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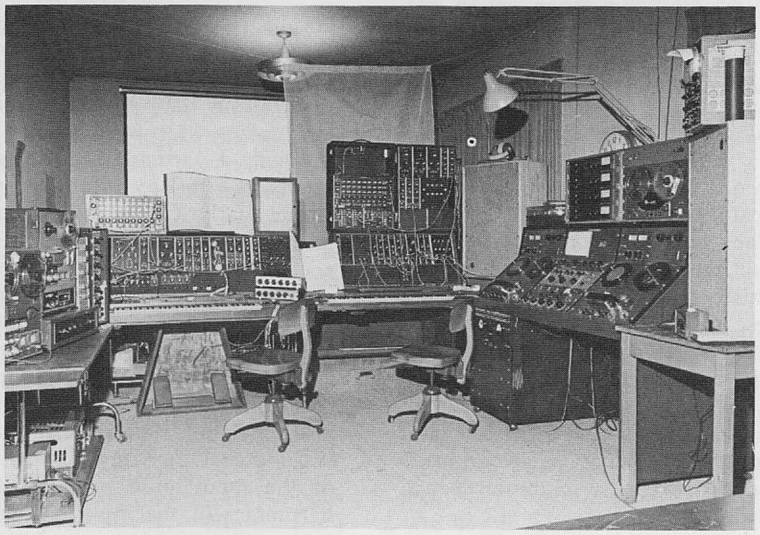
Live Electronic Music Equipment

In this article, the author explores some of the techniques and the equipment necessary to create and perform electronic music and/or electronic music plus live music in concert.



A typical? electronic music setup. This one is at the North Texas State University Electronic Music center. A lot of recognizable audio equipment is there as well as the two Moog synthesizers.

WE ARE TODAY witnessing the gradual electrification of music. Little by little, all the familiar forms of music are being taken over by electronics: organs, guitars, electric pianos, amplified instruments, synthesizers, recordings with electronically-synthesized reverb. and stereo and finally totally electronic music. Electronic music has generally been produced in studios through much effort in tape splicing and dubbing and certainly not in real time. Recently musicians, having gained a better idea of what electronics can do for them have begun experimenting with equipment on stage for real live electronic music, either alone or with more familiar instruments.

Today, the Moog synthesizers are concert instruments as are many other brands of synthesizers. Also, the rock groups, *avant-garde* improvisation ensembles, and the local "Pops" concerts have presented live electronic music in concert. Usually the instrument employed is a synthesizer; however, the synthesizer is far from an ideal instrument for live performance. Most are too slow in set-up and changing time. Most have complex functions that take many tries to get the best result which should then be spliced into a finished composition on tape. The amateur musician desiring to add electronics effects to a live performance is better advised to consider certain specific devices that will produce some reliable effects for him. This article describes several such devices.

FREQUENCY DIVIDERS

A device on the musical market at the present time is an electronic attachment for reed instruments[1]. With a contact microphone pick-up on the mouthpiece of either a clarinet or a saxophone the player has at his fingertips (at the push of a button) the sounds of any clarinet, saxophone, double reed instrument, or cello. This device is intended for the popular music player. However, there is a basic principle at work here which serious composers can use for their own more diverse interests.

The basic principle is that of an electronic organ (most brands) and is as follows: A basic, one octave, set of oscillators is provided to generate the highest octave of notes for the instrument. Then, in all the lower octaves, instead of having more oscillators, the circuits used are frequency dividers, which take their inputs from the next higher octave. Thus, all octaves are exactly the same in intonation and the complexity of the tuning procedure is greatly reduced. One thing to note about this is that there

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[1]Selmer's Maestro is one of many such devices on the market.

