ADVANCED APPLICATIONS

THE ART OF SAMPLING
by Craig Anderton

Recording good samples is not always easy; creating a really super set of sounds requires patience, practice, and skill. Sure, you can get musically useful results within a few days after working with the Emax II, but as you learn your craft the quality of your samples will improve dramatically.

Sampling is a multi-stage process, and consists of:

■ Taking the best possible sample.

■ Manipulating the sample within Emax II for maximum musicality (looping, splicing).

■ Efficiently combining samples into presets.

Part One: TAKING THE BEST POSSIBLE SAMPLE

What comes out of the Emax II can only be as good as what you put in: Strive for maximum fidelity when sampling. Here are some ways to increase sample quality.

GENERAL TIPS

■ ***Live Sampling***: Sample "live" whenever possible, rather than recording a sound on tape then sampling from the tape.

■ ***Beware of Overloads***: Emax II is a digital recording device. Unlike analog recording devices, distortion does not increase slowly past a certain level; rather, it increases rapidly above the overload point, and produces a "non-musical," splattering type of distortion. Monitor your levels carefully.

■ ***Use a Sampling Rate Twice the Highest Frequency Produced by the Sound Source:*** Because the Emax II can use different sample rates, you can optimize the sample rate for the particular sound. For instance, if the sound source is a kick drum, a 20k sample rate can be used to conserve memory because the kick drum does not have very much high frequency energy over 8k Hz. Whereas, if a cymbal were to be sampled, a 39k sample rate might be needed to capture the very high frequencies produced by the cymbal. Although this example is theoretically correct, the distortion produced by low sample rates can sometimes enhance the sampled sound. Use your ear for final judgement!

■ ***Sampling Direct vs. Sampling via Microphone***: Whenever possible, sample electronic instruments directly into the Emax II. Avoid using a recording console, direct box, or other device in between the instrument and the Emax II, unless the device is of very high quality (ie. low noise).

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■ ***Maintain the Instrument to be Sampled**:* Tune your instrument and carefully check the intonation; change strings (if applicable); and check that the instrument doesn't have any loose parts that rattle, squeak, or make other noises.

■ ***Sampling Acoustic Instruments**:* Choose the microphone and mic placement as carefully as you would for any recording project. Musicians sometimes note how hard it is to get a good "sound" in the studio, but that's what sampling is all about...getting a good sound, and once you've got that sound, keeping it.

■ ***Play Naturally**:* Sometimes it's a lot harder to play one note than several hundred. Unless you're trying for a special effect, play the sample as you would normally play the instrument. It's often a good idea to play several notes, and use truncation to zero in on the best of the bunch.

■ ***Avoid Ground Loops:*** Ground loops occur when electricity can take two different paths to ground. If there is a resistance difference between the two paths, this can generate a signal (consisting mostly of digital "hash" or hum) that can work its way into the recording chain. *With all the instruments patched in place and connected to each other*, turn off the Emax II and insert a ground lift adapter between the Emax II plug and the wall.

CAUTION: Removing the ground connection defeats the safety advantage of using a three-wire plug, so use this technique only as a last resort and exercise extreme caution. *Make sure the Emax II chassis has some other path to ground* (usually via the audio input and output cables). Having two paths to ground can cause ground loops, but having no paths to ground can cause a potential shock hazard if there's an equipment malfunction in the studio.

Using Signal Processing While Recording

Just as with regular recording, there are no absolutes about recording signals with or without signal processing. Some engineers feel that tracks should always be recorded flat to allow for the maximum number of options during mixdown; others prefer to record with a bit of processing, especially if the processor might be needed on another track when mixing. Here are some thoughts about the use of various types of signal processors when recording samples.

■ ***External Preamp**:* If you need to preamplify the signal being sampled, set the sample gain to 00 dB and use an external, high-quality mic preamp to increase the signal level going into the Emax II. Emax II's preamp noise level, while acceptable, may not be as quiet as an external mic preamp.

■ ***Limiting Dynamic Range**:* Limiting the signal being sampled can put a higher average signal level into the Emax II, thus improving the already excellent signal-to-noise ratio. Also, the limit point can be set just under Emax II's overload point, which makes level setting less critical.

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■ ***Equalization***: If using EQ gives a better instrument sound, then use it. You want the best possible sound going into the Emax II, and if a little EQ is indicated, it's best to add it while recording rather than having to think about it during playback.

■ ***Flanging & Chorusing***: There will be no problem using time delay effects with unlooped sounds, but looped sounds are another story. The worst case is that the time-delay processing creates such a complex, non-repetitive signal that it becomes virtually impossible to find a good splice point for the loop. The best case occurs if the effect is used to give cyclical sweeps, since you can often loop one sweep of the LFO with good results. Remember too that Emax II can add chorusing effects all by itself (**DYNAMIC PROCESSING 20**).

■ ***Compression***: Compression can be helpful when looping sounds since it evens out level changes, thus making it easier to find an equal-level splice point. If appropriate, use Emax II's VCA and AHDSR or velocity control to restore the original signal dynamics.

■ ***Audio "Exciters"***: To brighten up a sampled signal, use a psycho-acoustic enhancement device (such as the Aphex Aural Exciter or EXR Projector). These kinds of devices add a high-end "sheen" without adding the stridency encountered with excessive high-frequency equalization.

■ ***Using Noise Reduction***: A number of single-ended noise reduction units (such as the Rocktron Hush) are now available; these do not require that the noisy signal have been previously encoded, as is the case with Dolby and dbx. If your source signal is noisy, one of these devices can help to greatly clean up the overall sound quality as you sample.

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Session No. 1: SAMPLING FROM TAPE

Maintaining Sample Quality with VCRs

One excellent way to sample involves using a VCR and digital audio adapter (such as the Sony PCM-F1) for recording different sounds. Compared to using conventional reel-to-reel tape recorders, this approach offers several advantages:

■ ***Digital-quality Fidelity:*** A VCR/audio adapter combination adds virtually no degradation to the signal being sampled. Signal-to-noise ratio, wow, flutter, and dynamic range exceed that of all but the very finest analog recorders. These tapes of source sounds can serve as a “master” library.

■ ***Lower Tape Costs:*** VCR cartridge tape is very inexpensive; you can record hours and hours of samples for only a few dollars.

■ ***Portability:*** Using a portable VCR and battery powered audio adapter makes for a very compact and portable package.

Although splicing is not easy with VCR-based audio systems, you can generally do the required signal splicing and manipulation within the Emax II. A good alternative to the VCR/adapter approach is to use a Sony Beta Hi-Fi recorder—the audio tracks are of excellent quality. VHS Hi-Fi gives comparable results.

■ ***Digital Audio Tape*** recorders (DATs) use the same type of technology, but are optimized for audio recording and are generally smaller and more convenient to use. All of the advantages of VCR/PCM recording apply to DATs, with the added advantage of DAT recorders being fully integrated, small-size units.

Maintaining Sample Quality with Reel-to-Reel

Follow common-sense recording practice—be extremely careful about mic placement, use noise reduction when recording samples on to tape, limit the signal going on to tape rather than limiting the signal coming off the tape into the Emax II, and add aural enhancement if desired.

■ ***Using Variable Speed Tape Recording:*** Variable speed can help time-compress samples. For example, suppose you have one second of bank memory left and want to sample a two second sound. Record the sound on tape, and play back the tape at twice normal speed into the Emax II. This gives you a one second sample. Assign the original pitch one octave above the sample’s pitch, the low note one octave below the sample’s pitch, and the high note as desired. Playing the low note, which is an octave-lower version of the sample, will produce a two second sound at the desired pitch. This technique is also useful for speech compression.

Variable speed can help increase high frequency response as well. Record a sample on tape, then set the recorder to half speed and record the sound into the Emax II. Assign the original pitch to one octave below the sample’s pitch, the low note also one octave below the sample's pitch, and the high note at the sample's pitch. This high note will give the original sample sound with an extremely clean high end.

■ ***Creative use of multi-tracking***: Try multi-tracking sounds on a conventional recorder, then sampling the combination sound into the Emax II. Recording a half-speed piano note along with an electric bass and a little bit of analog synth in the background gives a pretty outrageous bass sound...

Session No. 2: SAMPLING ACOUSTIC INSTRUMENTS

Sampling acoustic instruments is not easy, which you've already discovered if you've tried it. It's not impossible, but you need persistence and patience. Here are some tips that will help make matters easier.

Use Proper Miking Techniques

Prepare for a sampling session as if you were intending to record a Grammy-winning album. This means matching the right mics to the right instrument. The general rule is to use condenser mics for instruments that need delicacy, high end, and good transient response. Dynamic mics seem to work best when capturing the sound of instruments that exhibit raw power, such as bass drums, some singers, and stacks of Marshall amplifiers. Some mics are acquiring a reputation as good sampling mics. What this usually means is that the mic has a very flat and accurate frequency response, as opposed to, say, vocal mics that boost the high end for more vocal presence.

Mic placement is crucial. Sounds that come from several directions, as is the case with acoustic instruments, which tend to radiate sound, tend to be out of phase with respect to each other. If the mic picks up these multiple signals, some will reinforce each other, while others will tend to cancel. This reinforcement/cancellation effect produces frequency response anomalies that can make a sound thin (with excessive cancellation) or muddy and boomy (with excessive reinforcement).

One way to minimize phase problems is to use a PZM (Pressure Zone Microphone, Crown's trademarked name for this particular type of mic technology). A PZM mounts the mic element over a small gap, facing a broad rectangular plate. This mounting arrangement makes the mic relatively insensitive to receiving multiple out-of-phase signals from an acoustic sound source.

I've often determined proper mic placement by sitting in the control room and taking notes while the engineer, wearing headphones, tries a variety of miking positions. There are some standard mic placements, but use these as points of departures only. Every instrument, every microphone, and every player is different. Adapt to the circumstances at hand.

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Miking specific instruments requires specific techniques. For piano, mounting a PZM a few inches above the round holes in a piano's cast iron frame produces a high level output suitable for sampling. Another option is to mount a PZM to the cover of a grand piano, facing down into the strings. For nylon-string acoustic guitar, pointing a mic up towards the sound hole, from below the player's right hand, usually gives good results. (However, most acoustic guitars have a body resonance that you may want to attenuate with a parametric equalizer in order to get a higher average signal level into the Emax II.) Steel string guitars tend to produce more output, which implies that you might want to place the mic a little further away from the body to pick up more of an overall sound.

The job of miking an instrument is greatly simplified by sampling in mono. In some cases one mic will be all you need. However, on occasion mixing several mics together will give a better sound, such as with piano, where you may need two mics to cover the full range of the keyboard. It's important to monitor in mono, though, to check for phase cancellation. And if you're sampling in stereo, it's extremely important to check for phase cancellation. Two mics will, by definition, pick up signals with different phase relationships. Listening to the combined result in mono will indicate whether there are any serious phase cancellation or reinforcement problems.

Contact microphones can also be useful as supplements to standard mics or pickups. Attaching a contact mic to, say, an electric guitar will pick up some of the string and finger noises usually lost through the pickup, and when mixed in with the standard pickups, gives a more natural sound. (Don't overdo it, though, or the sound could become gimmicky.) Basses tend to gain more snap when you attach a contact mic to the body. I've had good results by attaching a contact mic to the headstock, or by mounting it in some of the free space underneath the pickguard. And of course, contact mics can be attached to garbage can lids, suspension bridge cables, steel girders, and just about anything else you might want to sample.

