H. BODE

Psychoacoustical Aspects and Musical Applications of an Infinite Phaser.

As you recognize from the title of my today's presentation, I am going to discuss the psychoacoustical aspects and musical applications of an infinite phaser. However, since the infinite phaser has been preceded by other means and methods to produce infinite effects, such as infinitely stepping and sliding tones, I shall also briefly deal with the history of these phenomena.

In December 1964 a paper by Roger Shepard appeared in the Journal of the Acoustical Society of America with the title "Circularity in the Judgments of Relative Pitch". In this paper Shepard reported among other things about a sequence of ever ascending and descending half tone steps, which was generated at Bell Labs on the IBM 7094 computer by a computer program developed by M.V. Mathews for the synthesis of musical sounds.

In order to create the infinity effect, a cluster of tones was generated, consisting of sinusoidal components in one octave intervals and covering a 10 octave range. The amplitudes were largest in the center of the audio range and tapered off towards the low and the high end to disappear below the threshold of perception. This is illustrated in Fig. 1, where the "envelope" of the sound levels is represented by a bell shaped curve. In this picture the solid vertical lines represent the individual sinusoidal components at a given time and the dotted vertical lines the sinusoidal components in the following time increment. - Recording No. 1 -

For visually symbolizing this tonal sequence, Shepard used the picture of a "circular"staircase (M.C. ESCHER), shown in Fig. 2, which gives the optical illusion of being ascending when going counterclockwise and descending, when going clockwise. In a slightly different form this picture was presented by L.S. Penrose and R. Penrose in the British Journal of Psychology (1958) under the title "Impossible Objects: A Special Type of Visual Illusion".

Through the expedient of using the bell shaped loudness envelope the tonal structure was the same at the beginning and the end of a one octave cycle, so that the same computer program could be used for going through each of the following octave sequences.

As reported by L.A. Hiller,"Risset, until recently on the staff of IRCAM in Paris, also worked at Bell Laboratories in the 1960's. Among other tasks, he assembled a substantial compendium of complex sounds that could be generated by the computer sound synthesis techniques originated by Max Mathews and formalized as MUSIC5."

In contrast to Shepard's restricted sets of discrete intervals Risset generated complete tone sweeps. He found, that these glissandi had at least "the same psychologically disturbing effect of Shepard's discrete note pattern." Like in the case of Shepard, Risset's program "presents a little more than one octave of an endless glissando, which could be pursued indefinitely since it is back to its original point after an octave descent (or ascent)".

After Risset and other workers L.A. Hiller developed more elaborate programs and compositions in collaboration with John Myhill of the Department of Mathematics at SUNYAB and Robert Brainerd, who wrote an algorithm for their MUSIC5 program.

A major composition by Hiller evolved under the title "Electronic Sonata and Midnight Carnival" in 1976. One of the key elements of this composition was a complex set of glissandi with octave and other changing intervals and ... changing speed. Also Juxtapositions of up and down glissandi were used.

As a working basis a so-called "normal set" of eleven parallel glissandi was chosen, all one octave apart. The frequency range from 20 Hz to 20,000 Hz was almost completely utilized with the exception of using low pass filtering at the very high end to avoid "foldover" or beat frequencies with the sampling rate, which was chosen at 20 kHz. The production runs were made on a CDC-6400 computer utilizing a MUSIC5 program written substantially in COMPASS, a fast assembly language. - Recording No. 2 - FIG. 3 -

Within the framework of a larger composition still in progress Jay Lee generates sliding tones in intervals of octaves, fifths and triads to be compatible with classical harmonic structures. Lee also uses the bell shaped amplitude envelope known from Shepard's work to enhance the infinity illusion. He generated algorithms written in PASCAL and encompassing one octave intervals. His work was performed on a PDP-11 computer at a sampling rate of 20 kHz. Due to the use of the bell shaped envelope there was no danger of the high end of the frequency range interfering with the sampling frequency. - Recording No. 3 -

Now, after we have learned a few things about the psycho-acoustical aspects and effects of infinitely stepping and sliding tones, it will be interesting to find out, what happens, if this infinity phenomenon is moved from the domain of tone generation to sound modification, such as by selective filtering.

This is, where the so-called "infinite phasing" effect comes in. In order to implement the infinite phasing, the "Barberpole Phaser™" was created as a multifunction sound modification device. In the Barberpole Phaser™ the infinitely moving multiple filter effect is combined with a number of complementary modulation capabilities to optimize its versatility and effectiveness. -FIG. 4-

It is the purpose of this presentation to stay within the limitations of the subject title of the "Psychoacoustical Aspects and Musical Applications of an Infinite Phaser." Therefore a description of the technical details will follow in a later publication. - Also, I think you will be quite anxious to hear, what infinitely moving filtering or phasing sounds like, so I am going to give you a brief demonstration through the means of tape recordings.

-FIG. 5- (Recording No. 4) - First you will hear pink noise, which is filtered so that the peaks are approximately one octave and a third apart.

(5) - Next you hear the same effect with prerecorded drums.

(6) - Now I am playing a complex chord on the Polymoog and processed through the Barberpole Phaser™.

-FIG. 6- (7) - Next I am superimposing a sine wave modulation on top of the up sliding filter peaks.

(8) - Next I am superimposing a step function on top of the sliding filter peaks, first moving up and then down.

(9) - Next you hear a chord sequence phased up with a quasi saw tooth modulation and stereo panning.

(10)- Last you hear a Bass guitar phased up with a fast frequency modulation superimposed upon the phasing effect and with stereo panning.

I hope, that I have been able to demonstrate some of the potential possibilities of infinite phasing.

In closing I want to thank those who have helped me make this presentation possible. My special thanks go to Lejaren Hiller and Jay Lee who provided me with priceless literature information and sound recordings and to Tom Rhea, who was instrumental in the human engineering and appearance design of the Barberpole Phaser™.

- Thank you -

List of Recordings of Infinite Sliding and Stepping Tones and Infinite Phasing Effects.

1.- Shepard's Tone by R.N. Shepard (around 1963).

2.- Excerpt from Electronic Sonata by L.A. Hiller, 1976.

