edge of an exciting new world of sound, technology, equipment, computers, experimentation, and even newly created musical forms. This is the most exciting time in history to be a drummer! Perhaps the most wonderful part of all is that for the first time drummers can be in control of the final expression of their music.

Let me explain what I mean by "in control." You can sit down with a piece of paper and a pencil, and enjoy the creative activity of drawing. Your creation may not be great, but you've had the experience of working out a final version of your inspiration. You didn't have to rely on someone else's talent, technique, or problems. You did it all by yourself.

You may decide instead that you want to work with a piece of clay and try your hand at making a sculpture. You mold it, play with it, change it, adjust it, shape it, and swear at it, until it takes the form that you see in your mind's eye. Again, you are in control of how the sculpture turns out. If it's great, you take the credit; if it's a piece of junk, you take the blame! If you really hate it, you can take a hammer to it and start all over.

Until just a couple of years ago, if you wanted to create music, you first had to commit yourself to several years of practice on your instrument. You then had to wait for a group of compatible musicians to come along who wanted you to play with them, and only *then* could you create your art. At best, your role in the creative process would be designing a drum part to go along with the rest of the music. But you had little control over the sounds that the other players were going to produce. Even if you wrote your own music, you had to rely on others to interpret the music for you. Sometimes you would be happy with the result, sometimes you wouldn't. Meanwhile, it's been five to twenty years from the time you started learning your instrument to the time you create a finished product.

That was the way music was made back in the dark ages (anything prior to 1983). Those times are gone. Today's electronic drummer can have a musical inspiration, sit down at an electronic kit, and compose all the parts to the entire song. You can then orchestrate the entire piece, selecting any instrument to play any of the parts within the composition. From pads or acoustic drums, you can play violins, cellos, flutes, electric bass, or any sound you have the imagination to hear. You can perform the piece at any tempo, record the finished product, and print out the sheet music, all in the same day. Every aspect of every part, even down to the smallest detail, is in the hands of the electronic drummer. The artistic, creative process is complete, and the control is in the hands of a single person: you, the drummer!

Let's say you don't want to compose, and all you want to do is play. Should you still get involved with the electronic percussion scene? I believe that you should. These electronic instruments are becoming a significant part of the music industry. The more flexible you are, the more opportunities you will have for employment. Before drummers were involved with drum machines, record producers were using keyboard players to program the drums for recordings. Do you want your gig to go to a keyboard player? If you know how to work with electronic drums and drum machines, *you* may get the gig. But if you don't know anything about them, you can be sure that you won't be called.

When you hear people say, "Drum machines are going to take jobs away from other drummers," they may be right. But drum machines can't play by themselves! Drummers who understand how to program the machines *will* take jobs away from drummers who refuse to learn the new technology.

Some people may say that electronic drums and drum machines are a fad: "Everybody is using them now, but soon, the public will be bored listening to them." This may be true, but are they going to get bored next week or in ten years? Could it be that this is just a rationalization? I can hear them now: "The electric bass is just a fad; I'm not going to get involved because the bass violin is the only 'real' instrument." What ever happened to the clavichord anyway? New instruments come along that make older instruments obsolete!

Other drummers might say that electronic drums are just not as expressive as acoustic instruments. Wanna bet? It takes a lot of time, a lot of knowledge, and a lot of work, but it is possible to make electronic drums just as expressive as acoustic drums (it's not the instrument that is expressive, it's the player). Keep in mind that these people may have been playing acoustic drums for ten years in order to achieve this level of expression. Very few people have been playing electronic drums for more than five years because the instruments simply weren't around. As the technology progresses and grows, more electronic instruments will be designed to make expression easier to program. Who knows, maybe someday electronic instruments will even enhance our musical expression beyond the current acoustic limitations.

The bottom line is that music is changing. This should be no surprise to anyone. Music, just like any of the other arts, has had a history of change from one style to another. The Renaissance, Baroque, Classical, Romantic, and Twentieth Century periods have all had their different instruments, styles, and musical languages that defined the particular period. I believe that we have entered a new era of music, both on the popular side and in the world of "art music." Composers like Steve Reich and Philip Glass are inspiring musicians from every facet of the music world, from the symphony hall to the arena.

You have a choice to make. You can either "go with the flow" and embrace the new music of our time, or you can ignore it and hope that it will go away. If you choose the second path, put this book down now! If you choose the first, then embrace these new technologies with all the enthusiasm and open-mindedness of a child. You will find that music will mean much more to you than ever before, that this stuff isn't as scary or as difficult as you might think, and that controlling your art is a very satisfying experience!

THE BASICS OF ACOUSTIC SOUND

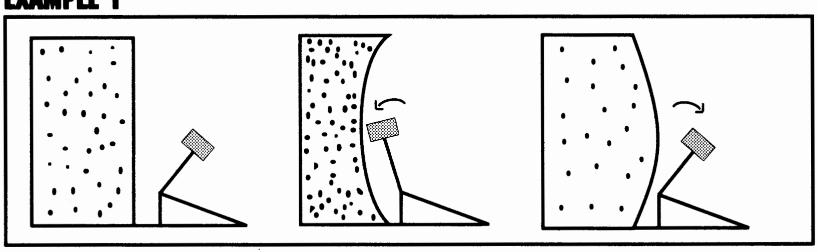
WHAT IS SOUND?

As musicians, our ears are our most valuable tools. Almost every musical decision that we make depends, in part, on how we think a particular musical idea will sound. Many of us take our ears for granted. Yet they are absolutely incredible machines! They are so sensitive that they can detect sounds softer than a jelly bean dropped on the carpet, or louder than a roaring 747 jet. They can identify your best friend's voice from a single word over the phone, and they can easily break down the complex sounds of the largest symphony orchestra into each of the individual instruments.

Just exactly what are your ears responding to when they hear something? What is sound?

Sound is nothing more than organized patterns and movements of the air around us. Air is made up of different types of molecules—little atoms of different gases that we learned about in grade school and have long since forgotten. Let's take the example of a stick hitting a drum and imagine that you could see the movement of air. When the stick hits the drum's surface, the drumhead makes a series of short, fast movements back and forth. These movements actually push the stationary air that was surrounding the instrument before it was struck. These patterns of movement are called **sound waves**.

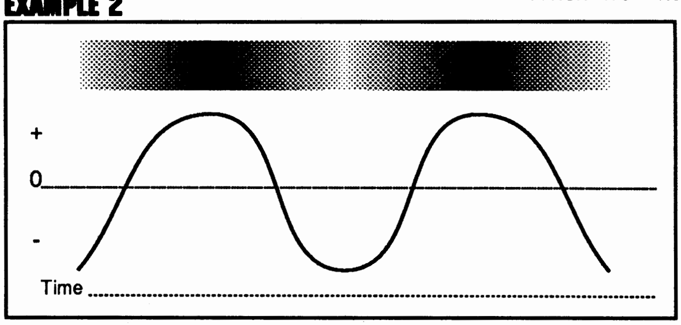
EXAMPLE 1



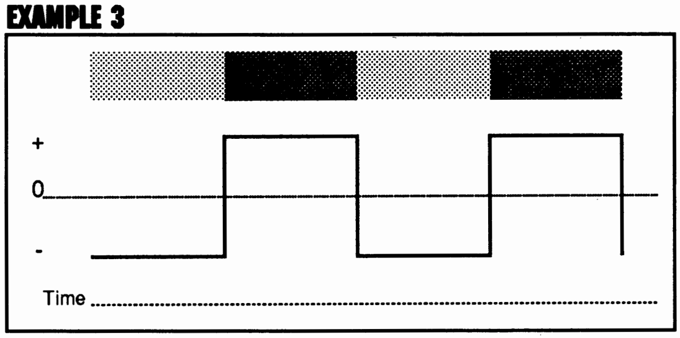
Imagine the movement of a drum head on a bass drum. As shown in *Example 1,* when the drumhead pushes forward, it causes a **compression** of the little molecules that make up the air. They move closer together than they were in their normal at-rest state. When the head vibrates back in the other direction, it causes a **rarefaction** because the molecules are less dense (further apart) than normal.

It's difficult for us to visualize air pressure because we're not able to see it. In order to make things easier to understand, sound waves are often represented as a graph. Looking at *Example 2,* you can see the air pressure and its corresponding graphic representation. The up/down movement on the graph represents the different levels of air pressure. Since the compression and rarefaction of the air is more or less than the at-rest state of the surrounding air, it can be shown by using positive and negative numbers to represent the relative air pressure. The left/right movement on the graph represents a period of time. The wave in this particular example is called a **sine wave**, and is the result of a smooth and equal transition from compression to rarefaction over time. *Example 3* shows the air pressure and the graph of another type of wave called a square wave. Most basic sound waves take their names from their shape, such as rectangle, triangle, sawtooth, etc.

EXAMPLE 2



EXAMPLE 3



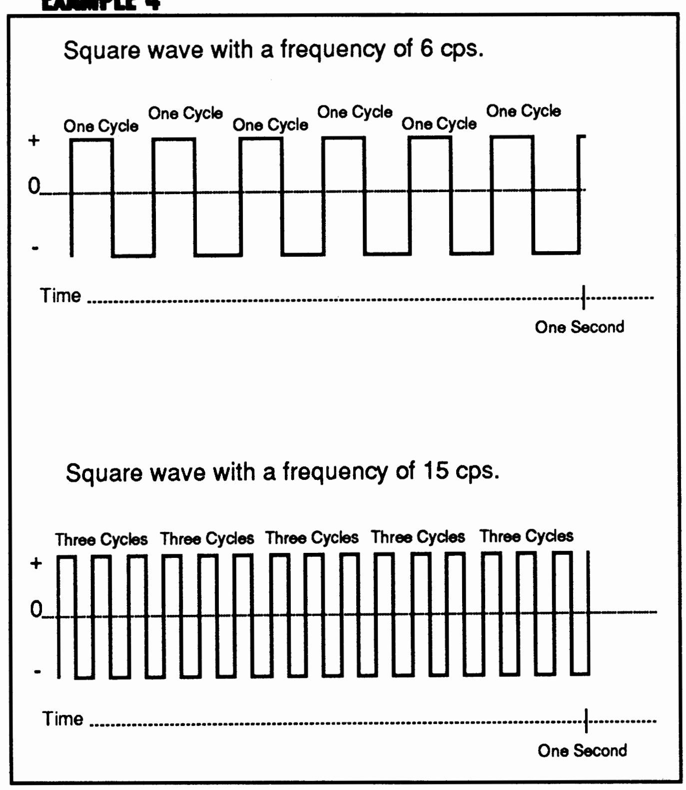
Our ears are extremely sensitive to these different levels of air pressure. When we hear sounds, our ears are picking up the different pressure readings and sending them to our brain to sort out. When we interpret musical sounds, we break them up into four different ingredients: pitch, amplitude, timbre, and duration.

PITCH

Pitch can be defined as the highness or lowness of a sound. If you play your floor tom, it will usually sound at a lower pitch than your mounted toms. How do these pressure levels indicate pitch? Remember when I said that sound was an ordered pattern? If the pressure patterns happen at a slower rate, then the perceived pitch will be lower. If the patterns occur at a faster rate, then the pitch will sound higher (see *Example 4*).

In general, human ears can hear sounds that range from about 20 patterns per second to around 20,000 patterns per second. These recurring patterns are called **cycles** and are measured by a term called **Hertz**, after a 19th-century German physicist. Most often abbreviated as Hz, Hertz is a measurement of a sound's wave, or number of cycles, occurring per second. (kHz indicates kilohertz, or 1,000 hertz.) This is the sound's **frequency**. A complete cycle of a wave form (also called a wavelength or a period) consists of the highest *and* lowest points of air pressure. A positive and a negative reading are included in a single cycle. A pitch of 440 Hz means that there are 440 complete cycles of the waveform during every second of time. As you might expect, a wave that is 440 Hz is going to reach our ears twice as often as a wave of 220 Hz.

EXAMPLE 4



AMPLITUDE

Amplitude is just a fancy word for volume. However, there is a slight difference between the two. Amplitude is the measurement of the density of the molecules in the air. Volume is simply a measurement of how loud a sound is. Amplitude plays a larger part in the sound's color (identity) than volume does. Think about the sound of a floor tom when struck softly. At a soft level, the head barely vibrates. Now, imagine the sound when you really whack it. The more forceful vibrations are going to move the air in a slightly different way. The sound is not only louder, it is different. This change in amplitude causes the drum's tonal color to change.

If the molecules range from very dense to very sparse, then a sound is said to have a higher amplitude than if the sound's pressure readings are less drastic. *Example 5* demonstrates a square wave of low amplitude and high amplitude.

Amplitude is measured by **decibels**. A decibel is the least amount of change that the human ear can hear. Generally speaking, we can hear from one decibel to over 120 decibels (this is the threshold of pain). By the way, an increase of 10 decibels will make any sound seem twice as loud.

TIMBRE

Timbre (rhymes with amber) is a word that is used to describe a sound's tone color. This doesn't translate easily into words, but it is the timbre that determines whether you distinguish a flute, a clarinet, or a xylophone sound. While these three instruments can play a note of the same pitch and the same amplitude, they will always have a different timbre. The sound's timbre is how our ears can differentiate between instruments.

The shape of a waveform plays a big part in the sound's timbre. Different wave shapes (such as square and sawtooth) are created by combining different frequencies (harmonics or overtones) along with the fundamental frequency. The **fundamental** is usually the frequency we hear as the pitch. It is the lowest frequency contained in the sound. **Harmonics** and **overtones** are multiples of the fundamental's frequency. If the fundamental's pitch is 100 Hz, then harmonics might occur at 200, 300, 400, 500, 600, or 1000 Hz.

Sine waves are the most pure waves, as they contain no harmonics along with the fundamental frequency. The flute comes very close to playing a pure sine wave. A sawtooth wave contains all of the harmonics along with the

FUNDAMENTALS, OVERTONES, AND HARMONICS

I often hear the terms fundamental, overtone, and harmonic thrown about. Are there any differences between these terms?

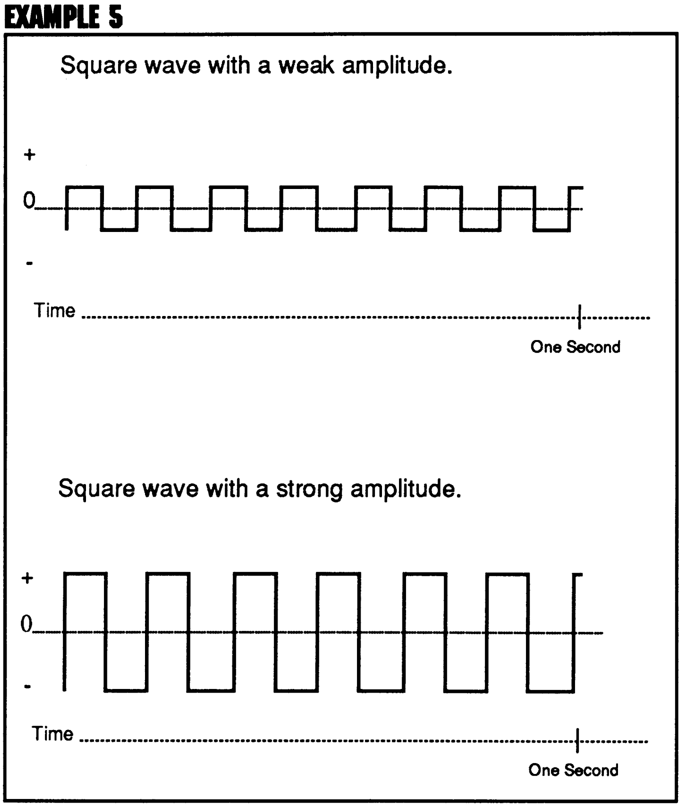
Yes there are. Remember that the frequency of a sound is the number of cycles that occur during one second of time. The **fundamental** frequency is usually the lowest frequency contained in any sound. It is the frequency that determines the pitch, and we tend to hear it stronger than any other. Whenever you whistle the melody of "Happy Birthday," you are whistling the fundamentals.

Overtones and harmonics are both multiples of the fundamental's frequency. The main difference is that harmonics are whole number multiples (such as 2, 3, 4, 5, etc.) while overtones can be any type of multiple: whole numbers or fractional numbers (such as 2.546 or 12.072).

Let's see how this might work. If you have a fundamental frequency of 440 Hz, then you are playing the pitch of "A." By adding the frequency of 2640 Hz, you would be adding a harmonic because 2640 (440 X 6) is a whole number multiple of the fundamental. But, if you added the frequency of 2534.4 Hz, (440 X 5.76) you would be adding an overtone, because this is not a whole number multiple. By adding either of these frequencies to the fundamental, you change the timbre of the sound.

In the real world, most harmonics are just a little out of tune. It is very difficult to design an acoustic instrument with totally pure harmonics, but that is what gives many instruments their distinctive timbre. A cymbal has very few harmonics but many overtones. This is why a cymbal has its characteristic "crash" sound. A vibraphone tone contains several harmonics and only a few overtones, and that is why it sounds so "focused."

EXAMPLE 5



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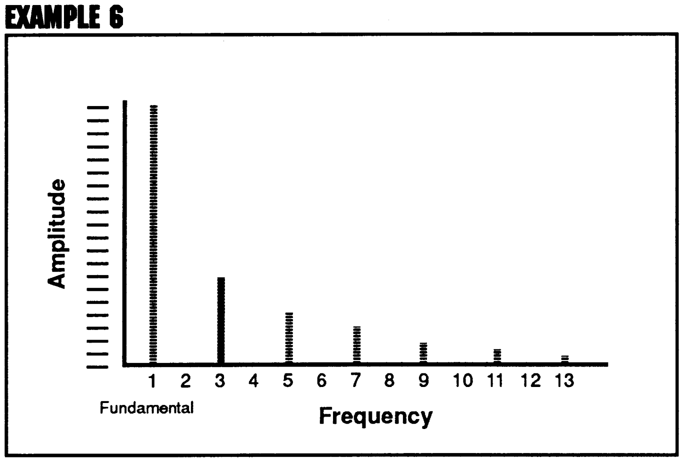
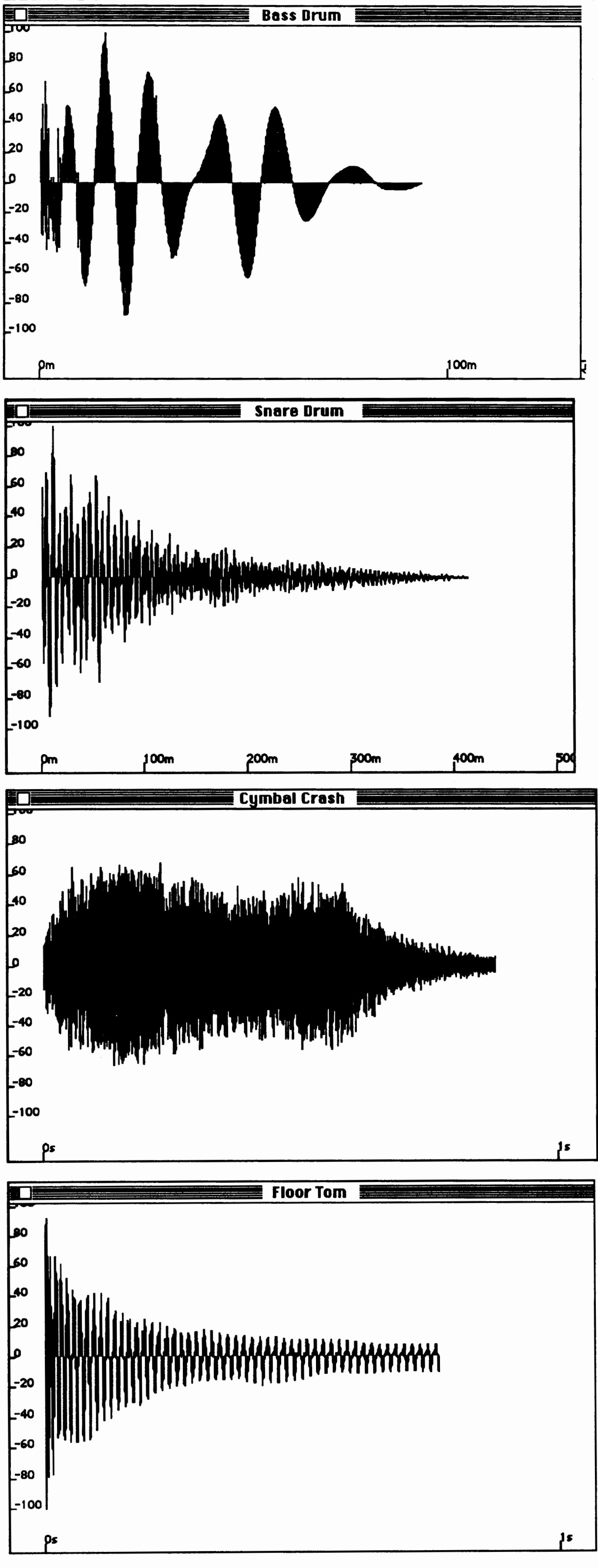


EXAMPLE 7



fundamental frequency. Brass instruments tend to produce waves that are sawtooth in quality. Square waves and triangle waves are very closely related because they include only the odd numbers of harmonics. The difference between them is in the amplitude of the different harmonics. In other words, some of the harmonics are stronger than others.

