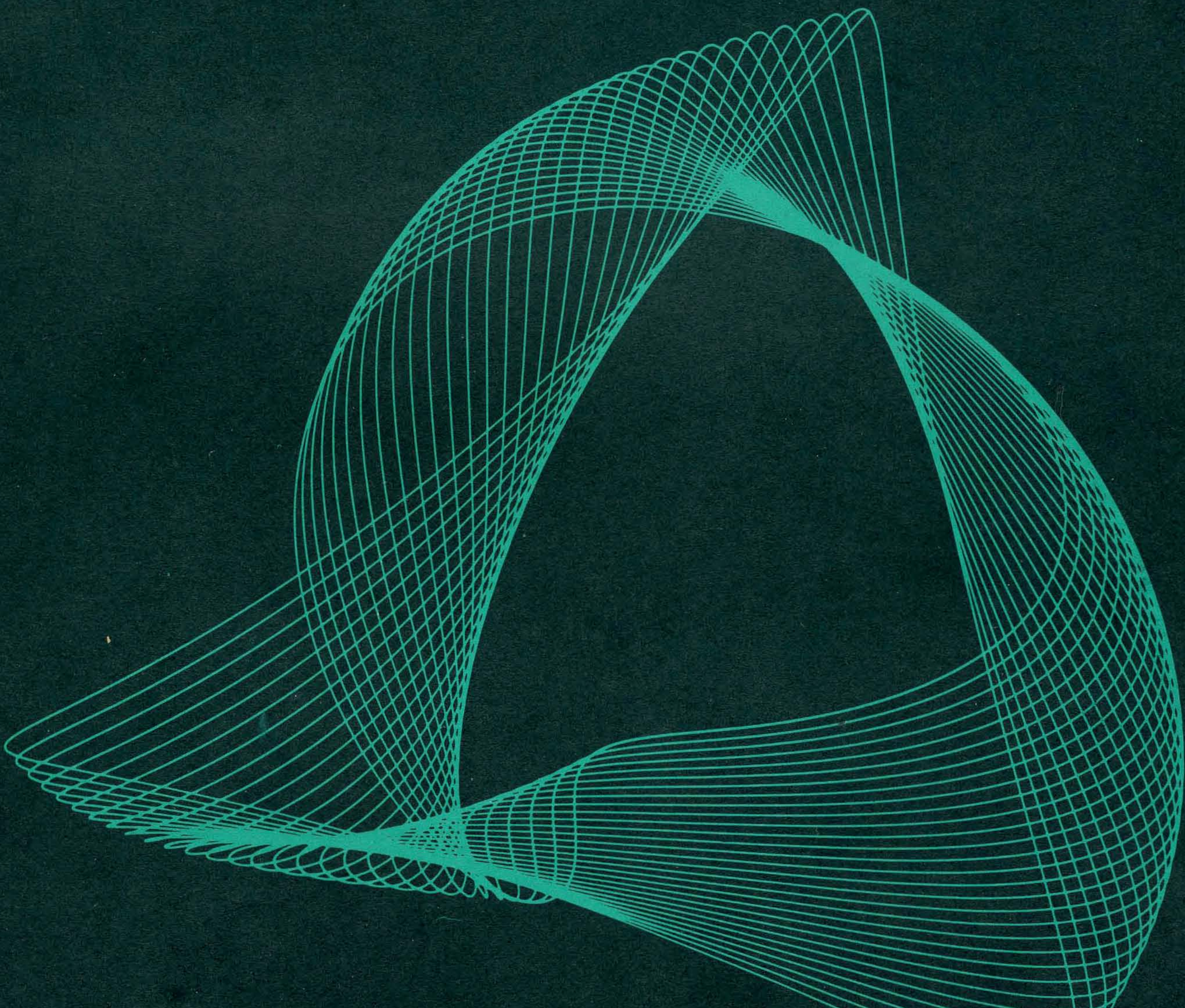
Electronic Music Review

No. 4 October 1967



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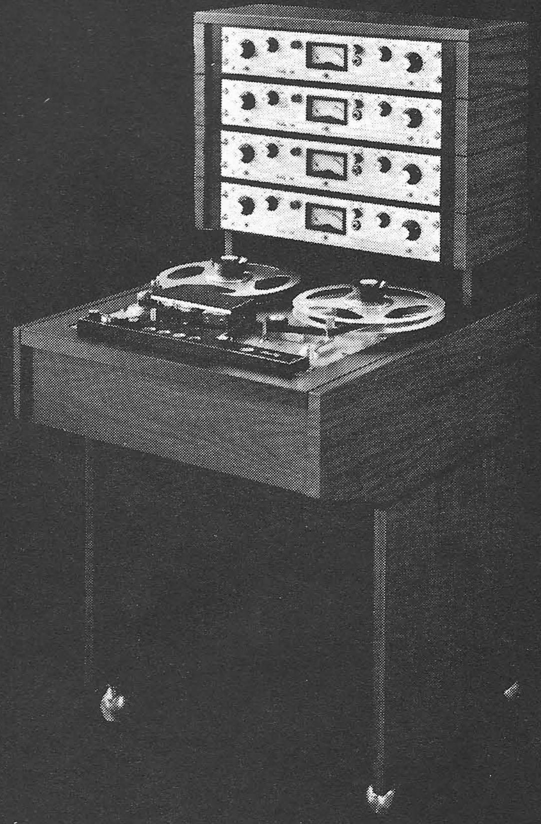

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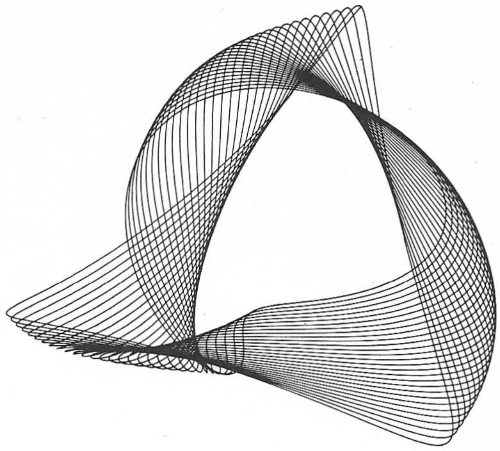
Electronic Music Review



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Reynold Weidenaar, Editor; Yael Gani, Associate Editor; Robert A. Moog, Technical Editor.

EMR is published quarterly by the Independent Electronic Music Center, Inc., Trumansburg, N.Y. 14886. Personal subscriptions are available through IEMC membership (annual dues $6); institutional subscriptions, one year $8, two years $15. Outside North America, add 50¢ per year, payable in U.S. funds.

EMR is indexed in Music Article Guide and in International Repertory of Music Literature (RILM).

EMscope

Due to the fact that EMR Nos. 2/3 (*Répertoire International des Musiques Electroacoustiques / International Electronic Music Catalog*) has undergone further production delays and will be unavailable for about two months, we have decided to go ahead with publication of this issue (EMR No. 4). Production of forthcoming issues will continue on schedule, and we hope that all 1967 members and subscribers will receive their copies of Nos. 2/3 in February. We deeply regret these continuing delays, but we must emphasize that this special issue is truly a massive undertaking, and will doubtless be one of the most unique and useful publications in the field of electronic music.

THE ADVISORY COUNCIL

The IEMC has established a large and active Advisory Council comprised of leading musicians and engineers. Through individual communication and consultation, the Council members provide criticism, comments, and recommendations to the Editors regarding the content and direction of the magazine. Most of the Council members are directly involved in electronic music. The few that are not were asked to join in order to provide a broader perspective of evaluation from their positions as leading representatives of contemporary music in general. EMR is concerned with electronic music as a useful medium for many styles of musical expression, not as a single esthetic credo. The diverse membership of the Advisory Council reflects this concern. The members of the Council are:

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Gottfried Michael Koenig - Composer; Director, Studio voor Elektronische Muziek, Rijksuniversiteit te Utrecht.
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Fritz Winckel - Composer; Director, Studio für Elektronische Musik, Technische Universität Berlin.
Charles Wuorinen - Composer, Columbia-Princeton Electronic Music Center, New York City.
Iannis Xenakis - Composer, Indiana University, Bloomington.
La Monte Young - Composer, New York City.

MEMBER SERVICES

STUDIO - The IEMC electronic music studio is now available for rental by members. Information on rates and equipment facilities may be obtained upon request.

INFORMATION SERVICE - Members may submit questions of a technical or general nature to be answered either by letter or in the pages of EMR. The Editors will either answer such inquiries in full or refer the member to organizations or reference literature where an answer may be obtained.

EQUIPMENT EXCHANGE - Members who wish to buy, sell, or trade equipment may list such information in EMR. The cost is $5; maximum 50 words.

COMPETITIONS

The Department of Music, Dartmouth College, announces the Dartmouth Arts Council Prize ($500) for an outstanding composition of electronic music. The judges of the competition will be Milton Babbitt, Vladimir Ussachevsky, and George Balch Wilson. Deadline for submission of works is March 1, 1968, and the results of the competition

will be announced April 6, 1968. Rules and regulations may be obtained by writing Mr. Jon Appleton, The Dartmouth Arts Council Prize, Griffith Electronic Music Studio, Dartmouth College, Hanover, N.H. 03755.

The International Federation for Information Processing announces medals for three musical works composed by a computer; the theme may be supplied to the computer, but the finished composition must be determined entirely by the action of the computer. Entries will be judged by a panel of musicians and computer programmers, and winning ones will be performed at the IFIP Congress at Edinburgh in August 1968. Further details are available from the U.S. Committee for IFIP Congress, 345 E. 47 St., New York City 10017, or from the IFIP Congress, 23 Dorset Sq., London N.W.1, England.

SEMINAR

David Tudor will conduct an informal seminar in live electronic performance at the Tape Music Center at Mills College, December 3 - January 20 and March 24 - April 21. The fee is $50. Further information is available from the T.M.C., Mills College, Oakland, Calif. 94613.

RECENT PUBLICATIONS

Beauchamp, James W. *A Computer System for Time-Variant Harmonic Analysis and Synthesis of Musical Tones*. 1967. Experimental Music Studio, School of Music, University of Illinois, Urbana, Ill. 61801. Softbound - free.

Cage, John. *A Year From Monday*. 1967. Wesleyan University Press, Middletown, Conn. 06457. Hardbound - $7.92.

Contactorgaan Elektronische Muziek (Prospectus). 1967. C.E.M., Marius Bauerstraat 30, Amsterdam 17, Netherlands. Free.

Cross, Lowell. *Bibliography of Electronic Music*. 1967. University of Toronto Press, Toronto 5, Canada. Hardbound - $5.00.

E.A.T. News (first issue). January 1967. Experiments in Art and Technology, Inc., 9 E. 16 St., New York City 10003. Free.

Küpper, Leo. *Couleurs Sinusoïdales et Potentiels d'Attraction des Sons Sinusoïdaux*. 1966(?). Studio de Recherches et de Structurations Electroniques Auditives, 26, Av. Jeanne, Bruxelles 5, Belgium. Softbound - free.

Meyers, Robert G. *Technical Bases of Electronic Music*. In Spring 1964, Winter 1964, and Winter 1966 issues of *Journal of Music Theory*, Yale School of Music, New Haven, Conn. 06520.

Olson, Harry F. *Music, Physics and Engineering*. 1967. Dover Publications, Inc., 180 Varick St., New York City 10014. Softbound - $2.75.

Schaeffer, Pierre. *Traité des Objets Musicaux*. 1966. Editions du Seuil, Paris.

RECENT STEREO LP RECORDS

CBS 32110044 - Karlheinz Stockhausen (*Mikrophonie I; Mikrophonie II*).

COLUMBIA MS 7051 - Milton Babbitt (*Ensembles for Synthesizer*), John Cage (*Variations II*), Henri Pousseur (*Trois visages de Liège*).