3.- Barberpoles by Jay Lee, 1981
   a) Octave slides,
   b) Fifth slides,
   c) Triad slides,

4.- Infinite Phasing with pink noise (Bode, 1981, Barberpole Phaser)

5.- Infinite Phasing with prerecorded drums.

6.- Infinite Phasing with polyphonic synthesizer - UP.

7.- Chord phased UP, sine wave superimposed upon infinite up phasing.

8.- Chord phased UP, then DOWN, with step function superimposed upon phasing.

9.- Chord sequence phased UP with saw tooth FM and stereo panning.

10.- Bass guitar phased up with fast sine FM and stereo panning.

11.- Bass guitar phased DOWN with sine wave superimposed upon down phasing.

12.- Electric guitar phased DOWN straight.

12b.- Electric guitar phased DOWN with FLIP feature, (switching back and forth between peaks.)

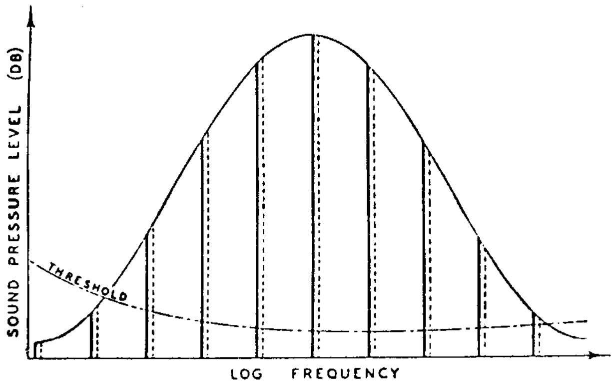


FIG. 1. Sound-pressure levels (in dB) of 10 simultaneously sounded sinusoidal components spaced at octave intervals. (The dotted lines correspond to an upward shift in the frequencies of all components.)

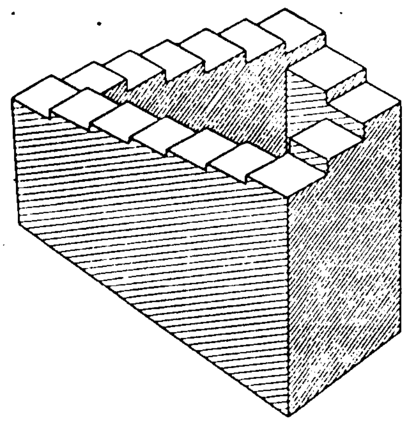


FIG. 2. "Circular" staircase illusion.

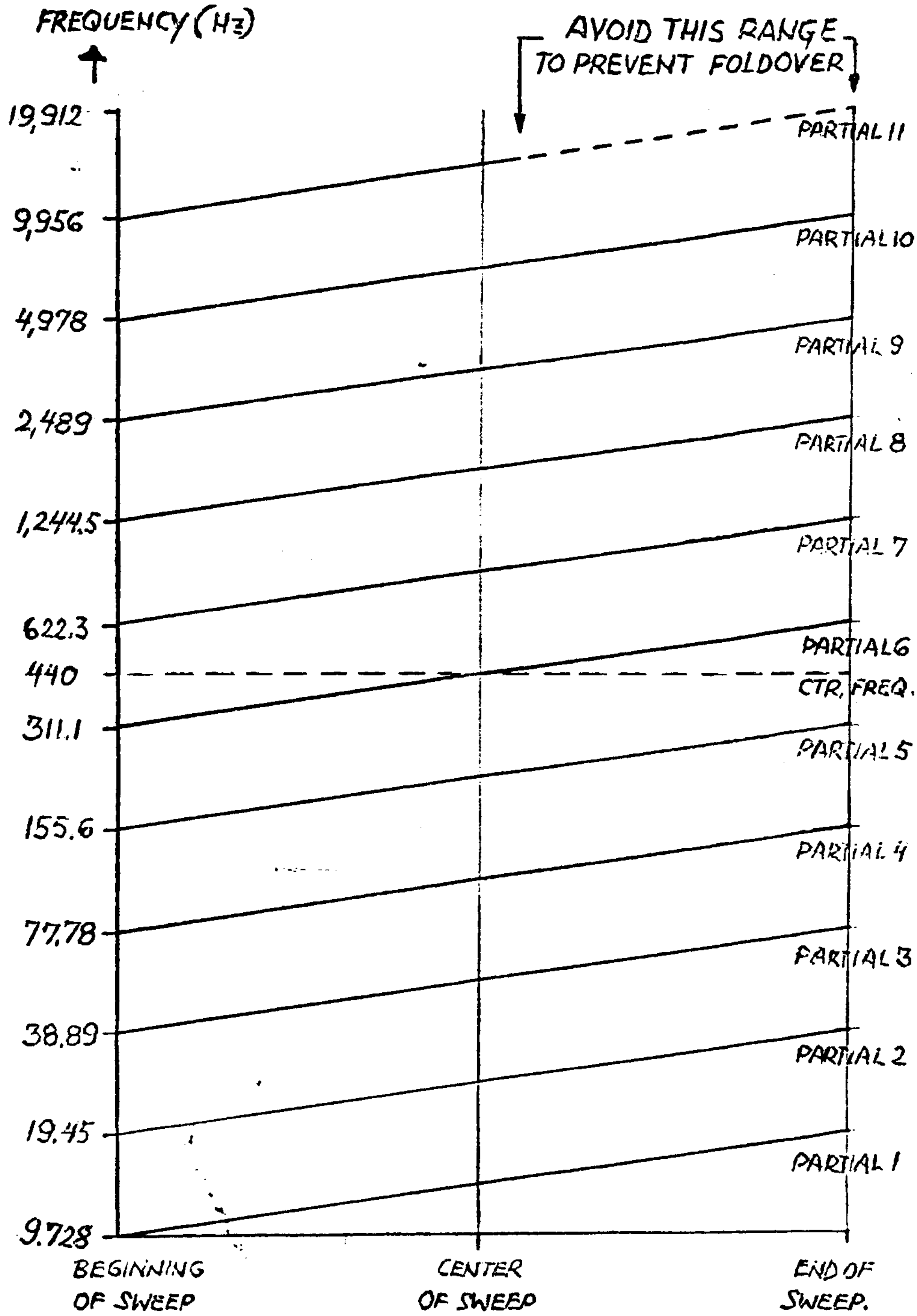


FIG. 3 BASIC PLAN OF OCTAVE GLISSANDI CENTERED AROUND CONCERT A.