A square wave for example, contains the odd number of harmonics, and the amplitude (or strength) of each harmonic decreases as its frequency increases. As you can see in *Example 6* (sometimes called a spectrum plot), if a square wave had a fundamental frequency of 100 Hz, then the harmonic at 300 Hz would have an amplitude 1/3 as strong. The harmonic at 500 Hz would have an amplitude value of 1/5 of the fundamental, and the harmonic at 700 Hz would be 1/7 as strong as the fundamental. The clarinet, playing in its lowest range, comes closest to a pure square wave. *Example 7* shows many different types of waveforms.

DURATION

Duration is the length of a particular sound. At its most basic level, the duration of a sound begins when you first hear it and lasts until the sound fades away to silence.

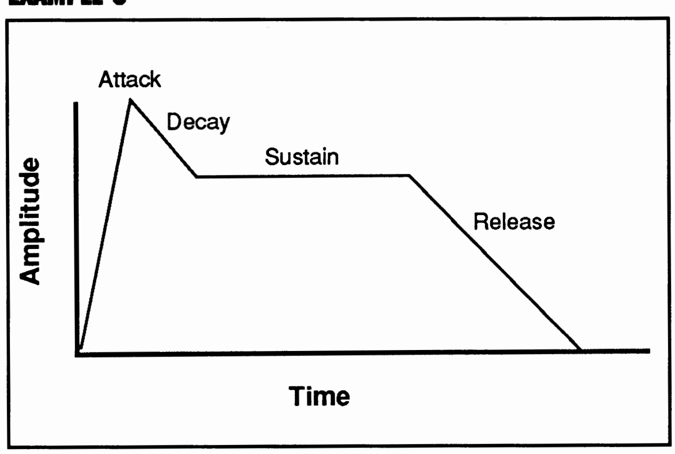
Duration, since it is linked to time, can be described in relation to time. How does the sound begin? Does it begin with a sharp, strong attack, like a snare drum, or does the sound seem to grow after the attack, like a large gong struck with a soft mallet? Does the sound die away quickly, like a woodblock, or does it ring for a long time, like a church bell? The changes in a sound's overall movement through time are called its **envelope**. *Example 8* is a typical envelope graph of a sound.

But duration can exist at several levels. Instead of indicating the overall length and quality of the entire sound, the duration may also measure a single portion of the sound.

If you strike a cymbal, you will hear the timbre change over the course of time. At first, all of the harmonics and overtones that make up a cymbal crash are included with the fundamental frequency. But as the note starts to decay, some of the overtones will die out quickly as others remain. It is possible to isolate the envelope of each individual harmonic and overtone contained in the sound.

As you can see, your ears do an amazing job of sensing the changes in air pressure, and your brain brilliantly identifies the different qualities that make up any particular sound. Trying to imitate the various sound parameters with an electronic instrument is no piece of cake. I'm glad that other people do most of the hard work for me! How do they go about building these electronic sounds from wires, circuits, and chips?

EXAMPLE 8



drummer some!" Of course, your sound generator would have to be able to produce those different sounds in the first place.

DATA STORAGE

Data storage devices are used to record and store electronic information for future use. There are three main types of information that you will want to store with your electronic system.

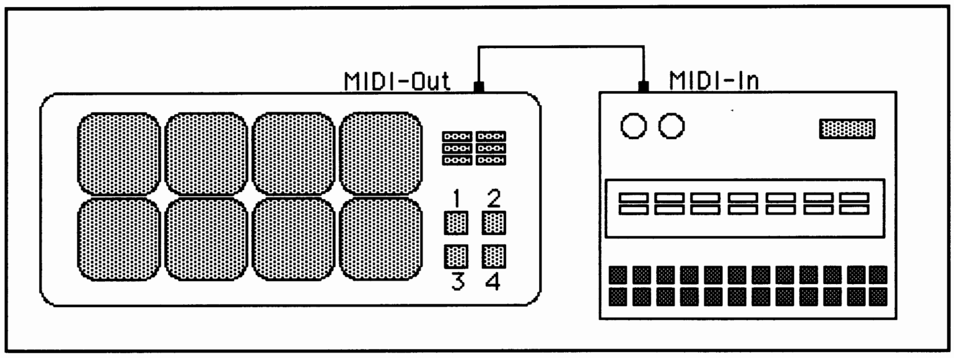
The first type of information is audio. You will probably want to record the sounds that you create using your electronic instruments. A hand-held tape deck can be purchased for about $25. If you desire a more flexible unit that will let you do overdubs just like the big guys, you can purchase a portable four-track cassette recorder for under $500. If you have the desire and the bucks, you could pop for a sixteen-track digital recorder. The purpose of all these different machines is the same (although the quality will be different): to get the sound of your music onto tape so that other people can hear it. After all, it's hard to get all your fans into your living room at the same time!

MIDI recorders store another type of information. Because MIDI commands are digital and travel too fast to be recorded on tape, they require a different type of system. A computer or a dedicated MIDI recorder is the only option available. MIDI recorders record the MIDI data stream as it happens. They can play these MIDI messages back at a later time through MIDI instruments. In other words, MIDI recorders are not recording any sounds at all; instead, they are recording the MIDI information that instructs the other devices in the system to make the sounds. In order to listen to a recording of MIDI messages, you must have them plugged into MIDI sound generators.

One of the great advantages of MIDI recorders is that once the commands are recorded in digital form, they can be easily altered. If the messages are played back at a faster rate, then the music will be performed faster. If you erase a certain amount of messages, you've performed a splice in the musical material. If you want to hear your song played by an ensemble of twelve marimbas, just send the messages to a sound generator that can produce marimba sounds.

The third type of information that you may want to store is data. Data can be recorded to several different formats. Some devices let you store your data off to cassette tape, memory cartridges, memory cards, or even to a computer if you have the proper software. Data is different than MIDI commands because it contains non-performance information about the sounds that a synth might produce. As an example, let's say that you own a drum machine, and you fill up all of the available locations for songs. Do you just buy a new one so that you can program more songs? Not unless you buy a new car anytime the ashtray gets full!

EXAMPLE 12



Just as you would empty the ashtray of cigarette butts, you can empty your drum machine of song data. Instead of dumping the data into the trash, you dump it into a tape machine, a computer, or some other type of data storage medium. This frees up your drum machine's memory, and you can create more songs. If you ever want to get the old songs back, just dump the current ones off to a different tape (or other medium), and load the older ones back into the machine. It's that simple!

PLAYBACK SYSTEMS

A playback system is required whenever you want to hear what is coming out of the audio-out jacks of your sound generators. A playback system consists of an amplifier, which increases the voltage level of the signals, and a speaker system, which transforms the voltages into moving air. You can plug your electronic drums into a boom box with a pair of headphones, or you can spend several thousand dollars on mixers, preamps, biamp systems, crossovers, and modular speaker cabinets. The final result is the same (but again, the quality is different) no matter what type of playback system you use. Playback systems get the sounds from the machines to your ears.

EXTRA GOODIES

Items that fall under the classification of extra goodies include additional pieces of gear that are required (or at least helpful) in an electronic drum studio. Cables, cassette tapes, computer disks, and data cartridges are an important part of any electronic system (none of this stuff will work without cables). This category also includes the little things that make your life easier, such as multiple-outlet strips and surge protectors for power requirements. Storage cases, racks, and covers for your hardware and software also come under this roof.

Another type of extra goody is the "problem solver" box. If you have too many different MIDI devices serving as master controllers, get a MIDI mixer. Are you trying to run sixteen different sound generators? Buy a MIDI patch bay. Are you trying to control a pre-MIDI sound generator with your MIDI drum pads? See if someone makes a MIDI retrofit system for that model synth. Synchronization boxes that convert one type of timing code to another just might make two incompatible devices become close friends. Special pieces of hardware that interlock MIDI devices to audio tape can come in handy under special circumstances. If you've got a problem, you can rest assured that someone else has had the same problem before you, and that some company has designed "just what the doctor ordered" to fix it.

DRUM MACHINES: BACKGROUND

Drum machines have had a short but incredible evolution. In the beginning, there were little items called "beat boxes." They contained poorly synthesized (by today's standards) versions of drum and cymbal sounds, and performed a few basic beats. Push a certain button and you could select from all the winners, like Mambo, Cha-Cha, Rock, Swing, and Polka. Many organ manufacturers included these beat boxes along with their instrument so that someone could play the organ, and these little electric wonders would provide the background groove. I'm sure that you heard them in your local mall, providing the drum part for the salesperson playing *Lady of Spain* with colorglow keys and one-button chords.

The only problem with these early units is that they were not programmable. If you didn't like the Samba beat that came along with the machine, you were out of luck. These little machines found a lot of favor with amateur musicians who played for fun in their homes, and several professional musicians who worked as "singles" in nightclubs and bars.

In mid 1981, everything changed. Roger Linn created something called the *LM-1*. It was advertised as a drum computer, and leaped to world-wide attention when Elton John used it (programmed by Jeff Porcaro) for the hit recording of "Nobody Wins." Machines were picked up by Frank Zappa, Stevie Wonder, Paul McCartney, Paul Simon, Oscar Peterson, Larry Carlton, Harvey Mason, Jim Keltner, and several other influential performers and writers. The *LM-1* had a suggested retail price of $3,995.00, and caused quite a commotion in the A.F.M. (the American Federation of Musicians). For a while, many people thought that the *LM-1* was going to replace all human drummers. Unions tend to get upset when they fear that people will loose their jobs to technology.

This new drum computer let the programmer (the person who knew how to work one) come up with his own patterns instead of just using the ones that were programmed at the factory. It also incorporated some of the latest technology of the time to store real drum sounds in the machine instead of the synthesized sounds used in the beat boxes. Less than ten year later, machines are on the market that do everything the *LM-1* could do (and more) for a fraction of the price.

WHAT DRUM MACHINES ARE

If you own a drum machine, you have a complete electronic percussion system in your possession. All drum machines come complete with input devices (the little buttons you hit that play the sounds), sound generators (the internal circuits and chips that create the sounds you hear when you hit those buttons), a data-storage device (to store your rhythm patterns for future playback), and a playback system (if your drum machine comes with a headphone-out jack). Some even have signal processors built into them that perform some of the more basic audio tricks of the trade such as flange and delay. A few even have data processors that will change certain MIDI commands into other commands. In short, these little boxes are complete, self-contained electronic wonders.

The main function of any drum machine is to let the user create his own beats and save them into different memory locations (called **patterns**) inside the machine. While the early beat boxes let you play a few factory programmed beats just by pushing a button, the modern drum machine can store up to a hundred different patterns that are created by your mind's ear. If it's Guagancò, Reggae, or whatever the newest style happens to be, you can have it on your drum machine.

All drum machines also have the ability to combine the various patterns into **songs.** When the song function is used, beats (along with their variations) and fills are put together to perform a complete composition. No longer do you have to use the same pattern for every measure. The song feature gives today's drummer all the variation you need.

HOW DRUM MACHINES WORK

Inside every drum machine is an extremely sophisticated **sequencer**. The sequencer records whatever you play so that the machine can repeat it later. In order to reproduce your ideas, a sequencer must be able to play the correct sound at the correct time. A sequencer uses two vital pieces of equipment in order to do this: a clock and a counter.

All current drum machines contain a digital clock. This clock is like a metronome because it relates its speed to the speed of a quarter note. If the tempo is slow, the clock moves more slowly. If the tempo is faster, the clock moves faster. But, unlike the standard metronome, it "clicks" at a much faster rate. Instead of simply clicking once at the start of each quarter note, it subdivides each quarter into a number of smaller parts. Drum machines typically use clock rates of either 24, 48, 96, or even 192 divisions for each quarter. This corresponds to the traditional note values of 64th-note triplets, 128th-note triplets, 256th-note triplets, and 512th-note triplets. In other words, this clock can go much faster than you can play!

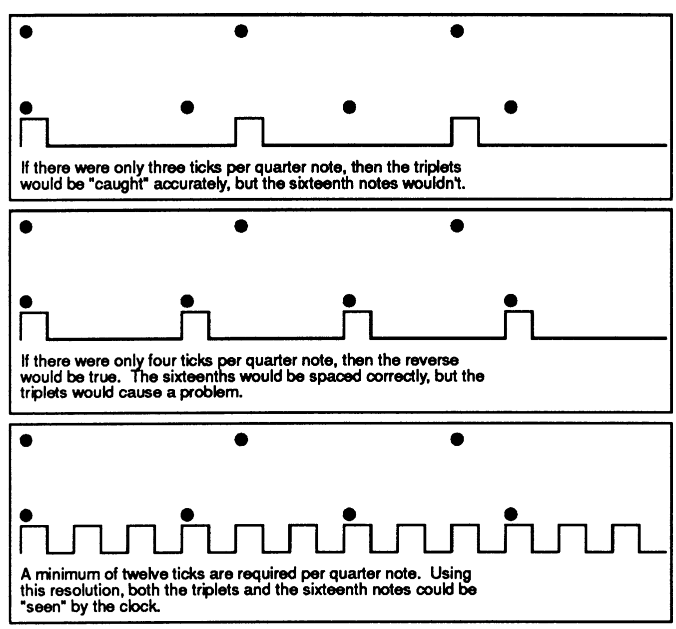
In addition to an accurate clock, drum machines also have a counter that keeps track of the clocks as they occur. The purpose of the counter is to measure the distance (in clocks) between one event and another. Let's say that you program a bass drum note on the first beat of a pattern and a snare drum note on beat two. If your drum machine's clock runs at 24 divisions (ticks) per quarter, then the bass drum note will occur on the first tick of the measure. The snare drum stroke will be on the twenty-fifth tick of the measure.

The microprocessor that acts as the sequencer's memory doesn't only record the message that the bass drum and snare drum played, it also records the number of clocks in between each message. When you have the sequencer play back the pattern, the microprocessor uses the same clock to determine when the sounds should be played. If you increase the tempo, you speed up the internal clock and the messages will happen faster. When you decrease the tempo, the clock slows down and there is more space between the clocks. The great thing about this is that the time *relationship* between the sounds is always the same, no matter what the tempo.

Why do we need clock rates that are so fast? Let's look at what would happen if you owned a drum machine that ran at only four ticks per quarter. You would be able to play as fast as 16th notes, but couldn't program 8th-note triplets. Remember that the clock is a digital clock. In other words, the digital clock is either at the first tick or the second tick, but it is never between them. In order to play a triplet, the clock must be able to divide the beat into three equal parts.

If a drum machine is expected to "capture" the placement of 8th-note triplets and 16th notes, the minimum resolution would be twelve ticks per quarter (see *Example 13*). If the clock can't move this fast, then the sequencer can't accurately record the exact timing of the message to play the sound. By increasing the speed of the clock, your performance can be more accurately recorded. Since most music doesn't use values faster than 32nd-note triplets, a clock speed of 24 ticks per quarter is usually sufficient, but higher clock rates can be preferable in some situations.

EXAMPLE 13



If there were only three ticks per quarter note, then the triplets would be "caught" accurately, but the sixteenth notes wouldn't.

If there were only four ticks per quarter note, then the reverse would be true. The sixteenths would be spaced correctly, but the triplets would cause a problem.

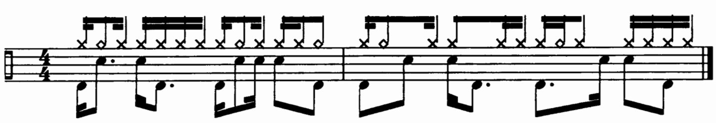
A minimum of twelve ticks are required per quarter note. Using this resolution, both the triplets and the sixteenth notes could be "seen" by the clock.

Moving up to pattern #20 (the one I will use for the transition into the chorus), I erase everything except the bass drum. Since the chorus is going to increase in intensity, I change the auto correct to quarter-note triplets and program a fairly funky fill. During the last count of the fill, I push the roll button and the snare drum button at the same time. By increasing the pressure on the snare drum's button, I build a nice smooth crescendo, which will lead into the chorus. I then change the auto correct back to 8th-note triplets and record a few well-placed notes in real-time. This additional rhythm creates a nice degree of tension.

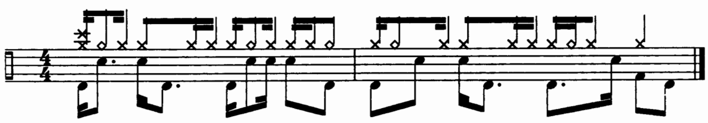
Pattern #20

Variations: Since I've got the same material in patterns four through eighteen, I spend the next several minutes making small changes to each of them. On some of them, I add additional bass drum strokes. On others, I add or remove some snare strokes, add a tom-tom note here and there, and vary the hi-hat a little bit. The purpose of this is to create some subtle changes in the basic pattern of the verse. Nobody wants to hear the same two measures repeated over and over, and these small changes help make the music flow more naturally and seem "alive."

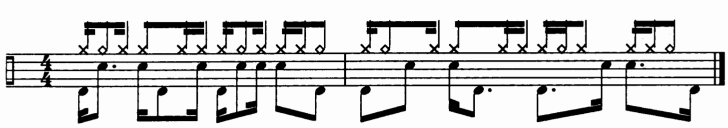
Pattern #7

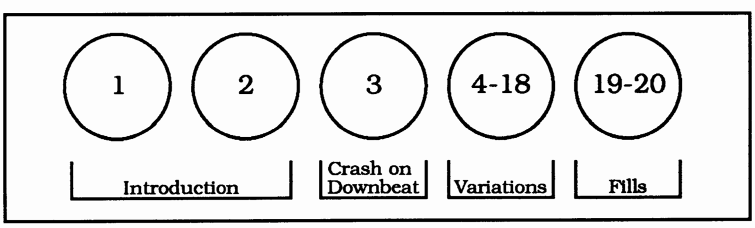


Pattern #12



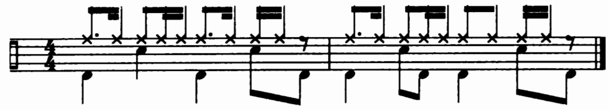
Pattern #16





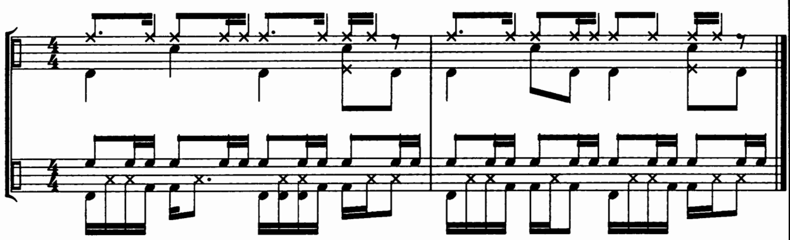
CREATING THE CHORUS

Basic Pattern: Now I'm ready for the chorus. Since the chorus changes grooves, I'm going to alter some things. First, I copy pattern #4 (my basic verse pattern) into pattern #21. I change the bass drum rhythm slightly, and erase all of the snare drum strokes except those that land on counts 2 and 4. The reason that I'm taking so many sounds out of the pattern is that I'm going to add others, and need to leave some space so that it doesn't sound too busy. Next, I hit a button called swap sounds. Pushing the button for the closed hi-hat and then the shaker, I've switched those two sounds in the pattern. Now I've got the shaker playing the same rhythm that the closed hi-hat played. Lastly, I erase all the open hi-hat strokes.



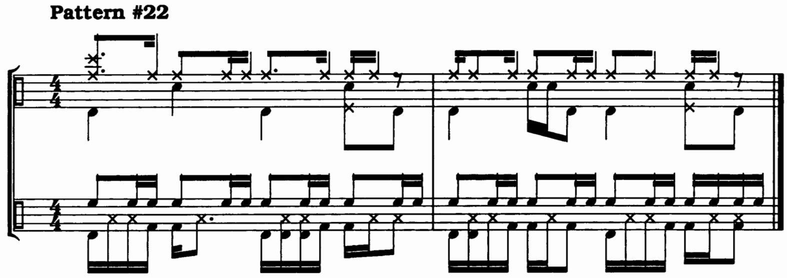
During the chorus, I'm going to add tambourine, handclaps, congas, and timbales.

Pattern #21

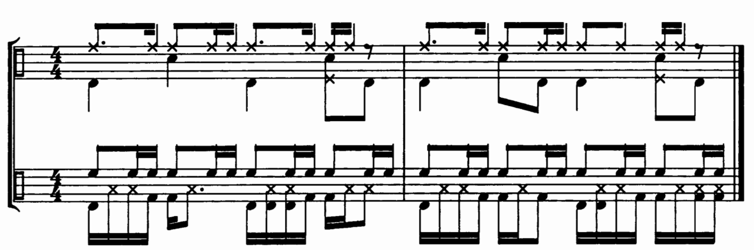


Variations: Copying pattern #21 into all the patterns from #22 to #40, I again make subtle changes in some of the voices: a cymbal crash here, some additional snare strokes there, a tom or two scattered about.