ELEKTRA EKS 74009 - Mort Garson (*The Zodiac*).

FOLKWAYS FMS 33436 - Robert Aitkin (*Noesis*), Victor Grauer (*Inferno*), Jean Eichelberger Ivey (*Pinball*), Hugh Le Caine (*Dripsody*), John Donald Robb (*Collage*), Myron Schaeffer (*Dance R4÷3*), Val T. Stephen (*Fireworks; Orgasmic Opus*), Arnold Walter / Harvey Olnick / Schaeffer (*Summer Idyl*).

HELIODOR HS 25053 - Lejaren Hiller / Robert A. Baker (*Computer Cantata*), Hiller / Leonard M. Issaacson (*Illiac Suite*).

NONESUCH 71174 - Morton Subotnick (*Silver Apples of the Moon*).

ODYSSEY 32160156 - Robert Ashley (*She Was a Visitor*), John Cage (*Solos for Voice 2*), Alvin Lucier (*North American Time Capsule 1967*).

ODYSSEY 32160158 - Gordon Mumma (*Mesa, for Cybersonic Bandoneon*).

ODYSSEY 32160160 - Richard Maxfield (*Night Music*), Pauline Oliveros (*I of IV*), Steve Reich (*Come Out*).

OWL ORLP 11 - Tod Dockstader / James Reichert (*Omniphony I*).

RCA VICTOR VIC(S) 1239 - Henri Pousseur (*Rimes pour différentes sources sonores*).

SVERIGES RADIO (Box 955, Stockholm 1, Sweden) RELP 5023 - Ralph Lundsten (*EMS NR 1*), Leo Nilson (*Aurora; Skivsida 2*), Lundsten / Nilson (*Tre elektroniska "pop"-stycken; Visioner av flygande tefat*).

TURNABOUT 34177 - Luciano Berio (*Omaggio a Joyce*), Jacob Druckman (*Animus I*), Ilhan Mimaroğlu (*Six Preludes for Magnetic Tape*).

VANGUARD VSD 79222 - Gershon Kingsley / Jean-Jacques Perrey (*The In Sound from Way Out*).

PLEASE NOTE

Information on recent records and publications, establishment of new studios, forthcoming concerts, lectures, seminars, special events, etc., should reach EMR no later than one month before month of publication

Superserialismus–Is There a Cure?

Tristram Cary

The basic criteria of musical (or indeed any artistic) judgement do not change. Human Art is by definition a human affair, invented by Homo Sapiens to please other Homines Sapientes. We can still enjoy the plays of Sophocles and Aristophanes because they are talking about exactly the same things as we are talking about today.

The 20th century flatters itself that it has invented new things, new devices for artistic communication, new horizons of creative surveillance. But, of course it hasn't, except in the most superficial sense. Certainly a musician has today a wider range of raw materials, but I am reminded of a composer who visited my studio and tried out a few manipulations of sound. "Marvelous," he said, intoxicated with power as he twiddled all the wrong knobs and filled the room with great roars and bangs, "you can make any sound you like." But you can't really; you can only make more than before. The same will apply to the most sophisticated studio of the future which we can picture in our wildest flights of science fiction imagination. The kind of comment made by my visitor might have been made if Monteverdi had been able to listen to Berlioz's orchestra. It might have seemed to him that the resources of instrumental sound had been so enlarged that anything was possible.

But the parameters of music as an art are firmly and irrevocably tied to the human imagination as creator, and the human ear and brain as receptor and interpreter. Naturally the composer uses the resources of the age into which he happens to be born, which is why a lot of us spend time chopping up little bits of brown ribbon instead of composing cantatas; but our problem is no different — we must have something to say and we must (even if it's a long time later) get through to someone what we are talking about.

It is axiomatic that many composers tend to be out of step with their public, and also that a good 90% of the work of any age is evanescent tripe which will die with its period. Some of the musical public apparently think that this time we really have gone too far, but of course this is not a new view either — one critic pronounced after the first performance that Tannhaüser had not a single recognizable tune from beginning to end — an opinion about a highly melodic work with which we find it hard to agree. But there does all the same appear to be a different slant to our thinking, and this is caused by the extramusical factors which intrude into much of our work. The conventional composer naturally concerns himself with the fingering the clarinetist must use and other necessary technical matters, but he doesn't want to know (why should he?) what waveform the instrument is producing — he only wants to know what it sounds like.

In our workshops we do want to know, indeed must know, what our raw materials are made of, but we must also guard carefully against making our little green pictures or our calculations an end in themselves. Musically speaking, it is only the sound that means anything. When I was studying basic harmony, we were actually encouraged to do our exercises visually, by chord shape on the paper, and certainly this enabled one to do a correct exercise when the pandemonium of a music college prevented the mind's ear from hearing clearly. But it had little to do with training a composer.

The evidence of a lot of work that is going on rather supports this method, however. We tend to suffer from Superserialismus, or the pursuit of the perfect crossword puzzle. Ever since Schoenberg, the pitfall of serialism has been that it provides a refuge of acceptable academicism for the creatively underendowed (this does not apply to Schoenberg — needless to say). The working out of an idea to its bitter technical end is now more than ever possible, and we can call on computers to make jolly sure that no stone is left unturned. It is clear that a large number of people think we are enlarging our artistic horizons by this kind of means, but I doubt it. I use the word "artistic" advisedly — there is no doubt, of course, that we are increasing our knowledge of musical physics.

It is impossible for anyone to assess the permanent value of work going on around him. There seems, though, to be a rather alarming lack of actual contact not so much with the listening public as with the people who might be expected to be more receptive — one's fellow musicians. It is interesting to note that early concrete pieces, like Symphonie Pour Un Homme Seul, are still tremendously effective because they use the new language as a fresh, lively, "happening" thing, instead of as a laboratory of sound. The impact is musical and direct, and we do not feel underprivileged if we haven't been told all about the modus operandi. We can hear the machinery if we stop and think, but it is the direct experience of the sound that counts.

This is not an impassioned advocacy of empirical over serial methods, but simply an observation of intelligent people listening to things. "Serialism" I am using in the strict sense of organization of some or all parameters (not necessarily just linear pitch progression), applied to a piece of music and seen through to its conclusion. A lot of serial instrumental music is not serial by this standard, and much electronic music which would like to be is not because it is not accurate enough. But what we have to question is the esthetic validity of such procedures. Simply put, the difference is this: — Q: Why did you make that sound? A: (from serial composer) Because its pitch is number 4 in a thus and thus series which I have calculated; because its loudness is part of a controlled program of dynamics on thus and thus principles; because its length is determined by a thus and thus metric structure. This is the sound, friend — there is, there could be no other sound in this place. A: (from non-serial composer) Because it seems musically right at this point; because I have been writing music for X years and am prepared to back my experience and intuition; because I rather like it.

The practical composer almost always partakes a little of both philosophies, but I think it is incontrovertible that electronic composers tend more to schizophrenia on this matter than others — because accuracy is expected of them, and because in the absence, very often, of clear esthetic guidelines it is easiest to fall back on serial justifications of procedures. "I did it because I like it" takes more courage to maintain in the face of "You like that!!" than "I did it because the principles of this style of composition demand it."

In our haste to explore the physics and mathematics of what we are discovering, we perhaps tend to forget the psychology. There is a lot of work to be done on the subjective effect of sounds that are taken out of context but in themselves carry a charge of familiar suggestion. At its simplest, we can do what the pop artist does by painting a can of soup ten feet high. By presenting a sound in a special context we create by that act alone a special dimension for it, and cause the listener to hear it anew. If we splice together, say, a distant natural sound, a human vowel, an "unknown" electronic sound, and a close natural sound, we place the recognizable sounds in a brand new light, although our recognition is still there. Our studios enable us to integrate the total aural experience in a way which could never be done before, and we cannot ignore this capability even when we are deliberately aiming at abstraction. Composers have broken

down and wept when, after weeks of working at a new, subtle, and exact sound-mixture, the first person to hear it says it sounds like some sort of organ.

The more sophisticated studios now have such complete resources that it requires enormous self-discipline by the composer not to be merely pyrotechnic and astonishing, but also thoughtful and interesting. If you write both sorts of music there is a moment in your career when for the very first time you are scoring for full symphony orchestra, and it is a heady experience. As your ideas develop you have to make your own disciplines and restrictions to prevent the richness of the possibilities from getting out of hand. I seem to detect in some electronic music that the Martians have begun to take over, and the composer is letting the machinery have its way. This road leads to total abstraction and total lack of communication, because the essential mind-to-mind contact is replaced by a machine-to-mind contact, which may or may not be successful, and will certainly be unpredictable.

Experiment and research have a great fascination in themselves, and every possible way of doing everything should be tried, but I think we must recognize that a lot of our results may be of minimal interest as permanent works of art. The definitive great works of First Period Electronic have yet to be written (or I have missed them), and if they are too long in coming we must re-examine the premises of our arguments. Changes are afoot in instrumental music, and there is even a return to — oh horrors — recognizable tunes. It is certain that the fundamental work now proceeding will add to knowledge and understanding, but only our great-grandchildren will know whether it added anything to Art in its time, or went to join the Great Magnetic Junkheap.

While it is true that there is nothing new, only new clothes for old ideas, we have certainly made for ourselves a gaudier and costlier suit of clothes than ever before. The difficulty is that many musicians are uncertain how to wear them, and many engineers are designing new styles which may or may not be musically interesting. The tail has often wagged the dog in human affairs, but let us hope to make an exception in this case.

O Electrons (as Sophocles never said), O Recalcitrant Holes and all Ye Assembled Nand Gates, Keep Your Appointed Places — under my thumb.

Introduction to Mixers and Level Controls

Robert A. Moog

Basically, a mixer is a device with at least two inputs and at least one output. The signal at an output is the linear algebraic sum of the signals applied to the inputs, in proportions determined by setting the level controls and routing switches. These level controls and routing switches may either be panel-mounted mechanical devices, or may be voltage- or light-activated components. Filtering (usually called equalization in mixer applications), reverberation, and other strictly linear operations may also be available.

A level-control amplifier (or voltage-controlled amplifier) is a device with a signal input, a signal output, and one or more control inputs. The magnitude of the signal output is directly proportional to that of the signal input, the constant of proportionality being determined by the magnitudes of the voltages applied to the control inputs. A mixer that uses level control amplifiers in place of mechanical level controls is called a voltage-controlled mixer, and is useful primarily for rapidly varying the signal level, or rapidly changing the signal routing.

The ideal mixer would have as many inputs as there are signal sources in a given situation, and as many outputs as there are channels of recording or monitoring. It would accept signals over as wide a range of levels as possible, and introduce no noise or distortion. It would offer smooth, convenient control over the amplitude of the various inputs, and the facility for routing any combination of inputs to any of the outputs. The ideal level control amplifier would accept signals over an extremely wide range of levels, and would introduce no noise or distortion. Its gain would be voltage-variable over a very wide range, and would respond instantaneously to changes in control voltage. However, control voltage variations would not feed through to the output.

The objective specifications by which existing mixers and level control amplifiers are described serve to indicate the extent to which the device in question departs from ideal behavior. Basic characteristics which are generally specified are listed below. Since most of these specified quantities have been adopted because of their use in specifying conventional audio equipment, their significance within the framework of electronic music composition will be indicated.