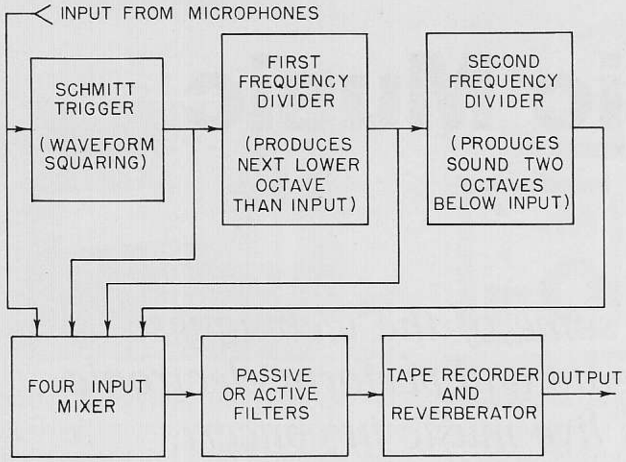


Figure 1. A block diagram of the electronic device discussed in the text.

will be no output from any octave if the original oscillator is turned off.

Now, what would happen if, instead of an oscillator, we used the output of a musical instrument with a microphone attached? We could produce any sub-octave of the basic instrumental pitch. In fact, we could have any or all octaves simultaneously. This is the principle of registration on electronic organs. These various octaves are mixed to produce various tone colors. Then, filters may be applied to the results as well as the other common signal processing devices: modulators, reverberators, mixers, and so forth. (Some of these other devices will be described later in this article.) The final result may be reproduced through a high fidelity sound system, a high-power p.a. system, or recorded on tape.

The output of the dividers may also be fed to an envelope follower or a ring modulator which is being controlled by another instrument. These will also be described subsequently.

The illustration in FIGURE 1 gives an explanation of how the circuit works. The schmitt trigger circuit converts the input waveform into a square wave. The two following dividers reduce the octaves one per circuit for a total of two octaves if two dividers are employed. The four signals (the input, the squared input, the first octave and the second octave) are all fed to inputs of a four input mixer so that any signal or any combination of signals may be selected.

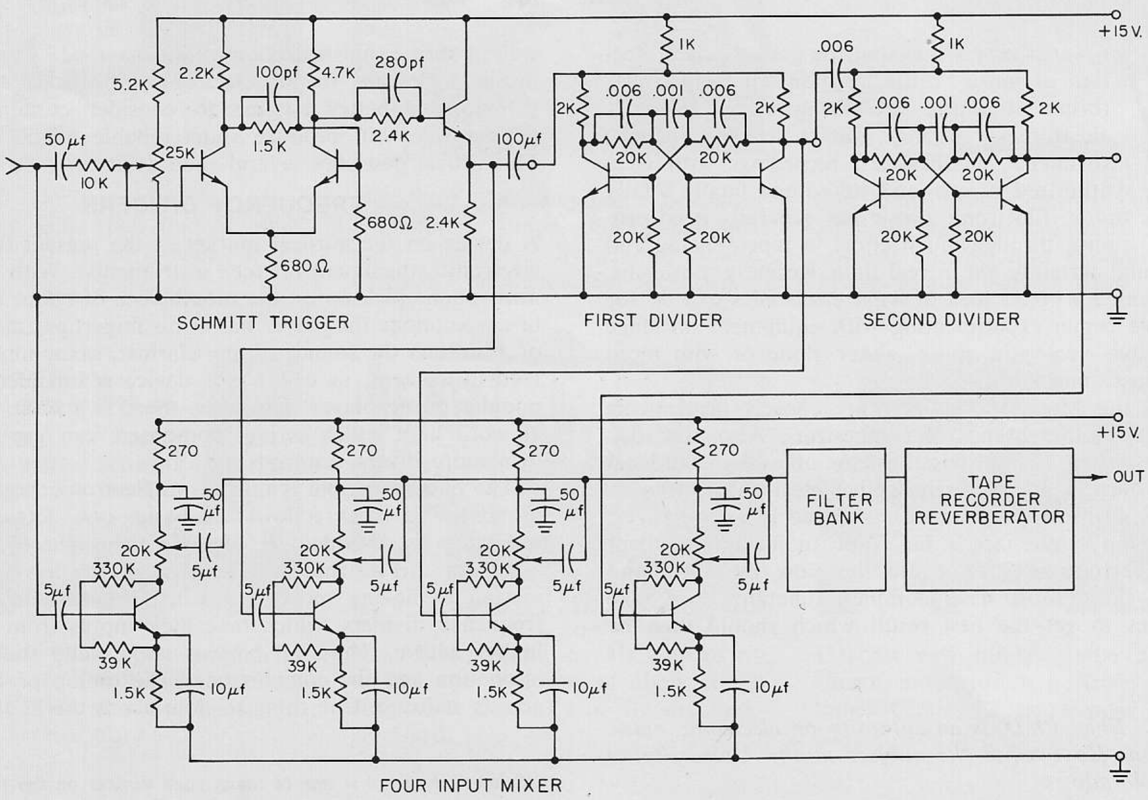
FIGURE 2 gives a possible circuit diagram for a frequency divider set, whereas FIGURE 3 illustrates how the fixed filter bank may be designed.