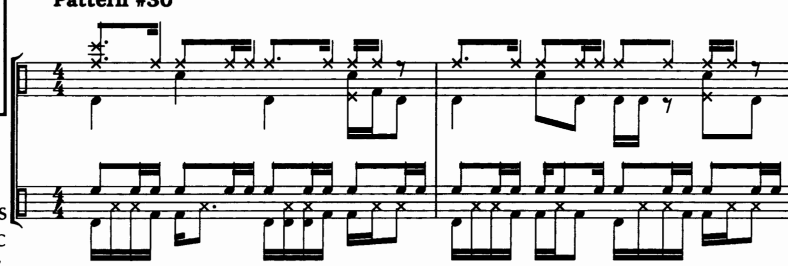
Pattern #22



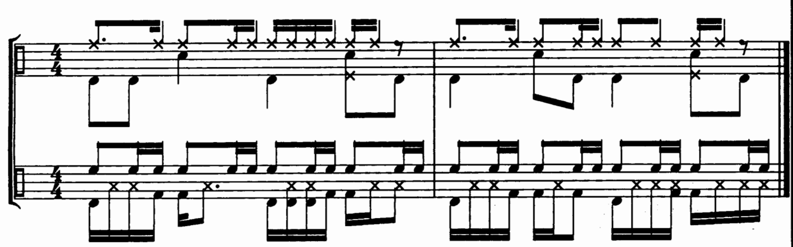
Pattern #25



Pattern #30



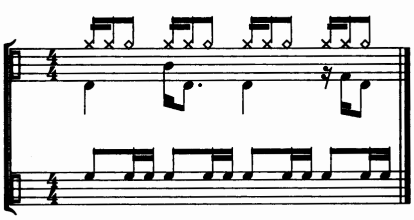
Pattern #34



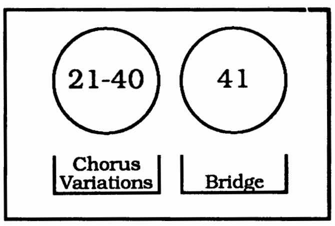
CREATING THE BRIDGE

For the instrumental bridge, I need something really special, so I'll add a few "special effects." I call up my fattest snare drum sound (I've named it the "Peterbuilt Truck"). I then activate the effects module, and I program a series of multi-strike echoes for this particular voice. I ask the machine to repeat this voice five times after the initial attack. Each repeat will be separated by 50 milliseconds, gradually fall in pitch (only about a half-step total change), decrease in volume, and bounce back and forth between the stereo field. This sounds so good to me that I decide to write the entire instrumental bridge around this particular sound.

Pattern #41



STEP 1



BUILDING THE SONG

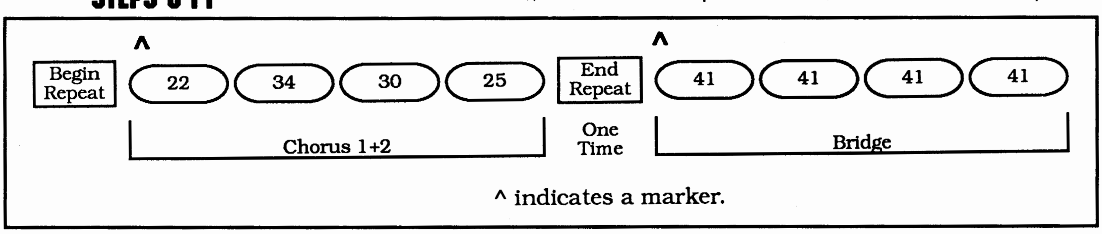
Step 1: Since I've programmed all the musical material for the individual parts of the song, it's finished. All that remains is to put the patterns together into a song. I activate the song mode on the drum machine and hit the record button.

Step 2: My first step is to set the initial tempo of the song. I'm not sure how fast it should go in numbers, so I use the tap tempo button to figure out the speed. I also ask the machine to average out the tempo of my last four taps. Just a few seconds later, the machine tells me that the average tempo was 112.5 beats per minute.

Step 3: The next steps go as follows: pattern #1 (first two bars of the song), pattern #2 (last two bars of the introduction—the one with the wide flam on the fourth beat), and pattern #3 (the basic groove for the first verse-also has a cymbal crash on the downbeat).

Step 4: Next, I select two more patterns from the ones I programmed with a few minor variations. I decide that pattern #16 and pattern #12 go pretty well together.

STEPS 8-11



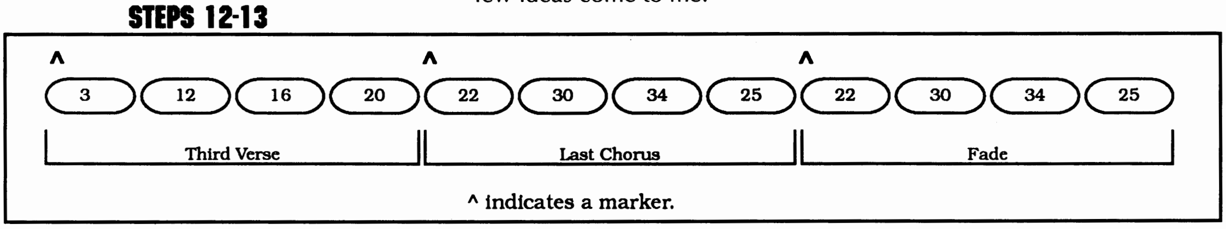
Step 5: Those three patterns (#3, #16, and #12), of two bars each, comprise the first six measures of the verse. My next step is pattern number #19 (the first fill pattern leading back to the second verse). I exit the song record mode and listen to to the results.

Step 6: For the second verse I begin with pattern #3 (the one with the crash). Since I'm going to have something pretty funky for the last two measures of this verse, I decide to pick only one of the variation patterns and play it twice. I find one that sounds good at this point, so my next two steps are pattern #7 and pattern #7. While I'm here at the second verse, I step back to pattern #3 and place a marker at this point, which is called "Verse 2."

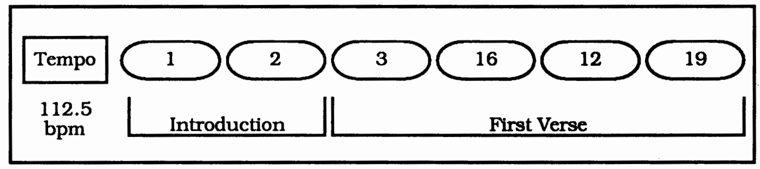
Step 7: Now its time for the fill that brings the second verse into the chorus. I program pattern #20 in the machine.

Step 8: The two choruses are going to be the same, so my next step is a command to begin a repeat. Again, while I'm here, I place another marker called "Chorus 1+2."

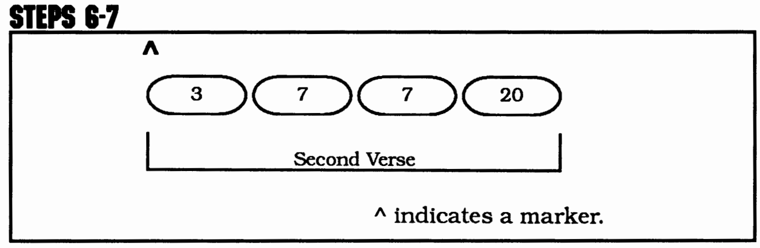
STEPS 12-13



STEPS 2-5



STEPS 6-7



Step 9: I've already programmed twenty slightly different versions of the basic chorus pattern, so I just pick four that I feel go together well. The steps for the chorus consist of pattern #22, #34, #30, and #25.

Step 10: Then, I move to the next step and press the repeat button. The machine is smart enough to know that this is an end-of-repeat command, and the display asks me how many times I want to repeat the material between the two commands. I enter "1" on the keypad.

Step 11: Now it's time to program the instrumental bridge. Another marker is placed at this point called "Bridge." Since this whole bridge is built around that snare drum effect, I program the next four steps separately: pattern #41, pattern #41, pattern #41, and pattern #41.

Step 12: We're all the way up to the last verse now (the marker is called "Verse 3"), and I choose pattern #3 (the one with the cymbal crash), pattern #12, pattern #16 (reversing these two patterns from the first verse), and the fill from pattern #20.

Step 13: The last two choruses will simply be repeats of the first chorus, but I again swap two of the patterns. The chorus consists of patterns #22, #30, #34, and #25. I plan to do something different with the last chorus, so I decide not to use the repeat command, and just enter those four pattern numbers again. Two markers have gone in. One at the beginning of the first #22, called "Last Chorus," and the other at the beginning of the second statement of pattern #22, called "Fade."

Step 14: My song is now complete. I exit the song record mode, and hit the play button. Sitting back and listening to the result a few times, a few ideas come to me.

CREATING ADDITIONAL VOICE ASSIGNMENTS

Now the song is complete, and it's time to assign the voices. First, I'll work with the stereo placement of each voice. The snare drum used for the bridge simply *must* come from the center of the stereo field. Since I've got 31 different positions (-15 to +15), I program that voice to a position of "0." That will make the echoes bouncing from side to side all the more effective. The bass drum is going to be placed at the -1 position, and the snare voice used for the majority of the song will be placed at +1. Other sounds are distributed accordingly around the field: tambourine +8, timbales +4, hi-hats -6, handclaps -12, etc.

Next comes assigning the sounds to different output channels. Four of the drum machine's channels have filters on them. These filters cut out a certain amount of the high frequencies, darkening the sounds of the voices that go through them. I've decided that I want to filter the bass drum, tom, and snare drum sounds (the "truck" is already dark enough!). I assign only those voices to the filtered channels.

CREATING THE MIX

Now to the drum machine's internal mixer. I play pattern #3 (the basic pattern for the verse), and as I'm listening, use a series of sliders to adjust the balance of each individual voice. By being able to mix each voice, I can achieve just the balance I'm looking for: bass drum very hot, hi-hat cymbals back a little bit further, etc. I define the end result as "Mix #1."

I then do a separate mix for the chorus. Since I've added several instruments, I need to adjust their balance, too. How "up front" do I want the timbales, the handclaps, the tambourine, etc.? I define this mix as "Mix #2."

I've got other plans in mind for the mix during the instrumental bridge. After getting all the various voices adjusted to their proper proportion, I define this as "Mix #3." Taking only the slider that controls the "Peterbuilt," I move it down just a little and call it "Mix #4." Move it down a little more and define that as "Mix #5." Still softer, "Mix #6." Now I've got four different mixes that are basically the same thing, except that fat snare drum sound that pans back and forth is also getting softer in each subsequent mix.

ADDING THE MIX

Step 1: Going back into song record mode, I make sure I'm at the first step and hit the insert button. I then hit another button called "change mix" (defined earlier) and tell the machine to use "Mix #1" as the first step of the song.

Step 2: Skipping over the next marker to the one called "Chorus 1+2," I insert another mix change (Mix #2).

Step 3: Jumping to the marker called "Bridge," I insert mixes #3, #4, #5, and #6, just before each one of the statements of pattern #41. Not only does the "Peterbuilt" snare have this great echo effect bouncing around the stereo field, but the drum fades further and further into the background of the mix each time the measure is repeated.

Step 4: Next I insert the proper mix-change commands before "Verse 3" and before the "Last Chorus."

Step 5: Moving on to the marker called "Fade," I insert a command for a volume change before the last statement of pattern #22. The machine's display asks me how much louder or softer I want the overall volume to change. I enter that the volume should get softer by an amount of 25%. Next, the display asks me if the volume change is supposed to be a smooth change or an instant change. After entering that I desire a smooth change, the display asks me: "Over how many beats?" I enter "8" on the keypad. This creates a smooth diminuendo over the first two measures of the fade out.

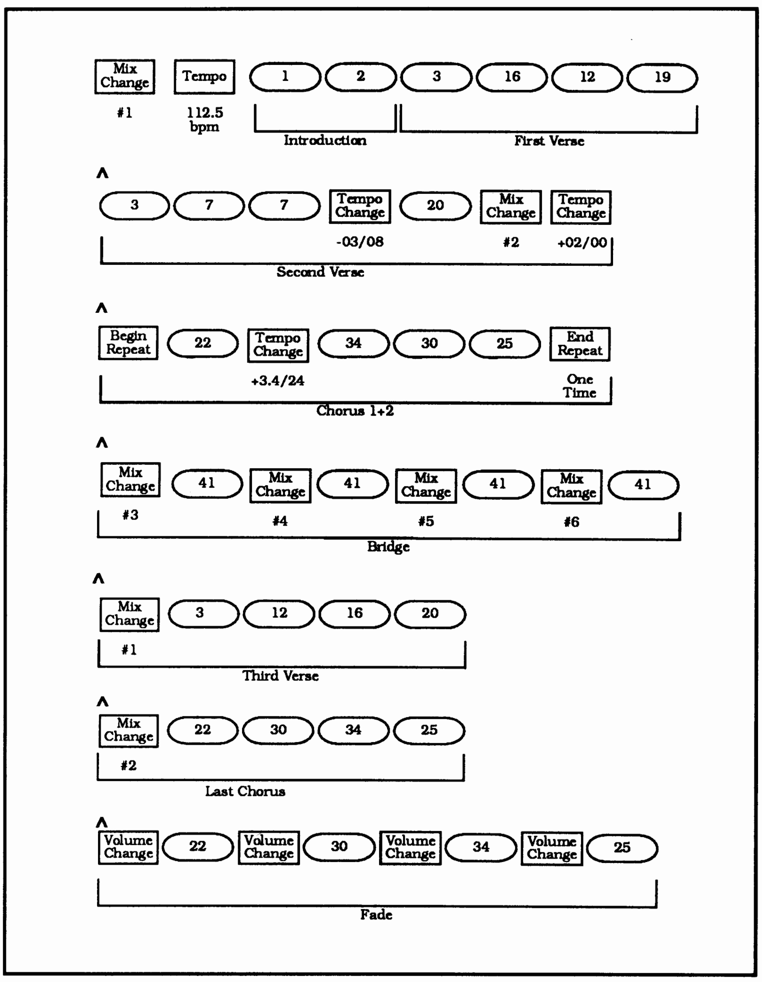
Step 6: I follow the exact same procedure before patterns #30, #34, and #25. Now I've created a perfectly smooth fade-out that spans the length of an entire chorus, and changes from the starting volume to no sound at all. With all my mixes inserted into the proper spot, I listen to the song a few times. It's not bad, but it's not quite there yet. I've got a few ideas that may help.

ADDING TEMPO CHANGES

Moving to the marker called "Chorus 1+2," I move back two steps. Now, I'm at the beginning of pattern #20 (the one with the quarter-note triplets and snare drum roll), and I hit the insert button again. This time, I hit another button called "tempo change." The display asks me if I want to go faster or slower. I key in the command for slower and the display asks me another question: "Change and Rate?" I key in the numbers "03" and "08." By doing this, I've told the drum machine that I want it to slow down by three beats per minute over a span of eight beats. Then the display asks another question: "Smooth or Curvature?" I key in the command for using a curvature, then the amount of "+66." By using a positive curvature, the machine will slow down a little more at first (making the quarter-note triplets seem a little more "stretched"), and less toward the end.

Next I move ahead a step to the beginning of the repeat command and insert another tempo change. As I said earlier, the chorus has a higher intensity than the verses, so I'm going to have my "machine drummer" get a little excited. This time, I ask the machine to insert a tempo change that gets two beats per minute faster over "0" beats. The zero rate means that the machine will change tempos instantly.

The end result is that the chorus is still one beat per minute slower than the verse. I step ahead to the next pattern (measures three and four of the first verse) and insert another tempo change. This time the change is an accelerando of 3.4 beats per minute over the span of 24 beats (the remainder of the first chorus). Since the second chorus will simply repeat all of the steps that comprise the first chorus, the accelerando is repeated, too. My tempo at the end of the choruses is 118.3 bpm, slightly faster than the initial tempo of the song. Since that tempo change isn't so drastic, and the energy level is still up for the instrumental bridge, I decide not to take the tempo back down for the third verse.



ADDING SOME TRICKS

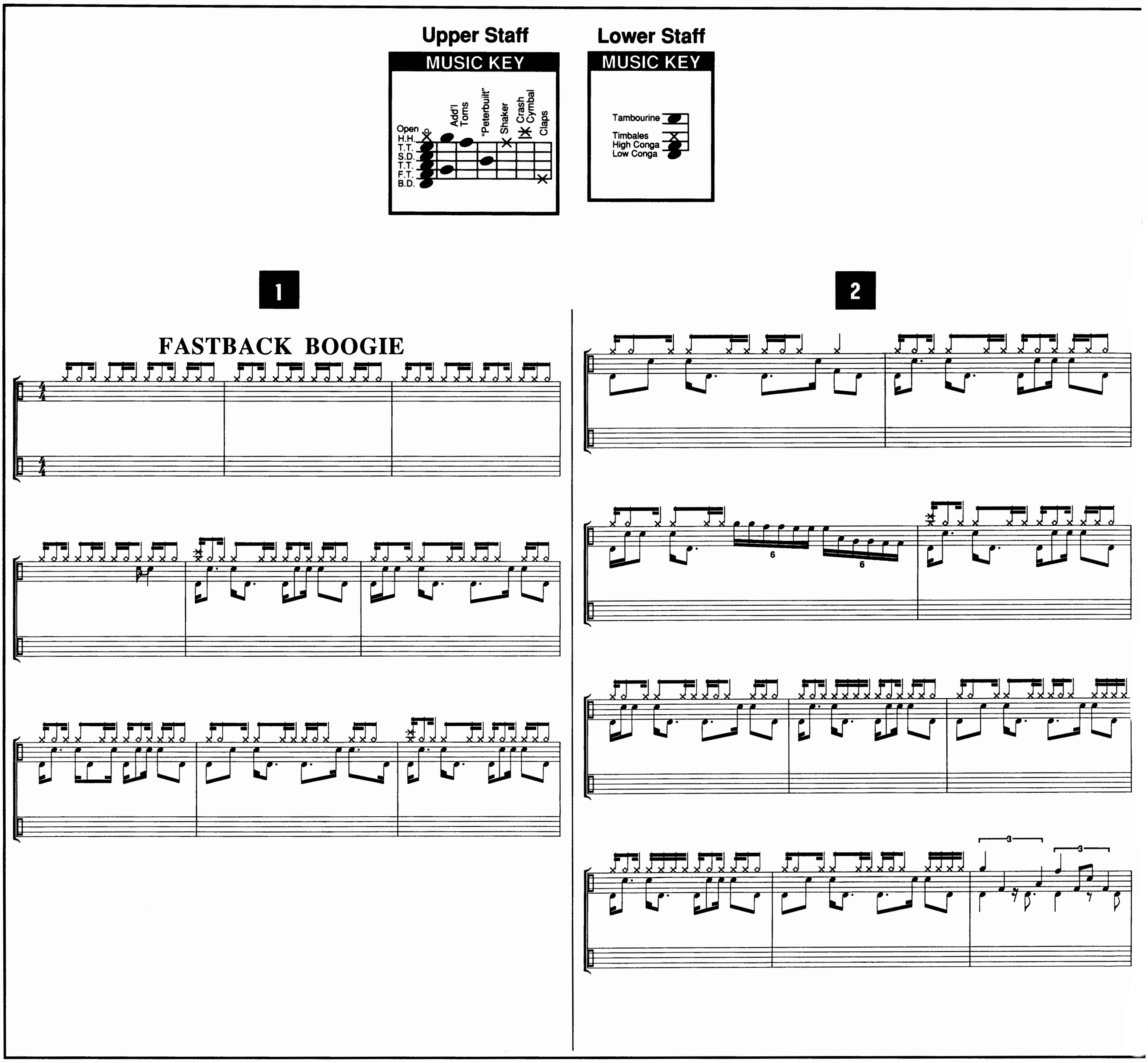
There is one more thing that I want to do to the chorus to get it "just right." Going back to the voice edit mode, I select the handclap and send it to the effects processor. In order to make it sound like more people clapping, I send it through the multi-tap processor and program eight repeats, each one being only 6 milliseconds apart. In all, I've now got nine people clapping their hands, and the claps are 63 milliseconds "thick." This sounds a little better.

Returning to pattern mode, I call up pattern #22 (the first pattern of the chorus) and punch a button called "time shift." The display asks me which voices I want to shift, and I tap the handclap instrument button. The display then asks me if I want to shift forward or backward in time. I select that I want to shift the handclap backward in time by two-96ths of a beat. The result is that the handclaps "come early and stay late."

In order that the entire chorus has the same feel, I time shift the handclap in the other three patterns that make up the chorus (#34, #30, and #25). Returning to song mode, I hit the play button and listen to the overall result.

FINISHING OFF

Calling up the command "name song," I enter "Fastback Boogie," and all is finished. I save the songs, segments, and edited voices onto the built-in disk drive. This way, whenever I want to play it again, I only have to enter a command that says "Load All." The drum machine loads everything I need for the song from the disk to its internal memory: instrument-button assignments, mixes, levels, pan settings, output assignments, as well as all the pattern and song information.



higher levels of force. See *Example 27* for a graphic description of this feature.

Dynamic Bend — Some converters let the drummer alter the amount of pitch bend depending on the force of the stroke. When you hit the pad lightly, the pitch won't bend as far as when you strike harder. When struck at the most forceful level, the full value of the pitch-bend range is heard. In many ways, this is similar to what happens when playing acoustic drums.