Avoid Including Ambience in the Sound

Of course, sometimes ambience is part of the sound, like recording a snare drum in a massive concrete stadium, or in a stone castle. But remember that once you record ambience, it is part of the sample. It's often best to record a dry sound then add ambience later, as desired, using electronic signal processors.

Adding Vibrato

Use the sampler's ability to add vibrato rather than playing the instrument with vibrato. To my ears, playing vibrato on the instrument sounds better, but there are some problems with this approach. First, when you transpose a sample, the vibrato speeds up or slows down, depending on whether the sample is transposed up or down respectively. With a multi-sampled sound where each sample doesn't need to be transposed very far, this is not a problem. But if you transpose the sound over a wide range, the vibrato will change from seasick-slow to Loony Tunes fast. Besides, the Emax II allows you to add vibrato anyway using the modulation wheels or other controllers. This gives you more flexibility, since you can alter the speed as required and bring the vibrato in and out as desired.

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Record Natural Playing

Record a musician playing naturally: runs, scales, solos, etc. It seems logical that when sampling, a musician would play just one note, and you would sample that note. However, it's very hard to play the same note over and over again naturally. It's often a good idea to play several notes, and use truncation to zero in on the best of the bunch. Some of my better samples have come from pulling parts off one track of a multi-track recorder, with the musician's knowledge, of course.

The problem then becomes how to capture the performance, since the Emax II only allows a couple of minutes of sampling even with maximum memory. As you might expect, the best sound does indeed come from playing a sound directly into the Emax II, but that is only practical for certain types of samples. Recording a musician on quality tape represents only a small compromise in sonic quality, and gives you much more material to work with. More affluent musicians do their sampling with PCM adapters connected to VCRs, RDATs, or on the audio tracks of Beta or VHS Hi-Fi recorders (whose performance often exceeds that of analog tape), or quality reel-to-reel recorders with noise reduction.

Music School Resources

Don't overlook students as possible sources of acoustic instrument sounds. Many music school students will be happy to work in a high-tech environment so that they, as well as you, can learn about sampling. Remember, though, that this kind of work is very difficult. It takes a particular breed of musician to play parts and notes over and over again while you adjust levels and try to get a good sound. It helps if the musician knows a bit about sampling, and while playing a note tries to give you a few seconds of highly consistent tonal quality to make looping easier.

Sampling Sources

Sampling from records, tapes, and CDs is good for practicing. However, I have a strong philosophical bias against actually using these sounds. The purpose of the sampler is to create new and exciting effects, not just imitate what already exists. You can take Phil Collins' drum sound off a CD...but so can everybody else. Still, for learning what makes a good sample, sampling from existing source material works very well.

Multi-sampling Frequencies

When multi-sampling, choose the frequencies at which you sample carefully. For example, with a guitar, the same note can be played at several places on the neck. Since most parts are played with fretted strings, it's often best to sample fretted, rather than open strings. Fretted strings also have a shorter decay, and tend to be somewhat easier to loop since the act of fretting dampens some of the harmonics.

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Session No. 3: SOUND EFFECTS

One of the most wonderful aspects of sampling is the ability to take real-world sounds, stuff them in a sampler, and play them back as actual pitched notes. Animals, industrial noises, acoustic environments, and much more can be used in all kinds of musical contexts. Hitting a metal garbage bin with a crowbar and sampling that sound makes quite a backbeat for your rhythm tracks.

Sampling these kinds of sounds, though, requires a different outlook from sampling acoustic instruments. The basic idea is to go out on a sampling expedition armed with some kind of quality recorder, and as always, have lots of patience.

Professional sound designers will often take something like a Nagra portable tape recorder (known for its ability to record high-quality sound in the field) or a complete PCM-F1/portable VCR combination out into the field. The PCM-F1 is a battery-powered digital audio adapter that records digitized audio on the video tracks of a VCR. Since it can be battery-powered, teaming it with a portable VCR frees the user from power lines.

Since I don’t do sound design for a living, I saved some bucks and invested in a Sony Walkman Pro recorder. This is about the size of a standard Walkman, but it records in stereo, handles chrome or normal tape, has a pause control and index counter, includes Dolby noise reduction and an LED peak VU meter so you can get a sense for whether a signal is being recorded at a proper level, and comes with a reasonably high-quality condenser mic. It also has a transport that is quite immune to vibration. One of my favorite features, though, is size. I can stick the Walkman, headphones, extra set of batteries, microphone, and a couple of cables in a small bag and carry it with me wherever I go. Sometimes I just leave the thing strapped around my shoulder, ready to record. In many ways, it’s like carrying around a camera.

The best way to learn what makes a good sample is to do as much sampling as possible. Sample industrial sounds, people, TV, commercials, whatever. Listen back to the tapes and keep a log that correlates the index counter setting to particular sounds. Then call up sounds that seem well suited to sampling and stuff them into the Emax II. Just about anything works for percussive effects, but some sound effects can also be tuned for melodic and chordal uses.

For example, I recently had a chance to tour one of Yamaha’s piano factories. Sensing that this might provide some good samples, I took my Walkman along. While many of the machine sounds were spoiled by background noise, I did get a stunning sample of a piano testing device that sequentially played all the keys at a very high rate of speed. Some drill presses and other sounds were also pretty hip.

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Sampling Suggestions

If you want to go sampling, I have some suggestions:

■ Always Carry Your Recorder With You

Craig's Law: The best possible sample you have ever heard will always happen when you don't have your tape recorder, or don't have the tape set up and ready to go.

■ Keep A Log of Your Tapes

This is important so that you can locate particular samples fairly rapidly. Careful, though; there's a world of sound out there and you don't have to save everything. Unless you need sound effects on a professional basis, try to keep your collection of tapes manageable.

■ Keep A Windscreen Handy

A lot of outdoor samples are ruined by wind noise. And if you take samples while you are walking or moving, a wind screen is a virtual necessity.

■ Try To Get As Much Level As Possible On Tape

If you're using an inexpensive tape recorder, any extra noise can be a real problem, although you can use the Emax II's internal filter, VCA and envelopes to reduce and sometimes eliminate this noise.

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Part Two: MANIPULATING THE SAMPLE

Minimizing Timbre Differences

There are several ways to minimize timbre differences between adjacent samples as you play across the keyboard.

■ ***Record Lots of Samples***

This is known as multi-sampling, and is most practical with relatively short samples.

■ ***Overlap Samples and Use Positional Crossfade***

This can help even out the sound to a great extent; refer to **PRESET DEFINITION 5** (“Positional Crossfade”) for further information.

■ ***Carefully Adjust Filter Tracking***

When recording the sample, add some extra brightness via equalization. When adjusting the voice filter parameters on playback, set the filter tracking somewhere between 0.00 and 0.50, then adjust the filter cutoff for the desired overall timbre. This helps keep an even timbre across the entire voice. After assigning the voices to a preset, “tweak” the filter settings (if necessary) to insure consistent timbre as your playing transitions from one sample to the next.

■ *Combine Any or All of the Above Approaches*

Reducing Noise With Envelopes

You probably already know how envelopes can be used to create dynamics in an otherwise static sound. However, the envelope generators on the Emax II can do much more than just add dynamics. One of their most useful purposes is to help reduce any noise that may be in the sample itself.

To illustrate how this works, let’s say you have a plucked sample from a guitar, bass, or piano recorded on tape. It’s a great sample, but has quite a bit of residual noise in the background which becomes most apparent as the sound decays.

One approach to dealing with this problem is to even out the level of the decaying part of the signal going into the Emax II, either by manually turning up the Emax II’s sample input gain control, or via compression (limiting) of the original signal. This produces a signal with more limited dynamics.

We can now use the amplitude envelope generator to restore the original dynamics. Set the decay and release to simulate the original decay curve, and you’re set. This reduces noise for the same reason that a standard audio expander reduces noise; as the decaying part is attenuated, any noise that’s part of the signal is attenuated as well.

For the ultimate in noise reduction, it would be even better to loop the sustained part of the sample, also saving memory, and then envelope the looped portion. However, if the

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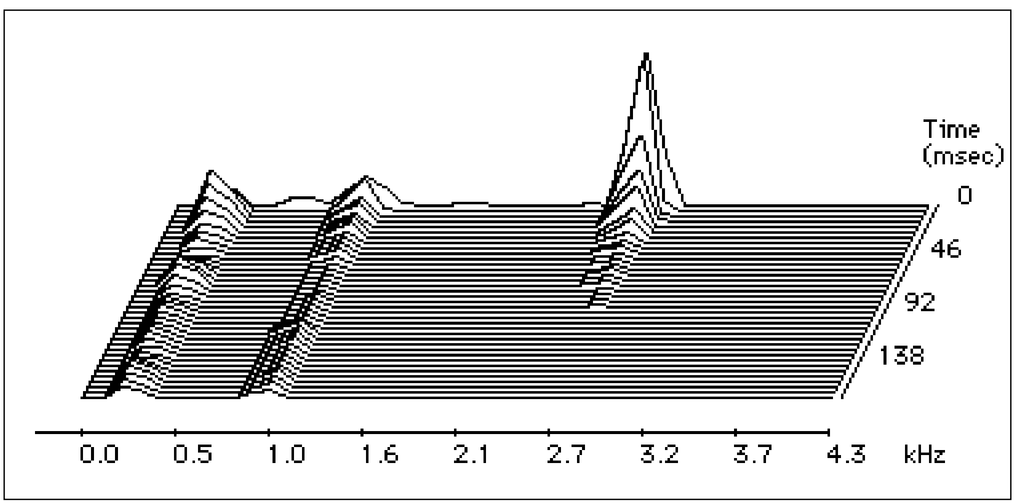
looping isn't good, the sound would be less realistic than recording the regular sound along with its natural decay.

Suppose we sample a sound that has a fairly bright pluck-type attack, then settles down into sustain. Further assume that the noise is noticeable during the sustained portion of the sound. Carefully setting the filter envelope can minimize this noise. Since this kind of problem is most common with bass (there are fewer high frequencies to mask any noise), we'll use a bass sound to illustrate this technique.

First, since you're dealing with a plucked, percussive sound, set the filter envelope for a short attack. Also, since we have a bright attack, we want to make sure that the envelope interacts with the filter cutoff in such a way that the filter passes the full bandwidth of the signal during the initial attack.

In many ways the initial attack, which contains complex harmonic and amplitude changes, is the most important part to the ear in identifying a sound. It takes only about 50 to 150 milliseconds to recognize the difference between, say, a trumpet and a guitar. After this period of time, the signal tends to settle down into a more repetitive pattern that is of less interest to the ear. If some other instrument begins playing during the decay of the original sound, your ear will tend to focus its attention on the new sound.

We can take advantage of this psycho-acoustic phenomenon to reduce the apparent amount of noise that occurs after the attack. Consider the diagram below, a spectrum analysis of the signal being sampled. Here we can see that the initial pluck consists of lots of high frequency energy—energy that tends to mask any noise. After about 100 milliseconds of this initial burst, the signal loses high frequencies and consists mostly of lower frequencies. Most of the high frequencies are contributed by the noise. This means that we can lower the filter cutoff frequency and therefore, cut out the high frequencies, where the noise is most noticeable,while leaving the low frequencies, where the signal is, virtually unaffected.