LEVEL: Level is synonymous with amplitude or strength in describing a signal. A device input is rated according to the optimum level of signal for which it was designed, and sometimes also for the maximum level that it can accept without exceeding a certain percentage of distortion. A device output is rated according to the maximum level of signal that it can deliver without excessive distortion. Two units of level are commonly used: the volt and the decibel. The volt is a measure of electrical force, while the decibel, strictly speaking, is proportional to the logarithm of the electrical power. In present day audio practice, the concept of power transfer between two devices is not often invoked except in very large installations, and the decibel is used instead as a measure of the logarithm of the voltage: 0dB is usually taken to be that voltage which appears across a 600-ohm resistor that is dissipating one milliwatt of power, or .774 volt. An increase of 20 dB in the signal level increases the voltage by a factor of 10; an in-

crease of 6 dB increases the voltage by a factor of 2. The standardization of signal levels among the instruments in a studio contribute to operating convenience and minimization of noise and overload problems. The standard level in a given studio is called "line level". Most professional audio equipment is designed to work with line levels of either +4 dB or +8 dB, since these signal levels are readily produced and yet are high enough to effectively mask the spurious noises which are usually present. Microphones, phonograph pickups, and tape reproduce heads normally produce signals of much lower level than +4 dB, and their signals require preamplification to be consistent within a studio. On the other hand, speakers require power levels in excess of +30 dB, and power amplifiers are used to raise line level signals to suitable power levels.

IMPEDANCE: Impedance is a measure of the voltage required to cause a standard amount of current to flow. The output impedance of a device is the impedance which would be observed at the output if the device were producing no signal. As with level, it is advantageous to standardize impedance within a studio. The standard, or line, impedance of professional audio equipment varies from 50 to 600 ohms, and is selected on technical considerations. The matching of impedances of an output and the input which it feeds is necessary only if the transmission line is long, or if the signal level is low; e.g., microphone impedances must be matched. With line level equipment, however, it is usually permissible to have all inputs high impedance, and to match impedances by connecting appropriate resistors across all outputs. This arrangement enables one output to feed many inputs.

FLOATING, BALANCED, AND UNBALANCED LINES: A floating output is one in which the voltage produced is not referred to ground or any other fixed electrical reference. A balanced output is one in which two voltages of opposite polarity are produced; only one voltage is produced in an unbalanced output. Outputs may be either floating, balanced, or both. Each reduces the amount of extraneous noise induced in the transmission line.

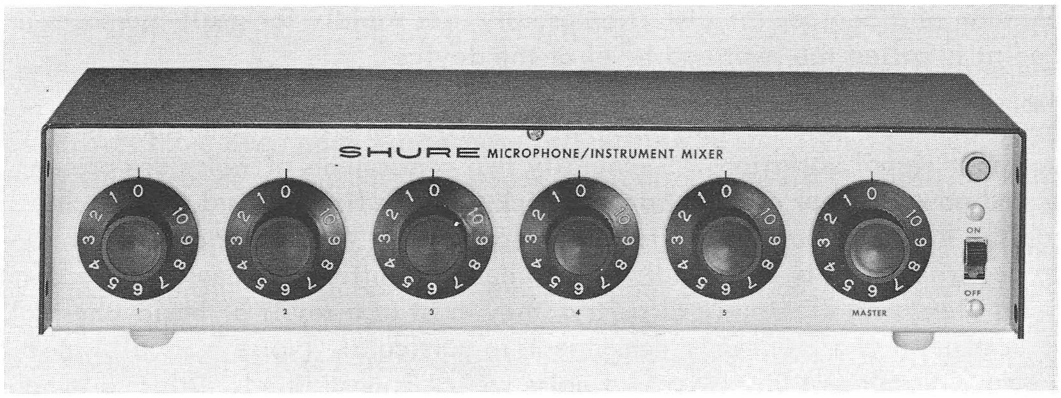
DISTORTION: In any device using non-linear components (this includes virtually all studio equipment) the signal is distorted and additional frequencies (harmonics or modulation products) are produced. Distortion is generally listed as a percentage of the desired signal. While some types of distortion are more audible than others, a distortion of less than 1% is generally inaudible, while a distortion of greater than 2% is audible. Distortion is added to a signal by every device through which it passes. Therefore, large studios using many signal-handling steps require very low distortion figures for each signal-handling device. On one point on a distortion vs. signal level graph of a device, the distortion usually rises rapidly for small increases in signal level. This point is called the overload level of the device.

NOISE: Of the unwanted output of a device, noise is the portion where the amplitude is constant with respect to input signal variations. Generally two components of noise are present: random fluctuations and power line or other extraneous pickups. The first is heard as a pitchless hiss, crackle, or roar; the latter is heard as a pitched hum or whine. While it is possible to reduce hum pickup to negligible levels by shielding and other straightforward measures, random fluctuations are always present in electronic circuitry; they exist as a result of basic physical properties of matter in general and electronic components in particular. Noise is usually specified in one of two ways. Sometimes an equivalent noise source is postulated. This is a hypothetical signal generator that, when connected to the input of a device, produces the noise observed at the output. The magnitude of the hypothetical generated noise signal is called "equivalent input noise", and is usually on the order of a few microvolts. Noise can also be specified in terms of the ratio between the noise level at the output of the device and the signal level produced at the output at the onset of overload. This ratio is called the "dynamic range", and may lie between 60 and 120 dB.

GAIN: Gain is the ratio of the magnitude of the output signal to that of the input. Line level mixers usually provide a small amount of gain (perhaps 20 dB), while microphone mixers provide 60 or 70 dB. Unlike noise and distortion figures, gain is not a measure of the merit of a mixer. The optimum amount of mixer gain for a given studio is that which produces line level output from the lowest level source which is likely to be used.

The arrangement of controls in a mixer varies according to the use of the mixer. There are three main functional sections of a typical mixer: input level controls to determine the proportions of the mixture, routing controls and switches to determine which inputs are to be fed into a given output channel, and output level controls (sometimes called master gain controls) to determine how much of the final mixture appears at an output. Small mixers are usually constructed as integral units; larger mixers are often assembled from widely available modular subassemblies. Fig. 1 is a photograph of a simple high quality mixer with five inputs and one output. This type of mixer is relatively inexpensive, and is suitable for a small studio or live performance. Fig. 2 shows a modular input amplifier and equalizer, and Fig. 3 shows a 12-input, 4-output professional audio mixer assembled from such modules.

The articles in this symposium discuss various aspects of the use of mixers in electronic music composition. James Seawright discusses the basic technical features of mixers. Gerald Shapiro describes a collection of simple modules that are particularly appropriate for electronic music. Hugh Le Caine's article on level controls includes an extensive discussion on control devices. Frederic Rzewski describes a performance-oriented mixer using photoresistors as level controls, while Fernando von Reichenbach discusses a photoresistor mixer designed for the realization of pre-programmed musical material. Finally, Robert A. Moog tells how to build a simple 2-output, battery operated mixer suitable for both composition and performance.



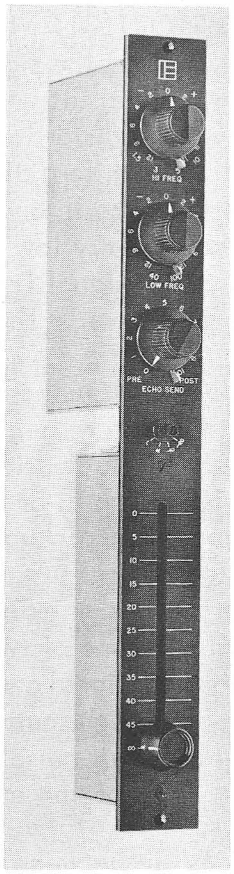


Fig. 1 (above). The Shure Model PE68M Microphone/Musical Instrument Mixer.
Fig. 2 (right). The Electrodyne Model 709L Equalizer-Amplifier Module. The input level control along the lower half of the panel is a slide-type attenuator, calibrated in decibels of attenuation.

Fig. 3. The Electrodyne ACC-1204 Audio Control Console. Note input level and equalizer controls (lower left), routing switches (upper left), and output controls (right). VU meters (top) indicate levels of outputs.

Definitions

Note: Definitions of many basic terms appearing in this symposium may be found in the January 1967 issue of EMR.

An audio signal is transmitted as an alternating (AC) voltage; control signals and power supply voltages are usually direct (DC) voltages. The level of a DC voltage refers to its instantaneous magnitude; the level of an AC voltage refers to its RMS average magnitude. "Level" is not used in referring to power supply or other voltages that carry no information. Level may be measured directly in VOLTS (a unit of electrical force) or in DECIBELS (dB), which are proportional to the logarithm of the signal magnitude.

An INPUT or LOAD receives information; an OUTPUT or SOURCE supplies information. The IMPEDANCE of an input is a measure of the voltage across it required to produce a given amount of current through it. The unit of impedance is the OHM (Ω). In a 1-ohm load, 1 volt across it produces 1 ampere through it. An AMPLIFIER is a device that increases the level of a signal; an ATTENUATOR reduces the level. The standard level in a studio is LINE LEVEL, to which signals are brought before they are manipulated. A PREAMPLIFIER brings low level signals to line level; a LINE AMPLIFIER restores attenuated signals to line level; a POWER AMPLIFIER increases a line level signal to a level appropriate for driving speakers or similar transducers.

GAIN is the ratio of the output level to the input level of an amplifier, and is usually expressed in units of dB. Similarly, ATTENUATION is the ratio of the output level to the input level of an attenuator. INSERTION LOSS is the minimum attenuation. EQUALIZATION is the tailoring of the frequency response of a device, usually to correct uneven frequency response of another device.

Mechanically activated attenuators are referred to as CONTROLS, or, more precisely, POTENTIOMETERS (or POTS). The TAPER of a pot is the relationship between the mechanical rotation or position and the attenuation. The level of a LINEAR TAPER pot is directly proportional to the mechanical rotation or position; the level of an AUDIO TAPER pot is proportional to the exponential of the mechanical position or rotation for at least a portion of the device's motion. The SLIDER of a pot (sometimes called WIPER ARM) moves along the resistance element across which the input is connected. Two or more pots are said to be GANGED when their sliders are mechanically linked to move together. A PANNING POT has one input and two outputs, and is used to continuously "move" the signal from one output to the other. A SEGUE POT has two inputs and one output and is used to "fade" from one input to the other.

The PHASE of an AC signal is a measure of the position of the signal in time with respect to some reference, and is usually measured in degrees: a 360-degree phase difference equals one period of repetition. Two signals are IN PHASE when the phase difference between them is zero or a multiple of 360 degrees; otherwise, they are OUT OF PHASE. In a common but less correct sense, two signals are said to be in phase when no phase difference exists between them, and out of phase when no phase difference exists between them but one is the algebraic negative of the other. This usage is applied to amplifiers, speakers, and other devices where polarity, rather than bona fide phase shift, is involved. An OPERATIONAL AMPLIFIER is a very high gain amplifier, the output of which is out of phase with the input. Resistors or other electronic components are connected between output and input to determine the characteristics of the device. A PUSH-PULL circuit (usually an amplifier) is a symmetrically designed amplifier that needs both in-phase and out-of-phase inputs.

A PHOTOCELL is any component whose characteristics depend upon the intensity of light incident upon it. The resistance of a PHOTORESISTOR decreases as the incident illumination increases. A PHOTOVOLTAIC CELL is a DC voltage source; the voltage increases as the incident illumination increases.

An INTEGRATED CIRCUIT is a semiconductor device that operates as a complete circuit. For example, complete low-power line amplifiers are now available as single, very small components. An OR GATE is a bi-state device with one output and two or more inputs. The output will be "on" when at least one of the inputs is "on". A CROSS BAR SWITCH is an array of single switches which allows the connection of any output to any input in any combination. A VOLUME COMPRESSOR AND EXPANDER is a level control amplifier whose gain is made to depend on the level of the input signal so as to either decrease or increase the dynamic range of the signal. A SCHMITT TRIGGER, or SCHMITT LEVEL DETECTOR, is a bi-state device that changes states when its input goes through a preset voltage.