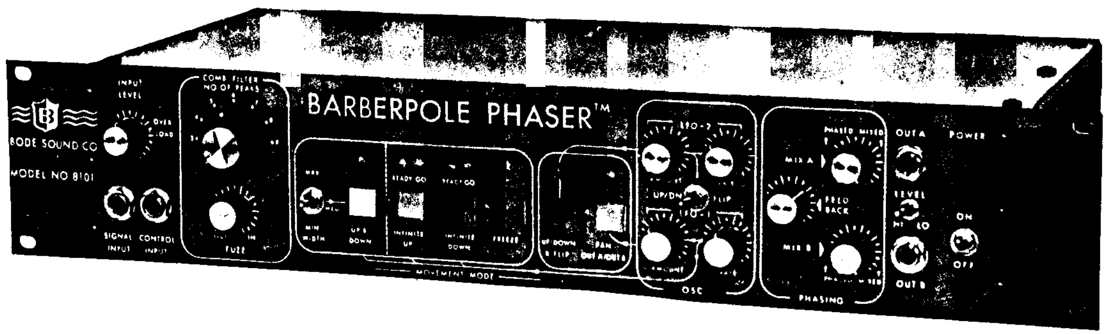


FIG. 4 FRONT PANEL CONTROLS OF BARBERPOLE PHASER™.