THE BODE-MOOG RING MODULATOR

An instrument designed by Harald Bode and manufactured by the R. A. Moog Co. is the multiplier-type ring modulator with squelch. Producing a much lower distortion output than equivalent circuits set up on the Moog synthesizer, this unit is ideal for live performance and produces a particularly attractive and musical output. The inputs may be any combination of audio signals; however, if one of the two is a live acoustical instrument, and one an electronic sound source, attractive variants on the acoustical signal result. The technique is desirable for use in real-time electronic music performances.

The multiplier-type ring modulator generates very small amounts of harmonic and intermodulation distortion (other than the desired sum and difference). Thus, when two sine waveforms are presented to the two inputs, the output is also a pair of sine waves, the sum and the difference of the two inputs. Other ring modulators may generate harmonics of the modulation products as well as products of the harmonics of the input signals. These roughen the sound

Figure 2. A possible choice of circuitry to implement the block diagram of Figure 1.



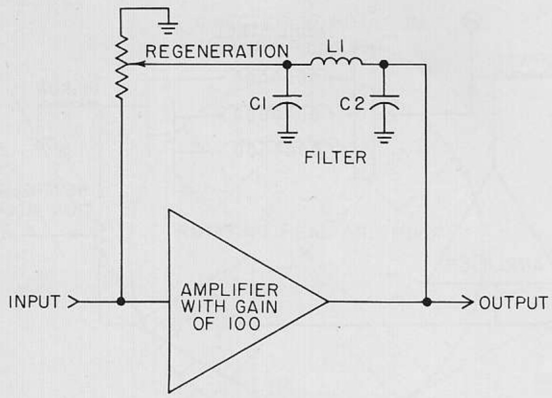


Figure 3. The use of an amplifier as a filter, regenerative filter, and oscillator.

quality of the output and make it much less attractive to the listener.

The reader is referred to the *Electronic Music Review, Volume 1, Number 1*[2] for detailed discussion of techniques that may be performed with the ring modulator. Some of these techniques will be repeated here for convenience. All of these techniques are suitable for live performance.

If one input is a band of white noise, and the other a square or sawtooth waves, having an infinite series of partials, the resultants will be two infinite series of partials, one the sum series and the other formed by the difference.

If one input is a band of white noise, and the other a sine wave, the band of white noise will appear in the output, but twice as wide, or in two segments, formed by the sums and the differences, respectively. If the sine wave is made to sweep, the white noise in the output will also sweep, producing the effects of howling wind.

Sounds picked up by a microphone may be ring-modulated with themselves if the overtones are filtered from one input and the entire spectrum is presented to the other input.

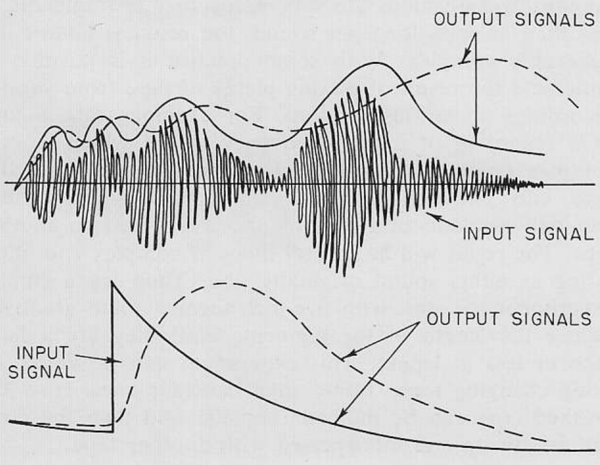
The ring modulator produces a particularly attractive tremolo effect when connected in the same manner as standard tremolo or vibrato on an amplitude modulator. If the unmodulated signal is reproduced through one loudspeaker and the modulated signal through another, the result is a sort of amplitude-phase modulation in space which is very attractive.