Symposium: Mixers and Level Controls

Fundamental Concepts of Electronic Music Mixers

James Seawright

One of the most frequently performed operations in the electronic music studio is that of mixing. The operation is probably performed a great deal more often than the composer realizes, for many studio devices such as tape recorder preamplifiers, oscillators, reverberation units, filters, etc., actually incorporate mixers in order to increase their versatility. The mixer is perhaps taken for granted more than any other studio device, with the possible exception of the tape recorder, yet its potential usefulness extends far beyond its function as a combiner of signals and can affect the composer's entire attitude toward the organization and production of a composition.

Mixing is basically a process of adding signal amplitudes algebraically. At any given instant, the sum or output voltage must equal the input voltages (or at least be proportional to their instantaneous sum). Consider the case in which two signals of the same frequency and amplitude are mixed. If they are exactly in phase (Fig. 1A) the resultant will be a signal of twice the amplitude; if they are 180 degrees out of phase (Fig. 1B) the resultant will be an exact cancellation. Now consider the effect of mixing two signals of slightly different frequencies, but with equal amplitudes (Fig. 1C): Note that as the two signals go from an in-phase condition to an out-of-phase condition, the resultant amplitude goes from a maximum value of twice the amplitude of either input to a complete cancellation. The rate at which this happens will correspond to the difference in frequencies of the two signals. This is the reason that audible differences in amplitude, or "beats", are heard when two instruments are played at slightly different fre-

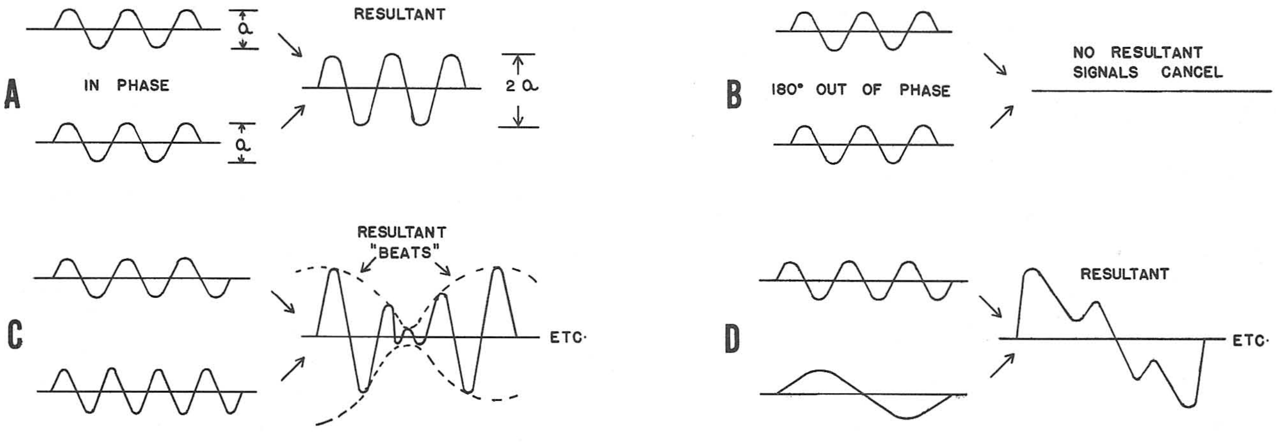


Fig. 1. Representative waveforms resulting from mixing sine waves of various frequency and phase relationships.

quencies. A final case, that of mixing two signals of greatly differing frequencies, is illustrated (Fig. 1D): Note that in every example the resultant is always the instantaneous algebraic sum of the inputs.

There are several ways in which the effect of mixing may be achieved with electronic circuitry. The usual way is by means of a resistive network which allows signals to combine and then appear, mixed, at a common output, yet minimizes unwanted interactions between signals or signal sources. Such a network is shown in Fig. 2A. In order to achieve the minimum of interaction between sources, the network must be designed in such a way that there is a large net loss in signal amplitude through it. If a mixing network has a large number of inputs, the mixing loss may be as high as 40 or 50 dB. In order to offset this loss, an amplifier is almost always included in the mixer to restore the signal output level to the nominal level of the inputs.

Before going into more detail, let us consider for a moment the way in which sounds are mixed acoustically. When an instrument is played, its vibrations are transmitted to the air. The air may be considered as a load against which the instrument works. Sound waves in the air produced by one instrument will naturally affect other instruments just as they affect our eardrums, by causing them to vibrate, but the effect, when considered as an interaction, is exceedingly slight. A second instrument being played in a room where the air is already vibrating from the first instrument simply imposes additional vibrations upon the air. The mixing process takes place in the air; at any given instant, the amplitude of the sound wave at a given point in the room is the algebraic sum of the various sound waves of the individual instruments which are being played (allowing for attenuation caused by propagation, and taking into account the phase differences caused by propagation time).

In the circuit of Fig. 2A, an analogous process takes place. Input signals applied to the input volume controls appear as voltages across the controls. Moving the slider on a control causes a sample voltage of an amplitude proportional to the position of the slider to be applied to the appropriate mixing network input. This network is seen by the voltage as a multiple-resistance path to ground (Fig. 2B). Depending on the values of the mixing resistors R_1 through R_6, and the resistor R_{in}, a voltage will appear at point A having a value proportional to the value of the voltage at the volume control slider. If more than one voltage is applied to more than one input, the mixing effect of the network will result in the voltage at point A being proportional to the algebraic sum of all the applied input voltages. Note that if the sliders of volume controls 2 through 6 are at ground, the resistances of R_2 through R_6 will have a certain net value as resistance in parallel with R_{in}, thus lowering the effective value of R_{in}. The fraction of the

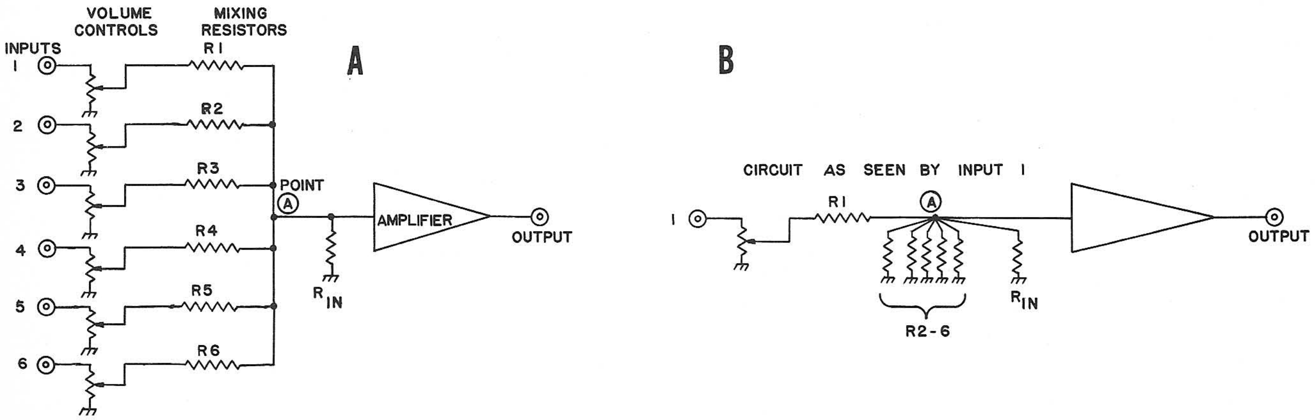


Fig. 2. Basic resistor-network/amplifier mixer.

input voltage at R_1 which will appear at point A will depend on the ratio between the value of R_1 and the net parallel resistance of R_2 through R_6 and R_{in}. Raising any of the other volume control sliders, as would be necessary to introduce another signal for mixing, adds resistance between point A and ground and raises the net parallel resistance value. Thus the actual value of the loss incurred by any given signal depends on the setting of the other inputs, and this interaction is minimized by making the mixing resistances R_1 through R_6 relatively high in value. However, the higher these values become, relative to the value of R_{in}, the higher the net mixing loss, so that a compromise has to be reached in all practical cases.

The resistance R_{in} in many cases is not an actual resistor, but the input resistance (impedance) of the amplifier which follows the mixing network. This amplifier, whose task it is to restore the gain lost in the mixing network, is the source of most of the signal distortions and noise found in mixers. The resistance network itself, ideally, is perfectly linear in operation. The amplifier, however, may not be, and if not will introduce distortion. The necessity of having upwards of 70 dB gain, to overcome mixing losses and have a reserve of gain, requires that the signal-to-noise characteristic of the amplifier be excellent.

So far we have considered a mixer which, in the example of Fig. 2, is capable of mixing six signals in any proportion to form one resultant. Such a mixer might, in practice, be used to combine oscillator outputs to form complex timbres, or to mix several tape loops running at different speeds. The usefulness of such a mixer would be increased considerably, however, by providing means whereby input signals could be mixed at several outputs, either separately or simultaneously. With this type of mixer, various signals could be mixed in differing proportions for a two-channel output to a stereophonic tape recorder.

There are several approaches to this type of multi-channel mixer, and in each case certain peculiarities. The first approach might be as shown in Fig. 3A. Here, each input channel may be switched after its volume control to the output channel at which it is desired. Note that the arrangement of the switch is such that the input of the mixing network not being used is grounded. A mixing network and amplifier will be needed for each output channel, but the arrangement offers economy and simplicity of construction.

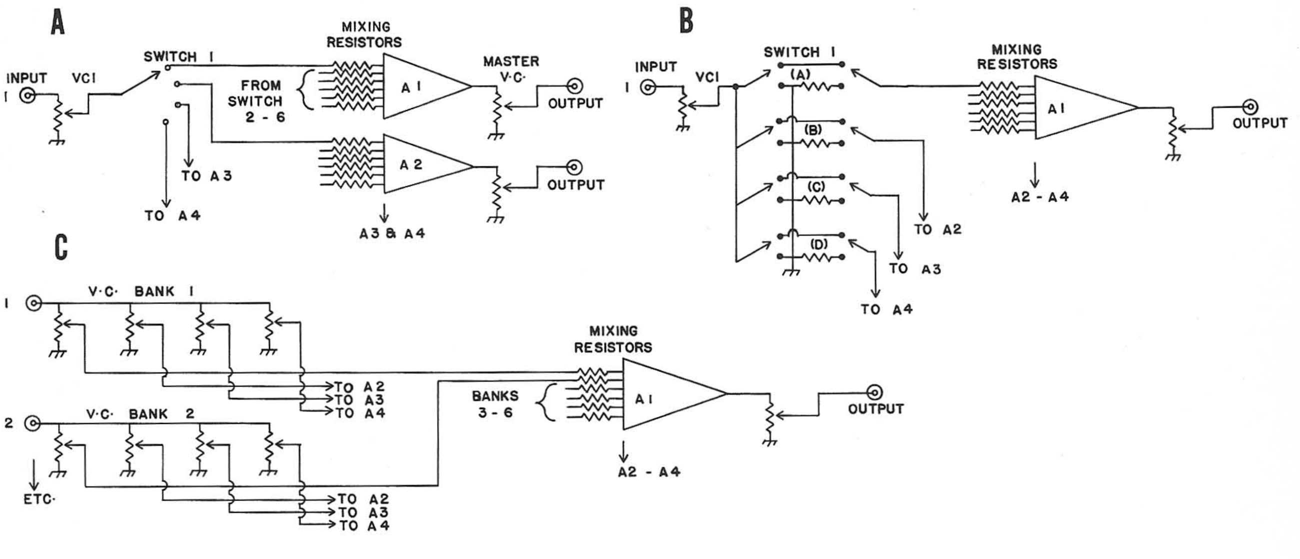


Fig. 3. Representative multi-channel mixer configurations.