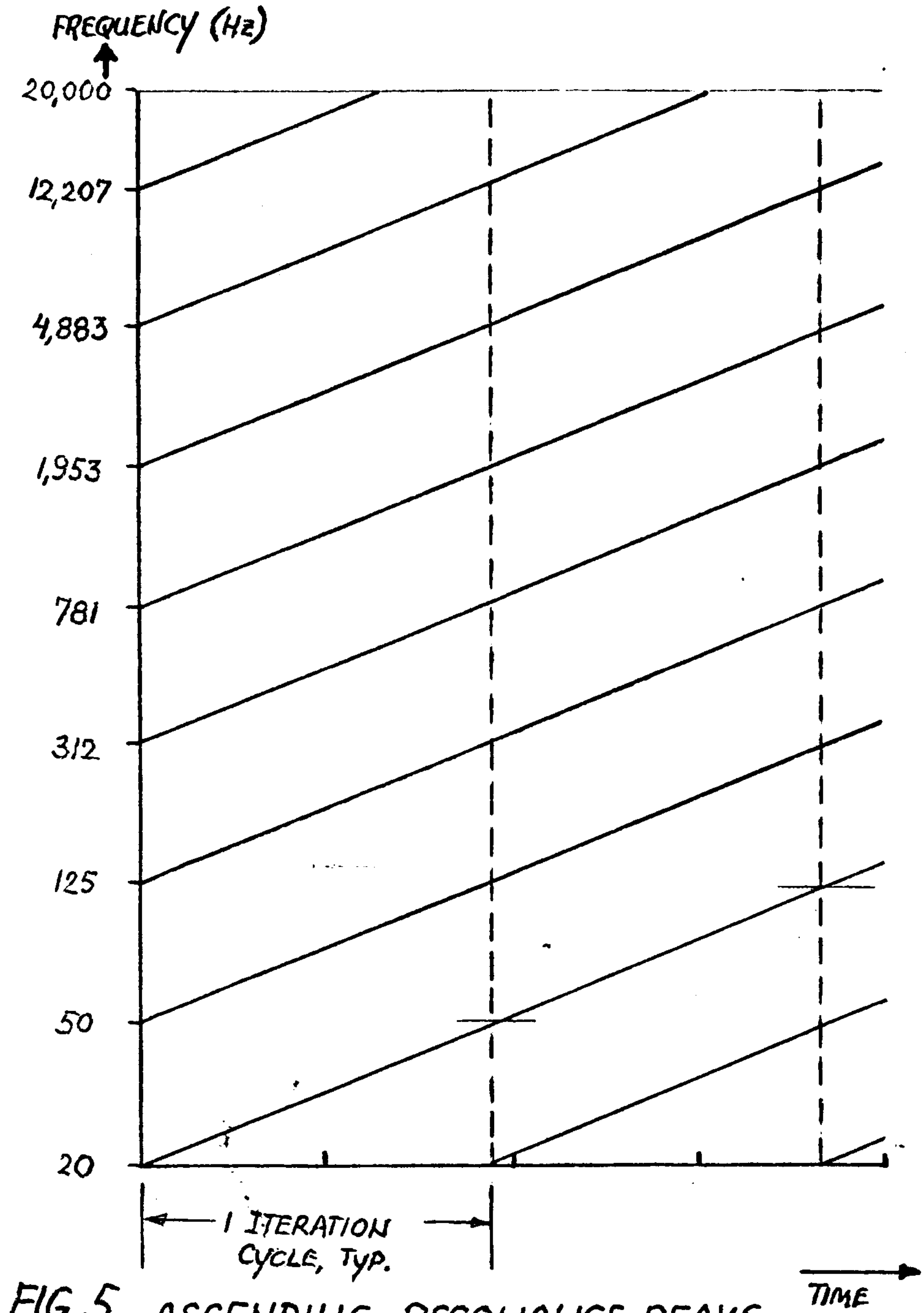


FIG. 5 ASCENDING RESONANCE PEAKS 1 1/4 OCTAVE APART

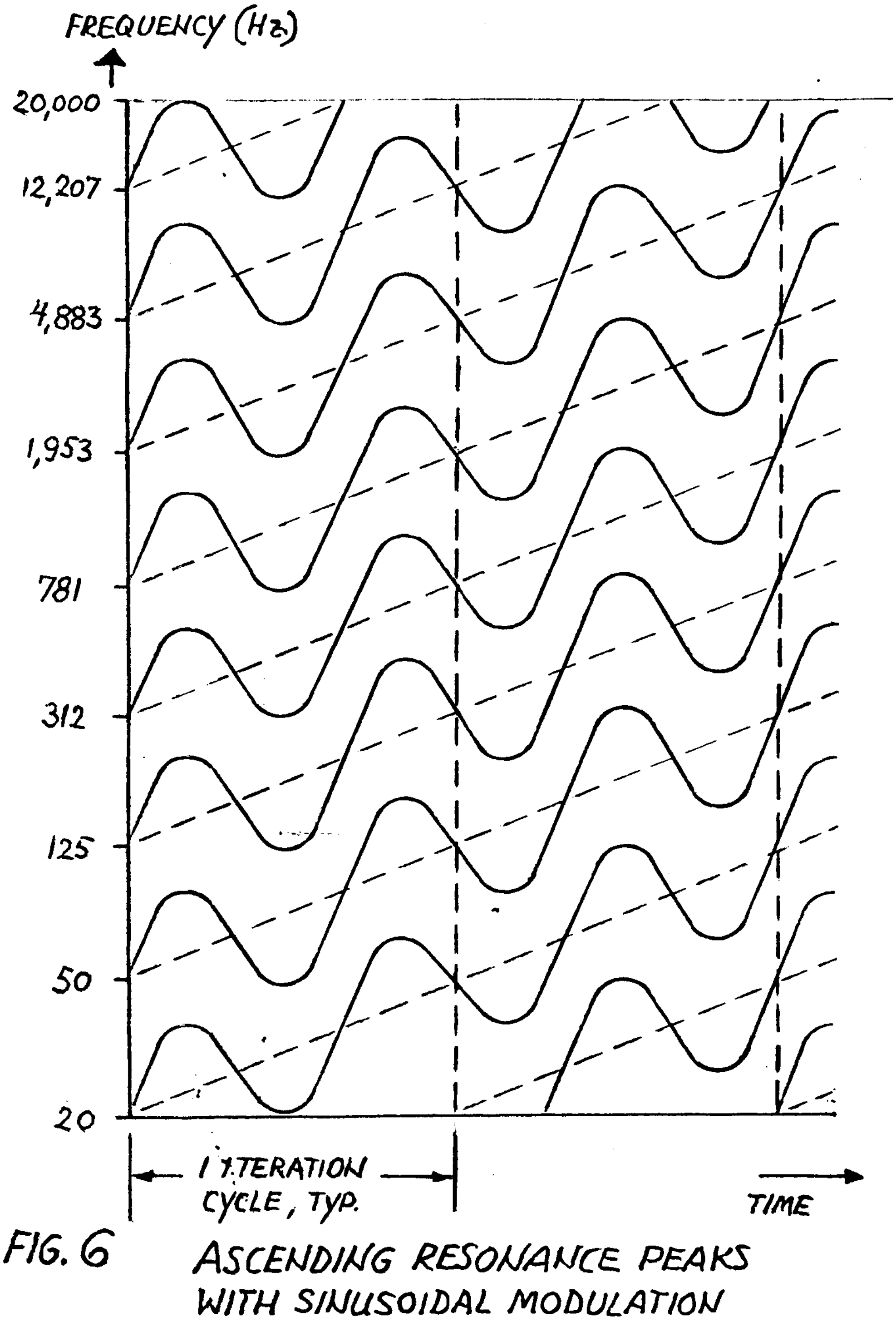
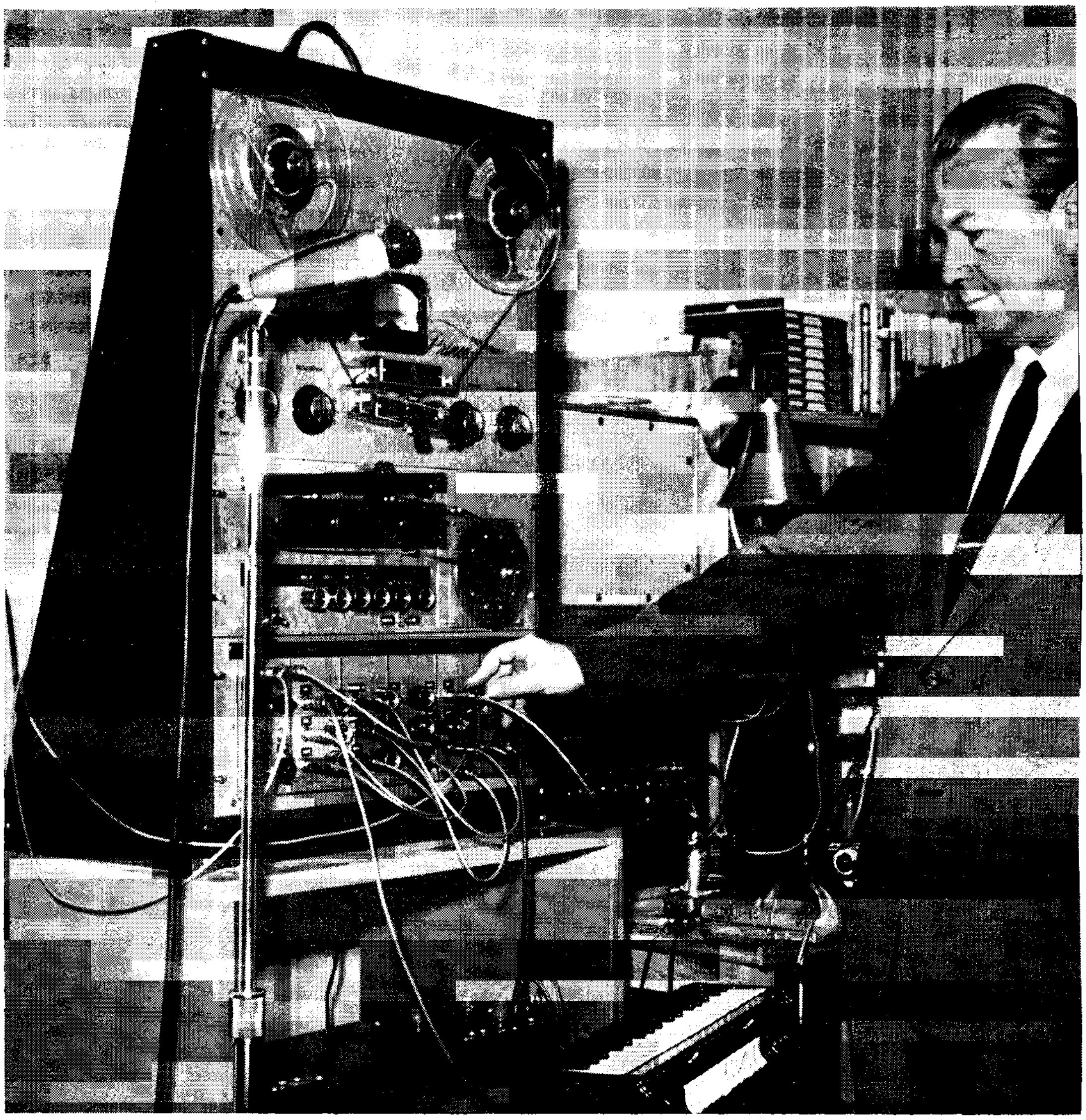