Any sound may be transposed (up and down simultaneously) by modulation by a sine wave. Up transposition is produced by the sum tones and down transposition by the different tones.

The Bode-Moog Ring Modulator is also available in dual units for stereo operation and for other techniques requiring two modulators.

[2]Electronic Music Revie, Volume 1, Number 1, Various authors. Discussions of applications of Ring Modulation.

Figure 4. Examples of outputs from an envelope follower with fast or slow response times. Fast response is shown with a solid line, while the dashed line indicates slow response.



ENVELOPE FOLLOWER

An envelope follower is an adaptation of a device common in radio applications known as a detector. Its function is simply to detect the lower of two modulated signals. That is, to separate the lower of two signals from the modulation product after modulation has been performed. Although the lower signal is usually selected, a detector for the upper signal could be designed and built. A rule to remember in using detectors or envelope followers is that the signals to be separated must be several octaves apart. This is because the actual separation is done by filters, and filters have a limited selectivity (half-octave or more, usually).

The envelope follower, while identical in circuitry to a detector, is usually employed to separate more complex signals, that is, signals with more than two component elements. These signals are often complex audio signals containing hundreds of component frequencies. Such signals are not the resultants of a modulation process, however but are treated by the envelope follower as if they were.

In actual practice, the envelope follower is used to extract a low-frequency signal from a complex audio signal. This low-frequency signal follows the general outline of the complex audio signal with a greater or lesser degree of accuracy depending on the settings of certain filters which are usually made variable and which may be labeled *slow, fast,* etc.

The envelope follower output signal is employed as a control voltage on voltage-controlled modules, thereby imparting the envelope extracted to other audio signals. This is illustrated in FIGURE 4.

TAPE RECORDERS AND REVERBERATION DEVICES

Once the experimenter has assembled a group of sound generating and modifying components (such as those previously described, or a professional synthesizer) he will probably find that his biggest problem is the obtaining of a suitable recording system. No matter what convenient or inexpensive means are used to generate the sounds, they are usually recorded for presentation as sound tracks, phonograph records, and so forth. Thus, tape recorders are required, and for professional applications a professional tape recorder will be needed. These cost much money and can not be improvised inexpensively. Most professionals recommend that the electronic music studio have at least two recorders and one of these should have four or more channels. For live electronic music one machine may be used as a reverberation unit and another for recording the performances.

The amateur should be able to purchase two good tape decks and build recording and playback amplifiers to use with them. He can purchase four-track heads for either quarter-inch or half-inch tape and the head manufacturers will usually provide circuits for amplifiers. This is the only economy move recommended since tape decks are notoriously difficult to design and less than high quality here shows up in poor speed regulation and other problems.

The diagrams of FIGURES 5 and 6 illustrate the head configuration of a professional tape recorder with *sel-sync* (an Ampex trade name) and a method for adding an extra

	Impedance	Voltage	Current
Recording head	high	high	low
Playback head	low	low	high

Characteristics of tape recorder heads in professional recorders.