Layer/Touch Control — Some converters are designed so that you can adjust the different note numbers on a layered pad to "kick in" as you strike the pad harder. Using the five sounds that we had before, if you hit the pad very softly, you would only hear the rimshot. Hit it a little harder and the first bass drum comes in. When you give the pad your most forceful stroke, all five sounds would be heard.

Dynamic Note Shift — This feature is really hip! As you increase the force of your strokes, different MIDI note numbers are sent out from the converter. This is much like the first analog electronic drums, where the force of the stroke determined the voltage that was sent to the oscillator that controlled the pitch. This feature makes it possible to play a complete melody on a single pad. Or, if you connect the converter to a drum machine, it lets you control many different voices from one pad. The drawback of this feature is the amount of control required by the performer. If a single pad is going to fire as many as twelve different sounds depending on the velocity of the stroke, then you are certainly going to need a great deal of dynamic control in order to get the sounds you want at the proper time.

ADDITIONAL FEATURES

Memory — Like many other devices, external converters can save their patch information off to memory cards, cartridges, or cassette tape. By saving your patches, you can free up other patch locations. Some converters can expand their patch memory by accessing a memory card or cartridge along with their RAM. If a unit can store 64 patches internally, and access another 64 on a RAM cartridge, there would be a total of 128 different patches available on the machine at once.

Chains — Remember how drum machines used the chain command to play songs in a specific order? Well, the chain feature on a converter calls up a certain order of patches. Chain one might contain 20 steps, each step being a different patch (#44, #74, #14, etc.). You can quickly move from patch to patch by moving forward through the steps of the chain. This is a very valuable feature for live performance.

Footswitches — Like the footswitches for drum machines, these are optional and rarely ever come with the converter when you buy it. Most often, a footswitch is used to step forward or backward through patches and chains (if the converter supports them). Some converters use footpedals to control pitch-bend changes, modulation changes, and sustain. The pitch bend and modulation pedals are exactly like the ones found on a keyboard synth, except they're moved by your foot instead of your hand. The sustain pedal operates the same way as on a standard keyboard synth: push the pedal down and the sound sustains; let up on the pedal and the sound stops.

Using footpedals to their fullest advantage requires some different thinking. Since pedals force you to stop playing the bass drum and/or hi-hat with your feet, you have to make some adjustment. You may want to have a pad always assigned to a bass drum voice so that your hands can take over while your foot is busy controlling the pedals.

Two MIDI-Outs — If your converter has two MIDI-Out jacks in the back, you might be able to program a pad so that it sends different information out each jack. As an example, you may have one pad sending MIDI note number 55 over channel three from one MIDI-Out port, and note number 87 over MIDI channel six from the other port. If you are using the converter to control many different MIDI sound generators having two outs can increase your flexibility.

INTERNAL CONVERTERS IN "BRAINS"

If you buy an electronic drumset as a total package, it will consist of the pads and the brain. Of course, the pads are what you hit. The brain contains all of the necessary hardware and software required to read the voltage signals from the pads, convert those signals into MIDI data, *and* produce the sounds.

Because these brains produce their own drum sounds, they are less dependant upon external MIDI devices. For this reason, their MIDI capabilities are generally less expansive than a pure pad-to-MIDI interface. Aside from all the bells and whistles that the machines may have to create and alter their internal sounds, most of their MIDI controls are pretty basic. You should be able to control the pad's MIDI channel, its MIDI note number, and its gate time. You might be able to assign a program-change message to be sent over the MIDI cable when you call up a particular patch. But that's about all.

As an example, some of the most sophisticated brains can assign a pedal to control pitch-bend range or sustain, but these commands are not sent over MIDI. Instead, the pedals only control the internal sounds included in the unit. The Roland *DDR-30* doesn't send any MIDI data other than note numbers and MIDI channel assignment. It will send program-change messages, but only the same program number as the patch (when you call up patch #17, you send program change #17).

An exception to simple MIDI implementation in drum brains is the Korg *DRM1*. This rack-mountable unit can send and receive more detailed MIDI instructions. However, most of the expanded MIDI features are due to an internal sequencer. Since it is possible to use the *DRM1* as a type of drum machine, it sends and receives all of the System Real-Time Messages. It will also let you set the gate time of the MIDI note to a different value than the gate time of the internal sounds.

In general, most drum brains are designed to be the sound-producing device when used in conjunction with pads. Most brains do what they were designed to do very well: They all produce great drum sounds. But they should not be considered as full-blown MIDI "control centers" for an elaborate system. Don't get me wrong; they are extremely useful. But if you are looking for a great deal of MIDI control from your pads, you may be better off considering a dedicated converter.

WORKING WITH MEMORY

I'm having trouble getting my interface to memorize changes that I program into my patches.

There may be a problem in your machine, but the problem could be "operator malfunction." I can't tell you how many times I thought one of my machines was acting up, just to finally realize I forgot one simple step or neglected to push the right button at the right time.

RAM memory, since it is not burned into the chips inside the machine, is a temporary memory. Whenever you turn your unit off, all the RAM goes to "Digital Heaven." If this happened every time you flipped the switch, it would be a gigantic pain in the neck. Almost every electronic instrument uses some sort of battery to keep power going to the RAM even though the machine is turned off or unplugged.

Wherever you make changes (of any nature) to a patch, the patch is instantly moved to another memory location called the Edit Buffer. The changes that you're making are not changes to the patch, but instead are changes in the edit buffer's patch. To say this another way, if your converter has 64 patches, then there are really 65. The extra patch is the buffer. If you neglect to save your changes from the buffer to your patch, all will be forgotten when you either turn the machine off or make edits in any other patch.

I suggest that you follow a simple procedure as if it were a religion. Anytime you complete an edit, save it. You might even want to consider saving anytime you make a partial edit. If you only change one thing, save it! If you are certain that you are saving all your changes, and you still feel that the machine is not remembering your patches correctly, take it in for service. You might only need a new battery.

By now, you are aware that the sounds you actually hear are produced by sound-generating hardware, not the triggers or the MIDI interface. In order to look at the different possibilities that are available, we will examine three different configurations.

ALL-IN-ONE UNITS

The original Syndrum and Synare were the ancestors of these instruments. By "all in one," I mean that the pad and the sound generating system are all included in the same physical instrument. Usually, since they only contain a single pad, they do not have any MIDI capability. You simply hit the playing surface, and the audio signal comes out the back of the pad, ready to amplify and send to speakers.

The Simmons *SDS1* was a good example of an all-in-one unit. This drum had a single playing surface and reproduced a single digital recording of an acoustic sound, which was stored in a PROM chip. **PROM** stands for Programmable Read Only Memory, and is a chip with a sound "burned" into its memory. Whenever you struck the *SDS1*, its internal circuitry would read the digital recording from the PROM and convert it into analog voltages that could be sent to an amplifier.

The *SDS1* could read PROMs of two different memory sizes, 8k and 16k. The smaller chip stored 8,000 bytes of information and was used for short sounds like a snare or cowbell. The other chip could keep twice as much information in memory and was used for longer sounds like toms and cymbals. Whenever you wanted a new sound on the *SDS1*, you would go to your local dealer and buy a new PROM chip, plug it in, and go.

PROMs had a retail price between $29 and $38 each, but you could buy blank E-PROM chips from Simmons for about $14. An **E-PROM** is an erasable PROM. Once you recorded your own sample onto the chip, you could use it for a while, erase it, and then sample another sound. Simmons made a unit called an E-PROM Blower. The *EPB* could sample any sound and then burn the digital data into the chip. The drummer then took the chip and placed it in the *SDS1*. Even though this was fairly expensive, it was a bargain at the time (1985). If you wanted a large library of sounds at your disposal, the *EPB* along with some blank E-PROMS was the way to go.

Once the desired sound was inserted into the *SDS1*'s socket, the drummer had a fair amount of control over it. Controls included a sensitivity and volume knob, a switch to tell the machine which size PROM you were using (8 or 16k), pitch control, pitch-bend range (either bending up or down and also affected by the force of the strike), and a run generator.

A **run generator** is an internal timer that progressively lowers the pitch of the sound over an adjustable time span. The speed that the pitch lowers is adjusted by the run time (from a fraction of a second to several seconds) and the overall bend amount is adjusted by a another control. The run generator can make a single pad seem like several because each subsequent stroke will have a lower pitch. Another nice effect can be achieved by setting the run time as fast as possible. Whenever you strike the pad, the pitch will be determined by a certain degree of randomness.

The trend for today's all-in-one units is slightly different. Instead of having replaceable chips, the samples stay inside the unit. Because memory is getting less and less expensive, some chips may be able to store up to six different sounds, such as two bass drums, tom, chimes, gong, and steel drum. These little pads can be placed anywhere around a kit, making it easy to incorporate them into an existing setup. If you're looking for a few special effects, or maybe just want to add several new sounds to your kit, these pads might be the way to go. They offer high sound quality for minimum dollars.

DRUMSET BRAINS

There are many good electronic drum brains on the market. All come complete with the required inputs and outputs for standard operation. Some have only stereo outs and some have individual outs, some read voltage levels from six pads and some read twelve pads. They all perform a similar function: see the voltage, make a sound.

A few brains use an analog style of sound creation, but most currently incorporate PCM samples for highly realistic drum and cymbal sounds. Along with editing the pitch, duration, envelope, stereo placement, bend range and rate, and maybe adding an LFO or two, most drum brains include other options to enhance the timbre. Typical options for sampled sounds are covered under the "Creating Voices" section of the "Drum Machine" chapter. It now seems like a good time to include additional voicing features that show up in some analog units.

TYPICAL ANALOG CONTROLS

Noise — While this term may have negative connotations to some people, noise refers to "white noise." White noise is a sound that includes all of the frequencies perceptible to the human ear. A brain that includes noise as part of its sound has to have special noise-generation circuitry. Because white noise contains so many overtones, no particular pitch or fundamental can be distinguished. Adding a certain amount of white noise can make a drum sound more realistic, especially if it is used during the attack portion of the envelope or if it is used to create a snare drum sound.

Noise/Tone — This control is used to mix and balance the amount of amplitude between normal sound of the drum and the noise generator.

Click — Click is a special classification of tone that is designed to imitate the sound of a stick striking the surface of a drumhead. When you add click to a sound, you add a degree of "punch" to the attack. Some units, instead of *adding* the click, *replace* the beginning portion of the normal tone with the click sound.

Attack — Some brains let the drummer program a very short amount of white noise right at the beginning of a sound. In this regard, it is a little like the click control. This is another way to create more realistic sound.

Filter Pitch — This knob changes the cutoff frequency of a filter. When this control is set to a low value, only the lower frequencies will be heard. As the cutoff frequency increases, more of the higher overtones and harmonics are included in the sound.

Filter Resonance — By increasing or decreasing the filter resonance, you can add or subtract higher harmonics and overtones to the pitch. It can be used to make the sound appear brighter or darker.

Filter Bend — This control can be used to "sweep" the filter cutoff frequency up to or down from a designated level. This is a very nice effect, as the tonal quality of the drum changes over a period of time.

Emphasis — On some brains, this term is used to add a low-end boost to the sound. It can be used to "fatten up" bass drum and tom sounds.

MIDI REAL TIME CONTROLS

Since most brains support the MIDI standard, new ways to use pads as controllers are surfacing all the time. The goal is to supply the drummer with all the creative real-time-performance control that a keyboard player enjoys. Toward this end, additional MIDI features are being implemented in a variety of ways. Since drummers can't use aftertouch to alter the amount of pressure after the stick hits the pad, and additional pedals are a pain to deal with, velocity is being used as the "ultimate controller."

Velocity Switch — If a brain supports velocity switching, one sampled sound can be assigned to any dynamic below a certain level, with a different sample assigned to any dynamic above a certain level. Let's say that you've assigned one pad to play a floor tom at a velocity reading of 64 or less, and a high tom at any velocity over 64. You can trigger two different sounds from the same pad, adding a lot more versatility to your pads. If your pad has four triggering surfaces, it can trigger up to eight different sounds.

Velocity Assignment — Instead of having the velocity indicating only changes in dynamics, a few brains let the drummer assign or "map" the velocity reading to other musical parameters. Mapping velocity to pitch allows you to make a single pad playing, say, a tom sample sound like five or ten toms. You might decide to map velocity to stereo placement. When you play softly, the sampled sound comes from the right channel; when you increase the stroke's force, the sound will move to the left channel. Mapping velocity to decay can change the sound from long to short while you are performing.

The most intriguing feature of velocity mapping is when two or more maps are combined at the same time. Velocity might control the switching of samples, dynamics, stereo placement, duration, *and* pitch all at once. Imagine what you can do with only a few pads that are assigned in this way!

SAMPLERS

The most accurate representation of acoustic drum sounds is achieved through sampling. I've mentioned sampling several times during the course of this book, and now its time to take a look at what it is and how it works.

Actually, sampling is a magic trick for your ears. You know that when you watch a movie, there really aren't little people on the screen moving around and talking to you. Movies are the result of individual photographs flashing across the screen. If the photos follow each other in close succession, then your eyes are fooled into thinking that there is real, smooth, continuous movement.

When a sound is sampled, the wave's shape is "drawn" as individual pictures that flash in front of your ears in quick succession. But, get this: Film runs at only twenty-four frames per second, while sampled sounds can run up to 44,100 (or more) "samples" per second. Your ears are much harder to fool than your eyes!

Sampling is the result of PCM technology. **PCM** stands for Pulse Code Modulation, and has its original roots in C.E. Shannon's article "A Mathematical Theory of Communication" back in 1948. Not too long ago, a musician would have had to ask the defense department to borrow their 20-ton computer system to take to the gig. It wasn't until recently that these same principals and theories could be implemented into musical instrument systems. In a nutshell, sampling takes the acoustic soundwave and turns it into digital numbers.

Digital information can be transmitted with much more clarity and precision than analog information. In an analog system (whether it's a TV signal, radio broadcast, or people to people communication), information is subject to interference and distortion. If you have a snowy picture on your TV, you've got interference. If you're trying to talk to your friend in a noisy bar, you've got interference. Whenever there is interference, there is the chance that it will be strong enough to destroy the communication and information. We already know how to deal with digital style information. If you've ever played the game of twenty questions (where you answer questions with yes or no), you've dealt with digital information. Digital signals are either off or on, black or white, yes or no. There is much less chance of distortion in the signal, because there is no ambiguous grey area.

Sampling follows a few simple steps. (The steps are quite simple; the technology to pull them off isn't.) First, the original soundwave is sampled by a fixed frequency. If the sampling frequency is 20 kHz, then the original signal will be "looked at" 20,000 times every second. If the sampling frequency is 44.1 kHz, then the signal will be read 44,100 times per second. At each one of the sampling cycles, the original signal's level is measured as higher or lower voltages. The speed at which the original sound is sampled is called the **sample rate**.

Higher sample rates are preferable to lower sample rates. There is a rule called the "Nyquist theorem" which states that there must be at least two samples of any sound in order to get an accurate representation of the wave. At the very least, you must have a reading during the positive (compression) and the negative (rarefaction) portion of the waveform. This means that if you want to sample a waveform that is 4 kHz, you must have a sample rate of 8,000 samples per second. While this may be a high enough rate to capture the fundamental frequency of the wave, what about all the harmonics and overtones? They will not be accurately measured. Since human hearing goes up to about 20 kHz, sampling rates of 40 kHz or faster are not uncommon. As a point of reference, the standard audio compact disc uses a sample rate of 44.1 kHz.

Second, these levels are converted into the binary number system. All samplers, because they deal with computers, must use binary digits to represent values. Inside each sampler is an A/D converter (analog to digital) that assigns each of the levels to binary numbers. The maximum number of bits available to the microprocessor is going to determine how many different numbers can be assigned to the voltage levels.

Let's say that a machine used a two-bit microprocessor to convert voltages. Since two bits can only represent four different values, all of the continuously varying voltages would have to be assigned one of these four numbers. This wouldn't give you a very accurate representation of the wave. If a voltage level falls between any two digital numbers, it has to be rounded off (digital numbers can't handle fractions). As the number of bits increases, so does the **sample resolution**. Resolution refers to the maximum number of available levels that a microprocessor can assign to the voltages. An eight-bit microprocessor could assign any of 256 different numbers to the voltage, while a twelve-bit machine has 4096 possible values. The highest resolution samplers on the market today (that musicians can afford) use sixteen-bit microprocessors. Sixteen bits result in 65,536 possible values. As a general rule, each additional bit of information adds approximately 6db to the signal-to-noise ratio. This would mean that a twelve-bit A/D converter would sound about 24db cleaner than an eight-bit converter.

Examples 28 through *30* show how the sample rate and resolution work together to measure the original sound. In *Example 28A*, the sample resolution is three-bit. This means that there are only eight different levels that can be assigned to the wave. The sample rate is twenty samples for one complete cycle. If the frequency of the original wave were 200 Hz, then the sample rate in this example would be 4,000 samples per second. *Example 28B* shows how the wave would look translated into digital values. You can easily see that the resulting waveform would not be an accurate representation of the original. In *Example 29*, the sampling rate has been increased to forty times per wave. This time, if the original was 200 Hz, then the sample rate would be 8,000 Hz. The sample resolution has also been increased to four-bits, giving sixteen possible levels. *Example 30* uses a five-bit resolution (32 levels), with a sampling frequency of sixty times per wave (12,000 Hz). It is easy to see that as the sampling rate and resolution increases, digital representations can more closely duplicate the original waveform.

After the sound has been stored as digital information, it can be played back by using a converter going in the opposite direction (D/A or digital to analog). The bits and bytes are converted back into voltage levels, and their sound image comes out the speakers.

Are you hearing the "actual sound" of the sampled instrument? Not really. Samplers are just fancy digital versions of tape recorders. They do sound very good, and better samplers have the same sound quality as a compact disc. But there are some problems with samplers. When you listen to an acoustic instrument, such as a trombone, each note is slightly different. Pitches in the low register have a different timbre than pitches in a higher register. If you want to get a great trombone sound from a sampler, do you sample every pitch in the instrument's range? All samplers have a finite amount of memory for storing the digital information, and you will likely run out of memory before you run out of notes. The solution is called multi-sampling.

It is possible to take one sample of a sound, and spread it across a large range of notes. When you do this, however, the digital numbers that make up the waveform are played back slower or faster in order to change the wave's frequency. But it's a lot like changing the speeds of a record player. As the sample moves up in pitch, it becomes a little nasal sounding as it "plays through" its digital information more quickly. When the pitch is spread too far down, it begins to sound like a record player after the power plug is pulled out of the wall. When you use multi-sampling, you sample several different pitches in the instrument's range. This way, each sample covers only a few notes, and the timbre changes will be more realistic.

Samplers are great for drummers! Percussion sounds are usually pretty short and don't require massive amounts of memory for storage. It's quite possible to place a different percussion sample under each note of a sampler (called "splitting" the keyboard) without running out of memory. Using a multi-pad or a keyboard percussion controller, you can control up to sixty different drum sounds at once.

There is another reason why I think that samplers are well suited for drummers: Drum timbres are not as standardized as those of other instruments. Just about everyone is familiar with the sound of a piano, a trumpet, or a flute. People know how these instruments are supposed to sound, and can hear the differences when samplers spread their sounds too far. ("That trumpet just doesn't sound right.") Hardly anyone knows just how a cowbell, snare drum, or floor tom should sound. A floor tom's sound will vary depending on its size, construction materials, head type, tuning, etc. A floor tom sample will generally sound good even if it is brought up to the range of a mounted tom, or down to the pitch of a bass drum. By sampling a single temple-block stroke, you can get many instruments, from log drums to claves.

If you are considering a sampler for your system, look for one with a fast sample rate of at least 30,000 or more samples per second, a high sample resolution (at least 12-bit), and as much memory as you can afford. In addition, since you will be working with percussion sounds, you want a sampler that will let you assign an unlimited amount of splits (or at least fifty). Samplers with the same rates and resolutions may still sound different. How good are the A/D and D/A converters? How useful are the filters that the sampler incorporates? When buying any piece of gear, your ears should be the final judge—not the spec sheet.

SYNTHS

With MIDI, drummers can use any type of sound source as their sound-generating hardware. Synths of all types are just as accessible to you as they are to keyboard players. In fact, you can plug right into a keyboard player's synth. When you're tired of having your drums play standard percussion sounds, you can plug them into any type of synth and play any sound you have the programming chops to create. Any type of synthesis is at your disposal, and all those great sounds you hear on the radio, recordings, and movie scores can be fired by your drum pads.

How would you like to have your snare drum sound like the laser gun used in the latest space saga? Would you be interested in your bass drum pad sounding like a squadron of helicopters descending on a herd of mutant gerbils? How about using your tom pads to double the melody of the guitar player, but playing timbres that are something between a trombone and an electronic music box?

If you plan to compose by using MIDI messages and a sequencer, you are going to need a few melodic sounds in addition to drums. If you're looking to expand your tonal pallet, think about picking up a MIDI synth. They often provide a new world of sounds for a reasonable price.

With the world of MIDI at your disposal, any type of synthesis that is currently available is under your control. What follows is a brief explanation of some of the most popular synthesis techniques.