FIG. 6 ASCENDING RESONANCE PEAKS WITH SINUSOIDAL MODULATION

To Steina and Woody Vasulka with great admiration and compliments from Harald Bode

electronics
December 1, 1961



Author sets up tonal effects on synthesizer. Conventional tape deck at top, tape-loop reverberation below, and plug-in modules at bottom

Sound Synthesizer Creates New Musical Effects

Well-known electronic circuits are used in system combinations so that unconventional sounds and patterns can be produced from ordinary audio. For example, a singer can generate his own string bass accompaniment simultaneously with his voice

By HARALD BODE The Wurlitzer Company, North Tonawanda, New York

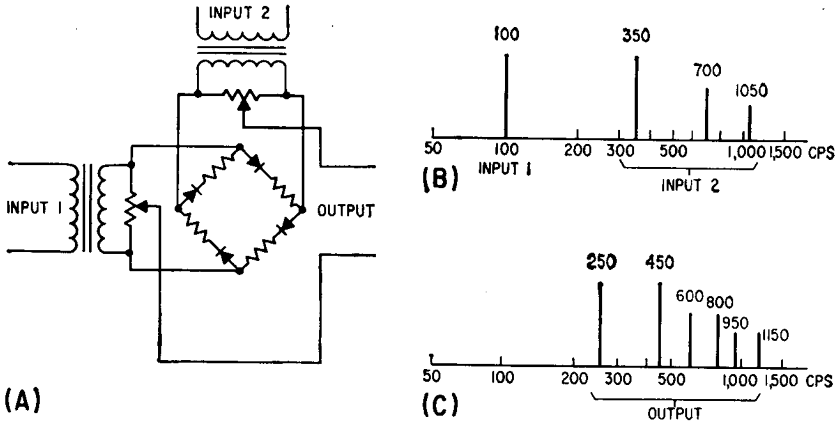


FIG. 1—Ring-bridge modulator (A) with input signals (B) and output (C)

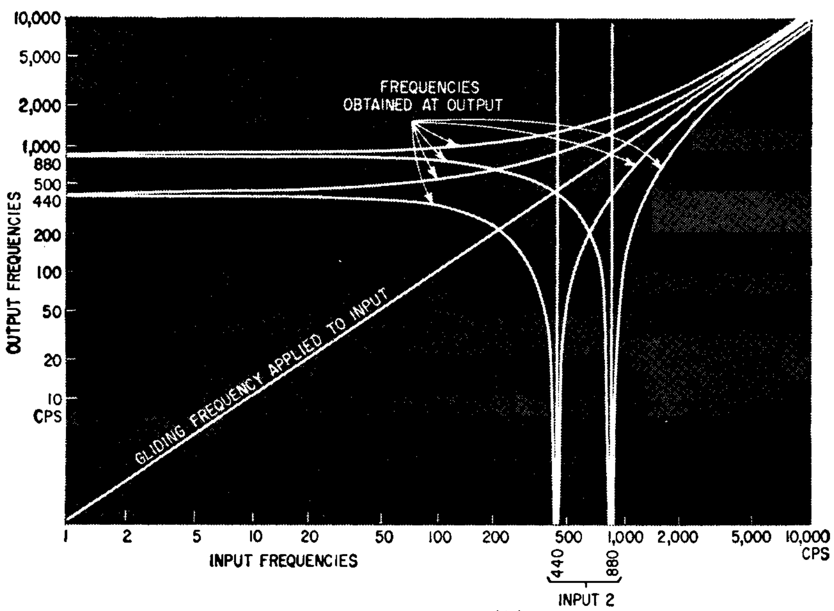


FIG. 2—Output of ring-bridge modulator with gliding frequency applied to one input and two harmonically related frequencies to the other

NEW SOUNDS and musical effects can be created either by synthesizing acoustical phenomena, by processing natural or artificial (usually electronically generated) sounds, or by applying both methods. Processing acoustical phenomena often results in substantial deviations from the original.

Production of new sounds or musical effects can be made either by intermediate or immediate processing methods. Some methods of intermediate processing may include punched tapes for control of the parameters of a sound synthesizer, and may also include such tape recording procedures as reversal, pitch-through-speed changes, editing and dubbing.

Because of the time differential between production and performance when using the intermediate process, the composer-performer cannot immediately hear or judge his performance, therefore corrections can be made only after some lapse of time. Immediate processing techniques presents no such problems.

Methods of immediate processing include spectrum and envelope shaping, change of pitch, change of overtone structure including modification from harmonic to nonharmonic overtone relations, application of periodic modulation effects, reverberation, echo and other repetition phenomena.

The output of the ring-bridge

NEW FRONTIERS IN ELECTRONIC MUSIC

Music is the art of making pleasing, expressive or intelligible combinations of sounds and making such combinations into compositions of definite structure and significance according to the laws of melody harmony and rhythm. It is also the art of inventing, writing or rendering such compositions, whether vocal or instrumental.

Ancient Greek music was limited in expression by the primitive instruments used, mostly of the lyre and flute types. However, it did establish the diatonic scales based on the tetrachord as a unit and gave us the rudiments of key relationships.