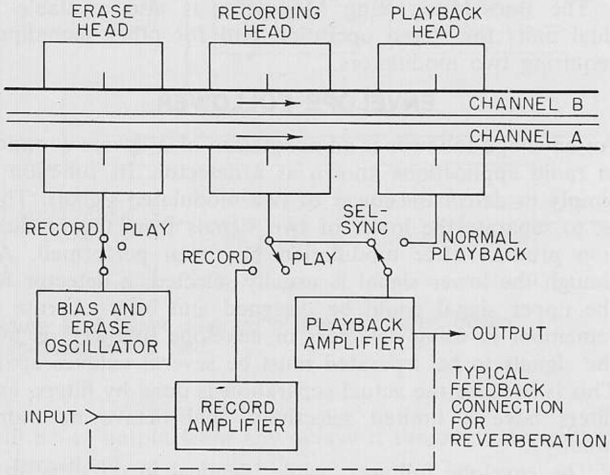


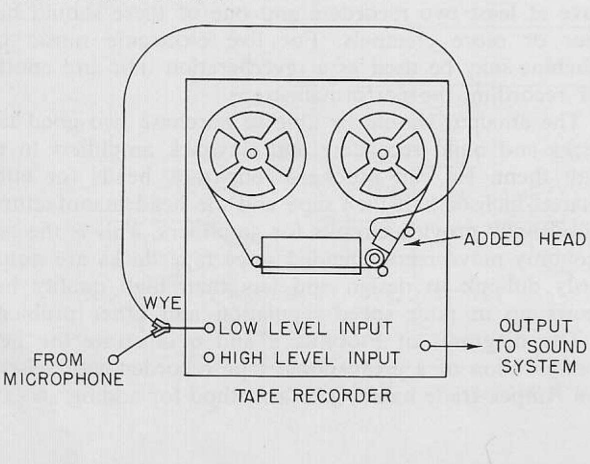
Figure 5. A typical tape recorder with sel-sync and three heads as used in professional electronic music studios.

head to a two-head machine to create tape-recorder reverberation. The illustration of the steel spring in FIGURE 7 shows another popular reverberation device that may be used.

The studio with sufficient funds is advised to invest as much as possible in high-quality tape recording equipment. An expensive but useful addition to the professional tape recorder is the Eltro Pitch and Tempo Changer which can be adapted to most professional machines. This device (FIGURE 8) has a rotating head assembly containing four heads and a variable speed capstan. It can be used to change both pitch and tempo or either one independently. A less expensive and less versatile device is the variable-speed tape capstan drive oscillator-amplifier manufactured by various companies, including the R. A. Moog Co. With this device, the pitch and tempo are continuously variable together but not independently.

The technique of mixing the two signals from two recorders, one with variable speed and one without may be used to produce the phase-shift effect popular in many current rock recordings. The cancellation frequency is shifted by changing the signal delay in the recorder with variable speed and beating it against the unchanged signal in a simple mixer. If both recorders are running at exactly the same speed there will be no phase difference and no

Figure 6. The method of adding a head to a tape recorded to use for reverberation.



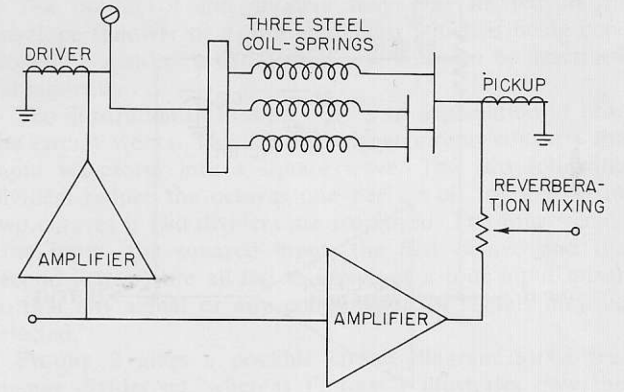


Figure 7. A typical steel spring reverberation device.

phase cancellation effect. This technique causes difficulties in live performance, however, due to the inherent time delay involved. A similar but not delayed effect may be achieved with either a ring modulator or a resonant voltage-controlled band-pass filter (both have been described earlier in this article).

MONITOR SYSTEMS

Two types of amplifier-speaker systems must be considered for electronic music playback. The first are smaller units for studio use. These may be regular high fidelity units. Vacuum-tube versions are recommended because they may often be obtained inexpensively today and also because they will stand up to experimental and accidental misuse better than transistor amplifiers.