Additive Synthesis

Additive synthesis is the easiest to understand, and at the same time, the most difficult to master. Additive synthesizers contain several different oscillators that are mixed together in order to construct the final sound, just as a painter might create his colors by adding and mixing different pigments. Additive synths use sine waves (the most pure wave—no additional harmonics) to create their sounds. If you've ever played with the drawbars of a Hammond organ, you should have a pretty good idea of the additive technique.

Back in the days of analog machines, true additive synths would have cost an arm and a leg. Chaining multiple oscillators together didn't come cheap! With digital electronics, the sound generators now used for additive machines are software oscillators. A software oscillator doesn't really exist. Instead, the microprocessor inside the synth uses an extremely complex mathematical formula to build the resulting waveform. It is as if the microprocessor said to itself: "What would a wave look like if it was the result of combining this many oscillators tuned to these specific frequencies?"

Additive synths can be very powerful, allowing the programmer to create an untold variety of sounds. But, creating a desired sound from scratch requires that you have a good understanding of how sounds work. If you want to create some new, weird, bizarre timbre that has never been heard before, you can just grab the synth and start goofing around. If instead, you are trying to closely imitate the sound of an English Horn, and you don't know exactly how that instrument sounds, good luck!

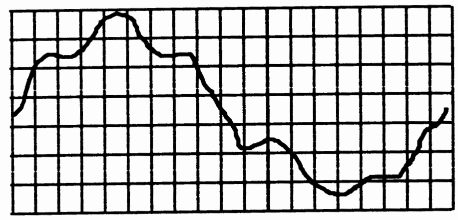
Subtractive Synthesis

Found on Roland modular synths, older Oberheim instruments, and the popular *Prophet 5* by Sequential Circuits, subtractive synthesis (as you might expect) is the reverse of additive synthesis. Instead of combining several sine waves to create a complex waveform, subtractive synths begin with a complex wave (such as square, sawtooth, or noise) and filter out certain harmonics and overtones to create other complex waves. Most often, these filters are controlled by some sort of envelope, which would sweep the filter's frequency over a period of time.

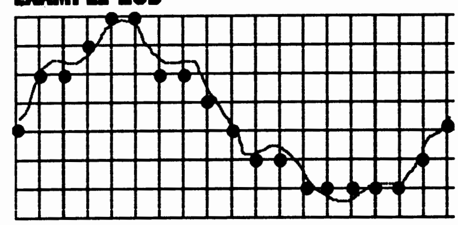
FM Synthesis

FM (frequency modulation) synthesis took the world by storm when it was incorporated into the Yamaha DX7. The DX7 became the most popular synthesizer ever to be produced.

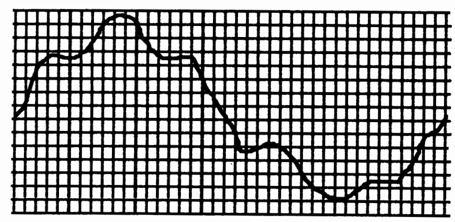
EXAMPLE 28A



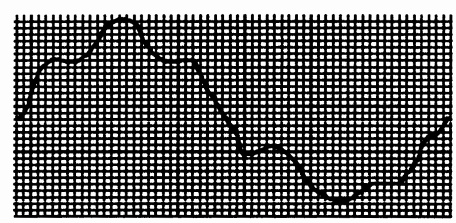
EXAMPLE 28B



EXAMPLE 29



EXAMPLE 30



Yamaha has made FM synthesis their trademark sound. As you get more involved with electronic drums, you're going to run into one of these instruments (if you haven't already).

Perhaps you have heard some bad rap about FM synths. It seems that everyone likes the sounds that they produce, but no one likes to program the beasts. Potential programmers are faced with a whole new terminology. Words like algorithm, operator, modulator, and sidebands tend to turn people off before they get turned on. Programming an FM synth is not really all that hard. Problems are most often the result of buttons having multiple meanings. Just like some drum machines, you may have to push seven different buttons and switches just

MIDI TO COMPUTER INTERFACES

*I already own a computer, and I want to connect it to my synth. The people at the store are telling me that I need something called a MIDI interface. I though MIDI **was** the interface!*

MIDI signals run at a particular voltage level, while computers often read a different voltage level. Remember that MIDI is a standard for musical instruments, not computers. Currently, there are only two computers on the market that include a MIDI port along with the more common printer and modem ports. In the future, maybe all computers will include a MIDI port as a standard feature.

If your computer doesn't include a MIDI port, you need a MIDI interface. Its job is to receive voltages from the MIDI instrument and convert them into the types of voltage signals that your computer can read. When buying an interface, be certain that it is compatible with your brand and model of computer. Because all computers read and send differing amounts of voltages (due to the lack of any computer standards), an interface intended for one brand of computer will not work with any other brand.

work with each individual sample. If your sampling rate is 44 kHz per second, you can edit that sample with a resolution of 1/44,000 second!

Let's say that there is some dead space at the beginning of your sample and you want to cut it out. Sure, you can use your ears to hear when a sound actually begins. But as accurate as your ears are, they're not as accurate as your eyes looking at each individual sample point. You can actually see the exact sample number where the sound begins, and cut out all the samples in front of that point. Try doing that by ear!

In addition to cutting out certain portions of the sample, visual editors are much like word processors for sounds. You can copy, paste, erase, merge, mix, loop, and do all types of digital processing to different parts of the sample. After all, the sample is only digital information just like MIDI data, and once it's on the computer disk, the program just plays around with the numbers.

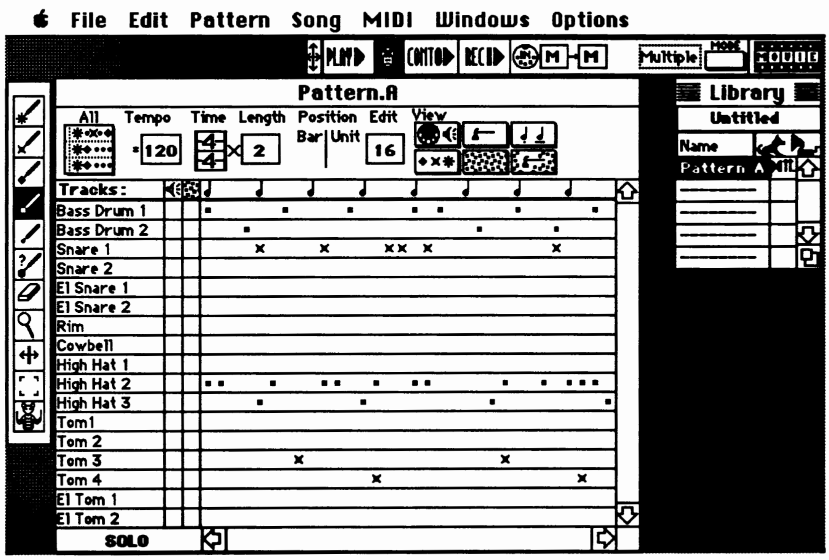
Sample editors are still somewhat expensive, but recent visual editors (thanks to the MIDI Sample Dump Standard) can communicate with several different brands. This means that you won't need to buy a separate program for each sampler that you own. Visual sampling editors give you a great deal of power. Only a few years ago, computer systems that could perform these tasks cost close to half a million dollars. Now you can have one in your studio!

INTELLIGENT PROGRAMS

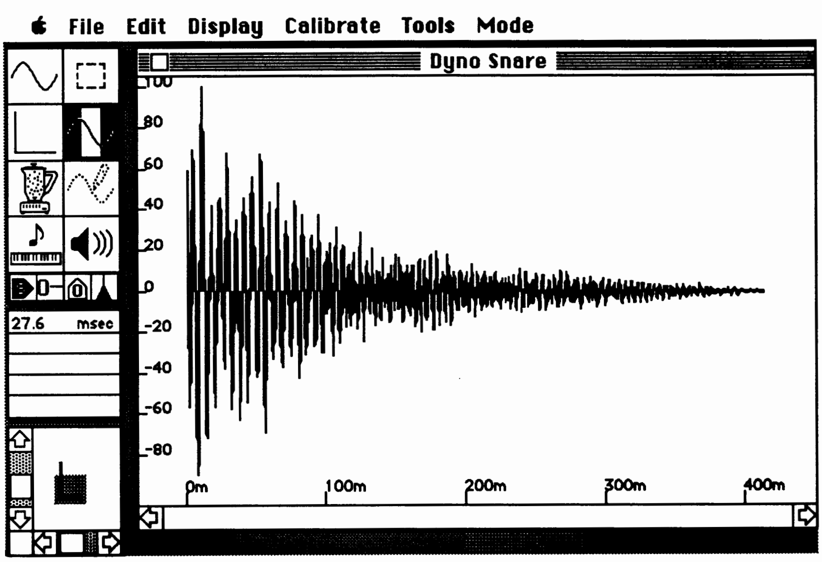
What are intelligent computer programs? They are a class of programs that use a complex set of instructions to help you become a more creative musician. In fact, you can use them to "compose" or "generate" entire musical pieces.

How can this be done? Well, let's say that you play a pattern consisting of seven different sounds into the computer using your drum machine (See *Example 44*). You can then instruct the computer to repeat the pattern over and over again exactly as it was played, or to add a certain degree of randomness to it. Most often, these random changes to your original input are handled as a percentage. In other words, if you want to alter the order of sounds that you played, you might specify something like twenty percent. With that instruction, the computer will play your sounds back, but with twenty percent of the notes out of order. If the program is an advanced one, the order will be infinitely changing rather than altering the same notes each time the pattern repeats.

What sound is going to replace the sound in the original order? Some programs will even let you specify this factor into the equation. You might say something like: "When the original order is altered, use this sound ten percent of

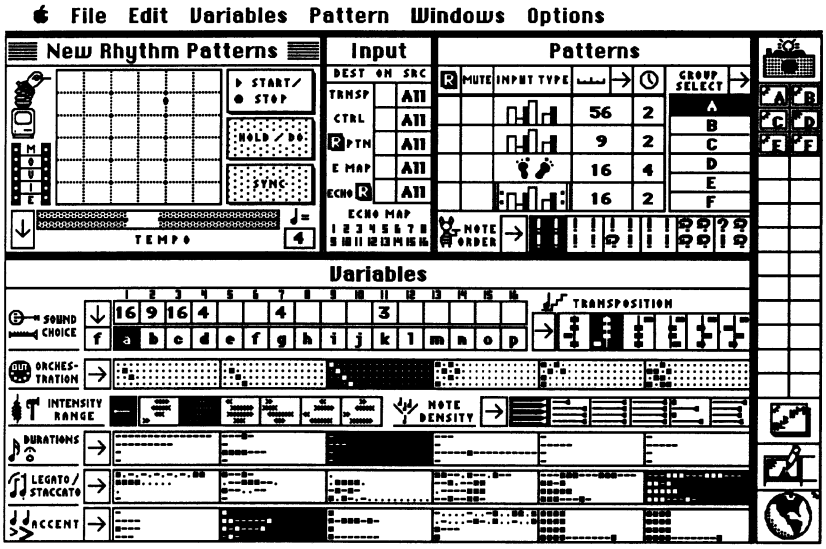


EXAMPLE 44



EXAMPLE 43

EXAMPLE 45



the time, this other sound fifty percent of the time, and a third sound forty percent."

As well as changing the order of the sounds, intelligent programs allow the composer to alter the dynamics, the duration, the MIDI output channel, the tempo, and several other parameters of the data. A few of these programs let you change several parameters in real-time while the computer is playing the music. In essence, this gives you the power to use the computer as a musical instrument. *Example 45* shows the main screen of an intelligent composition program. Notice the number of variables that can be called into play.

Do these things work? Well, yes they do, if you use intelligent programs intelligently. While you may not like to listen to ten minutes of a computer-generated performance (even with all these variables, they tend to get a little repetitive), you can certainly use them to give you new, creative musical ideas. These tools serve best when used as a springboard to get your artistic juices flowing. Many professional composers use these programs to generate ideas that later become the basis for a piece of music. Intelligent programs are fast, powerful, and "smart." But they are not as intelligent or creative as you.

SOFTWARE SYNTHESIZERS

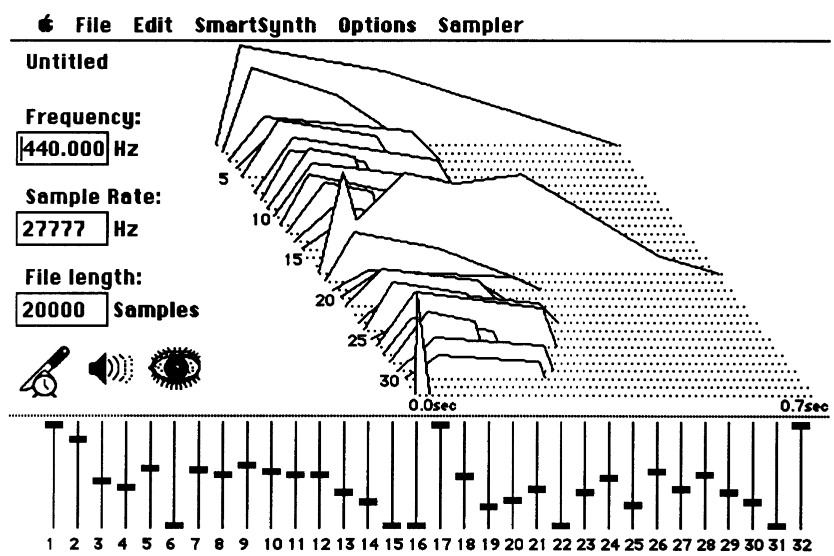
There is a category of program that uses the microprocessing power of the computer to generate complex musical waveforms. Earlier, we briefly mentioned software synthesizers when discussing additive-synthesis instruments. But now, the actual mathematical calculations that create the waveform can be done outside the instrument, inside the computer. Generally speaking, computers are faster and more powerful than the microprocessors used in electronic musical instruments.

So, if the computer is going to take a bunch of instructions and formulate the resulting wave, how does it get that wave back into an instrument? The answer lies in the Sample Dump Standard commands of the MIDI specification. In order to use these programs, you must own a sampler. The computer then turns the newly designed musical wave into digital information and sends it to a sampler through the MIDI cable.

Example 46A shows the main screen of an additive-software synthesizer called *Softsynth* by Digidesign. Notice that there are thirty-two different "oscillators" whose output can be adjusted from the sliders at the bottom of the screen. And each oscillator's envelope is shown in the middle of the screen. *Example 46B* is the portion of the program where each oscillator can be individually adjusted. With this particular program, there can be up to forty different stages to the amplitude's envelope, and forty stages to the frequency's envelope. In addition, the oscillator's waveform and tuning ratio can be adjusted.

Along with additive synthesis, new software programs are being released that act like FM, subtractive, and wavetable synthesizers. They even include a certain amount of digital sound processing like filters, digital reverbs, and pitch shifters. With one of these programs and a sampler, you can create just about any sound you wish.

EXAMPLE 46A



EDUCATIONAL SOFTWARE

Earlier, I mentioned that some of the first computer music programs were CAI based, and you can still buy programs that will teach you music. You can buy a program that is geared to adults, or if your child wants to use a program, some programs use little animals to teach the lessons.

You can find programs that will drill you on rhythmic reading, sight singing, intervals and their transpositions, chord structures, and all other aspects of music theory. There are programs that let you play games with pitch-pattern recognition, and programs that let you improve your tuning skills. Others show you where to place your fingers to play a certain chord on a guitar, or let you play crossword puzzles using musical terms.

Some of these educational programs will work with MIDI. In other words, a program that asks you to name an interval might be able to play those intervals on your synth by sending the appropriate MIDI commands. A few of them will even read MIDI messages and let you answer the computer's questions by playing the notes on a synthesizer. Most of these programs are specifically designed for use with keyboards.

The software available for music education is pretty extensive. If it can be taught, it can be taught with a computer, and there is probably a program around that can teach it! But computer assisted instruction is only the tip of the iceberg as far as education is concerned.

Along with using software that is specifically designed to teach you music, you can learn a lot working with software primarily intended for performance. You can use a sequencer to record your playing, and then you can listen back in slow motion to hear any inconsistencies that may show up in your technique. Do you think that your fast single strokes are a little uneven? Using the sequencer in this manner is a quick and easy way to tell for sure.

In addition to slow-motion performance, the sequencer can play bass lines and keyboard parts that serve as your own custom-made rehearsal band. Set both tracks to loop, and play your drums along with your "computer musicians." Use these same techniques to learn how to improvise on vibes or marimba. Just sequence a basic drum part along with some twelve-bar blues and start jammin'. If you want to practice improvising in different keys, you can copy the progression and transpose it to all twelve keys in a flash.

EVALUATING SOFTWARE

How do you decide which software to buy? Since software packages can be pretty expensive, you don't want to find out after the sale that it isn't going to do what you had in mind. Unfortunately, buying software is not the same as buying anything else. If you don't like it, you may not be able to take it back. Most software guarantees only cover the physical disk itself, not the code that makes up the program or the quality of that code. Here are some hints that you can use before you make your software purchase.

1. Read as many reviews as you can about the software you are considering. Computer magazines often carry articles about music software, and electronic-music magazines are a great source of information.

2. Make certain that the software is compatible with your computer. Ask questions about the minimum memory required, the brands of printers that the program supports, and if any special hardware devices (like a mouse or a hard disk) are required.

3. Write or call the company that makes the program, and ask them to send you a product brochure.

4. Try to get an in-store demo of the programs you're considering.

5. Try to get your hands on the manual for the program. Is the manual legible and understandable? If it seems like some propeller-head wrote it after a lost weekend, it's not going to be much help.

6. Find out what kind of product support is available for the software. Don't believe any salesperson who tells you that the software is so good that you won't need any support. It's just not true! Some companies offer a toll-free number for support service, while other companies charge you big dollars. Will the store selling you the program teach you anything about it and answer your questions? Do you have friends that use this program who can help you in a pinch?

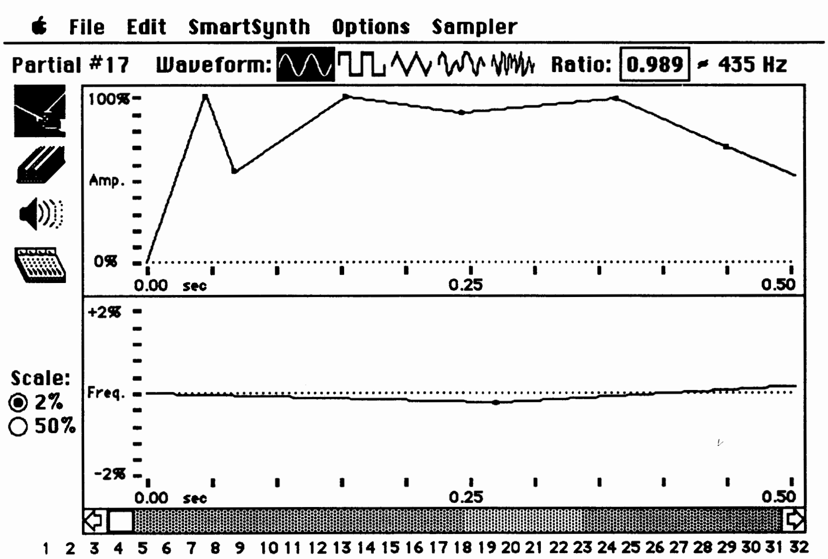
7. Is the program copy-protected? I'm not going to get into the pros and cons of software protection, but copy-protected programs cannot be duplicated. If your program is ruined, you're out of luck! If the program is protected, are backup disks available from the company for free or a moderate charge?

8. There are a few software rental stores that will let you evaluate a program on your own computer in your own home. If you like it, buy it. If the program doesn't do what you want, simply return it and pay a rental fee instead of the full purchase price.

9. Some software dealers have a guarantee that allows you to return software that doesn't meet your needs. Try to buy your programs there.

10. Find out the company's policy on updates. All computer programs are in a constant state of improvement. Most companies offer free updates to any registered owner of the program. This is a good deal, as you know that your software investment will not be obsolete in two months. Other companies charge a reduced update fee (although it can still be pretty expensive). Never buy software from a company that is going to charge you full price for updates.

EXAMPLE 46B



A sequencer can teach harmony and orchestration. If you've ever wondered how a tuba doubling a cello part would sound, now you can find out. This is a much faster, easier, and less expensive way to learn than hiring a studio full of musicians just to try out your ideas.

How about using a sampler editing program to analyze your tone quality on timpani? You could sample a recording of your favorite timpanist from a compact disc and look at it in the sample editor. Then, sample your own playing and compare them on the computer's screen. Does the recording contain more overtones? Try moving your playing spot a little closer to the edge. Does the recorded sound have a slightly slower attack than your sound? Try using a softer mallet.

Actually, running any type of synthesizer editing program is an education in itself. One of the best ways to learn about synthesis is to fire up an editing program, experiment with the controls, and listen to what happens. It won't take long to realize that raising the ratio of an FM modulator is going to create a sound with higher harmonics. Messing around with the envelopes for a while really gives you the feel for what a release or sustain rate is doing to the sound.

Musicians learn about music whenever they experiment. If you play around long enough, and pay attention to what you're doing, you're going to learn something.