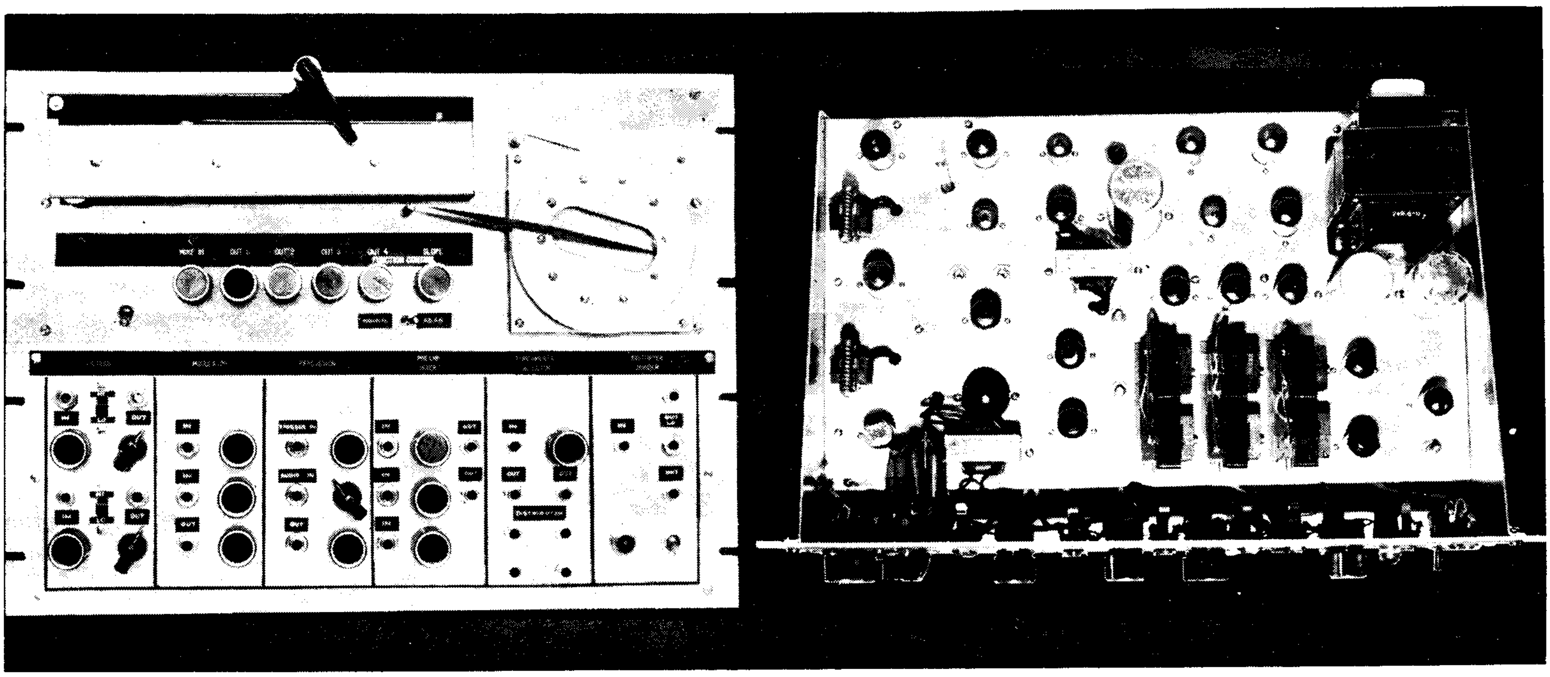
Early church music gave us the neumes to indicate pitch, the development of staff notation, and the superceding of the tetrachord unit by the hexachord. The practice of descant, or simultaneous melody, gave rise to mensurable music which in turn gave birth to counterpoint.

Establishment of modern major and minor scales with the octave as a unit and of equal temperament made possible modulation into any key and led to the development of harmony.

Great improvements in instrument making marked the growth of purely instrumental music. The modern symphony originated in the time of Haydn and others about 1770.

Recently, increasingly dissonant linear counterpoint, Schoenberg as an example, is becoming popular. The French gave us polytonality and the Russian composers have given us barbaric, dynamic rhythms while American jazz has contributed to the music of the dynamic era in which we live.

Modern proponents of musical thought and expression have often complained of being musically restricted by instrument range and



Front panel of tape-loop reverberation and modular units (left) with top view of module chassis (right)

modulator shown in Fig. 1A yields the sum and differences of the frequencies applied to its two inputs but contains neither input frequency. This feature has been used to create new sounds and effects. Figure 1B shows a tone applied to input 1 and a group of harmonically related frequencies applied to input 2. The output spectrum is shown in Fig. 1C.

Due to operation of the ring-bridge modulator, the output frequencies are no longer harmonically related to each other. If a group of properly related frequencies were applied to both inputs and a percussive-type envelope were applied to the output signal, a bell-like tone would be produced.

In a more general presentation, the curves of Fig. 2 show the variety of tone spectra that may be derived with a gliding frequency between 1 cps and 10 Kc applied to one and two fixed 440 and 880 cps frequencies (in octave relationship) applied to the other input of the ring-bridge modulator. The output frequencies are identified on the graph.

Frequencies applied to the ring-bridge modulator inputs are not limited to the audio range. Application of a subsonic frequency to one input will periodically modulate a frequency applied to the other. Application of white noise to one input and a single audio frequency to the other input will yield tuned noise at the output. Application of a percussive envelope to one input simultaneously with a steady tone at the other input will result in a percussive-type output that will have the characteristics of the steady tone modulated by the percussive envelope.

The unit shown in Fig. 3 provides congruent envelope shaping as well as the coincident percussive envelope shaping of the program material. One input accepts the control signal while the other input accepts the material to be subjected to envelope shaping. The processed audio appears at the output of the gating circuit.

To derive control voltages for the gating functions, the audio at the control input is amplified, rectified and applied to a low-pass filter. Thus, a relatively ripple-free variable d-c bias will actuate the variable-gain, push-pull amplifier gate. When switch S_1 is in the gating position, the envelope of the control signal shapes that of the program material.

To prevent the delay caused by C_1 and C_2 on fast-changing control voltages, and to eliminate asymmetry caused by the different output impedances at the plate and cathode of V_2, relatively high-value resistors R_3 and R_4 are inserted between phase inverter V_2 and the push-pull output of the gate circuit. These resistors are of the same order of magnitude as biasing resistors R_1 and R_2 to secure a balance between the control d-c signal and the audio portion of the program material.