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The way to set the envelope is as follows: With no envelope applied to the filter, play the sampled sound and wait for the sustaining portion of the sound. Now adjust the filter cutoff so it lets through most of the signal, but attenuates the noise. This may take some trial and error before you obtain the desired result.

Now increase the effect of the envelope on the cutoff frequency. Start off with the attack at 0 time, sustain at 0 level, and the decay set to some minimum, but noticeable value. The release is not all that important for now. Repeatedly hit the note and adjust the decay until it is long enough to let through the sounds and high frequencies associated with the initial pluck. Adjust the envelope level to make sure that the cutoff is high enough to let these sounds through, but not so high that the hiss gets through.

I've noticed that sounds with few harmonics are particularly prone to noise problems. Applying the right kind of envelope to the filter will help dramatically towards lowering the overall amount of noise.

Signal Processing with the Emax II

■ ***Pitch Transposer with Regeneration***: To make a voice sound as if it is being pitch-shifted downward and regenerated, hit three or four adjacent keys from the same sample at the exact same time (try this with a sampled sentence for an obvious example of how the effect works).

■ ***Time Delay Special Effects :*** Set a voice for non-transpose, then press several keys at almost the same time—you will hear all sorts of flanging, chorusing, and other delay effects.

■ ***Echo***: Loop a complete voice so that it begins repeating as soon as it ends. Then, set the VCA decay so that the repeated voices get softer as you hold down the key. The effect is very similar to echo, with the exception that different notes will have different echo rates. Or, set a voice for non-transpose, and hit a new key every time you want to hear an echo.

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Part Three: MULTI-SAMPLING

You would think that sampling an instrument would be a piece of cake, right? After all, you just play a note, sample it, and there it is—you're done.

Well, not quite. First of all, a single note transposed over the range of the sampler is going to sound very unrealistic. That's why most samplers allow for multi-sampling, or placing samples every few notes. This means that no note will have to be transposed over too wide a range. The tradeoff, of course, is that more samples use up more memory. As a result, it's usually best to concentrate the greatest number of samples toward the most-played range of the keyboard. For example, I've noticed that some bassy sounds can be transposed downwards to an octave or so without sounding too unnatural. Likewise, for some sounds like a cello, you're not going to play too much in the top octave. Therefore, one sample might suffice for, say, the top octave and a fifth. You might also be able to get away with a tighter loop on this sample as well since it will not be played as often as the other notes.

However, unless you take a sample for every note, there is going to be a noticeable difference in sound when you cross over from one sample to the next. For example, suppose you've sampled a sound at C3 and another at C4. You transpose the C3 sound up to G3, and transpose the C4 sound down to G#3. The note at G3 will tend to sound bright, and the note at G#3 will tend to sound rather dull, compared to their respective original samples. As a result, when you cross over from the G3 to the G#3 you'll notice a distinct timbral difference.

There are two common ways to minimize this problem. One involves the use of filter tracking, and the other requires equalizing the sample as it gets recorded into the Emax II.

Filter Tracking

If you're not sure what filter tracking is, refer to the Dynamic Processing module, VCF section. In this particular application, you will generally want to set the tracking so that the filter is open all the way (maximum brightness) for the lowest-transposed notes of the sample, then closes down as you play towards the highest-transposed notes of the sample. Thus, the lower notes are allowed to play as bright as possible, whereas the high notes, which tend to sound bright and sometimes shrill due to the transposition, are mellowed out a bit. This minimizes tonal differences as you transition from one multi-sample to the next. In fact I have used this technique so successfully with some analog synthesizer patches that even though there may be only five samples across the keyboard, which means that each sample has to be transposed over an octave, it is very difficult to tell where one sample begins and the next one ends.

One trick is to record the original sample with a bit of treble boost or aural enhancement (Aphex Exciter, EXR Projector, etc.). This extra brightness, when transposed down-wards, gets reduced to what is perceived to be a normal amount of brightness on the lower transposed notes. Sure, the high notes will be very bright, but that's what filter

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tracking is for. By setting the tracking to zero (i.e. the filter frequency stays the same regardless of where you play on the keyboard), or slightly higher, these high notes can be tamed to the point where they sound acceptable.

Parametric Equalization

One of the problems with transposing a signal up or down is that it changes the characteristic formant of an instrument. The formant is a frequency at which an instrument tends to have some kind of resonant peak that gives the instrument it's distinctive character. For example, I recorded a classical guitar once that had a very pronounced peak around 200 Hz. This gave a bassy, rich sound. However, if you sampled that guitar and transposed it up an octave, then the resonant frequency would transpose upward as well to 400 Hz, and the guitar wouldn't sound so bassy anymore. Conversely, transposing downwards an octave would lower the frequency to 100 Hz, which would be very bassy.

Sometimes it's possible to equalize the signal being sampled to minimize the formant shift problem. Generally, the tool of choice for this task is the parametric equalizer, an equalizer that can boost or cut a specific frequency (or group of frequencies) by a specific amount, usually +/- 12 to 15 dB. Reducing the formant doesn't affect the original sound very much, but can make a big difference on the transposed sounds.

Parametric Equalization is very much a trial-and-error sort of process, which makes a good argument for having samples originally recorded on tape so that you can experiment with them as you record them into the sampler.

The Art of Sampling is an excerpt from a forthcoming book by Craig Anderton and is used by permission.

FIX IT IN THE MIX
WITH A SAMPLING KEYBOARD
by Craig Anderton

Uh-oh. You're doing a mix and you start hearing things...like clicks, and pops, and out-of-tune parts, and even a fluffed note on the piano section in the second verse. And the singer didn't quite make that high D

In the pre-sampling days, these problems would have meant either redoing a track, or getting intimately involved with a razor blade and splicing tape. But no more. Although most people look upon samplers as sampling *keyboards*, this is only one way to look at that these wonder boxes. In the studio, for example, you might be better off thinking of a sampler as a bunch of little digital audio recorders, controlled by switches that—just by coincidence—come in a piano keyboard's configuration. With our consciousness thus properly shifted, we're ready to "fix it in the mix" and get into some serious cut-and-paste work. Most of these applications involve recording a part from tape into the sampler, manipulating it within the sampler, then recording the part back on tape in place of the flawed original. Thanks to16-bit sampling systems, the loss in fidelity that occurs during the transfer is minimal—and is certainly preferred to whatever glitch you wanted to eliminate.

STOP THAT PLOSIVE!

One of the quickest ways to spoil a great vocal take is with a popped "P" or "B" sound. Punching one word can be a Big Deal, since it's going to be difficult for the singer to get the same feel and match the level. Besides, what happens if sloppy engineering on the original session let through that exploding "P" sound, yet you have to solve the problem during the mix when the singer isn't around to re-do the part?

Simple. Sample the offending word, complete with pop, into the sampler. Next, use the truncation function to cut off the pop at the beginning of the sample. Well, maybe not the entire pop; usually you want to leave in just enough to retain a bit of the P sound. When the plosive has been properly tamed, send the truncated sample back into the recorder and punch it in over the original word.

What? You say it's a short word and who can do that quick a punch? Then transpose the sample down an octave, slow the recorder down to half speed, and do the transfer. Or if the word is embedded in the middle of a phrase, sample the entire phrase and copy it to a second sample. Then, truncate the end of the first sample just before the pop, truncate the beginning of the copy just after the pop, and splice the two samples back together again and record it over the original phrase. Usually any pops are short enough that the few milliseconds lost in the splicing process won't significantly affect the overall timing.

THE TRANSPLANT

Thanks to the repetitive nature of most pop tunes, melody lines and parts often repeat in more than one place. Recently, I was mixing an album project where, in the second verse, there was an audible electrical click on one of the notes in a part. The musician who played the part was on tour, so she wasn't available to replay the line.
I checked back in the song and sure enough, there was an almost identical line in the first verse. We sampled it, and recorded it over the line with the glitch. The process worked so well it's impossible to tell which is the "transplant" and which is the original.

FIX IT IN THE MIX

TESTING A SPLICE POINT

Do you get as nervous as I do when slicing up a multi-track master tape? It's no fun to slash a 2-inch tape into pieces, only to find that maybe that snare hit wasn't the right splice point after all.

The usual procedure is to test your splice by recording a mix of the multi-track on to quarter-inch tape, and then doing your "practice" splices on the quarter-inch version. Once you find the right place, you then do the "real" splice on the multi-track master. This is all well and good, but no longer necessary if you have a sampler with a splicing function. Start by sampling a couple of seconds around each of the potential splice points. Truncate the samples to the splice points, then use the splice function to glue the truncated samples together. If it works, great—go do the splice. If not, re-truncate the samples and try again. When you find the right splice point, proceed to slicing and dicing the multi-track master.

FIXING THE OUT-OF-TUNE NOTE

Here is where a sampler can truly save a session. The singer has just given a stunning performance except...that last note was just flat enough to blow the take.

Or was it? Sample the note into the sampler, and punch over the existing bad note. Only this time, artfully move the pitch bend wheel at just the right moment to bring the note up to pitch. This is the kind of salvage job that can make a singer your friend for life.

Incidentally, you can also use this technique during the recording process to help a singer hit a note outside of his or her range. Sample the highest note the singer can sing, and when it's time to hit that ultra-high note, play it on the sampler by transposing the sampled sound upward.

STEADY, STEADY...

What happens if the kick drum is just a shade off tempo? You guessed it: assuming the kick is on its own track, you can sample the sound, then during the transfer back to tape hit the sampler key at the right time so that the bass drum falls right into the pocket.

CURING THE NO-VIBRATO EFFECT

During mixdown, I heard a vocal line that was just crying out for a real heavy, thick, even vibrato at the end of a phrase. Unfortunately, not all singers can provide this...but a sampler can. Sample the note, and when you record it over the existing note, turn up the mod wheel for the desired amount of vibrato. You can even do some really bizarre things like simultaneous pitch bend and vibrato...pretty bionic sounding.

FIX IT IN THE MIX

BUT I'VE RUN OUT OF DDLS!

You've got one DDL chorusing the lead vocal, one on the drum overhead mic, and your very last DDL doubling a sax solo. At this point the rhythm guitarist muses about how it would be nice to have a little eighth-note slapback every time he hits that fancy B maj 7th chord...you know, a tail of about three or four echoes that fade out nicely into the drum fill.

Here's where velocity keyboards come in handy. Sample the rhythm guitarist's fancy chord, and play the *sample* when the echo is supposed to come in. Play it once for each echo, and play the note a little softer each time so that the echo's level diminishes properly. If you have an open track, the echo can be recorded there and you won't have to think about it any more. If all your tracks are full, find someone to play the part in real time during the mix itself.

I must confess to using this technique a lot, even when I haven't run out of DDLs, since it allows for echo effects that would be just about impossible to achieve with standard DDLs (such as strange polyrhythms, echoes that get louder and then softer, and so on).

TESTING, TESTING...

And since we're in a mix-oriented frame of mind, we want to make sure everything is aligned level-wise. Sample a 100 Hz, 1 kHz, and 10 kHz tone into the sampler; they won't take up much memory since all you have to do is loop a few cycles. Then make up a little sequence that spits out these tones in the right order for calibration. Some samplers will have a hard time dealing with a 10 kHz signal, but if you sample at a fast sample rate and sample the tones at a moderate level, you should have something good enough to use. Keyboard players who would like to endear themselves to guitarists can sample an E = 164.8 Hz tone into the sampler. When transposed down an octave, this produces the right frequency for tuning the guitar's low E string. When transposed up an octave, you can tune the high E. Notes for the other guitar strings (A, D, G, B) lie within this two-octave range. You can even make up a little tuning sequence—say, 10 seconds on each note needed for tuning.