SOUND REINFORCEMENT

WHAT YOU HEAR IS WHAT YOU GET

Electronic drums are great! Unfortunately, there is one drawback: You can't hear them without a sound system. Sound systems take the voltage signals from the electronic instruments, amplify them, and pass them along to speakers that will ultimately move the air so that you can hear what's going on. While this may seem to be an easy task, it is actually quite complex. Problems can arise from a lack of accuracy somewhere within the system. Obviously, if you're trying to make a single home-stereo speaker fill a 6,000-seat amphitheater, you're going to run into trouble!

It just doesn't make sense to spend $3,000.00 on a great sounding drum machine, and then run its output through a cheap, poorly designed audio setup. In general, you should get the cleanest sounding one that you can possibly afford. After all, it's the playback system that determines how electronic drums are going to sound. The right components can take an average drum sound and turn it into gold. With a bad system, a great drum sound can become garbage.

In its most basic form, a playback system consists of an amplifier and some speakers, but you can seldom get away with one that is this bare-bones. If you own two or more electronic instruments, you will require some sort of mixer, and it won't be long until you want to add a few external effects like reverbs or equalizers. After leaving the mixer, the audio signals travel to an amplifier and then on to speakers. (See *Example 47*)

What kind of sound-reinforcement system is best for you? It really depends upon what you are planning to do with your electronic drums. If your electronic setup is primarily played in your home as part of an electronic recording studio, the sound system needs to be the best you can afford. A home recording studio requires quiet mixers, clean power amps, and a pair of reference monitor speakers that are as "flat" as possible for even response throughout the entire audio spectrum.

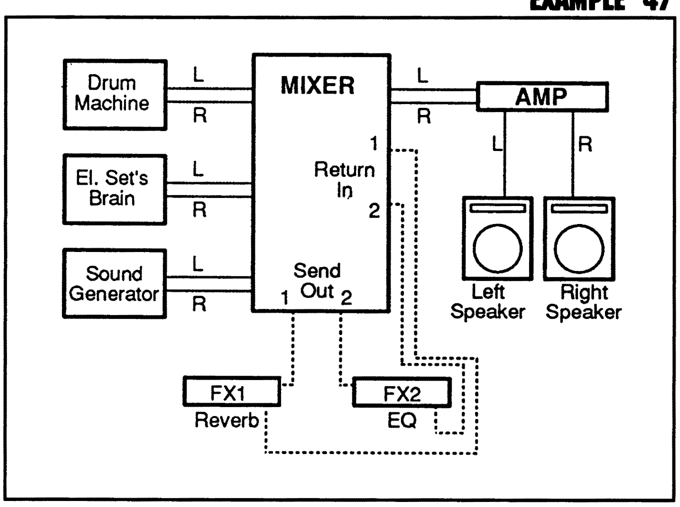
If you're going to be playing gigs with electronic drums, the sound system has different requirements. When playing in a small room, your amplified sound may be perfect by running all your instruments through an all-in-one system. Put the audio stack near the drums, and its sound will cover the stage and the hall at the same time. It may not be critical if the sound is mono or stereo, and clean sounding amplifiers often take a back seat to those with high power.

If you often play in larger rooms, you might actually need two distinct sound systems operating at the same time: one for the house and one for you. In order to have your sounds fill the hall, take the drum's outputs to the main PA system. Many groups, when playing big auditoriums or arenas, push all the sounds through the big speakers (guitar, electric bass, keyboards, drums, and vocals) to cover more space. While this is going to create the best sound for the audience, the members of the band won't be able to hear themselves properly.

Along with the sound system intended for the audience, there must be a stage mix. One solution is to use a monitor signal from the main mixing board that you and the other musicians in the group can hear on stage. In other words, point a few of the speakers toward the band instead of the audience. Depending on the capabilities of the main mixing board, this mixed signal can be the same as the house signal, or a completely different set of balances. The other solution is to have each member of the group play through his own stage-amplification system.

Like the electronic instruments themselves, some playback systems are complete within a single box. A complete system (such as a guitar or keyboard setup) might include a small mixer of up to four inputs, an amp, a spring-style reverb, and a few speakers, all in the same cabinet. There are also some products that are semi-complete. A powered mixer contains both a power amplifier and a mixer, and powered speakers include the amplifier in the same cabinet with the speakers. While these units may save some space in a crowded studio or on a tight stage, the majority of good systems use a component approach. We'll play divide and conquer again, looking at each ingredient of a playback system individually.

EXAMPLE 47



MIXERS

All mixers do primarily the same thing. They take several different input signals and combine them into one or more output signals. If you own more than two electronic instruments, you are going to need a mixer. The output signals from your electronic instruments are fed into different channels in the mixer. Each separate audio signal (in stereo, one for the left and one for the right) will take up a single channel. Mixers come in various shapes and sizes, and are usually named for how many inputs and outputs are included. You can purchase a four-in/one-out mixer, which would take four different inputs and combine them into a mono signal. If your system is running in stereo, and you need to mix more than four inputs, you might want a sixteen-in/two-out mixer.

The front panels of all mixers incorporate a series of knobs, sliders, and switches that control the input and output signals. Take a look at *Example 48*. This is an example of a four-in/two-out mixer. Let's run through the various controls one at a time.

Input Selector — Some mixers have a selector switch to choose between two different signal levels. Since microphones have a much lower output than a synth, this switch can give a little more control over the amount of signal going into the channel. Using this switch at the mic' setting is similar to adding a microphone preamplifier. If the switch is set to the line setting, it is ready to accept a signal from a synth.

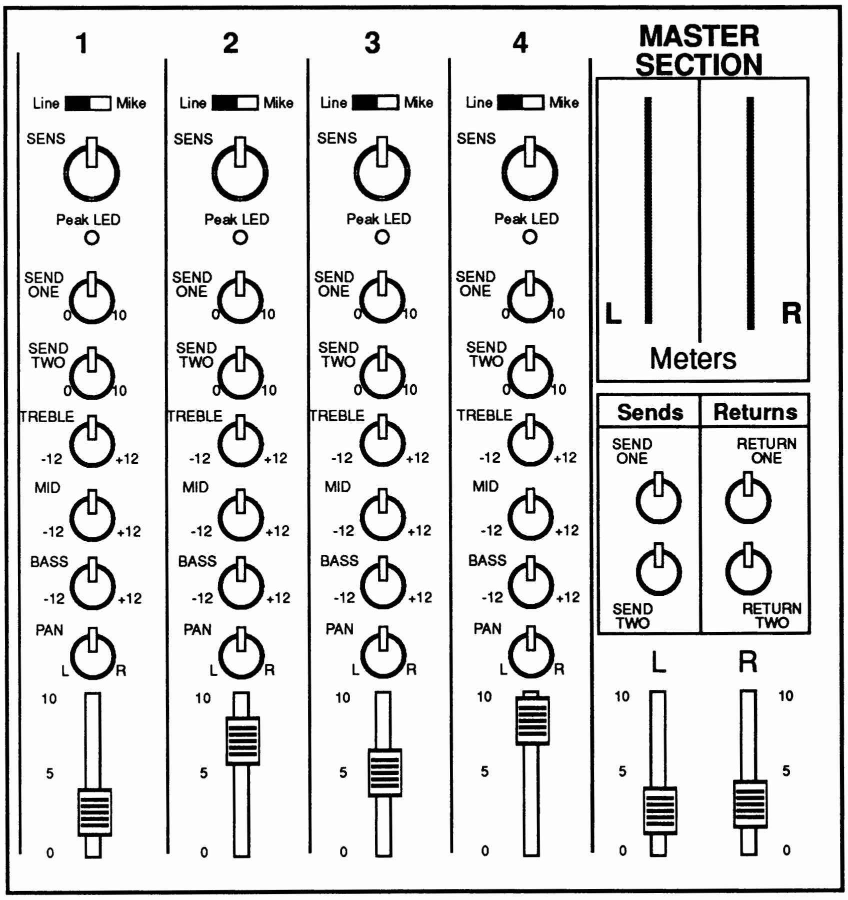
Sensitivity — Once you have selected which type of signal will be coming into this channel (mic' or line), this knob fine-tunes the sensitivity of the input.

Meter — Any good mixer will have a meter that tells you if the input signal on this channel is too high. Some meters are little LEDs that glow when they overload. Others are two-color LEDs that might be green when the signal is good and red if it's too hot and begins to clip. Generally, you want to adjust the sensitivity setting to get as much signal as possible, without distorting too often (just a little is okay). This provides a better signal-to-noise ratio.

Send One — Some mixers let you send the audio signal from a channel to an external sound processor or special effect. This external processor could be a reverb unit, gate, or some other type of device. This knob actually controls the amount of signal that will go to the processor. If the knob is turned all the way to the left, then no signal is sent. If the knob is turned all the way to the right, then the entire signal is sent to the processor. Any position in between these two extremes determines some percentage. These controls are often called "effect sends."

Sometimes, a send is also called a "bus." This term causes a great deal of confusion, but if you think about it for a minute, it makes a lot of sense. Take some of the signal from a channel, put it on a bus (like the ones that travel on

EXAMPLE 48



the street), and send it somewhere else. The signal then "rides" the bus to its new location.

Send Two — On this particular mixer, the same signal from this channel may be sent to two different processors. Other mixers may have as many as eight send and return busses.

Treble / Mid / Bass — These are tone controls that add a certain amount of EQ (equalization) to the signal on that channel. The trend in today's mid-priced mixers is to leave out tone controls. Most electronic instruments have their own fairly sophisticated set of controls, making those on the mixer unnecessary. In addition, those included on mixers are usually pretty basic (and somewhat noisy). If you need really accurate tonal control, send the signal from that channel to a dedicated unit by using one of the sends.

Pan — This control determines the stereo placement of the signals. When the knob is turned to the left, the signal is sent to the left speaker, and when the knob is rolled the other way, it goes to the right. By using the pan control, you can place the signal anywhere within the stereo field.

Fader — The fader controls how much of the signal from this channel will be included in the final output's mix. Most mixers use sliding pots instead of knobs. The main advantage in using sliders is that you can move several of them at the same time, and that the relative levels of each channel can be instantly visible.

After each individual channel has been adjusted, mixed, and possibly sent to some external processors, it's time to work with the mixer's master controls.

Master Sends — The individual send controls on each channel determine how much of that input's signal is going to a processor. The master send controls the levels of all signals at once. In other words, it acts as a master volume control for all signals going to the processor.

Master Returns — This knob determines how much of the signal returning from the processor will be mixed in with the original, unprocessed signals. Remember that the sends and returns act like a loop. The signal flows from the mixer, through the send to a processor, then the signal from the processor comes back into the mixer via the return lines. In a way, the returns can be considered another set of inputs, bringing more signals into the mixer.

Master Faders — After the relative volumes of each channel have been balanced the way you want them, the master faders control the amount of the combined signal that is going to the amplifier. By using the master faders, you don't need to adjust each individual channel fader when you want to bring everything up or down at once.

In addition to all these controls, some mixers have a headphone jack for monitoring the signal. Other, more advanced, mixers have another set of outputs called "monitors." A mixer with a monitor send can adjust the balance of each instrument and send these combined signals to a different amplifier. In other words, the monitor is another completely different mix than the main mix, but would still use the same set of inputs.

In a performance situation, the main mix would be sent to the main speakers that are "playing the hall," while the monitor mix might go to a little personal monitor placed near the drummer. If your mixer doesn't have separate monitor outs, use one of the effect sends to direct a signal to another amplifier.

In a recording situation, the main outputs would be sent to the tape deck, while the monitor outs would go to a set of headphones. This way, the player using the 'phones can request his own mix of instruments.

Why might you need a mixer that has eight or more inputs? You can use up those inputs faster than you think! It's not uncommon to see drum machines that have as many as eight or twelve separate outputs. Almost all electronic kits have individual outputs for each trigger. If you run a drum machine with twelve-output channels, and an electronic kit with six, all of a sudden, you need a twenty-channel mixer!

The advantage of using the individual outputs instead of the stereo or mono outs is more flexibility and control over your sounds. Depending on the situation, you may want to add a certain amount of reverb on your tom-toms, use gated sounds for your snare drum, and keep your bass drum sound "pure." Unless you're using the individual outputs, there is no way to add reverb to one voice without adding reverb to all the voices in the kit. With a multi-channel mixer, the snare drum could be sent through the first send and connected to a gate, while the toms are sent through the second send, connecting to a reverb unit.

Never buy a mixer that has *just* enough channels. If all your channels are full when you bring the mixer home, you're going to run out of inputs if you ever expand your system. If you *do* run out of inputs sometime in the future, you can "submix" certain instruments together. In order to submix an electronic kit, buy a smaller six-channel mixer, adjust all your levels, add your processing, and send the submixed output to the main mixer.

You have worked with filters before, and their operation should be somewhat familiar. When you're listening to your car radio, and you decrease the tone control called "bass," you've added a filter. In order to understand how a filter works, it is necessary to realize that filters operate within a frequency continuum. A single sound is actually made up of many different frequencies that are combined to form the sound's overall timbre. A filter's job is to resonate (increase) or attenuate (decrease) certain frequencies within some area of the timbre's frequency spectrum.

As more and more sounds are included in the mix, there are a larger number of frequencies. An overall sound of drums, bass, guitar, voice, and synth can easily cover a very wide range. In a recording studio, the engineer may add a series of filters (also known as adding EQ or equalization) in order to keep the different instruments sonically separated in the mix. In a live performance, EQ can be added to compensate for feedback or poor room acoustics. In the last case, the engineer will try to improve the sound by cutting out any "booming" lower frequencies or by making the high end brighter by boosting the upper frequencies.

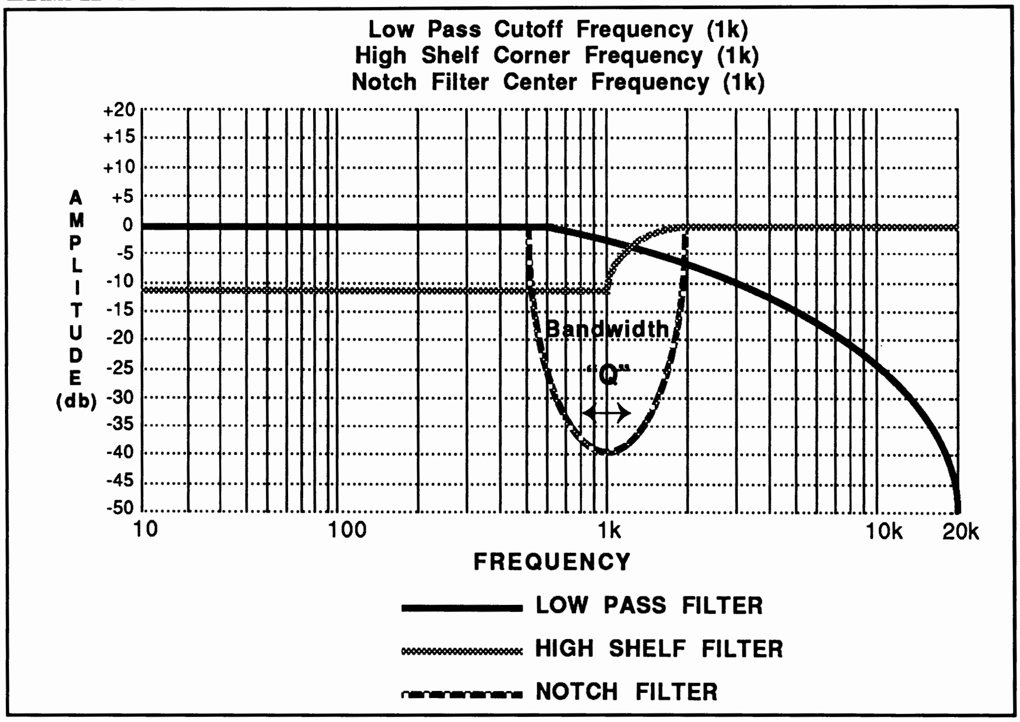
In an electronic percussion system, filters can be used to enhance a sound, but they can also change the overall harmonic content of a sound, thus changing its timbre. Is your bass drum a little "mushy"? Add some EQ. Does the snare need just a little more "crack"? Add some EQ. Creative addition of filters can take a normally dull drum sound and make it extremely individual and special. Have you ever heard a cymbal crash with most of the higher frequencies cut out? It doesn't sound much like a cymbal anymore, but it might be just the sound that you're looking for.

The tone controls included in a car radio or a home stereo system are not very sophisticated. They can boost or cut a portion of the sound's frequency, but what portion of the frequency response are they working with? Beats me. When you twirl a knob called "bass," you might be happy with the end result. But if you're not, then you are going to need a filter that offers a little more control.

So, where do you find filters? Even though some companies are currently making mid-line mixers without any EQ options, many mixers continue to include a variety of tone controls. All high-end mixers offer a great deal of flexibility in EQ. You will encounter filters of various types in drum machines, electronic sets, sound generators, synths, samplers, and tape decks. If you know the filter's capabilities, it will be easier for you to design your sounds.

What types of control are possible when working with filters? Well, filters are available in several different configurations. All filters are designed to let certain frequencies pass through them unaffected while suppressing or enhancing others. Which frequencies are going to be passed and which will be attenuated? This depends on the type of filter that is being used.

EXAMPLE 49



Lowpass Filter — This type of filter lets the lower frequencies pass through unaffected, while attenuating the upper frequencies. The upper edge of the passing frequencies is determined by the "cutoff frequency" (1 kHz in Example 49). If the filter allows a variable cutoff frequency, when this frequency is lowered, fewer highs get through the filter. The slope of the cutoff is not often variable, but depends more upon how the filter is going to be used.

Lowpass filters that are included as crossovers in a loudspeaker system (to drive the woofer) have very steep slopes. In a recording studio, the engineer might add a lowpass filter with a fairly steep slope to a floor-tom mic' in order to help eliminate bleed from a nearby crash cymbal. The slope is usually indicated by a certain number of decibels per octave. In other words, a 6-dB per-octave filter will lower the amplitude by 6 dB for each subsequent octave of sound. A 12-dB per-octave filter will have a much steeper slope.

Highpass Filter — A highpass filter is the opposite of the lowpass filter. The high frequencies are allowed to pass through the filter, and the lower frequencies are attenuated. Again, a cutoff frequency determines the point at which the filter begins to affect the sound, and the slope is determined by the filter's rating (dB per octave).

Low Shelf Filter — A low shelf filter is similar to a lowpass filter with one exception. Instead of completely cutting out the high frequencies at some point, the maximum amount of attenuation remains constant at all frequencies above the "corner frequency." Because the cut remains constant after the corner frequency, these filters sound quite different than cutoff filters.

High Shelf Filter — High shelf filters are the reverse of low shelf filters. Again, the amount of cut as well as the corner frequency can be specified.

Bandpass Filter — A bandpass filter differs from the lowpass and highpass filters because it operates somewhere in the middle of the frequency spectrum (instead of at the ends). All frequencies that lie within the filter's range will be passed through the filter, and all other frequencies will be rejected. Typical bandpass filters will let you specify the "center frequency" of the filter, along with the filter's bandwidth (often called "Q").

The center frequency is defined as the center of the filter's frequency response. The filter's bandwidth is defined as the amount of frequencies that are affected. A filter with a wide bandwidth (low Q) will pass a larger number of frequencies than a filter with a narrow bandwidth (high Q).

Band Reject Filter (Notch Filter) — While the bandpass filter rejects all frequencies that lie outside its range, a band reject filter operates in the opposite way. In other words, any frequencies that are *within* the filter's range are rejected, while all other frequencies are left alone.

Peak Filter — A peak filter boosts any frequencies that lie within the filter's range. Just like the notch filter, the center frequency, bandwidth, and amount of boost (in dB) can be specified. The main difference between a peak and notch filter is whether the affected frequencies are boosted or attenuated.

Graphic Equalizers — A graphic equalizer is nothing more than a series of several peak/

notch filters that are combined together into a single unit. Each one of these filters has a specified center frequency and a moderate bandwidth. Most often, the boost or cut of each filter is determined by moving a slider. When the slider is in the center position, the filter is not active. As the slider moves up, those frequencies will be boosted, and as the slider moves down, frequencies are cut.

Graphic equalizers are usually rated by how many different bands or filters they use to cover the entire audio spectrum (given in octaves). Since there are ten octaves between 20 Hz and 20 kHz, a one-octave equalizer will have ten bands and ten sliders (one per octave). A 1/3-octave equalizer will have thirty controls, and each filter will affect a tighter range of frequencies.

Parametric Equalizers — In order for a filter to be a true parametric equalizer, the musician must be able to control or tune the center frequency, the bandwidth or amount of Q, and the amount of boost or cut. Not only are all of these factors variable, but they should be *continuously* variable over a large range. Because all of the different parameters of the filter are controllable (that's why it's called parametric), these filters can be applied to an extremely wide variety of uses.

All this stuff about cutoff frequencies, center frequencies, and bandwidths can get a little wild. For those of you who are completely lost, take a look at *Example 50*. Here, a home stereo tuner, with treble and bass controls, gives a very simplistic view of different types of filters. While we can't see the exact frequencies that the filters affect (and they're not adjustable), the treble and bass controls can cut or boost by six levels each.