A slightly more versatile plan is shown in Fig. 3B. Here, the selector switch has been replaced with a bank of individual switches or push-buttons, so that a signal at a given input may be sent to any combination of outputs. Obviously, since the level of the input signal has already been determined by the volume control setting, the signal will appear at the same level at each of the outputs to which it is sent.

The most versatile arrangement is that of Fig. 3C. A matrix of volume controls, in effect a separate set of input volume controls for each output channel, allows the user to send an input signal to any combination of outputs at any combination of levels. This system is quite workable up to about six inputs and four output channels, but beyond this begins to involve such a demand for panel space for controls that it becomes cumbersome. In all of the three versions of multi-channel mixers described, master volume controls may be provided in the output lines, as shown.

In the actual design and construction of mixers, several important factors appear. First of all, a mixer can be designed to operate at optimum quality for only one general category of signals. A microphone mixer, for example, must be designed to match the output characteristics of the microphones with which it will be used. Sometimes a mixer may be arranged to have a variety of inputs, each suitable for a specific class of signals. A microphone input would then have to incorporate an additional stage of preamplification to raise the signal to a level compatible with other line level signals from other inputs with which it may be mixed. In commercial and broadcast audio practice, mixers are usually designed to have a 600-ohm input and output impedance. Since all the other devices in the commercial studio have 600-ohm outputs, they may be connected to the mixer inputs without loss. Only in the situation in which it is desirable to connect a 600-ohm output device to more than one 600-ohm input at the same time do difficulties arise. This situation occurs frequently in electronic music when, for instance, the composer wishes to set up several mixtures of the same set of signals through several channels of a mixer, and then fade from one to another. A tape channel may be connected to several inputs at the same time, each input being set to a different level, and routed to a different output. (This is very likely to occur with mixers of the type shown in Fig. 3A.) In order to avoid the loading effect which occurs when an output drives an excessively low impedance load (such as three 600-ohm inputs in parallel, a net resistance of 200 ohms), and the consequent loss in signal level and possible distortion, it is much more desirable to design the mixer to have high-impedance inputs; high-impedance inputs will place a negligible load on low-impedance outputs and may be paralleled almost without practical limit. The type of mixer illustrated in Fig. 3C is really not feasible except with high-impedance inputs.

It should be clear by this time that the mixer is really capable of much more than mixing. In a well-designed studio, the mixer may be arranged to serve as the coordinating unit, almost as a "switchyard" for signals. It also offers the logical place to locate a monitoring system for originating signals to the studio speaker-amplifiers. A typical arrangement for a small studio is shown in Fig. 4. Note the use of normal connections between the tape recorders and the mixer inputs and outputs. The tape input and output lines go from the machines to and from the mixer through pairs of jacks for each channel of input or output. Insertion of a patchcord plug in either of these jacks breaks the normal connection and establishes an alternate connection through the patchcord which may then be connected to another unit. In one case, one of the jacks establishes an input to the mixer, the other an output from a tape channel; in the other case, one jack establishes an output from the mixer, the other an input to a tape. The monitoring facilities provide for connecting to the input of speaker-amplifiers (high impedance) a signal from

any one of the numbered points (per channel); this may be by means of rotary or push-button switches. A mixing and routing facility based on this arrangement appears in Fig. 5.

The saving in time and the increase in operating efficiency with such a system cannot be overemphasized. Electronic composition is a tedious enough business as it is. With a mixer system arranged to coordinate the tape recorders and provide for monitoring, it becomes possible to carry out many of the routine studio operations such as editing, copying, etc., without any patching at all, even for monitoring. Yet the mixer is always available for the mixing of other

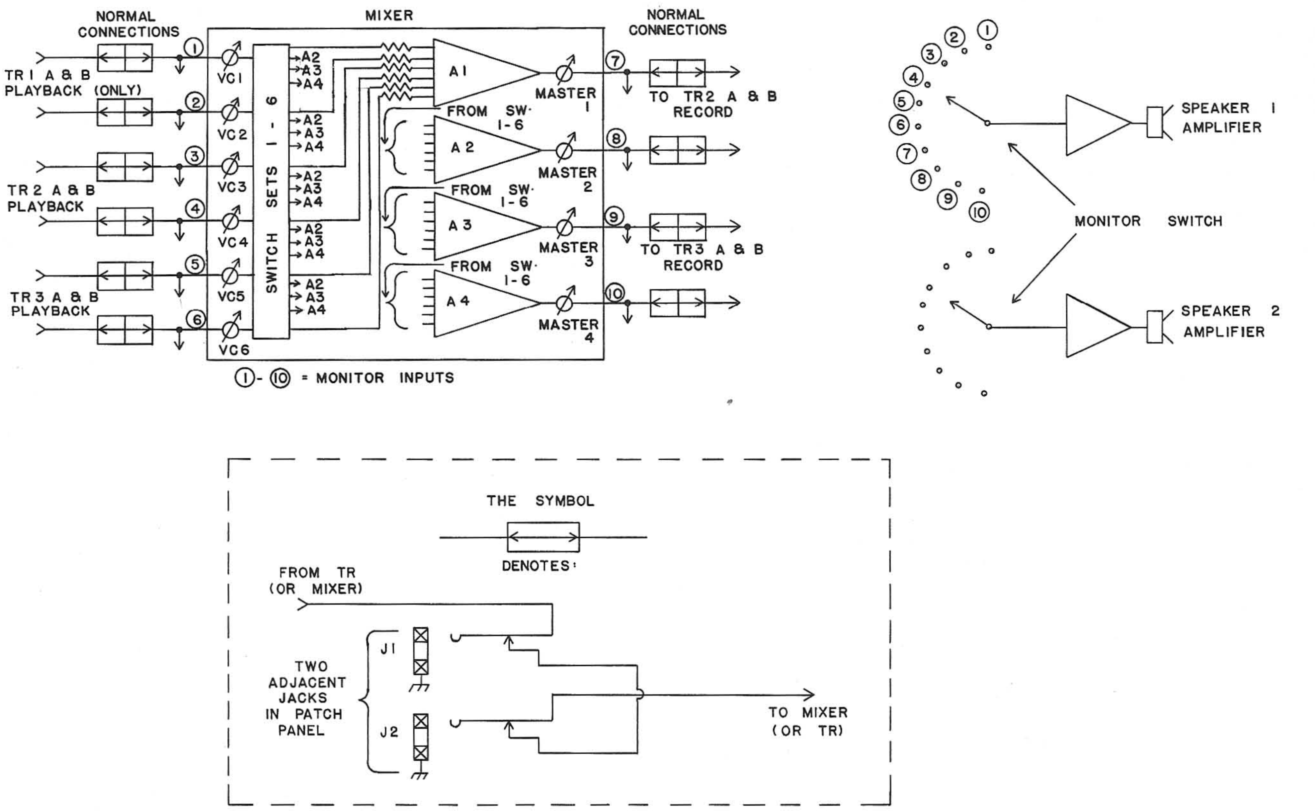


Fig. 4. Mixer configuration appropriate for a small studio.

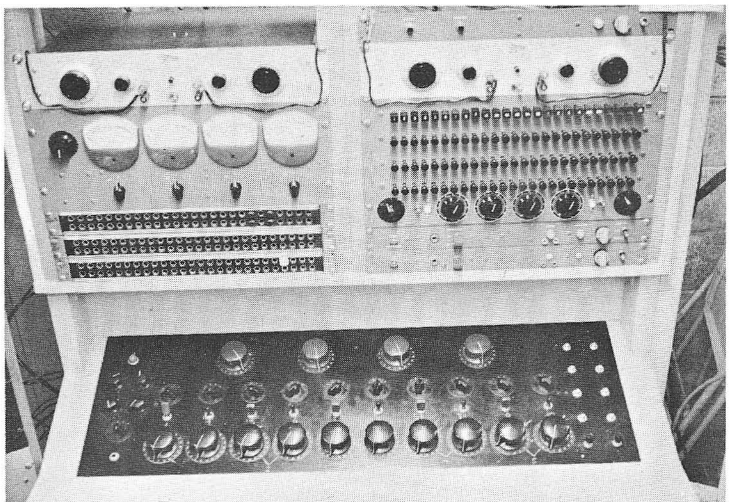


Fig. 5. Mixing and signal routing facility at the Columbia-Princeton Electronic Music Center. On the lower panel, input attenuators appear along the bottom, channel selector switches are in the center, and output attenuators appear along the top. The upper panels provide both patchcord and push-button signal routing.

signals, and the modification of signals between tape output and mixer input can be accomplished by patching other connections in place of the normal connections.

Sometimes it will be found desirable to have other separate mixers available for, say, mixing oscillator outputs without involving the main mixer. These mixers may be relatively simple in design, offering four inputs and one or two outputs. These mixers are sometimes referred to as sub-mixers, and may even be arranged to be a part of a larger general studio mixer, either as separate units on the same panel, or as a part of the large mixer normally, but capable of being isolated by appropriate switches, when necessary.

A great many practical details have been omitted from the above discussion, as the general intent of this article is not to provide actual construction information. The actual requirements for a mixer, especially a general-purpose studio mixer, are so dependent on the other facilities available in any studio that it is best to design each mixer to suit its application, rather than to try to offer more than the most general guidelines in an article such as this. It is, however, realized that this is an impractical attitude to take from the point of view of the independent composer or small studio user. If the user is unable to design and build his own mixer, or to afford the custom building by commercial sources, there is little in the way of modular or kit-form equipment which is really suitable for electronic music use. Thus many small studios are forced to rely upon limited mixing facilities which may, in many cases, be adaptations of commercial hi-fi-grade units, or to take advantage of the recently available mixers which offer signal characteristics of good quality but have a limited adaptability to studio coordination, at least as outlined above.

At present, there seems to be no prospect of automatic or programmed equipment which will supersede the classical techniques in the area of compositional organization. Since the mixer is also most useful in this area, it would seem that any improvement in this fundamental studio unit would be directly reflected in the quality of the music produced.

Functional Design of Electronic Music Mixers

Gerald Shapiro

The purpose of this article is to describe a new method of designing audio mixers for the synthesis and performance of music employing electronic media. The three most important concepts employed in the mixer to be described are: functional design, plug-in modular construction, and the use of passive circuits wherever possible. By using techniques derived from these concepts, it is possible to produce a high quality mixer with substantially greater flexibility than is currently available in commercial mixers.

In general, the current practice is to design an audio mixer as a single, complete instrument. The logic of the mixer is sequential. That is, the individual components of the mixer (inputs, mixing grids, etc.) are designed for their place in a sequence of operations. It is possible, however, to isolate several distinct functions in a complete audio mixer. The three most basic functions are: (1) amplification, (2) mixing-distributing, and (3) amplitude control. A mixer can be designed and built as a number of separate, functional (rather than sequential) plug-in modules, each performing one of the operations listed above. Several different types of modules can be designed for each function, and as many duplications of each as are necessary to form a mixer of the required capacity can be built. These modules can then be patched together in the most convenient sequence for a studio operation or a performance.

Each module should be complete and self-contained, built into a standard-size panel with all the necessary jacks for patching included. The only connections that should be made behind the front panel of any plug-in module are those to the power supply, for active modules such as amplifiers. These connections can be made with a printed circuit edge connector or similar easily removable connector so that the active modules are as easy to install and remove as the passive ones. If a modular studio system such as those of Buchla or Moog is already employed, the power supply, panel sizes, patching jacks, and other connections should be made to conform with this system. The modules can then be plugged into existing consoles, or a new cabinet can be built to house the entire mixer.

For a performance that requires the use of electronic equipment, as many mixing modules and other pieces of equipment as are necessary for the performance can be removed and installed in a separate portable cabinet, and taken to the concert hall. If a piece requires mixing facilities not already available in the system, a new module can be designed and built, and easily installed in the performance console. In this way, each performance is done with the "ultimate performance machine" that we all dream of, and enough equipment is left in the studio so that the usual work there can continue.