AND THERE'S MORE!

While technically not a fix-it-in-the-mix application, some samplers [Emax II, ed.] let you slow the sampling rate way down, thus allowing very long sampling times. This can be a real boon to songwriters. I don't know about you, but I often come up with words by playing a chorus or verse over and over and over again until I've got most of the words filled in. In the Dark Ages, this meant rewinding your tape a lot. Now TASCAM, Fostex, and others offer "block repeat" functions where the tape will rewind back to a set point, play forward to another set point, rewind, play, and so on. But why torture your tape deck transport? Sample the entire verse, chorus, or whatever, and loop it.

FIX IT IN THE MIX

END OF LINE

Samplers make great musical instruments, no doubt about that. But they also make versatile digital recorders with very useful editing capabilities. Next time you need to fix something in the mix, see if a sampler won't do the job in the quickest, simplest, and most cost-effective manner.

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MULTI-TIMBRAL OPERATION
By Gerry Bassermann

WHAT IS SUPERMODE?
Supermode means that you can turn your Emax II sequencer into a MIDI matrix, capable of routing any of 16 MIDI channels to any of 100 Emax II presets simultaneously. Since there are 50 available sequence locations, you can have up to 50 such setups.

TO ACCESS MULTIPLE PRESETS THROUGH MIDI:

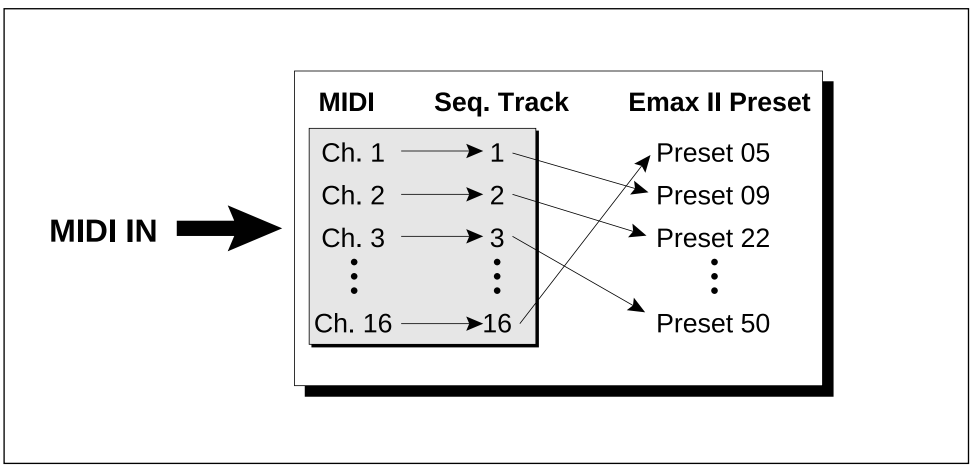
1. Turn Supermode on (Setup/#6).

2. Select Supermode Map (Sequence 00) (see below).

3. Hit Setup #2 (Track Preset).

4. Assign presets to the desired tracks.

When Supermode is on, all MIDI channel and mode settings within presets (Preset Definition #7) are ignored. Instead, incoming MIDI data will be individually channelled into respective sequencer tracks. Remember, MIDI channel numbers correspond to the same track numbers.



MIDI Channels are always routed to the same numbered Track.
The SuperMode Map directs a Track to the appropriate Preset.

WHAT IS A SUPERMODE MAP?
It is the sequence, labeled 00 on all E-mu factory disks, in which all 16 tracks have been recorded with nothing for one second. This is so that there exists a 'sequence' where you can assign presets to any one of the sixteen tracks. You can't make assignments to empty tracks.

MULTI-TIMBRAL OPERATION

It is also very easy to create your own Supermode Map. First, decide which MIDI channels the Emax II should receive. Then choose an empty sequence, turn Supermode on, put the tracks which correspond to those MIDI channels into record mode, then record one second of silence. Make the proper track/preset assignments, as described above, for whatever external sequencer project you're working on. It's a good idea to change the name from Supermode Map to the title of your piece, so that later, when you need to select the correct Supermode routing, you can easily find your work.

After you've slaved the Emax II sounds to your sequencer, and finished working on the music, it is possible to download all the data into the Emax II so you won't have to carry the dedicated sequencer around with you. In doing this, you're actually turning that Supermode Map into a real sequence. Remember, when downloading, that the Supermode Map was only recorded for one second and your download will terminate after that time if you don't first activate Auto Extend (Setup # 3).

TO DOWNLOAD SEQUENCES INTO EMAX II VIA MIDI:

1. Select SuperMode Map with proper track/preset assignments.

2. Turn Auto Extend on (Setup#3).

3. Turn MIDI Start/Stop on for current preset (Preset Definition #7).

4. Choose MIDI clock (Manage#2).

5. Hit record, play, and enter on the Emax II.

6. Hit play on source sequencer.

MULTI-TIMBRAL BANKS

One of the most powerful ways to use the Emax II sound library is to create an arrangement of sounds drawn from different disks and combine them all on your own disk. It is also one of the most asked questions of our customer service department.

HOW?

Creating a multi-timbral bank is easy. It's basically a matter of loading a preset from one bank and saving it into another.

1) Decide which presets you want in your multi-timbral bank.

2) Let's start with a clean slate, just to avoid confusion. So, clear the bank (**MASTER 4**).

3) Activate **PRESET MANAGEMENT 1**, Load Preset.

4) Use the data slider or inc/dec buttons to select the bank containing the first preset you want to use, then press **ENTER**.

5) Now use the data slider or inc/dec buttons to select the first preset you want, then press **ENTER**. You can now rename the preset if you desire.

6) Go back to step 3 and select the next preset that you want on your multi-timbral disk. Continue until you run out of memory or have enough presets.

7) Save the bank (**PRESET MANAGEMENT 8**), or your work will be lost.

NOT ENOUGH SAMPLE MEMORY

Problems come when you see the flashing error message;

"`Not Enough Sample Memory`".

This message is telling you that you have run out of sample memory. On an Emax II with only one Megabyte of RAM, this message will appear frequently when trying to load presets.

When this message appears, you have two alternatives;
1) Get more memory, or
2) Make room for the new preset by erasing unwanted presets, voices, parts of voices, making loops shorter or by using Sample Rate Conversion.

EXPAND YOUR MEMORY!

By far the easiest method is #1. Get More Memory! The Emax II can be memory expanded up to 7 or 8 Megabytes of RAM. To be honest, it's really a pain trying to create multi-timbral disks with only one Megabyte. On the other hand, creating a multi-timbral disk with 3 or more Megabytes is quick and easy. It's the only way to go.

MULTI-TIMBRAL BANKS

THE HARD WAY

No money left over for memory? OK, OK. Here's what you do.

Check the preset sizes of the presets you want to load (**PRESET MANAGEMENT 7**) before loading them. If the total number of samples in the presets you want to load exceeds the memory capacity of your Emax II (524,244 samples on a 1 Meg unit), you must take action.

The basic technique for combining impossibly large presets is Memory Reduction. Either erase voices and transpose your other voices further up and down the keyboard, erase voices that you don't absolutely need, make shorter loops or use Sample Rate Conversion in the Digital Processing module to reduce the memory required for each voice. Maybe you can use a combination of all three methods. All these methods take time, but will do the trick until you can save up for that memory expansion.

ORGANIZING YOUR MULTI-TIMBRAL BANK

If you are going to use the multi-timbral bank with an external sequencer, here are a couple of tips.

Create a system where certain types of sounds are assigned to certain preset numbers.

For example; **Preset 1 - Keyboard Sounds**
Preset 2 - Bass
Preset 3 - Drums
Preset 4 - Horns
etc.

Creating an organized system will allow you to easily swap sounds around if they don't work out in the sequence. You can try out several bass guitars, for instance, to hear which one sounds best. All you would have to do is "Load Preset" over the existing bass.

To make life even easier, assign the presets to their corresponding MIDI channels.

For example: **MIDI Channel 1 = Preset 1**
MIDI Channel 2 = Preset 2
etc.

Now you have one *less* thing to remember.

EXPERIMENT!

Combining sounds together is a true art. Take the time and try out various instruments in a given part. Many times an instrument that sounds thin and wimpy on its own will perfectly fill out the mix. And, too many "fat" sounds can create "MIDI Soup", a huge stifling sound. Keep experimenting until you get everything just right. Creating Multi-Timbral banks fulfills the promise of sampling technology by letting you have *any* sound, where and when you want it. Take advantage of it.

SPECTRUM SYNTHESIS

OVERVIEW

Sine waves are the most basic components of sound. All waveforms can be broken down to their component sine waves. *Additive Synthesis* is the process of constructing sounds by adding sine waves together in varying amounts.

The pitch of any sound is determined by its *Fundamental Frequency*. The fundamental is the lowest frequency component and usually the one with the greatest amplitude.

The other components of a complex sound are known as *Partials*. The way the different partials are mixed together with the fundamental frequency determines the tonal quality of a sound. Normally the partials are based on whole number multiples of the fundamental, 1x, 2x, 3x, etc. When partials are whole number multiples of the fundamental, they are called *harmonics*. Partials do not have to be exact multiples of the fundamental. In fact, many natural sounds have some partials which are not exact multiples. Bells and cymbals are examples of sounds in which many of the partials are detuned from a normal harmonic relationship.

So, sine waves can be added together in various amounts in order to form complex waves. But the waveform of most natural sounds varies quite dramatically over time. Simply being able to create any waveform is not enough. The waveform of a sound must *vary* over time or the sound is static and uninteresting.

Enter the Time Slice. Emax II's Spectrum Synthesis allows you to specify up to 24 complex waveforms, (called Spectrums) and to either step or fade through them during the course of the sound. The 24 waveform locations are called *Time Slices*, since they represent the waveform at that slice of the sound. This method allows more complex and interesting sounds to be created. In addition, each partial (sine wave) can have a 24 point envelope that determines the pitch of that partial over the course of the sound.

Each Emax II voice can consist of two synthesized timbres, a synthesized timbre and a sampled sound, or two sampled sounds. The resulting voice can then be processed using Emax II's extensive Dynamic Processing features and then placed anywhere in the stereo field. As an example, the primary voice could be the attack of an overblown flute while the secondary voice could be a synthesized timbre of your own creation. When played together, the total effect is that of a natural sound, but a totally NEW natural sound. The possibilities are endless!

1. Select **Digital Effects 6** and the display will say:

```
Synth Changes
Data!   OK?
```

If it's OK to write over the current voice, press **YES**. If not press **NO** to return to the module identifier.