If the treble control is set all the way down and the bass is left alone, a lowpass filter is imitated. By leaving the treble alone and cutting all the bass, a high pass filter is created.

While the audio signals of your electronic instruments are inside the mixer, you have the opportunity to send some of those signals to another device. An outboard effect is usually a modifier or special processor that alters the audio signal in some way. They are called outboard effects because they are not included within the mixer (board) itself.

Reverberation Units — Most electronic drum machines and drumsets use sampled sounds, but more often than not, these sounds are recorded "dry." In other words, the signal doesn't contain any trace of natural reverberation. In reality, all spaces from the smallest bedroom to a church cathedral have some sort of reverberation. But recording studios, by design, are very dry. The thinking is that sounds should be recorded without any reverberation, because it can always be added later by some piece of outboard equipment. Playing drum sounds directly "out of the box," without any reverb, may make the sound seem a little unnatural, tight, or dull. It's amazing how adding a touch of reverb can improve a drum sound.

Often, the terms "reverb" and "echo" are used to mean the same thing, but in electronic music systems, they are quite different. A reverb unit creates the impression of sound waves bouncing and reflecting off walls, floors, and ceilings, generally trying to imitate the sound of a concert hall or auditorium. Reverbs try to mimic all these reflections at once. If the reflections are spaced too far apart, then individual attacks will be heard, and the sound seems to "echo" from a single distant source, such as a hilltop.

Reverb units have different types of controls that help "tune" the sound to the desired result. You can make your drums sound like they are being played inside a gym, or sound as if they are coming from far away.

Reverberation time is one of the most important controls. It is defined as the amount of time between the exact moment that the sound begins, and the instant that the sound's envelope decays to a level of 60 dB below its original amplitude. In other words, the reverberation time controls the length of time that the sound takes to decay.

Don't get too carried away with long reverb times, as a little goes a long way. If the reverb time is overly long, some of your sounds will be masked by the reverb of a previous sound. As reverb time increases, your playing has to become more basic and simple. Otherwise, everything will end up sounding like a drumset in a glass-walled racquetball court!

Pre-delay controls are a little harder to understand. In a large concert hall, the sound waves take a little bit of time before they hit the walls and ceilings and begin bouncing around. Reverb units without this control begin the reverberation of the sound as soon as the signal enters the input. If the reverb box has a pre-delay control, it can be adjusted so that the reverberation effect will not begin until a few milliseconds after the original signal (usually anywhere from 20-70 ms.).

Initial reflection is another control that can be used to enhance a reverb's sound. The initial reflection is a setting of the relative strength between the original signal and the first echos that make up the reverb effect. If the initial reflection is high, the listener will feel that he is close to the source of the reflection. If this setting is low, a listener will feel as if he is further away from the reverberation field.

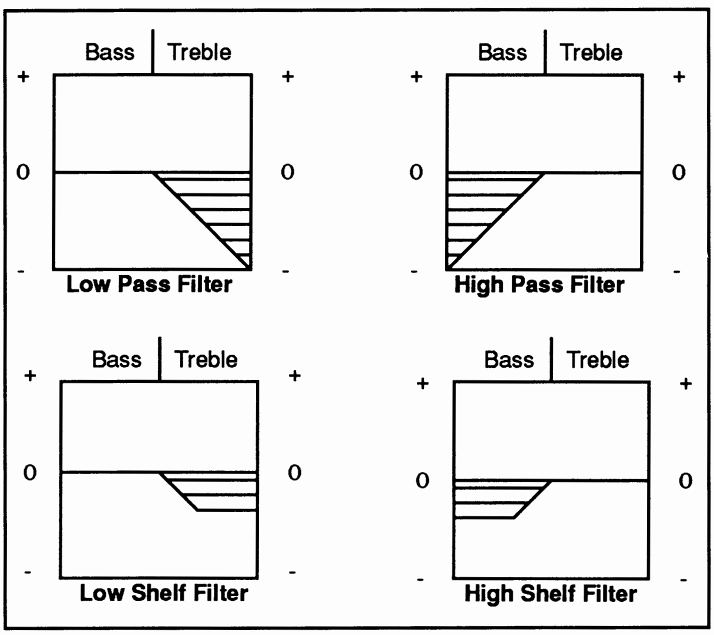
Another control, called the reverb level, is used to set the relative strength of the reverberation. Some units will even let you place the reverberation in different parts of the stereo field. For instance, you might want the original signal somewhere on the right, while most of the reverberation seems to come from the left side.

As you can tell, reverb units offer a lot of control and can be used to liven up a sound, make it seem more real, and improve the overall impression of a performance. This is probably the most important piece of sound-processing gear in your electronic drum studio.

Echo Units — True echo units are a little different than reverbs. The primary difference is the amount of time before the sound is repeated (echos normally have longer delay times than reverbs). Most units let you control the amount of time between the echos, the mix between the original signal and the reflected signal (the reflection is usually set to a lower amplitude), and filters. Adjusting the filter is a good way to achieve a more natural echo effect, as higher frequencies tend to disperse and be absorbed as they are reflected off different surfaces.

Chorus Effects — A chorus effect simulates an ensemble of musicians by delaying a sound by a small amount and detuning the delayed sound against the original. Think about a group of violinists. All members of the ensemble don't play their notes at the exact same time, and

EXAMPLE 50



each instrument will be slightly out of tune with the others. For some sounds, like strings and woodwind instruments, adding chorus can take take a fairly plain timbre and give it "body" or thickness, making it appear more lush and rich.

A stereo chorus does the same thing, but also sends the outputs from the effect to different sides of the stereo field. Chorus effects won't be too obvious on short sounds like snares and kick drums, but try putting it on toms and cymbals. Or, use a little reverb to make a snare drum sound longer, then put it through a chorus. Experiment until you find something that grabs you.

Gates — We've already discussed gates when using acoustic drums to trigger electronic sounds. Sophisticated audio gates let the user program many different aspects of its operation. First of all, the threshold level can be set so that the gate won't open until the signal reaches a certain volume. Then, once the gate opens, does it open instantly or gradually? How long does the gate stay open and how long does it take for the gate to shut? These parameters can be adjusted to suit your individual tastes and needs, and actually serve as another set of amplitude envelope controls.

Phase Shifters — A phase shifter is a cousin to the echo unit. These devices delay a signal by a very small amount and then combine the two signals together. Instead of ouputting both the original and delayed signal like an echo unit, the two signals are "summed," creating a single output. The end result is that some frequencies will be boosted while others will be decreased, changing the input's timbre.

Flangers — Flangers duplicate that famous jet-plane swooshing sound effect that was so popular in the '60s. You might try flanging cymbal or snare drum sounds in order for them to sound more "pitched." As the flanger sweeps up and down, these sounds will appear higher or lower.

Ring Modulators — A ring modulator takes two different inputs, and produces sum and difference tones from the input frequencies. If the initial wave is very complex, more sidebands will be generated. Ring modulators are useful in creating bell-like and metallic sounds.

Digital Delay Lines — Digital delays are extremely versatile. Depending on the amount of features included, DDL devices can create chorus, flanging, vibrato, reverberation, doubling, echo, and even more special effects. Basically, all digital delays work in a similar fashion. They read an input signal, turn it into numbers (the digital part of the name), and output the digitized signal anywhere from one millisecond to several seconds later.

Delay lines will usually contain an input level control, a delay time selector, a modulator (routed so that it varies the delay time), a feedback control (sending the delay's output back into its input, and also called regeneration), an output mix control, and a bypass switch. Some of the more advanced units will include an infinite repeat switch that takes the digitized sound and repeats it over and over, along with other features that make the unit more flexible and easier to control.

Multiple Effect Processors — The current trend in the effects market has been to include many different effects within the same box. These are usually rack-mounted units with effects like reverb, reverse reverb, gated reverb, delays, multi-taps, chorus, vibrato, and flanging. In addition to having several different effects, the musician can pass a signal through two or more of these effects at the same time (that's where the multiple part comes in). You might want to create a multi-tap delay with gated reverb going through a low-pass filter, or a medium-room reverb with delay and chorus. These processors provide a lot of bang for the buck, and can be a very good investment because they are so versatile. If you're only going to buy one outboard processor for your electronic drum studio, check these out.

Extra Effects — Guitar players have been using effects pedals and boxes for years. These little units can do some pretty incredible things to a sound, and they aren't too expensive (some under fifty bucks). Their small size means that you can stick them just about anywhere, and many of them sound pretty good with drums.

These small units contain effects such as delays, flangers, stereo chorus, graphic equalizers, sustainers, compressors, distortion boxes, fuzz boxes, noise gates, octaviders, and more. While they will not be as versatile or give you as much control over the different parameters as larger units, they might just do the trick.

MIDI CONTROL OF OUTBOARD EFFECTS

Is it possible to control my reverb unit through MIDI messages? I don't really see how this can be done, since a reverb can't play any notes.

The amount of MIDI control you have over your reverb depends on the reverb itself. Most units will let you call up different types of reverb configurations (small hall, large hall, gated reverb, etc.) by hitting buttons on the front panel. While the manual may call these different settings effects, presets, or patches, MIDI can reach them with program-change messages.

Let's say that your reverb has a large room in preset number fifteen, and a reverse reverb in preset number twenty. By sending the reverb a program-change message, you can force the unit to jump to the program you want. This can be great during a live performance! If your reverb unit is slaved to your drum brain, and your brain is capable of sending program-change messages (almost all are), calling up set twenty on the brain will also call up the reverse reverb setting from the effect. You might even want to assign a pedal or a pad to send a program-change message so that you can change effects while you are playing.

The trend today is to give the musician even more control over the effects during a performance, using continuous controllers. Yes, you are correct that a reverb unit can't play any notes, but that doesn't mean that the people who manufacture the device can't make it respond to other types of MIDI messages.

A common way to control a reverb with MIDI commands is to map one of the continuous controllers to the reverb decay time. If the controller is in its lowest position, the reverberation time is extremely short. When the controller is moved to higher values, the unit reads this message and alters the reverberation time to make it longer. This way, you can change the size of the room as you are playing (neat trick).

If the machine has been designed to be as versatile as possible, you should be able to assign any of the continuous controllers to any of the parameters of the effect. You might want the modulation wheel to control the low-pass cutoff frequency, or a data slider to control the flange feedback amount. Currently, most consumer effect devices will only respond to a single controller at a time. But as more companies realize that this is a feature that is desirable, units will be able to read several controllers at once.

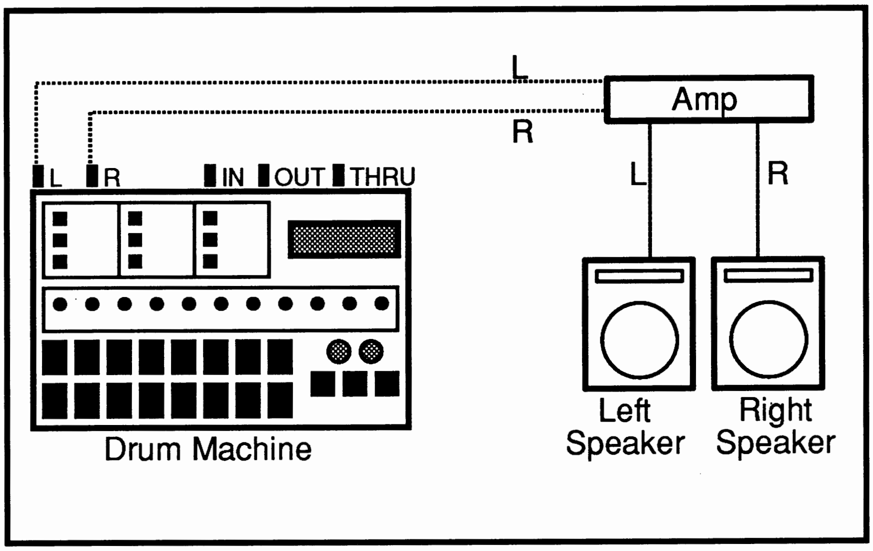
ELECTRONIC SETUPS

SYSTEM ONE

INSTRUMENTS
Drum Machine

SOUND SYSTEM
Stereo Power Amp, Stereo Speakers

CABLES
Four Audio Cables



If you're just getting started with electronic drums, this is the way to go. The initial investment is minimal, and there are only a few cables to deal with. The audio outputs from the drum machine travel to the inputs of the stereo amplifier, and a second set of cables carry the signals from the amp to the speakers. Since MIDI is designed for multiple instrument networks, and we only have a single device in this system, there are no MIDI cables.

LIVE PERFORMANCE

If you play with a group, the drum machine can add a great deal of flexibility to your performances. It can be programmed to play all the normal drumset parts, while you get a chance to go out front and play a little percussion: congas, shakers, timbales, etc.

If you're not too confident of your percussion chops, going the other way may be more beneficial. If the drum machine contains timbale or tambourine voices, program patterns that will be complementary to your live drumset parts. In essence, you're adding a percussionist to the group.

Another option is to program some of the drumset parts while others are played live, giving you an extra set of hands and feet. As an example, programming only the hi-hat parts would free both hands for snare and tom-tom work. Programming a few bass drum parts or maybe a floor tom stroke here or there will let you create beats and fills that might not be possible without the machine.

How about programming a second drumset part, one that might fit with, or even against, what is being played live? There are several groups that use two drummers, and there's no reason why one of them can't be a machine! Along the same lines, why not use the drum machine during an extended solo that showcases your playing ability? Use it as a partner in a duet, or even to trade fills.

Synchronization between the band and the drum machine may be a problem at times. Many machines have footswitch jacks in the back that can be used to control the tempo of the unit and the start/stop commands. To solve this problem, set two footswitches near your kit. As you count off the tempo for the band, tap the tempo footswitch with your counts. Then, on the very first beat of the song, hit the footswitch that starts the song playing. From this point on, you will have to follow the tempo of the machine, but with a good set of monitors (either speakers or headphones), this shouldn't be too hard. If the song you're playing contains tempo changes, they should be programmed along with the patterns. In other words, the drum machine will actually handle the time-keeping chores.

Along with using the pattern and song performance features of a drum machine, sounds can also be played from the instrument buttons on the front panel. Place the drum machine so that another member of the band can reach over and play some additional percussion parts when the inspiration hits.

EDUCATIONAL BENEFITS

If you own a drum machine, you've got the ultimate metronome. A few of the newer metronomes on the market can keep track of odd time signatures, such as five or seven. Drum machines can play in any meter that exists! How about working on your 15/16 time, or 11/8? Most machines will let you enter any number up to 99 as the upper digit of the meter, and any value from half notes to 32nd notes as the lower.

In addition to playing along with the drum machine for odd meters, it's possible to work with less common subdivisions of common meters. By using different sounds for different portions of the bar, 4/4 time can be phrased as 3+3+2, 2+3+3, or 3+2+3. Program a bass drum on the downbeat of each bar with differently pitched cowbells for the 8th-note subdivisions.

How about working with mixed meters? Standard metronomes can't alternate 4/4 measures with 3/8 bars. With a drum machine, it's a snap! Simply program one pattern in common time and another pattern in 3/8. When you're ready for some mixed-meter playing, go into song mode and connect the two patterns together. In fact, you may want to program fifty or sixty different meters (along with different phrasings) into the drum machine's memory. Whenever you need a certain combination of meters, program a song using the appropriate pattern numbers.

When you're practicing along with the drum machine used as a click track, you may want to program thicker sounds for the click. In other words, using a fat snare drum sound, instead of something short like a cowbell, means that the actual beat takes up a little more sonic space. By using a "wider" click, you can practice laying strokes a little behind the beat. If you want to work on pushing some of the counts, bring the drum machine into high-resolution mode, and place some snare drum strokes a fraction of a second in front of the beat.

Try programming an entire snare drum etude, exercise, or even an orchestral excerpt, then play along with the machine, and compare your performance with the "perfect" programmed performance. Are you working on a snare drum, multiple percussion, or drumset solo? Program the solo into the machine, and hear how it sounds before you begin working on it. Not only will you be able to hear a finished product, but the act of programming the solo will teach you about its structure.

Here's another little trick you may want to try: Alter several different voices in the drum machine to come up with something that you think sounds like a good rock 'n' roll drumset. Then create other voices and build a great jazz drumset, a heavy metal set, and a funk set. Experiment with different tunings, decay rates, mix levels, and anything else that you feel will help create the sound you're looking for. Now, go back to your acoustic drums and apply what you've learned to their tuning and adjustment. This will give you a better understanding of the different types of sounds that go into creating kits that suit different musical styles. Not only will you learn more about tuning and adjusting your acoustic drums, but you'll be better equipped to evaluate percussion instruments based upon their tonal quality, not just their appearance. What factors make a good sounding snare, cymbal, or bass drum? Obviously, you might not be able to turn a set with power toms and a 26" bass drum into a good jazz kit, but at least you'll have an idea why.

CREATIVE IDEAS

Remember that there are two ways to program a drum machine. You can try to imitate a human drummer's feel, groove, and style, or you can try to make the patterns sound like a machine. While many programmers strive to make the machine sound "human" or "natural," it's not the only way to go. The genres of techno-pop and rap music would never have come about without the drum machine. There is nothing wrong with creating patterns that sound like they were programmed.

If you want to program patterns that sound like a drummer, don't be in a hurry. Humans are capable of an enormous amount of variation and subtlety. Good drum machines are just as capable, but they require a programmer who knows what those subtleties are and is willing to take the time to create them. Instead of using the repeat button to enter the hi-hat notes, use the multi-level feature to spread the hi-hat voice over several buttons at different dynamics. Then analyze your own playing on acoustic hi-hats. Where are the strong and weak parts of the measure, and how many variations in dynamics are you using? I'll bet more than two or three! Play a steady beat and listen to the sound of your bass drum. Which notes are heavier, which seem to lead into others, and how hard do you play the bass drum when supporting a cymbal crash? Create each note in each pattern with as much attention, detail, and care as you do in your acoustic playing. Then you are really playing the machine, and it becomes a musical instrument instead of a robot. It's not the instrument that creates the music, it's the musician.

which makes them easier to play.

Let's say the multi-pad supports four different sets of note-number assignments. The first set might consist of eight pitched toms, the second could be eight cymbals, a third might be used for ethnic percussion, and the fourth could be a mixture of different voices. If the multi-pad supports a foot switch to change from one setup to another, the groups of voices can be changed during a performance. These voices can be played in real time, even if the drum machine is being used in pattern or song mode. If the drum machine is playing a song that has been preprogrammed, you can still reach up and hit the multi-pad to fire another crash cymbal, add some extra tom voices, or play a quickly moving bass drum lick with your hands.

CREATIVE IDEAS

With MIDI, drum machines can't tell whether voices are being triggered from the instrument pads on the front panel or from an external device that's sending note-on messages. Why not do all of your drum machine programming from the multi-pad instead of those little buttons?

Put the drum machine into pattern record mode, and play the multi-pad. If your drum machine reads note-on velocity levels (all but a few do), then the programmed dynamics will sound much more natural and "human." Dynamics play an important role in the style and feel of music. One of the first things that identifies a pattern created on a drum machine is the lack of any subtle dynamic nuance. By programming the patterns with sticks hitting drum-like surfaces instead of with fingers hitting buttons, the actual feel of the live performer will be captured more accurately.

Once the notes are in the machine, they can be edited with all the features that the drum machine supports. You can still erase a voice, change its stereo placement, adjust its balance in the overall mix, or alter a note's decay time.

If you plan to do drum machine programming with a multi-pad, here's another little trick you can try for a more natural impression. Most drummers have one dominant hand that is slightly stronger than the other. Whenever patterns are sticked "hand to hand," there will be a small volume or tonal difference between the sounds. To create this impression with a machine for a series of snare drum strokes, assign the same voice to more than one instrument button. Then take one of those sounds and change it by setting it a notch softer in the mix (this will simulate the weaker hand). If you can change the envelope or pitch, or add a filter, try it. Then, assign one of the multi-pad's surfaces to each one of these voices. Now you've got two surfaces for the snare drum, and you can use both of them to imitate the right- and left-hand strokes.

SYSTEM TWO

INSTRUMENTS

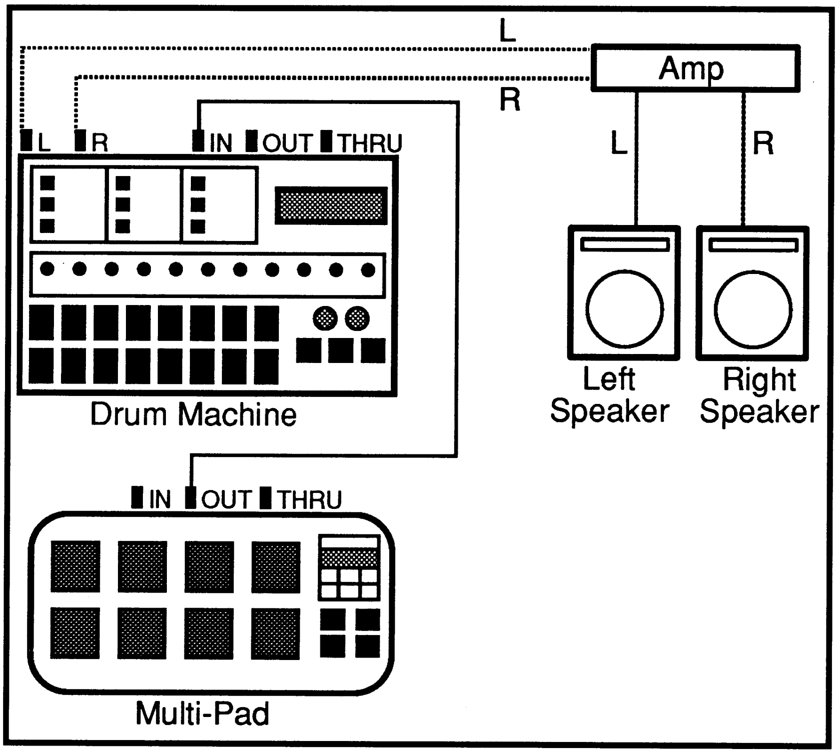
Drum Machine, Multi-Pad

SOUND SYSTEM

Stereo Power Amp, Stereo Speakers

CABLES

Four Audio Cables, One MIDI Cable



In this system, the addition of a multi-pad requires a single MIDI cable. Since the multi-pad makes no sounds of its own, it is only used as a remote controller for the drum machine.