In order to reduce the cost of this type of mixer, as many of the modules as possible should be passive. That is, they should be able to carry audio signals in either direction. Obviously, a mixing network, if carefully designed for the purpose, can become a distributing network. In the same way, a panning potentiometer which divides a signal between two channels can be used as a pan-segue control to mix portions of two signals into one channel.

The relatively low cost of high gain integrated circuit operational amplifiers makes it possible to design a single amplifier module for the mixer. Many duplications of this module will be needed. A rotary switch can be supplied to give various amounts of gain and perhaps equalized settings for tape head and magnetic cartridge. Balanced and unbalanced inputs and outputs can be included as required. A circuit might also be added for treble and bass equalization. Fig. 1 is a block diagram of a possible amplifier module.

The basic mixing-distributing units (Figs. 2, 3) are simple Y connections with the addition of switches and attenuators. Two types of basic units have been devised so far. Both are passive circuits as described above.

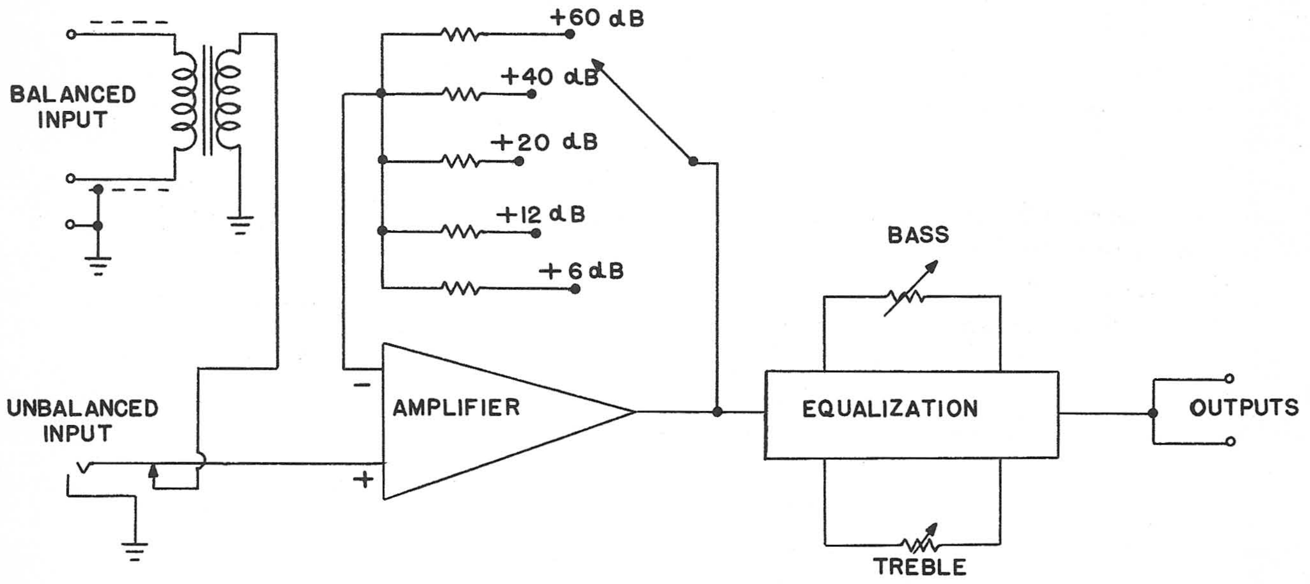


Fig. 1. Circuit of amplifier module.

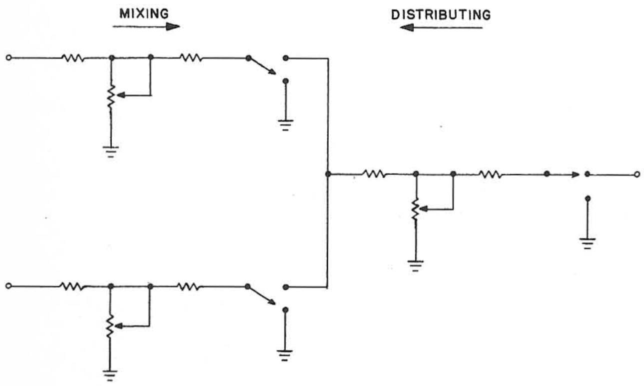


Fig. 2. Circuit of on/off mixing-distributing unit.

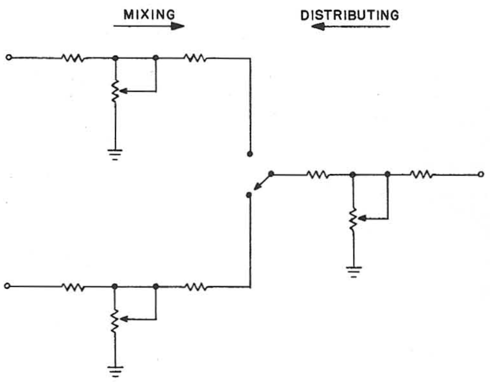


Fig. 3. Circuit of either/or mixing-distributing unit.

The addition of a voltage-controlled switch (Fig. 4) with a panel jack to accept a control voltage from an external source such as a sequencer allows the mixer to be programmed for a series of operations. If a voltage-controlled switch is included, it can be designed into an "or gate" type circuit to allow for manual control as well.

The most important part of the mixing-distributing unit is the switch, or combination of switches. The attenuators are included to pre-set levels. Although they might be employed during a performance or studio operation to manually vary the amplitude of a signal, this is not their primary function. The attenuation circuit (Fig. 5) employed in these units has been especially designed to work equally well in either direction.

R_1 and R_3 are fixed resistors of equal value. R_2 is an audio taper potentiometer of equal or smaller value. In determining the values of the three resistors, a compromise must be decided upon between preserving the taper of the potentiometer and the amount of insertion loss in the circuit. The determining factor is the ratio of either of the fixed resistors to the potentiometer. A large ratio preserves the integrity of the potentiometer but increases the insertion loss. A ratio of 1 : 1 yields an insertion loss of 6 dB and a taper that is 5.5 dB higher than the ideal at 50% rotation of the potentiometer. Lower ratios seem to be too destructive of the logarithmic taper of the potentiometer for adequate control. Fig. 6 illustrates the relationship of insertion loss and taper for an attenuating circuit between a low impedance output and a high impedance input.

Experience has shown that it is generally advisable to build mixing modules that are made up of clusters of basic units. Panels with small clusters or only basic units are more flexible but very many are needed and higher amplification is necessary to overcome the multiple insertion loss. Panels with larger clusters of the basic units are less flexible but easier to use and require less amplification. Circuits of two such arrangements are shown in Figs. 7 and 8. Several different types of clusters can be designed and included in the mixer for greater flexibility. Of course, new ones can always be added cheaply as required.

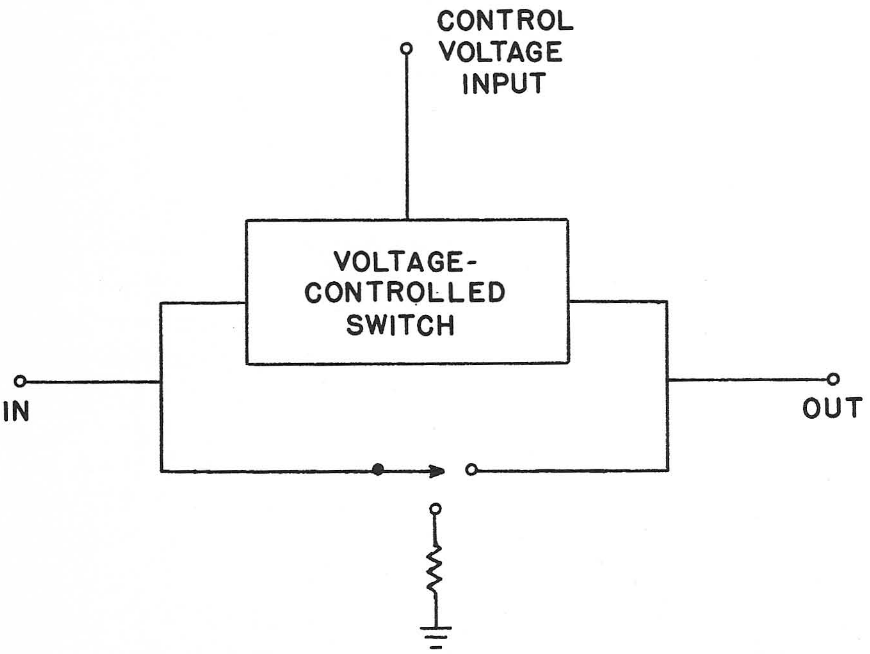


Fig. 4. Block diagram of mixer with voltage-controlled switch.

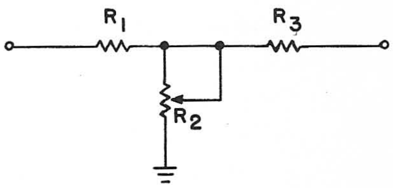


Fig. 5. Attenuation circuit of mixing-distributing unit.

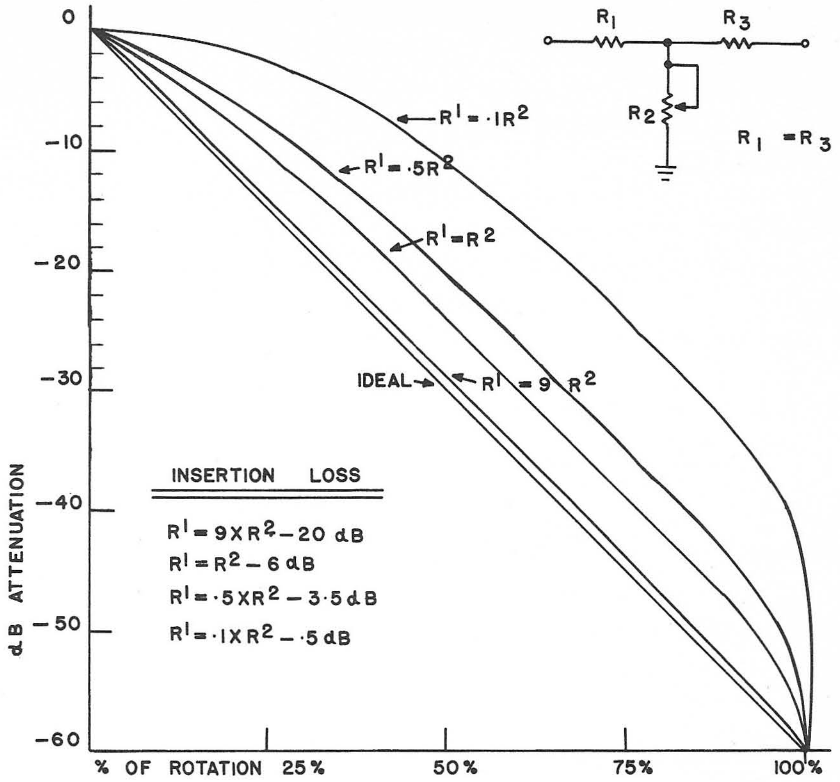


Fig. 6. Insertion loss versus rotation of the attenuation circuit, when inserted between a low impedance output and a high impedance input.