SPECTRUM SYNTHESIS

If you pressed **YES**, the display will now say:

```
Synth Options
[0-9] / Slider
```

2. Use the slider to catalog the various options.

0. Synthesize!
1. Freq. Spectrum
2. Time Slice
3. Ampl. Contour
4. Pitch Contour
5. Pitch/Ratios
6. Interpolate
7. Load Backup
8. Save backup
9. Erase

SPECTRUM INTERPOLATION SYNTHESIS

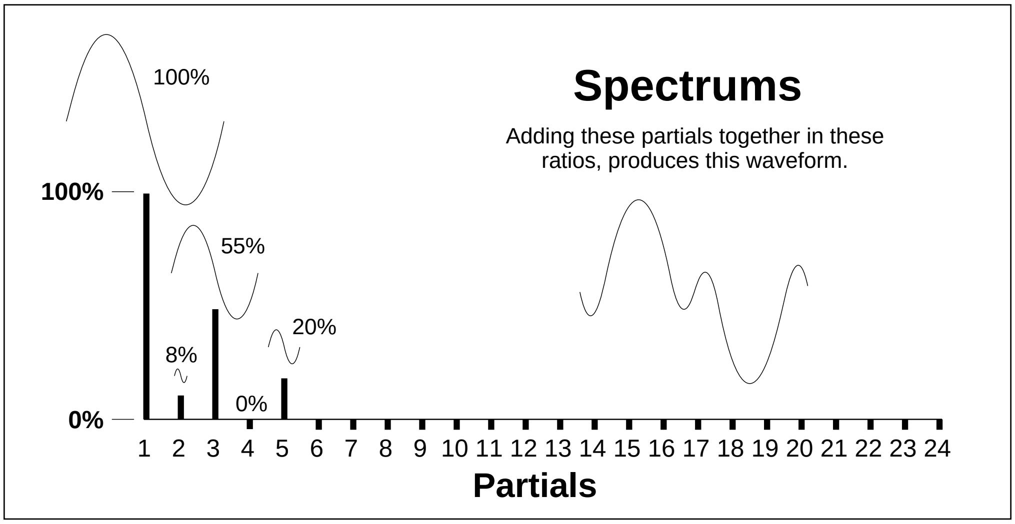
Spectrum Synthesis can create dynamically shifting timbres using additive synthesis techniques. Virtually any imaginable waveform may be created. Up to 24 sine waves are added together to form *Spectrums*. Spectrums are then placed at up to 24 *Time Slice* locations over the length of the sound. As the sound is played, it either steps or smoothly fades through the spectrums. Sounds may be any length (up to the maximum memory) and can be customized by using the sophisticated complement of edit parameters that are included. After a sound has been created it may be spliced, combined, and processed in the same manner as a sound that has been sampled into the Emax II.

ABOUT SPECTRUMS

The spectrum synthesizer development disk contains 95 pre-programmed spectrums for you to use, or you may create and store your own. Each spectrum may contain up to 24 sine waves, with each wave having its own specific amplitude (volume). These sine waves may be tuned to any ratio of the fundamental frequency. These ratios do not need to be harmonically related to the fundamental pitch of your sound, hence the sine waves are called *partials*, rather than harmonics. Varying the amplitude of a specific partial will result in a timbre change. Generally, the greater the amplitude of the higher partials, the brighter the resulting sound will be.

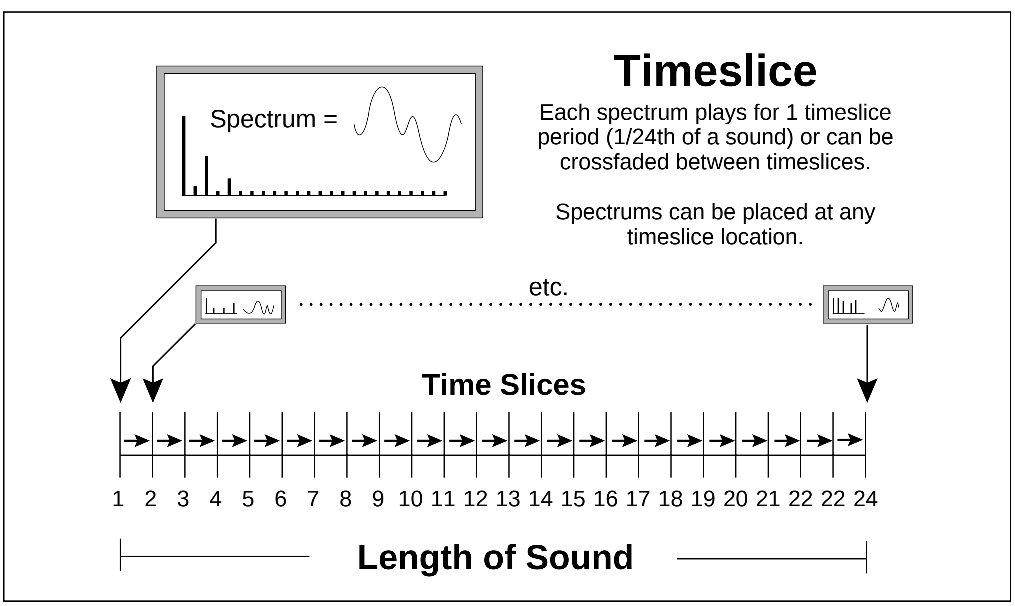
SPECTRUM SYNTHESIS

A spectrum may be placed at any of the 24 "time slice" locations. Placing a spectrum at a given time slice will determine the timbre of the sound at that point in time.



ABOUT TIME SLICES

Once you have determined the length of the sound that you wish to synthesize, Emax II automatically divides the sound into twenty-four equal sections. Each section is called a time slice. A time slice contains the spectrum at a given point in the sound. A *Time Slice is simply a Spectrum which has been placed at a Time Slice location.* You can create a time slice and save it as a spectrum, or place a previously created spectrum at a time slice.



SPECTRUM SYNTHESIS

CREATING SOUNDS

The best way to learn about the spectrum synthesizer is by creating a sound, so load the Spectrum Synth development disk into your Emax II and follow these simple steps...

DEFINE THE LOCATION

First, we must define the location and the length of the sound. This can be accomplished in one of three ways:

■ Select a preset 95-99 on the Spectrum Synth development disk. These contain empty samples.

■ **1.** Clear all memory. (**Master #4**)
2. Enter the **SAMPLE** module
3. Press **#4** on the keypad and select the length that you want the synthesized sound to be. (The longer the sound, the more time it will take to synthesize.)
4. Press **#7** on the keypad to force sampling.
5. Press the **SAMPLE** button to exit the sample module

■ Write over an existing sample. The synthesized sound will replace the sampled sound. (Remember to back up the sample first!)

START SYNTHESIZING

Now you that you have defined a location and length for the sound, its time to start synthesizing!

1. Enter the **DIGITAL PROCESSING** module and select the voice you just created.

2. Press **#9** on the keypad to access the **Digital Effects** menu.

3. Press **#6** on the keypad to access “Spectrum Synth”. The display will read “`Synth Changes Data! OK?`” Press **YES** to continue. (**NO** to exit)

4. Now you are in the Synth Options menu. Press **#1** (Freq. Spectrum) on the keypad andhit **ENTER**. The display will show Spectrum #1 and you will see a series of vertical lines. These lines represent the amplitudes of the 24 partials contained in the spectrum.

```
Spectrm: Select
SP03 ..................
```

Note: Because of the limited display resolution, each dot in the bar graph represents 14% of full level. If a partial has a level of less than 14%, the bar will appear to be 0%.

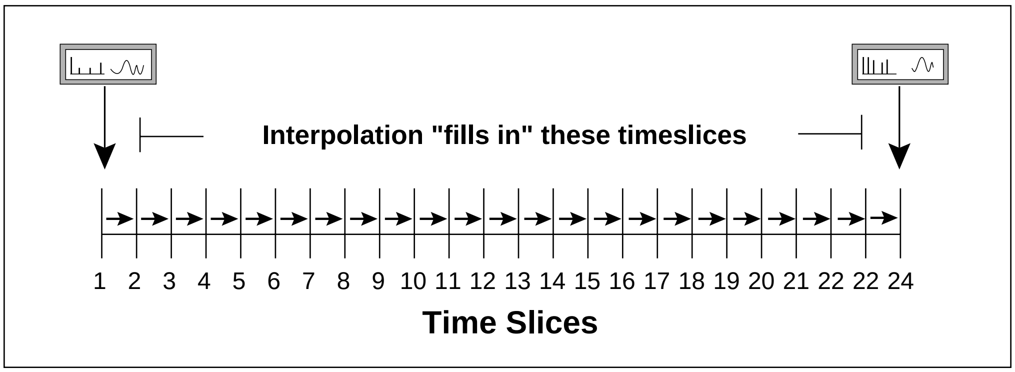
5. Press **NO** until the display says “`Spectrum->TS Y/N`”, then press **YES**.

SPECTRUM SYNTHESIS

6. The display now says "`Move SP01 to: TS01...`", and the ENTER light will be flashing. Press **ENTER**. Now Spectrum #1 has been placed at Time Slice #1.

INTERPOLATE

You have now defined a tonality for the beginning of the sound. To define the tonality at the end of the sound repeat steps #4 - #6 but in step #4 select Spectrum #2 instead of Spectrum #1. (Use the slider, yes/no buttons, or the keypad to select spectrums 1 - 99). In step #6, move Spectrum #2 to Time Slice #24. To fill in the "middle" of the sound, we use the "Interpolate" function. This tells the Emax II to create all of the spectrums between time slices 1 and 24.



1. Press **#6** (Interpolate) on the keypad. The display reads "`Interpolate from: TS01...`".

2. Press **ENTER**. The display now reads "`Interpolate to: TS01...`". Use the slider to select Time Slice #24 and press **ENTER** to execute the interpolation.

3. Now press **#2** (Time Slice) and slowly move the slider. You will see that the Emax II has created time slices 2 - 23. These time slices represent the gradual transition from time slice #1 to time slice #24. Move the slider all the way to the bottom. The display will show "`None - exit`". Press **ENTER** to return to the Synth Options menu.

SYNTHESIZE!

At this point it is time to tell the Emax II to generate the sound that you have defined. This is done by pressing **#0** (Synthesize!) on the keypad. There are two modes of synthesis - "Smooth" and "Stepped". "Smooth" will produce a very gradual tonal change from time slice to time slice. "Stepped" will result in more abrupt tonal changes, as each time slice plays for exactly one twenty-fourth of the sound and then jumps to the next time slice. Select the "smooth" mode by using the **YES / NO** buttons, then press **ENTER** to begin synthesizing. The display will show how many time slices are left to synthesize, then prompt you to play the sound when the synthesis is complete.

SPECTRUM SYNTHESIS

SPECTRUM SYNTH OPTIONS MENU

There are ten options listed in the spectrum synthesis menu. Each of these options covers a specific function or group of functions, and some contain sub-menus and functions. When a menu option contains multiple functions, answering **YES** to the prompt **" ... ? Y /N "** will execute the function and then take you to the next level of operation. Answering **NO** to the prompt takes you to the next option in the sub-menu. The menu options are as follows:

1. FREQUENCY SPECTRUM

In this menu, spectrums may be drawn, edited, copied, and placed.

DRAW a Spectrum
(Use for quick and easy creation of spectrums)

A. Press **#1** on the keypad to access the Frequency Spectrum menu options.

B. Use the slider to select the number of the spectrum you wish to draw. Press **ENTER.**

C. Press **YES** to start the drawing countdown. You will see the display countdown 5...4...3...2...1... When the countdown finishes, the drawing process starts. The cursor position automatically moves from left to right - each position of the cursor denoting one partial. By raising and lowering the slider as the cursor moves across the screen, you can draw the amplitude of each of the twenty four partials. The bar graph displays the relative amplitude of the partials.

D. Press **NO** three times to return to the Additive Options menu.

EDIT a Spectrum
(Use for precise definition of spectrums and partial amplitudes)

A. Press **#1** on the keypad to access the Frequency Spectrum menu options.

B. Use the slider to select the number of the spectrum you wish to edit. Press **ENTER**.

C. Press **NO** to get to the spectrum EDIT screen. Press **YES** to edit.

D. Use the slider to enter the amplitude of the partials. Amplitude is shown inpercentages on the right hand side of the screen.