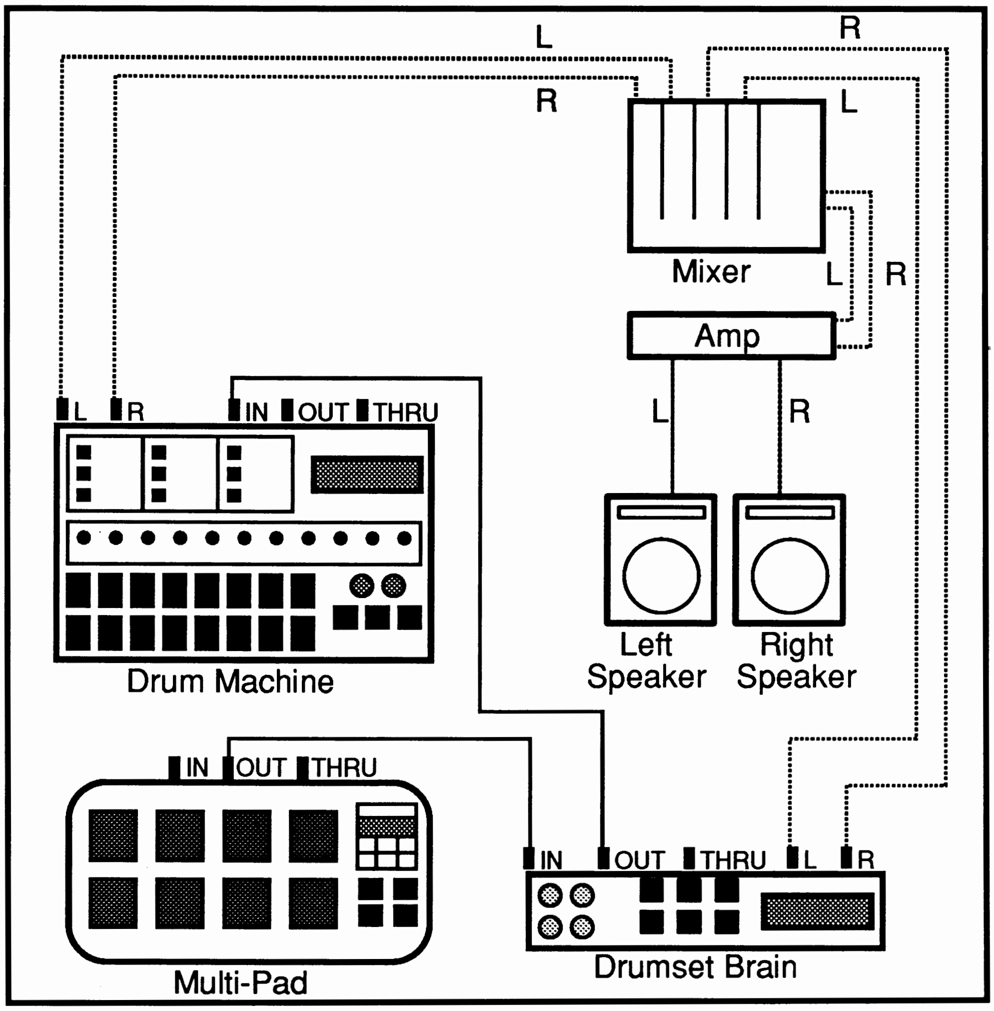
LIVE PERFORMANCE

Set the multi-pad near your acoustic drumset. Now you can play any of the sounds from the drum machine without having to use those little buttons on top of the unit. Multi-pads have larger surfaces, specifically designed for sticks,

SYSTEM THREE

System three adds two pieces of gear: the electronic drumset and an audio mixer. Since there are now four discrete audio channels (two stereo channels from both the drum machine and the kit), a four-channel mixer is the minimum requirement. Although the pads for the electronic kit aren't shown in the example, they will need to be connected to the kit's brain by the proper cables (most often using phone plugs or Cannon plugs).

The MIDI cable routing may look a little odd. The drumset brain is using a special feature called "MIDI mix." A feature of this type is included in several different drum brains, and merges the signal received from the MIDI-In port with the signals being created by the brain itself. This way, both the multi-pad and the electronic drums can serve as master controllers for the drum machine.



LIVE PERFORMANCE

As more instruments are added into the system, more care is needed in selecting the proper MIDI channels and listening modes. Let's assume that the electronic kit's pads are playing the sounds from the brain, and the multi-pads are triggering sounds from the drum machine. With the cables routed as shown in the example, the brain and the drum machine should be set to different MIDI channels.

If the multi-pad is sending its information out on MIDI channel one, and the brain is sending on MIDI channel two, both signals are going to be merged before going to the drum machine. Even though MIDI signals are merged, data retains the discrete channel assignment. If the drum machine is set to listen to MIDI channel one in poly mode, it will only fire notes played by the multi-pad. If it is set to channel two in poly mode, it will only listen to instructions from the electronic kit's brain.

If the idea is to use the sounds on the drum machine for both the multi-pad *and* the electronic kit, put the drum machine into omni mode. Now it will listen to all instructions, no matter what MIDI channel happens to carry them. Depending on the features available on the electronic kit, you may be able to send mode-change messages over the MIDI cable whenever you call up a particular patch on the kit's brain. Using a footpedal to send program-change messages to the brain means that you can change the listening mode of the drum machine during a performance.

In a system such as this, you may need as many as four different pedals. One might be assigned to the tap-tempo button of the drum machine, another controlling its start and stop functions, a third to move back and forth through the electronic kit's different programs, and the fourth moving the multi-pad through its presets. How in the world are you going to control four different footswitches? Keep a bass drum sound on one of the multi-pad's surfaces, and while your foot is busy with the switches, grab the bass drum notes with your hand instead.

CREATIVE IDEAS

Since multi-pads usually let you assign not only the note number of each surface but the MIDI channel as well, think about assigning different channels to some of the pads. Perhaps your drum brain is listening to MIDI channel two in a poly setting, while the drum machine is listening in an omni mode. If a surface of the multi-pad is sending its message on channel one, then only the drum machine will fire. If another surface is set to send on channel two, then both the electronic brain and the drum machine will fire. Depending on the note numbers assigned, one single surface on the multi-pad may trigger a bass drum sound from the kit's brain *and* a cymbal crash from the drum machine. This is typically called "layering" sounds. Meanwhile, since the drum machine is in omni mode, it will respond to whatever is played on the electronic kit. What about using a pad (either from the kit or the multi-pad) to trigger two snare drum sounds for a thicker and heavier texture?

Speaking of layering sounds, several drum brains are capable of sending multiple note-on messages with a single strike. You might be able to layer six sounds (one from the kit and five others on the drum machine) on a single pad. How about a bass drum, two crash cymbals, snare, floor tom, and electronic tom all with a single stroke?

If your electronic kit supports local on/off messages, changing from local on to local off will separate the brain's internal sound genera-

INSTRUMENTS
Drum Machine, Multi-Pad, Electronic Kit

SOUND SYSTEM
Mixer, Stereo Power Amp, Stereo Speakers

CABLES
Eight Audio Cables, Two MIDI Cables, Cables Connecting Electronic Pads To Brain

tor from its pads. Now the pads will only fire the drum machine sounds, but the multi-pad (being an external source) can still trigger sounds from both the kit and the drum machine.

By changing one of the MIDI cables, you can have the drum machine fire sounds from the electronic kit. Just run a cable from the MIDI-Out of the drum machine to the MIDI-In of the kit's brain. Now the drum machine has an expanded number of voices at its disposal. Perhaps you want to layer the snare sound from the electronic kit with the sound of the drum machine's snare. Be certain that the note number the drum machine sends for the snare drum stroke is the same as the note number assigned to the snare on the kit's brain.

Along the same lines, if you only want to hear the electronic kit's snare sound, adjust the volume of the snare voice on the drum machine to its lowest setting. The MIDI note number, along with its actual velocity reading, will still be sent through the MIDI cable, but the volume of that voice on the drum machine will be too soft to hear.

THE PRICE OF POWER

I think electronic drums are too expensive. Why should I spend two- to twenty-thousand dollars for an electronic system when I already own a set of drums?

All the instruments in system six can be put together for well under $12,000. Instead of buying a sampler that costs $16,000, buy one that sells for under $2,000. It seems like a lot of money, but not when put into perspective.

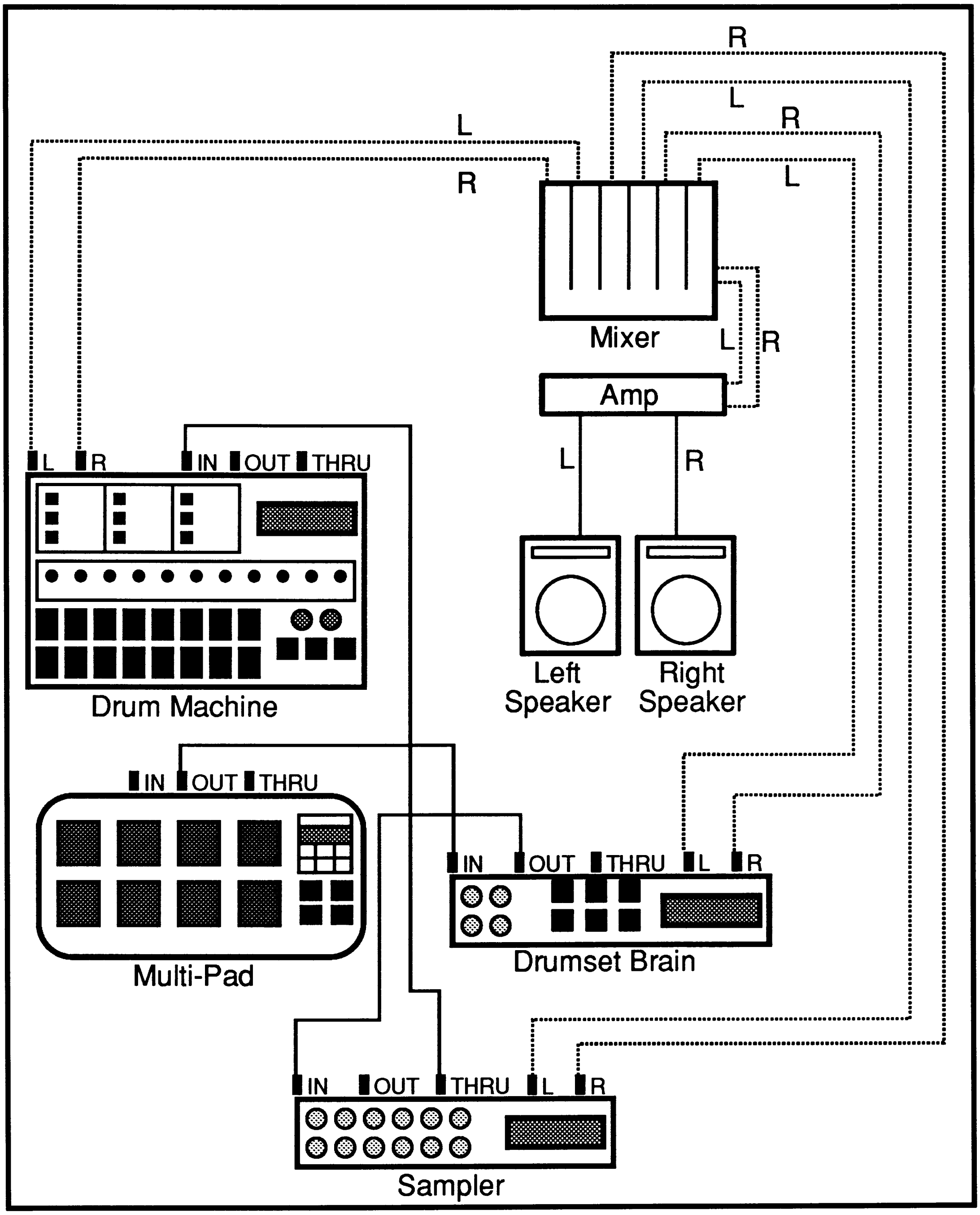
If you're currently playing a professional model drumset, you probably have three- to five-thousand tied up in drums, hardware, and cymbals. A good marimba costs over four thousand, and a set of high quality timpani are going now for over $16,000. That's sixteen grand for only five drums! You could be a harmonica player and buy your instrument for less than ten bucks, or you could be a concert violinist and pay up to a quarter of a million dollars for a great instrument. But, you're a drummer.

Unless you've got loads of spare cash lying around the house, you won't be able to buy your complete studio all at once. Get started with a drum machine or electronic kit, and let them make you some money before buying more gear. If you eventually get a big MIDI studio going, don't forget that you can then expand into other musical areas. Try recording some demos for friends, then advertise your system as a recording studio. Contact local radio and television stations, and see if you can get your foot in the door producing jingles and other background music for their programs. Put together a few spots to highlight your skills, then let them know that you're interested.

If you approach your drums as a hobby, you may not need an extensive electronic drum system. But, just like any other musical instrument, electronic drums are part of the tools of your trade. You can use them to make money, and fulfill your creative desires.

In a way, it's like a person who loves to work with cars and fancies himself a professional mechanic. He may own a wrench and a screwdriver, and actually do pretty good work with only two tools. But if he had the proper tools, wouldn't some jobs be easier and faster to complete, making him more productive? And wouldn't he have more fun?

SYSTEM FOUR



The fourth system adds a rack-mounted sampler. Because there are now two more stereo audio signals, a six-channel mixer is the minimum requirement. Notice that there is no MIDI cable connected to the MIDI-Out of the sampler. Since this instrument doesn't have any type of input device, no messages need to be sent from the sampler. Keep in mind that the signal coming from the MIDI-Out port of the electronic set's brain is a merged signal from both the brain and the multi-pad. This signal is sent to the MIDI-In of the sampler, and from there continues on to the drum machine by way of the sampler's MIDI-Thru port (an exact duplicate of the MIDI-In signal).

LIVE PERFORMANCE

Now that a sampler has been added to the system, the sky's the limit. We're going to assume that the sampler came with several disks of factory sounds, many of which are drums. Let's see what this electronic drumset can sound like!

Set the drumset's brain to send messages on MIDI channel two, and have the sampler listen to the same channel. Find a great-sounding snare drum on the sampler, and assign the note number sent from the brain to the note number that has the snare sound. Your electronic kit is now playing the sampler.

If your drum brain allows saving note numbers as part of the patch information that is stored in memory, you can do some pretty incredible things. Let's say that the sampler is extremely flexible. (You did buy a sampler that allows several split points, has lots of memory, and lets you assign any sample to any key, didn't you?) Load a bunch of drum sounds into the sampler, and assign them to six adjacent note numbers (or eight if your kit has that many pads). It might be easier if you follow a particular format, such as bass, snare, tom 1, tom 2, tom 3, and tom 4. Keep assigning drum sounds to note numbers until your sampler runs out of memory or split points. Now, build a patch on the electronic kit that sends those first six note numbers. Save that patch into internal memory and build another patch that uses the next six (or eight) MIDI note numbers. Get the picture? You're building new drumsets with new sounds that can be called up by changing the preset on the electronic brain. You can combine different sounds from the sampler by building more kits that have different combinations of note numbers. How about combining the bass drum from the first kit with the snare of the third and the toms of the fifth? You're going to run out of memory on the sampler long before you run out of patches on the electronic set.

You've still got all the flexibility that you had in the last system, so try playing around with local on and off messages, or sending multiple note-on messages with a single stroke. You may also want to layer sounds from the electronic drum's brain, the sampler, and the drum machine all together. Here's how. Set the brain, the sampler, and the drum machine to the same MIDI channel and fire them all with the multi-pad. You might want to make different pads send messages on different MIDI channels. By changing the MIDI channel and listening mode of the other three devices, you can have some pads trigger only the drum machine, only the kit's brain, only the sampler, or any combination of two of these units.

INSTRUMENTS
Drum Machine, Multi-Pad, Electronic Kit, Sampler

SOUND SYSTEM
Mixer, Stereo Power Amp, Stereo Speakers

CABLES
Ten Audio Cables, Three MIDI Cables, Cables Connecting Electronic Pads To Brain

CREATIVE IDEAS
Now that you've got a sampler, sample all your acoustic drums, cymbals, and anything else that you own. If you want to play the sound of your own acoustic snare drum from the electronic pads, you can. Contact a band director or percussion teacher at a local school or college. Spend an afternoon and sample instruments you couldn't possibly afford to buy yourself. Timpani, chimes, gongs, Latin percussion instruments, hand cymbals (*real* crash cymbals), additional suspended cymbals, marimbas, vibes, xylophones, log drums, and the list can go on forever. Sample as many sounds as you possibly can, save them on disk, and take them home to edit. Go to a museum and see if you can get permission to sample authentic instruments from Africa, South America, or the Far East. Adding a talking drum or gamelan to your sound library might start your creative juices flowing.

Change the MIDI cable configuration so that the drum machine's MIDI-Out port is connected to the MIDI-In of the sampler, and you've got a new drum machine. Simply place the samples under the proper note numbers and turn the volume of the drum machine off. The MIDI messages will go to the sampler, and the sampler will fire the sounds, but the original sounds of the drum machine won't be heard.

Since MIDI messages only tell an instrument to play a certain note at a certain time, just about any sound in the world can be fired from that note number. For some new ideas, play your old drum machine patterns and songs using marimba or electric guitar samples. Or, for that matter, turn your electronic kit and multi-pad into a drum synthesizer. If the multi-pad has eight surfaces and the kit has six, there are 14 different pitches to work with. This can be arranged in any number of ways, from over an entire octave of chromatic pitches to a certain melodic configuration. How about doubling the melody of a song by playing the pitches with your sticks, as well as the percussion sounds that might be layered by the drum machine?

Some electronic brains allow you to send one set of note numbers when a pad is struck softly and another set of numbers when it's struck harder. In essence, this can double the amount of sounds that are available at any one time. Instead of only 14 pitches, you can access 20 (and if the multi-pad also incorporates this feature, 28 pitches).

All along, we've assumed that the drum brain that is being used is capable of sending multiple note-on messages for a single strike. As well as using this feature to layer drum sounds, it can be used to create chords on the sampler when it is playing pitched instruments. By carefully laying out the samples under the different note numbers, you may have a drum sound *and* a chord from a melodic instrument firing at the same time.

In addition to all this stuff happening with the MIDI messages, most samplers are capable of some hip tricks of their own. Try taking a single sample and adding a low-pass filter, adding some modulation to the pitch, or even assigning modulation to the stereo placement of the sound. Some samplers will let you assign two different sounds under the same note number. (How many layers are available now?) These two samples can be fired at the same time, velocity switched, or velocity faded into each other.

Some samplers will let you set a delay time for the sound. Unlike a digital delay audio processor, this delay is from the time the note-on message is received to the time the sound actually begins. By delaying a sampled snare sound by just a few milliseconds and triggering it with the snare from the electronic kit, a type of slapback can be achieved. Now, send the electronic kit's snare through the left audio channel and the delayed sound from the sampler out the right audio channel. We're talking mucho slick here!

THE LEARNING CURVE
I've just been left a $15,000 inheritance from a long-lost aunt. I'm ready to get into electronic drums in a big way.

If you don't have any prior experience with electronic instruments, I suggest that you don't spend your $15,000 all at once. This kind of money can buy an extremely sophisticated electronic drum studio, but you're going to have a certain amount of trouble putting it all together. The more instruments and devices you try to connect, the more potential problems there are.

Buy a very nice drum machine or electronic set first, and put the rest of the money in the bank. Learn how to do everything you possibly can with the drum machine before buying any other gear. If you've got 20 pieces of equipment in your system, you're going to be faced with 20 manuals, 20 operating systems, and 20 different sets of terminology all at once. No one wants a nervous breakdown caused by information overload! Instead, learn all there is to know about one or two instruments before adding a third or fourth.

The learning curve on electronic instruments is fairly steep. Progress may be slow before you get a handle on how everything actually works and fits together. But after getting over the initial hump, it all falls into place quickly. Once you have mastered a few different instruments from different companies, learning a new electronic instrument only takes one read through the manual, and about two hours of hands-on exposure.