The second type of monitor system to be considered is the large, high-volume type necessary to produce the volume levels for filling large auditoriums and theaters. Amplifiers for such a system are best transistorized for portability and should produce at least 50-watts per channel. Acoustic suspension speakers are often used where portability is an important factor but these are low-efficiency systems and much more sound can be obtained from large horn systems. Popular music groups often use heavy-cone, fifteen-inch speakers in open-back cabinets. These produce a great deal of acoustical power but add an undesirable coloring and distortion to the sound for some applications.

TAPED ELECTRONIC MUSIC

Taped electronic music is most popular today on the college and university campuses and is the primary method used by serious electronic music composers. It should be obvious that it is possible to achieve a great complexity by means of splicing small segments of tape together. If each segment contains a complex sound and segments containing diverse sounds are spliced together in fragments of less than an inch long per sound, the result is almost unbelievably complex. With some practice it is possible to anticipate the result of mixing pieces of tape from various recordings in various fashions. For example, take a tape of a recording of a high-frequency sound which is very complex (such as the sound of a cymbal crash). Cut this tape into one-inch segments and splice these between one-inch segments of a low-frequency recorded on another tape. The result will be several times as complex and interesting as either sound originally was. Then try a similar experiment but start with five-inch segments and gradually reduce the length of the segments until they are a half-inch or less in length. This experiment can be varied by using changing tones rather than constant ones. Also the finished tape can be dubbed (copied) and then the copy can be cut up and interspersed with another tape.

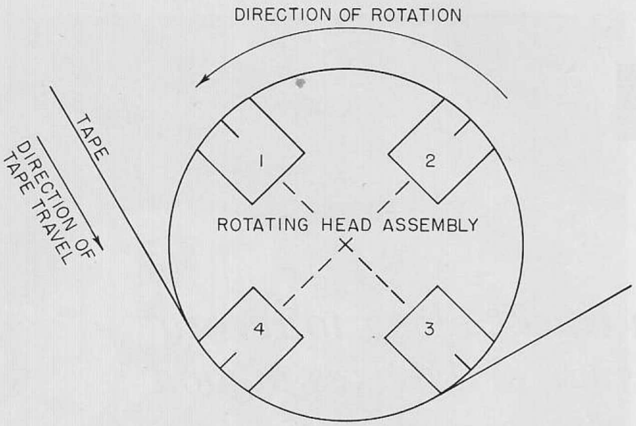


Figure 8. A tape player with a four-head rotating group used for independent changing of pitch and tempo of a recording. When the head assembly rotates clockwise, frequencies on the tape are increased as small segments are repeated; rotated counterclockwise causes frequencies to reduce and small segments are skipped.

Of course, the best results will be obtained in this sort of process if the person assembling the tape has specific goals in mind. For example, start with simple sounds and gradually build up to a climax in the musical composition by gradually increasing the complexity of the sounds, the shortness of the tape segments and the contrast between the materials on the tapes being spliced. Remember that you can change the speed of any sounds on your tapes by copying them at different speeds than they were recorded. Tapes can also be played backwards (particularly interesting for musical instruments and voices) and spliced together with alternating forward and backward sections.

Many techniques for splicing have been developed. For example, splice a long gradual splice by cutting a long diagonal cut in two pieces of tape and splice the two sections together with splicing tape. If one uses splices like this often it pays to construct a special template that will fit a professional splicer and make cutting straight edges easier. This can be a strip of metal the width of the tape with a cutting edge of the shape of the desired tape splice. Two pieces of tape cut with the template will match exactly when spliced.

Remember that it is possible to splice sections of blank tape in between sections of recorded tape and that these blank sections can be as short as a half inch for short interruptions within a continuous sound. This can produce interesting rhythmic effects.

As was mentioned earlier, for best results do your work at fifteen inches-per-second on half-inch tape with four channels. Still, good results can be obtained with home tape recorders; even if they are not up to broadcast standards they can be very satisfactory for amateur performances, background music for local theater groups, sound track for semi-professional movies and also home movies, local dance groups and so forth. They can also be quite adequate to provide a great deal of satisfaction and entertainment.

Taped electronic music is not live, but can often be mixed with live music to produce live results. For example, it may be used with an orchestra or as the background to a ballet or the theater. ■