E. Use the cursor (**<- ->**) buttons to move from partial to partial.

F. Press **ENTER** , then **NO** twice to return to the Synth Options menu.

SPECTRUM SYNTHESIS

COPY a Spectrum
(Use for creating a duplicate spectrum which can then be modified)

A. Press **#1** on the keypad to access the Frequency Spectrum menu options.

B. Use the slider to select the number of the spectrum you wish to copy. Press **ENTER**.
C. Press **NO** two times to get to the spectrum COPY screen.

D. Press **YES** to copy. The display reads "`Copy SP01 to SP01...`".

E. Use the slider to select the number of the spectrum that you wish to copy ***to.***

F. Press **ENTER**, then **NO** to return to the Synth Options menu, or YES to make another copy.

SPECTRUM -> TS
(Use for placing a spectrum at a Time Slice location.)

A. Press **#1** on the keypad to access the Frequency Spectrum menu options.

B. Use the slider to select the number of the spectrum you wish to move. Press **ENTER.**

C. Press **NO** three times to get to the **MOVE** screen.

D. Press **YES**. The display will read "`Move SP01 to: TS01...`".

E. Use the slider to select the number of the time slice where you want to move the spectrum.

F. Press **ENTER** to execute the move.

Note: After executing a function in the Frequency Spectrum options menu, it is not necessary to exit the menu before accessing another function. For example, you may DRAW a spectrum, then EDIT, COPY, and MOVE it before returning to the Synth Options menu.

2. TIME SLICE

In this menu, time slices may be drawn, edited, copied, and saved.

Note: This is exactly like creating Frequency Spectrums, except here the Spectrums are already placed at Time Slice locations.

DRAW a Time Slice
(Use for quick and easy creation of time slices)

A. Press **#2** on the keypad to access the Time Slice menu options.

B. Use the slider to select the number of the time slice you wish to draw. Press **ENTER.**

SPECTRUM SYNTHESIS

C. Press **YES** to start the drawing countdown. You will see the display countdown 5...4...3...2...1... When the countdown finishes, the drawing process starts. The cursor position automatically moves from left to right - each position of the cursor denoting one partial. By raising and lowering the slider as the cursor moves across the screen, you can draw the amplitude of each of the twenty four partials. The bar graph displays the relative amplitude of the partials.

D. Press **NO** three times to return to the Synth Options menu.

EDIT a Time Slice
(Use for precise definition of time slices and partial amplitudes)

A. Press **#2** on the keypad to access the Time Slice menu options.

B. Use the slider to select the number of the time slice you wish to edit. Press **ENTER**.

C. Press **NO** to get to the time slice EDIT screen. Press **YES** to edit.

D. Use the slider to enter the amplitude of the partials. Amplitude is shown inpercentages on the right hand side of the screen.

E. Use the cursor (**<- ->**) buttons to move from partial to partial.

F. Press **ENTER** , then **NO** twice to return to the Additive Options menu.

COPY a Time Slice
(Use for placing the same time slice at two different locations, or for creating a duplicate time slice which can then be modified)

A. Press **#2** on the keypad to access the Time Slice menu options.

B. Use the slider to select the number of the time slice you wish to copy. Press **ENTER**.

C. Press **NO** two times to get to the time slice COPY screen.

D. Press **YES** to copy. The display reads “Copy TS01 to TS01...”.

E. Use the slider to select the number of the time slice that you wish to copy ***to.***

F. Press **ENTER**, then **NO** to return to the Synth Options menu, or YES to make another copy.

SPECTRUM SYNTHESIS

T SLICE -> SP
(Use for moving a time slice to a spectrum location)

A. Press **#2** on the keypad to access the Time Slice menu options.

B. Use the slider to select the number of the time slice you wish to move. Press **ENTER.**

C. Press **NO** three times to get to the MOVE screen.

D. Press **YES**. The display will read “MoveTS01 to: SP01...”.

E. Use the slider to select the number of the spectrum where you want to move the time slice.

F. Press **ENTER** to execute the move.

Note: After executing a function in the Time Slice options menu, it is not necessary to exit the menu before accessing another function. For example, you may DRAW a time slice, then EDIT, COPY, and MOVE it before returning to the Synth Options menu.

3. AMPLITUDE CONTOUR

In this menu you may draw, edit, and copy the amplitude envelope of specific partials.

DRAW a partial’s Amplitude Contour
(Use for quick and easy creation of the volume envelope of a specific partial)

A. Press **#3** on the keypad to access the Amplitude Contour menu options.

B. Use the slider to select the number of the partial whose amplitude contour you wish todraw. Press **ENTER.**

C. Press **YES** to start the drawing countdown. You will see the display countdown 5...4...3...2...1... When the countdown finishes, the drawing process starts. The cursor position automatically moves from left to right - each position of the cursor denoting one time slice. By raising and lowering the slider as the cursor moves across the screen, you can draw the amplitude of the partial at each of the 24 time slice locations. The bar graph displays the relative amplitude of the partial at each time slice.

D. Press **NO** twice to return to the Synth Options menu.

EDIT a partial’s Amplitude Contour
(Use for precise definition of partial amplitudes)

A. Press **#3** on the keypad to access the Amplitude Contour menu options.

B. Use the slider to select the number of the partial whose amplitude contour you wish to edit. Press **ENTER**.

SPECTRUM SYNTHESIS

C. Press **NO** to get to the amplitude contour EDIT screen. Press **YES** to edit.

D. Use the slider to enter the amplitude of the partial at each time slice. Amplitude is shown in percentages on the right hand side of the screen.

E. Use the cursor (**<- ->**) buttons to move from time slice to time slice.

F. Press **ENTER** , then **NO** twice to return to the Synth Options menu.

COPY a partial's Amplitude Contour
(Use for copying the amplitude contour of a partial to one or more partials)

A. Press **#3** on the keypad to access the Amplitude Contour menu options.

B. Use the slider to select the number of the partial whose amplitude contour you wish to copy. Press **ENTER**.

C. Press **NO** two times to get to the amplitude contour COPY screen.

D. Press **YES** to copy. The display reads "Copy AC01 to AC01...".

E. Use the slider to select the number of the partial that you wish to copy the amplitude contour to.

F. Press **ENTER** to execute the copy, then **YES** to make another, or **NO** to return to the Synth Options menu.

Note: After executing a function in the Amplitude Contour options menu, it is not necessary to exit the menu before accessing another function. For example, you may DRAW an amplitude contour, then EDIT and COPY it before returning to the Synth Options menu.

4. PITCH CONTOUR

In this menu you may draw, edit, and copy the pitch envelope of specific partials.

DRAW a partial's Pitch Contour
(Use for quick and easy creation of the pitch envelope of a specific partial)

A. Press **#4** on the keypad to access the Pitch Contour menu options.

B. Use the slider to select the number of the partial whose pitch contour you wish to draw. Press **ENTER.**

C. Press **YES** to start the drawing countdown. You will see the display countdown 5...4...3...2...1... When the countdown finishes, the drawing process starts.The cursor position automatically moves from left to right - each position of the cursor denoting one time slice. By raising and lowering the slider as the cursor moves across the screen, you

SPECTRUM SYNTHESIS

can draw the pitch of the partial at each of the 24 time slice locations. The bar graph displays the relative pitch of the partial at each time slice, from plus 1 semitone to minus 1 semitone.

D. Press **NO** twice to return to the Synth Options menu.

EDIT a partial's Pitch Contour
(Use for precise definition of partial pitches)

A. Press **#4** on the keypad to access the Pitch Contour menu options.

B. Use the slider to select the number of the partial whose pitch contour you wish to edit. Press **ENTER**.

C. Press **NO** to get to the pitch contour EDIT screen. Press **YES** to edit.

D. Use the slider to enter the pitch of the partial at each time slice. Pitch is shown incents on the right hand side of the screen, from -99 cents to +98 cents.

E. Use the cursor (**<- ->**) buttons to move from time slice to time slice.

F. Press **ENTER** , then **NO** twice to return to the Synth Options menu.

COPY a partial's Pitch Contour
(Use for copying the pitch contour of a partial to one or more partials.)

A. Press **#4** on the keypad to access the Pitch Contour menu options.

B. Use the slider to select the number of the partial whose pitch contour you wish to copy. Press **ENTER**.

C. Press **NO** two times to get to the pitch contour COPY screen.

D. Press **YES** to copy. The display reads "Copy PC01 to PC01...".

E. Use the slider to select the number of the partial that you wish to copy the pitch contour to.

F. Press **ENTER** to execute the copy, then **YES** to make another, or **NO** to return to the Synth Options menu.

Note: After executing a function in the Pitch Contour options menu, it is not necessary to exit the menu before accessing another function. For example, you may DRAW a pitch contour, then EDIT and COPY it before returning to the Synth Options menu.

SPECTRUM SYNTHESIS

5. PITCH/RATIOS

In this menu you may determine the pitch of the note you are synthesizing, and define the ratios of all of the partials. Original pitch may range from CØ to C5. Partials may be any ratio to the fundamental pitch from 1.00 to 40.99. The default pitch is C3 (middle C). The default ratio tunings of the partials are harmonics 1 through 24.

To ***DEFINE*** the ***PITCH*** of the sound you wish to synthesize

A. Press **#5** on the keypad to access the Pitch / Ratios menu. Partial #1 appears on the screen and shows a default pitch of C3.

B. Press the right cursor button to move the cursor under the C3.

C. Use the slider or the YES/NO buttons to select the desired pitch.

D. Press **ENTER** to return to the Synth Options menu.

To ***DEFINE PARTIAL RATIOS***
Use for tuning partials to ratios other than the first 24 in harmonic series.

A. Press **#5** on the keypad to access the Pitch / Ratios menu.

B. Use the slider to select the partial whose ratio you wish to define.

C. Press the right cursor button to select the partial tuning “tens” column.

D. Use the slider, YES/NO buttons or the keypad to enter the ratio you desire.

E. Press the right cursor button to select the partial tuning “hundreds” column.

F. Use the slider, YES/NO buttons or the keypad to enter the ratio you desire.

G. Press **ENTER** to return to the Synth Options menu.

Note: After defining a pitch or ratio tuning in the Pitch / Ratios options menu, it is not necessary to exit the menu before accessing another partial. For example, you may define the pitch of the synthesized sound, then set the ratios of up to 24 different partials before returning to the Synth Options menu.

6. INTERPOLATE

Use for creating spectrums between time slices, or for smoothing out the harmonic movement from one time slice to another.

A. Press **#6** on the keypad to access the Interpolation screens.

B. Use the slider to select the time slice you wish to interpolate ***from.***

SPECTRUM SYNTHESIS

C. Press **ENTER.**

D. Use the slider to select the time slice you wish to interpolate ***to.***

E. Press **ENTER** to return to the Synth Options menu.

Note: An easy way to create a sound is to place a spectrum at time slice #1, place another spectrum at time slice #24, then Interpolate from time slice 1 to time slice 24. This fills in time slices 2 through 23. If there is no spectrum at a given time slice, then there will be no sound generated when that time slice is played. It is necessary to define a spectrum at each time slice, or interpolate across "blank" time slices to synthesize an uninterrupted sound.