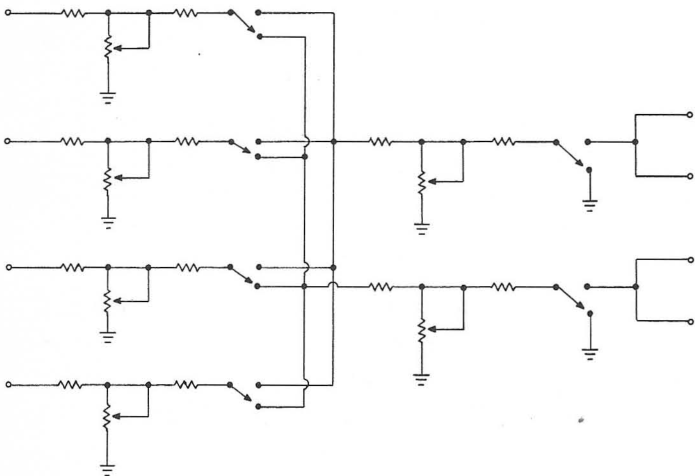


Fig. 7. Circuit of either/or larger mixing module.

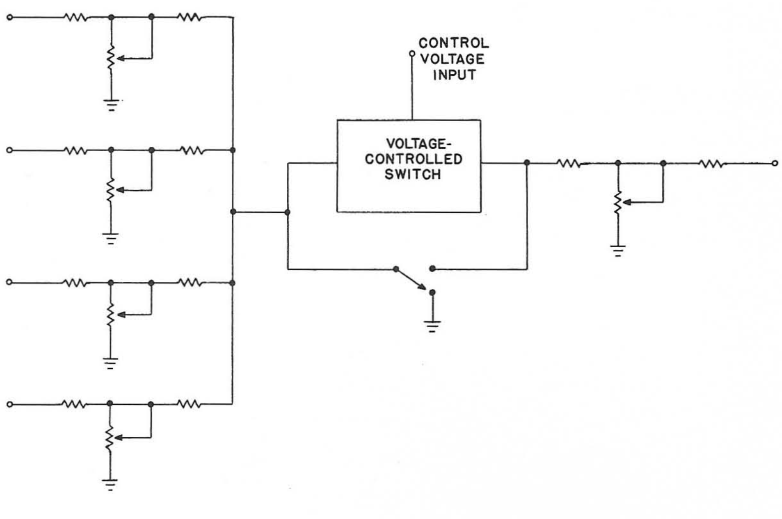


Fig. 8. Circuit of on/off larger mixing module with voltage-controlled switch.

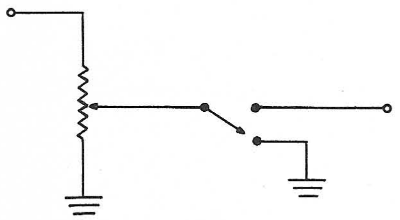


Fig. 9. Circuit of attenuator amplitude control module.

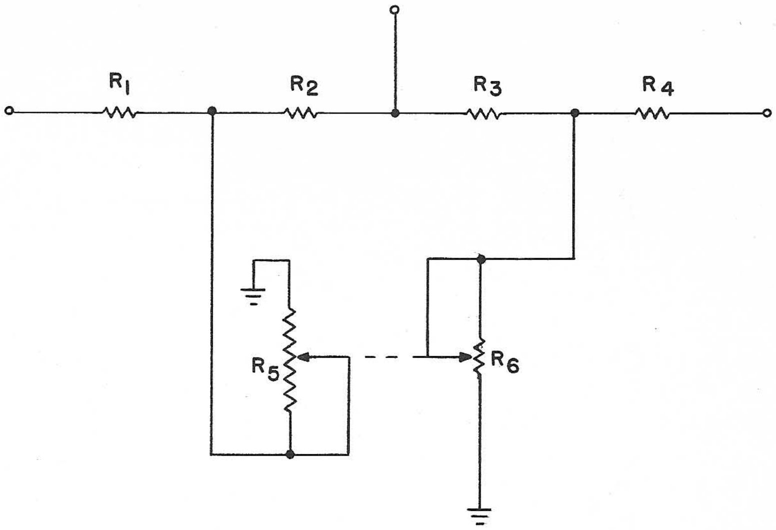


Fig. 10. Circuit of pan-segue amplitude control module.

Two types of amplitude control modules have been designed. The most basic one (Fig. 9) is a simple attenuator with an on/off switch added for convenience. The second type (Fig. 10) is the pan-segue control mentioned earlier in the article. It is made up of two identical passive attenuators. The two potentiometers have opposite tapers and are ganged.

Other types of modules can be designed and included in a mixer of this type. V.U. or dB meters are helpful if not essential in any mixer. One or several cross bar switches would help relieve the congestion of a complicated patching setup. Finally, monitor amplifiers can be included if necessary.

Many of the proposals in this article are tentative insofar as they represent thinking that has not yet been applied to the production of an existing piece of equipment. This is only a first step in the exploration of the possibilities of functional mixers. These possibilities are only limited by the ingenuity of the composers and engineers working in the field of electronic music. One important area not covered in this article is the design of the panels for the plug-in modules. Obviously, a great deal of attention must be given to the problem in order to achieve the greatest possible convenience of operation.

Acknowledgements:

Considerable technical assistance in the preparation of this article was provided by Karl Amatneek. Many of the original concepts utilized here were first worked out in collaboration with Bill Maginnis, the technician at the Tape Music Center at Mills College.

Some Applications of Electrical Level Controls

Hugh Le Caine

Electrical level controls have many uses in the electronic music studio. Some of those considered at one time or another by the writer will be described briefly. Most of the devices described were developed under the direction of Gustav Ciamaga and form part of the equipment of the University of Toronto Electronic Music Studio.

The most familiar electrical level control circuit (Fig. 1), which goes back to the early days of vacuum tubes, uses two amplifiers in push-pull. When transistors are used, the relation between gain and bias is closer to an exponential one than it is when vacuum tubes are used, and a better match is obtained between the two transistors.

In setting up the circuit of Fig. 1 the output balance is adjusted to minimize the change in DC level at the output, with control voltage. The gain balance is then set to minimize the second harmonic of the signal appearing at the output. If an instrument for measuring the second harmonic is not available, the gain balance may be set to give the most symmetrical waveform at the output using a sine wave input about three times the normal maximum value.

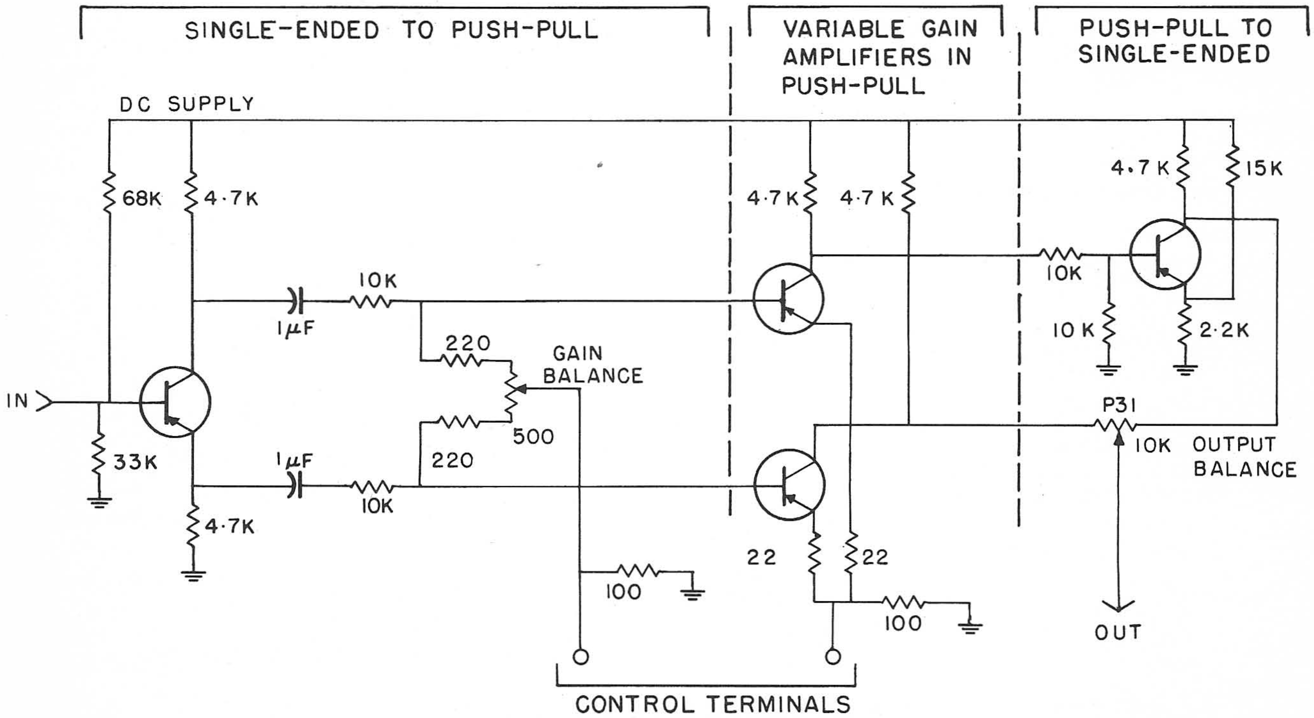


Fig. 1. Basic electrical level control.

The attenuator shown in Fig. 2 is simply made with printed circuit techniques to give a long-wearing control without contacts. The variable capacitors that control the voltage applied to the electrical level control consist of fixed plates arranged in pairs on printed circuit board and one movable plate for each pair. The construction can be seen in the photograph. The assembled attenuators are seen on the left. The cover plate has been moved over to show the pairs of fixed plates (center) and the movable plates (right).

The hand capacitance control for four speakers shown in Fig. 3 was used in the University of Toronto Electronic Music Concert of 1963. The grids are made by printed circuit techniques and covered with a layer of plastic. Soft tones are elicited by lightly touching the board, loud tones by a more extensive contact.

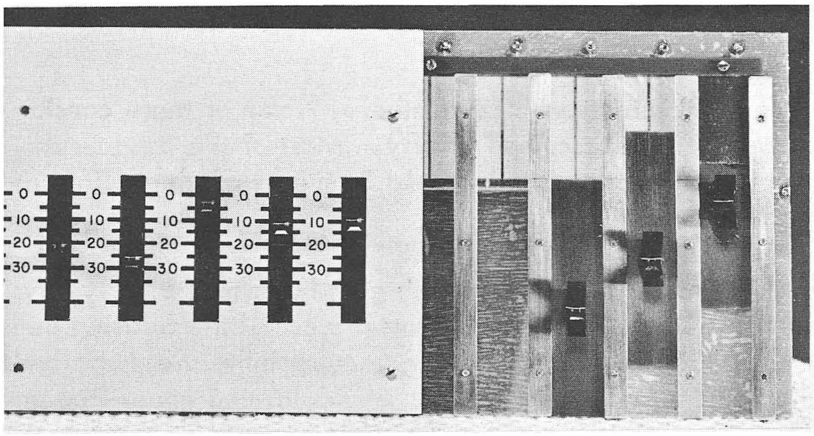


Fig. 2. Attenuator without contacts using the level control of Fig. 1.

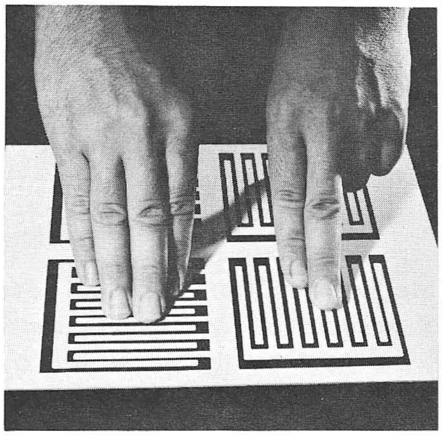


Fig. 3. Hand capacitance control.

The circuit of the hand capacitance control is shown in Fig. 4. Capacitor C is initially adjusted with hands off the board so that there is no AC voltage across the diode and no DC output. When the hands are placed on the board as previously described, a DC output is obtained that is used to operate four electrical level controls, one for each section.