The input circuits of V_5 and V_6

color; thus most of their compositions cannot express the emotions they feel and wish to communicate. New forms of music await new type instruments or the development of sound synthesizers that can produce all possible audio tones, in any combination, at any amplitude, and with any form of envelope shaping that will enable any imaginable combination of tones to be played.

Synthetically produced sound offers a broad canvas upon which all tones, regardless of their nature, can be painted, thus opening new avenues for the composers of tomorrow

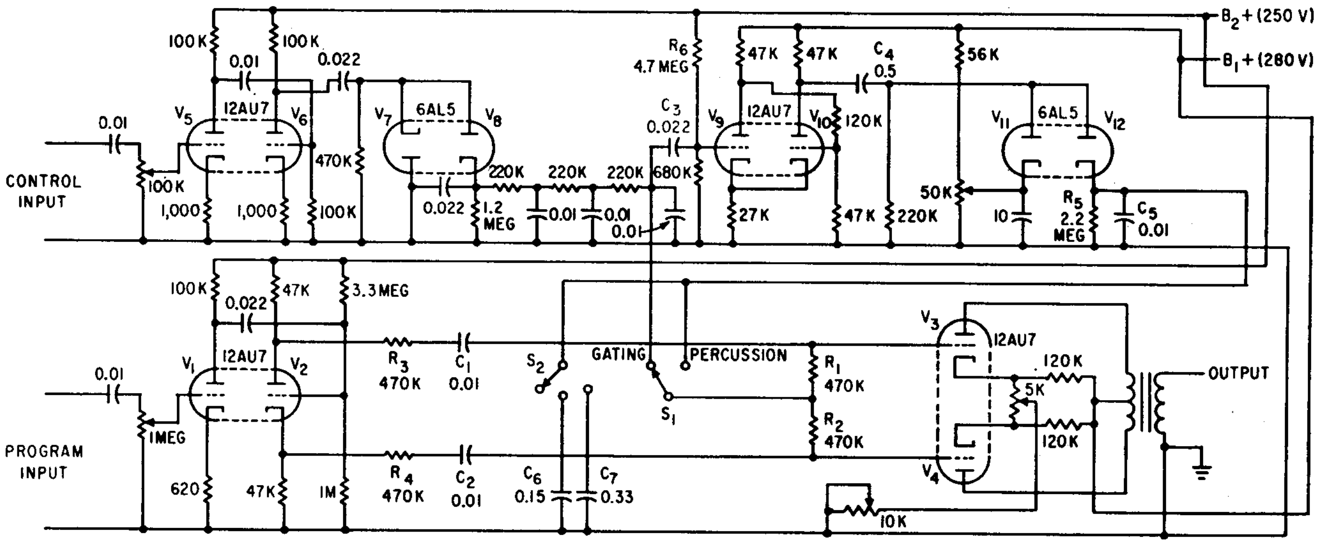


FIG. 3—Audio-controlled gate and percussion unit

act as a high-pass filter. The cutoff frequency of these filters exceeds that of the ripple filter by such an amount that no disturbing audio frequency from the control input will feed through to the gate. This is important for clean operation of the percussive envelope circuit. The pulses that initiate the percussive envelopes are generated by Schmitt trigger V_9 and V_{10}. Positive-going output pulses charge C_5 (or C_5 plus C_6 or C_7 chosen by S_2) with the discharge through R_5. The time constant depends on the position of S_2.

To make the trigger circuit respond to the beginning of a signal as well as to signal growth, differentiator C_3 and R_6 plus R_7 is used at the input of V_9. The response to signal growth is especially useful in causing the system to yield to a crescendo in a music passage or to instants of accentuation in the flow of speech frequencies.

The practical application of the audio-controlled percussion device within a system for the production of new musical effects is shown in Fig. 4. The sound of a bongo drum triggers the percussion circuit, which in turn converts the sustained chords played by the organ into percussive tones. The output signal is applied to a tape-loop repetition unit that has four equally spaced heads, one for record and three for playback. By connecting the record head and playback head 2 in parallel, output *A* is produced. By connecting playback head 1 and playback head 3 in parallel, output *B* is produced, and a distinctive *ABAB* pattern may be achieved. Outputs *A* and *B* can be connected

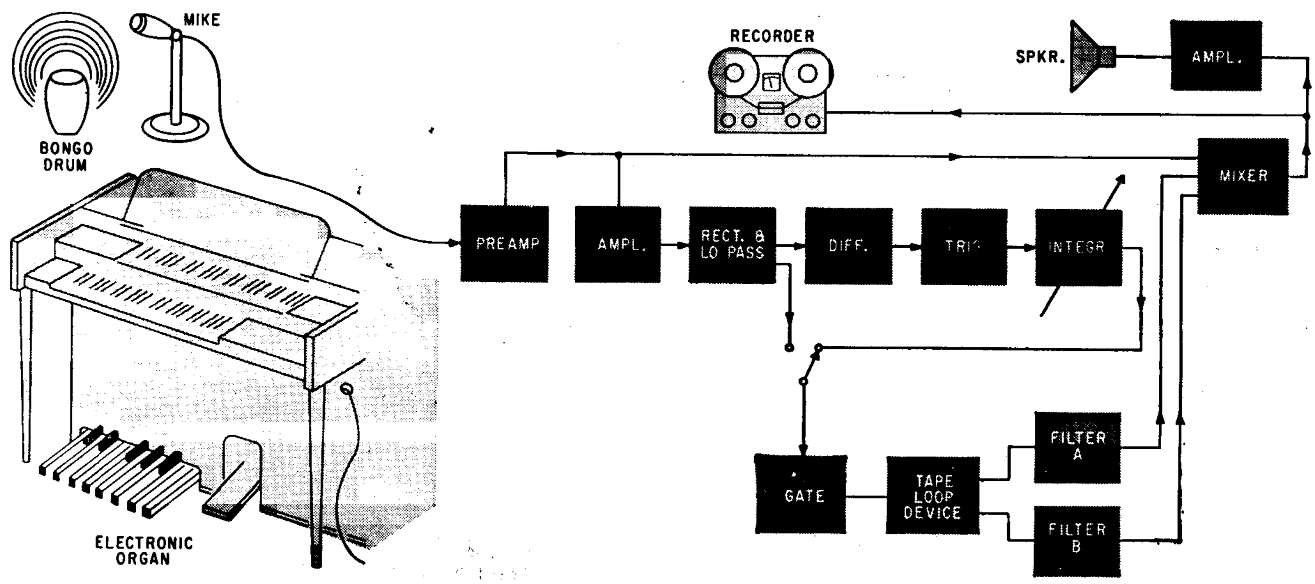


FIG. 4—Organ chords are given drum envelope to produce musical drum effect. Other instruments, including voice, may be substituted for the organ

to formant filters having different resonance frequencies.