7. LOAD PARAMETERS

Use to load Time Slices, Amplitude Contours, and Pitch Contours from one of the three backup banks into the current memory.

A. Press **#7** on the keypad to access the Load Parameters function.

B. Use the slider to select the backup bank you wish to load. (The display will show whether the bank is empty or full.)

C. Press **ENTER** to execute the load and return to the Synth Options menu.

8. SAVE PARAMETERS

Use to save Time Slices, Amplitude Contours, and Pitch Contours (All Parameters) into one of the three backup banks from the current memory.

A. Press **#8** on the keypad to access the Save Parameters function.

B. Use the slider to select the backup bank you wish to save the current memory to. (The display will show whether the bank is empty or full)

C. Press **ENTER** to execute the save and return to the Additive Options menu.

Please Note: Parameters are not saved to disk until you have "Saved Bank".

Also Note: The parameters of a sound are not automatically loaded when that sound file is selected in Digital Processing. If you believe that it will be necessary to edit a sound after you have already started working on another one, it is advisable to first save the parameters of the sound before starting on the new one. This is the only way that the sound's original parameters can be restored. Failure to save these parameters will prevent you from re-creating your sound!

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9. ERASE PARAMETERS

The following parameters may be erased:

None
Backup #1
Backup #2
Backup #3
Current Parameters
Spectrums (*entire* palette)
All! (everything listed above)

To ***ERASE*** parameters:

A. Press **#9** on the keypad to access the Erase screens.

B. Use the slider to access the erase option you desire.

C. Press **ENTER** then **YES** to execute the erase and return to the Synth Options menu.

0. SYNTHESIZE!

Use to generate the sound that you have defined.

A. Press **#0** on the keypad to access the Synthesize! menu.

B. Use the slider to select “smooth” or “stepped”.

C. Press **ENTER** to begin synthesizing.

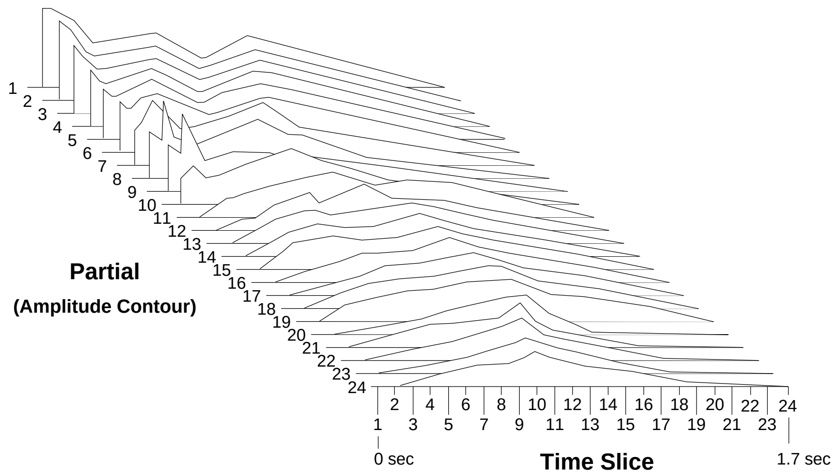
D. When Emax has finished its calculations, play your sound!

E. Press **ENTER** again to return to the Synth Options menu.

Note: After returning to the Synth Options menu, the sound will no longer be active on the keyboard. The only time you can hear your sound while in the Synth Options menu is immediately after it has been synthesized.

Also Note: There are two modes of synthesis - “Smooth” and “Stepped”. “Smooth” will produce a very gradual tonal change from time slice to time slice. “Stepped” will result in more abrupt tonal changes, as each time slice plays for exactly one twenty-fourth of the sound and then jumps to the next time slice. A sound may be synthesized in one mode and then synthesized again in the other mode. Use the mode that best suits the type of sound you are trying to create.

SPECTRUM SYNTHESIS



APPROACHING SPECTRUM SYNTHESIS

As with any type of synthesis, there are many ways of approaching the creation of sound. No matter how long you have worked with any synthesizer there are always sounds that have yet to be created. The palette of possible sounds is just so large that whole categories are bound to be missed. And remember that some of the best sounds are created by chance.

There is a great deal to be learned about sound. Spectrum Synthesis (and the Emax II in general) can be thought of as a powerful audio laboratory. Spectrum Synthesis is just one of the tools in your lab. Here we have presented some ideas so you will have a starting point with regard to your experiments. Remember to turn in your lab notebook by Friday.

SPECTRUM SYNTHESIS EXPERIMENTS

■ Start with a Spectrum at Time Slice location 01 and one at location 24. Interpolate between them, then synthesize. Add Spectrums to the sound in order to add complexity.

■ Create sounds by using the Amplitude Contour function. This may be a more intuitive way for you to synthesize sounds. Amplitude Contour lets you draw the amplitude envelope for each partial as shown in the Amplitude Contour diagram. Use combinations of Time Slicing and Amplitude Contouring.

■ Insert wildly varying Spectrums at each Time Slice location and synthesize using the Step function for wavetable synthesizer sounds.

■ Splice the sampled attack portion of a sound onto the beginning of your synthesized sound. Preset 01 (InstAttacks!) of the Spectrum Synthesis development disk contains a full range of various attacks just for this purpose. The attack portion of a sound carries a great deal of information to your ears about the nature of a sound. That sampled flute chiff might be just what your synthesized sound needs.

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■ Use the Pitch/Ratio function to set the upper 12 partials so that they are just slightly detuned from the first 12. For instance, set the ratio of partial 13 to 1.06, partial 14 to 2.06 and so on. This will cause beating between the oscillators and fatten up the sound just as it does on an analog synth. Spectrums 11 through 20 (marked “1-12” in their titles) on the Spectrum Synthesis development disk have their partials set up in this way.

■ Use the Pitch/Ratio function to change the ratios of the partials so that they are not whole number multiples of the fundamental. This will get you into the realm of bell and clangorous noises. Try changing the ratios of just one or two partials and note the effects.

■ Each partial has its own 24 stage pitch envelope. Many natural sounds have pitch variations occurring at various points during their evolution. For instance, plucked string instruments start off a little sharp, then return to pitch. Experiment by pitch bending different partials and noting the effect.

■ Mathematics and music always seem to be closely related. Try applying mathematical ratios and number series to the creation of sound. For instance, interpolate between prime numbered harmonics and non-prime numbered harmonics. Fibonacci sequences, ie. 1,1,2,3,5,8,13,21...., or squares, cubes, etc. You name it.

■ If you have Sound Designer by Digidesign, try taking a sampled sound and running a frequency analysis on it. Next, make a print out of the frequency plot and try to copy the partial envelopes using the Amplitude Contour function in Spectrum Synthesis. It probably won’t sound exactly like the real sound but will put you into the general area for further experiments.

OTHER TIPS

■ ***Save Time*** You have noticed that the Emax II takes time to perform the thousands of calculations necessary for Spectrum Synthesis. You can save your valuable time by working on short sounds (.6 sec to 1.2 sec), then lengthening it when it sounds the way you want. You will still be able to hear the sound’s characteristics and the calculation time will be cut dramatically. Also bear in mind that the fewer partials that are in the sound, the less time it will take to synthesize. Interesting sounds can be created with just a few partials.

■ ***Multisample*** Resynthesize your sound at at least 4 pitches and place them across the keyboard to minimize the negative effects of pitch shifting.

■ ***Save Parameters*** When you have created an interesting sound, use the “Save Parameters” function to save it to one the three backup banks. You can recall the parameters later and use them as a starting point to create new sounds. Remember to “Save Bank” or the parameters will be lost forever.

DIGITAL SOUND TRANSFER

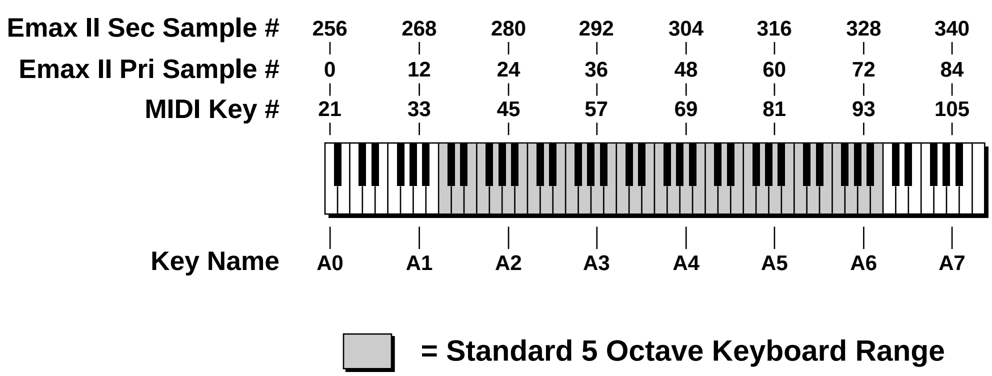
Digitally transferring samples into the Emax II is an efficient way to quickly build up your sample library. Because of the Emax II's near perfect pitch shifting, phenomenal S/N ratio, and 16-bit resolution, the result will be improved sound quality even if the samples were originally sampled on a 12-bit sampler.

MIDI SAMPLE DUMP

The MIDI Sample Dump Standard was created so that sound samples could be transferred between samplers regardless of brand. The Emax II has the ability to send and receive MIDI Sample Dumps, *but*, it cannot initiate a transfer from the front panel. It can only send and receive samples in response to a MIDI command. The *other* device must be the initiator.

In sending or receiving a sample dump, the Emax II uses *sample numbers* to identify which sample will be transmitted or where a sample will be placed. Emax II has an 88 note keyboard range. Primary Voices are sent and received using sample numbers 0-87 and Secondary Voices are sent and received using sample numbers 256-343. For example; If the Emax II is sent sample # 24, that sample will be placed in a primary voice on Emax II key A2. If it is requested to send sample # 280, then the secondary voice assigned to Emax key A2 will be sent. If there is no voice on Emax II key A2, the request will be ignored. The diagram below shows the MIDI key numbers in relation to the Emax II keyboard.

Emax II Keyboard Range



Sometimes after transferring a sound into the Emax II, the loop points may need to be adjusted. If there are clicks in transferred samples, try decreasing the Loop Start point by 2 (or 3) samples and increasing the Loop Length by 1 sample. This error is due to the way Emax II interpolates samples for pitch shifting.

DIGITAL SOUND TRANSFER

LOADING EMAX I DISKS

When Emax I disks are loaded into the Emax II, they are automatically expanded out into a 16-bit format. This dynamic range expansion coupled with the low noise of the Emax II, usually means that Emax I disks sound fantastic on the Emax II. In addition, all the preset parameters are retained. Sometimes, however, audio problems that were inaudible on Emax I, are quite noticable on the Emax II and must be manually corrected. There are also a few restrictions on Emax I sounds.

1) Emax I banks which are completely full, will not load into a 1 Megabyte Emax II. This is because the Emax II needs some memory to perform the 16-bit dynamic range expansion. The solution is to delete sequences, presets or voices until the disk loads.

2) The minimum Loop Length on the Emax II is 32 samples. If Emax I voices are loaded which have Loop Lengths of less than 32 samples, the loop may be turned off of the transposition range may be limited. The solution is to re-loop those voices.

3) The Emax II has a few features which are different from the Emax I.