The number of repetitions may be extended if a feedback loop is inserted between playback head 2 and the record amplifier. The output voltages of the two filters and the microphone preamplifier are applied to a mixer in which the ratio of drum sound to modified percussive organ sound may be controlled.

A presentation dealing more specifically with the tape-loop device and the sound-processing modules used in the synthesizer is shown in Fig. 5.

The program material originating from the melody instrument is applied to one of the inputs of the audio-controlled gate and percussion unit. There it is gated by the audio from a percussion instrument. The percussive melody sounds at the output of the gate are applied to the tape-loop repetition system. Output signal *A*—the direct signal and the information from playback head 2—is applied through amplifier *A* and filter 1 to the mixer. Output signal *B*—the signals from playback heads 1 and 3—is applied through amplifier *B* to one input of the ring-bridge modulator. The other ring-bridge modulator input is connected to the output of an audio signal generator.

The mixed and frequency converted signal at the output of the ring-bridge modulator is applied through filter 2 to the mixer. At the mixer output a percussive *ABAB* signal (stemming from a single melody note, triggered by a single drum signal) is obtained. In its *A* portion it has the original melody instrument pitch while its *B* portion is the converted nonharmonic overtone structure, both affected by the different voicings of the two filters. When the direct drum signal is applied to a third mixer input, the output will sound like a voiced drum with an intricate aftersound. The repetition of the *ABAB* pattern may be extended by a feedback loop between playback head 2 and the record amplifier.

When applying the human singing voice to the input of the fundamental frequency selector, the extracted fundamental pitch may be distorted in the squaring circuit and applied to the frequency divider (or dividers). This will derive a melody line whose pitch will be one octave lower than that of the singer. The output of the frequency divider may then be applied through a voicing filter to the program input of the audio-controlled gate and percussion unit. The control input of this circuit may be actuated by the original singing voice, after having passed through a low-pass filter of such a cutoff frequency that only vowels—typical for syllables—would trigger the circuit. At the output of the audio-controlled gate, percussive sounds with the voicing of a string bass will be obtained mixed with the original voice of the singer. The human voice output signal will now be accompanied by a coincident string bass sound which may be further processed in the tape-loop repetition unit.

The arbitrarily selected electronic modules of this synthesizer are of a limited variety and could be supplemented by other modules.

A system synthesizer may find many applications such as exploration of new types of electronic music or as a tool for composers who are searching for novel sounds and musical effects. Such a device will present a challenge to the imagination of composer-programmer.

The modern approach of synthesizing intricate electronic systems from modules with a limited number of basic functions has proven successful in the computer field. This approach has now been made in the area of sound synthesis.

With means for compiling any desired modular configuration, an audio system synthesizer could become a flexible and versatile tool for sound processing and would be suited to meet the evergrowing demand for exploration and production of new sounds.

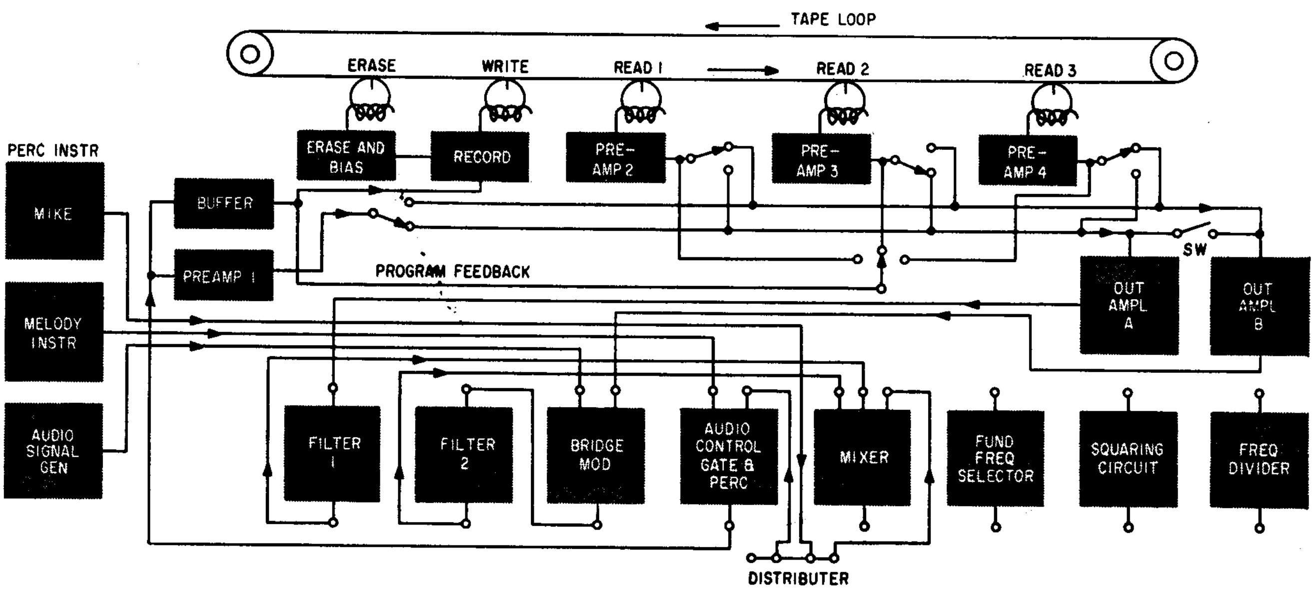


FIG. 5—Overall synthesizer system with tape-loop device and sound and envelope shaping units. Various combinations of shaping circuits can be patched together to generate desired tonal effects