Emax II uses polyphonic outputs instead of mono outs like the Emax I. Emax I voices with outputs 1-8 selected will be routed to the Main outputs on Emax II. If specific outputs *are* selected on Emax I voices loaded into the Emax II, the ouput routing will be as follows:

Emax I	Emax II
1->1	A
X->2	B
X->3	C
X->4	M
X->5	A
X->6	B
X->7	C
X->8	M

X = any channel number

Emax I voices which are in Dual Mode will play normally on the Emax II but will use two channels.

MIDI SUPPLEMENT

BASICS: QUESTIONS AND ANSWERS ABOUT MIDI

MIDI is causing a certain amount of confusion among musicians. Fear not-it's not all that hard to understand, and the Emax II makes it particularly easy to deal with. For those of you who aren't that familiar with MIDI, we'll first answer some common questions.

What does the MIDI cable do?

An instrument already has an AC cord that carries electrical current, and an audio cord that carries audio signals to an amplifier. Now you have a third connection: the MIDI cable. This carries neither audio nor power, but transmits information about the status of the instrument to, and receives "status reports" from, other MIDI instruments. This information is coded in a computer language...a somewhat primitive language with few words and several dialects, but a language nonetheless.

What does MIDI stand for?

MIDI stands for Musical Instrument Digital Interface. You already know what "musical instrument" means, so that takes care of the first half of this phrase. *Digital* means that the instrument's information is conveyed in digital, or computer, language. *Interface* is the term for the actual link between instruments, where data passes from one instrument to another. So MIDI is a link between musical instruments that speaks data in computer language.

How can information control a synthesizer?

First we need to know a bit about computers, since MIDI instruments have microcomputer souls...in fact, MIDI could not exist without microcomputers.

Computers are decision makers, and they base those decisions on the data they receive. However, to be useable by a computer any data has to first be translated into a number-based language that the computer can understand. Actually, when you press a keyboard key with a computer-based instrument, you are not directly controlling the sound source. Instead, each time you close a keyboard switch you're sending a number to the computer, and this *number* tells the computer what note you want *it* to play for you.

The computer's "window on the world", where it receives and transmits numerical data, is called its *data bus*. The computer looks to see whether any information is on the data bus, and if so, acts on this data. For example, if it sees a digital "word" that says "play F#" on the data bus, it will do as the data commands and control a sound source so that it plays an F#. However, note that the computer doesn't care whether this word is placed on the data bus due to closing a keyboard switch or striking a guitar string—once MIDI translates a note into computer language, the note becomes compatible with any device that speaks the same language. MIDI provides access to the computer's data bus and selects which device will be "on the bus" at any particular moment, thus letting you determine the flow of information from one MIDI device to another. This is why having a specification that manufacturers can follow is so important; it insures that a variety of otherwise incompatible devices will be able to communicate with each other over a common data bus.

MIDI SUPPLEMENT

How does MIDI differentiate between different MIDI instruments on the same bus?

MIDI provides 16 independent channels of information suitable for driving up to 16 polyphonic synthesizers or other MIDI devices. There are three modes that determine how each MIDI instrument responds to these channels.

In ***Omni*** mode, the Emax II (or any other MIDI keyboard) listens to all channels at once. No matter how many notes from how many sources make it through the MIDI bus into the instrument, when in Omni mode it will attempt to play all of them.

In ***Poly*** mode, the instrument can "tune in" to one MIDI channel — just like you can tune in one channel of your television.

Mono mode is considered a "multi-timbral" mode. In mono mode, *each* individual analog synthesizer voice can be driven by one specific MIDI channel of information to play a certain melodic line. Each voice can also have its own "patch" (program), so an analog synthesizer capable of MIDI Mono mode can therefore produce multiple timbres simultaneously. Since the Emax II does not have an equivalent to the analog synthesizer voice, Mono mode is technically not supported by the Emax II. However, in the Emax II Super Mode, each preset can be assigned to receive information through a separate MIDI channel, which provides Mono-like operation but unlike traditional MIDI Mono operation, each preset can respond polyphonically. Of course, you still cannot play more than sixteen notes on the Emax II regardless of what MIDI mode you select.

16 channels, huh? Sounds like a lot of patch cords to me!

MIDI information is transmitted *serially*, which means that all data is sent in sequence. As a result, a single line can carry the MIDI information for all 16 channels, with each instrument monitoring all the words that pass over the MIDI interface but responding only to the data on its particular channel. Typically, MIDI instruments include a MIDI IN, MIDI OUT, and MIDI THRU jack. The instrument receives data over the MIDI IN jack and transmits data over the MIDI OUT jack; the MIDI THRU jack provides a replica of the signal at the MIDI IN jack. Therefore, if you want to slave three keyboards to a master keyboard, you would patch the MIDI OUT from the master keyboard to the MIDI IN of the first slave, patch MIDI THRU from the first slave to the MIDI IN of the second slave, and connect MIDI THRU from the second slave to MIDI IN of the third slave. Note that the Emax II provides two jacks, MIDI IN and another jack which is software-selectable to provide either the MIDI THRU or MIDI OUT function.

What kind of words does the MIDI language include?

Spoken language is a collection of words that convey a particular set of meanings, hopefully understood by all those who speak the language. As soon as we start talking about a musical language, the first question is: What concepts do we want this language to define? Since musical instruments will be doing the talking, the language should consist of musical terms—pitch, tempo, and so on. In fact, let's set up a practical example and see what words our language would need.

MIDI SUPPLEMENT

Suppose we have two synthesizers, Synth A and Synth B, and want Synth B to precisely follow along with what is being played on Synth A to create a "doubling" effect. To do this, our "language" would have to convey several things.

First, Synth A would have to tell Synth B whenever a key was hit and which key it was ("hey Synth B, play F#!"). We'll call this the NOTE ON word.

This turns on the keys just fine, but now we need a way for Synth A to tell Synth B when a note is over so it can turn the key back off again. We can call this the NOTE OFF word.

Now we have the notes under control, but what happens if we do some pitch-bending on Synth A? As it is, Synth B wouldn't follow because it can only tell if a note has been turned on or off—not if its pitch has been changed. So, we need a new word that indicates a pitch bend CONTROL CHANGE.

Everything's fine until we switch patches, at which point Synth A changes but Synth B doesn't. We clearly need another word to signal PROGRAM CHANGE, so that if you call up a different patch on Synth A, Synth B will call up a different patch as well (which you have hopefully programmed to sound compatible with Synth A).

And the list of words goes on...what happens if both synthesizers can respond to dynamics? You'll also want to transmit that data from one synth to another. We also need some information about the system tempo, as set by a drum machine or sequencer...

MIDI's vocabulary encompasses all these considerations, as well as many others. For a more detailed description of the complete MIDI vocabulary, refer to the book "MIDI For Musicians" (AMSCO Publications, a division of Music Sales).

TYPICAL MIDI APPLICATIONS

■ ***Slave two keyboards together so that one doubles the other for lush-sounding string effects, choirs, and so on.*** To do this with the Emax II and another MIDI instrument, connect the MIDI IN from one machine to the MIDI OUT of the other instrument. Select Omni mode and the same basic channel for both instruments—either keyboard should be able to control the other instrument. If you program each instrument to receive "program change information", changing presets on the Emax II should change programs on the other MIDI synthesizer, and vice-versa.

■ ***Slave two keyboards together to create a composite sound.*** For example, you might like the sound of Emax II violins fading in as the attack of a digital synth violin patch fades out.

■ ***When composing, you can sequence parts via MIDI and experiment with changing timbres.*** How would that harmony line sound as a piano instead of a guitar? Re-assign presets on an Emax II sequence and find out.

MIDI SUPPLEMENT

■ ***Dynamic control of drum machines.*** If you have a drum machine with MIDI, you can program it dynamically from Emax II keyboard. Run a MIDI cable from the Emax II MIDI OUT to the drum machine's MIDI connector, then consult your drum machine operation manual for information on which Emax II keys control which drums.

ABOUT THOSE MIDI CONTROLLERS...

MIDI controllers cause a lot of confusion all by themselves, so hooking them up to other instruments doesn't make things any easier. Fortunately, though, the Emax II is quite helpful in setting up the MIDI controllers functions.

To understand what a "MIDI controller" is, we first must understand what a "controller" is. A "controller" is a device which provides a means of smoothly controlling a certain parameter. the Emax II has 6 such controllers: Left Wheel, Right Wheel, Pressure, Pedal, MIDI Ctrl A, and MIDI Ctrl B.

MIDI however, does not support a "Left Wheel", "Right Wheel", or "Pedal" controller because that would be too machine-specific. Instead, MIDI simply numbers the controllers 0 to 63 and lets the manufacturer decide how it wants to adapt these "MIDI controllers" to its machine. In addition to the controller numbers, MIDI supports two special controllers that are used so often they have their own commands. These are "pitch wheel" and "channel pressure".

The Emax II supports MIDI controller numbers 0 to 31, as well as "pitch wheel" (pwh) and "channel pressure" (chp), for a total of 33 possible MIDI controllers. The Emax II allows the user to map each of the 6 internal controllers to any of the 33 MIDI controllers (**PRESET DEFINITION 7**).

As an example, let's say the Emax II's Left Wheel is set in **PRESET DEFINITION 9** to control pitch, and mapped to MIDI controller #19 **PRESET DEFINITION 7**. Now, every time the Emax II's Left Wheel moves, the pitch changes and the wheel value goes out the MIDI OUT jack as MIDI controller #19. Also, if MIDI controller #19 comes in the MIDI IN jack, the Emax II will treat it the same as it does its own Left Wheel, i.e., change pitch.

The Emax II's MIDI Ctrl A and MIDI Ctrl B, unlike its other four controllers, are only accessible via the MIDI IN port. However, all 6 controllers can be recorded to the sequencer and all 6 will be sent on MIDI OUT during playback.

MIDI SUPPLEMENT

FINDING OUT ABOUT OTHER SYNTHESIZER'S MIDI CAPABILITIES

It's easy to find out the controller numbers for parameters on another synthesizer by patching the Emax II MIDI OUT connector to the other synth's MIDI IN. During MIDI Setup select the "Rt Wheel" setup step, use the slider to select a controller number, vary the right wheel, and note any changes. Select another controller number, vary the right wheel, and see what that affects. Most synthesizers will only have a few controllers assigned; note which control numbers affect which parameters, and write them down for future reference.

For a practical example of how to use this feature, suppose you want to use the Emax II footpedal to simultaneously fade out the Emax II and a MIDI'ed synthesizer. During MIDI setup, select the "Pedal" setup step, and use the slider to change controller numbers until you find the number that corresponds to overall level. Suppose the controller number is 07. If you assign the Emax II "Level" Real Time destination to the footpedal and also send the footpedal information out over the MIDI controller 07, both instruments should fade in and out together.

DEALING WITH "MIDIOSYNCRACIES"

There are occasional compatibility problems between MIDI gear from different manufacturers; however, many problems are created by operator error and/or a lack of understanding of how MIDI works. If you transmit information on one channel and have the "receiver" set up for a different channel, forget it. MIDI is quite unforgiving that way...you can hit a bum note and not too many people will notice, but send a computer a wrong number and it will most definitely notice.

MIDI is a lot of fun provided that you don't get discouraged when things go wrong. Many times there is a solution; sometimes there isn't. In any event, MIDI as it is today is far better than no MIDI at all. At the very least you can almost always slave two keyboards together, and drive MIDI keyboards from a MIDI sequencer. If you don't understand things at first, don't worry— keep experimenting and eventually everything will fall into place.