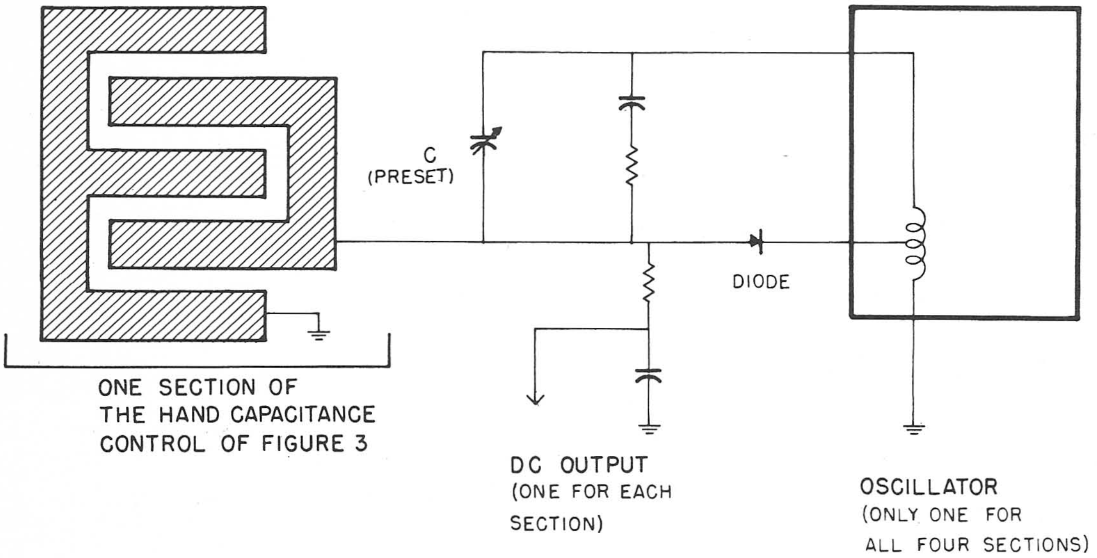


Fig. 4. Circuit of the hand capacitance control of Fig. 3.

The touch-sensitive key[1] is an indispensable envelope shaper for the electronic music studio. Electrical acceleration and electrical sustain[2] extend the usefulness of the simple touch-sensitive key (see Fig. 5). When the key is struck a sharp blow, electrical acceleration counteracts the inertia of the key by increasing the rate at which the level rises. When the key is operated more slowly, normal touch-sensitive control is obtained. The electrical sustain facilitates the production of uniform slow decays and is useful when it is desired to sustain the sound after the finger has left the key. The sustain may be put into action by operating a stop tablet, a sustaining pedal, or a wrist bar.

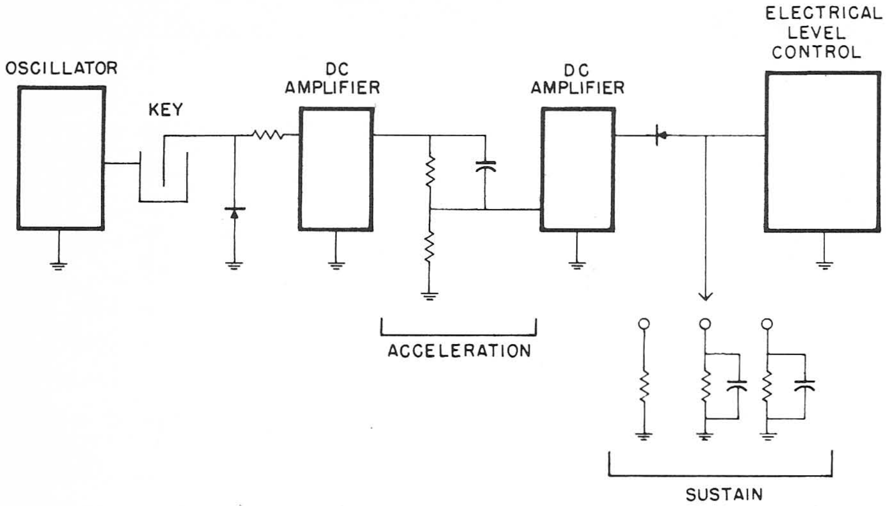


Fig. 5. Electrical acceleration and sustain on a touch-sensitive key.

The general purpose expander-compressor shown in Fig. 6 has a limited amount of usefulness in modifying the envelopes of existing tones, in changing the characteristics of a reverberation device, as an "amplitude filter"[3], and in noise suppression. The circuit shown in Fig. 6 is arranged to control the important variables in a convenient way. The control zone covers a constant range in terms of decibels at the input but may be moved closer to the maximum level (0 dB) or farther from it. The amount of expansion or compression may be usefully varied from none to a maximum of 30 dB. In an amplitude filter a greater range is required.

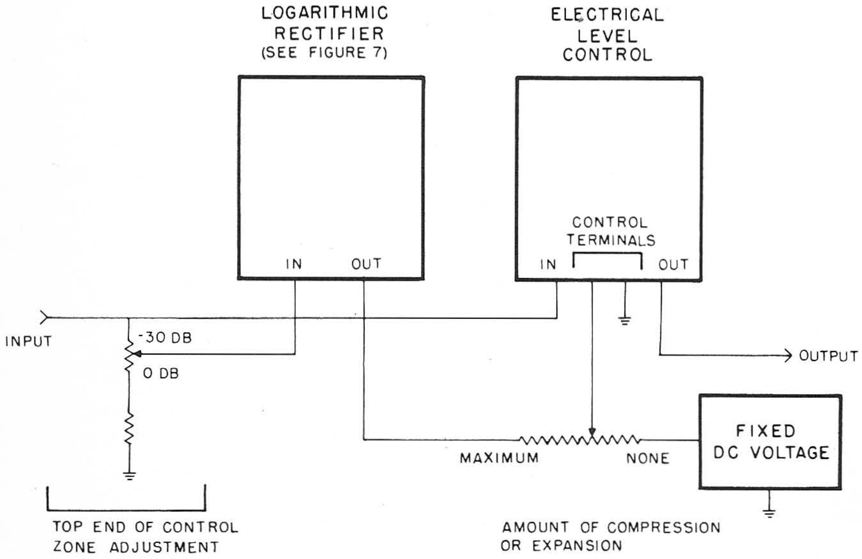


Fig. 6. General purpose expander-compressor.

In the block diagram of Fig. 6, the logarithmic rectifier of Fig. 7 is used to obtain uniform compression or expansion as shown in Fig. 8. The top end of the control zone is moved from 0 dB to -30 dB input level by means of a variable attenuator on the input to the logarithmic rectifier. The amount of compression or expansion introduced is varied without changing the output level for 0 dB input level by means of a potentiometer, one end of which is connected to a constant DC voltage equal to the maximum output of the rectifier. Compression or expansion is obtained by applying the voltage from the potentiometer to one or the other of the control terminals of the basic level control.

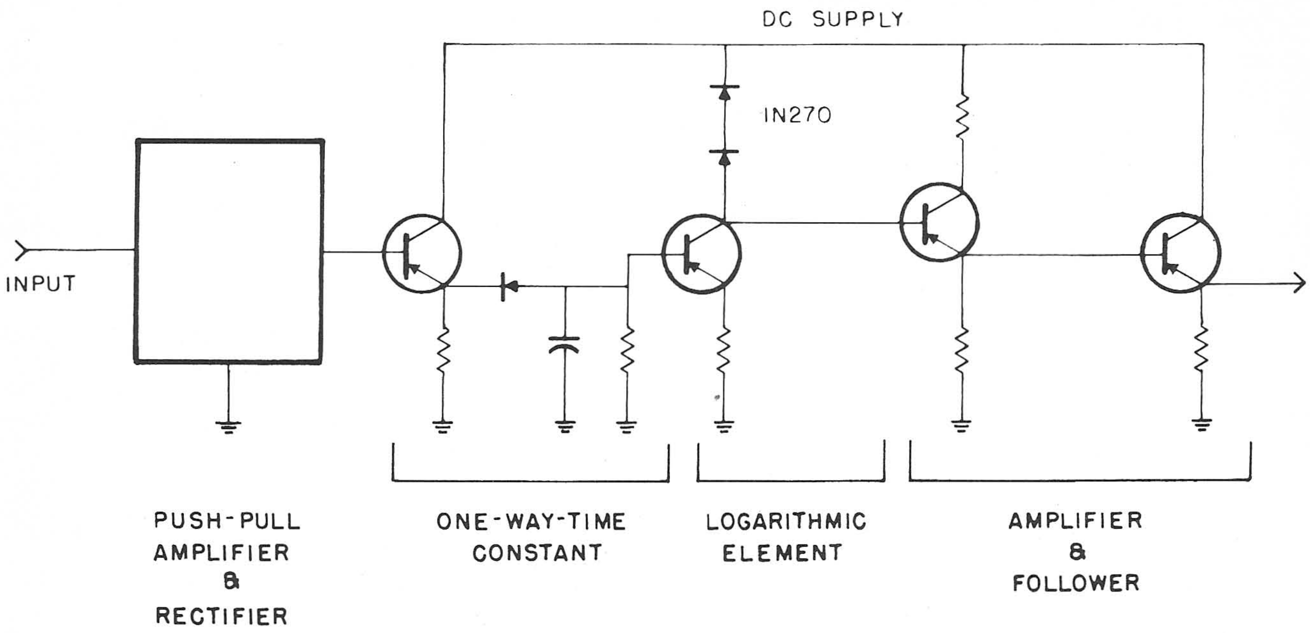


Fig. 7. Logarithmic rectifier suitable for use in the expanders and compressors shown here.

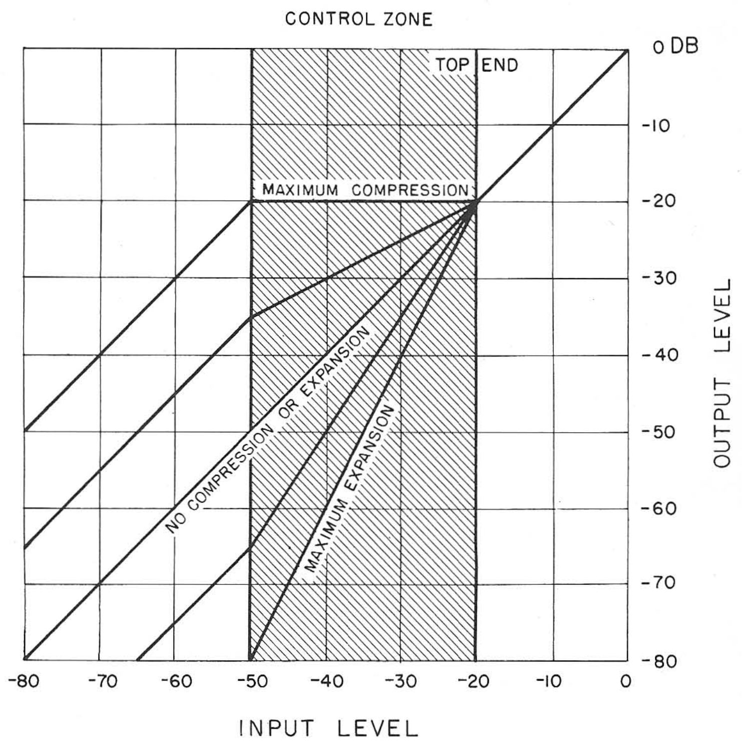


Fig. 8. Output level versus input level in the general purpose expander-compressor.

For systematic noise reduction where compression in recording and a complementary expansion on playback is used, location of the top of the control zone 20 dB or so below the maximum level is recommended since any misfit between compression and expansion is less noticeable at lower levels. A compression of 20 dB over the next 30 dB may be used with a corresponding expansion in playback. Though the improvement in noise level is dramatic, there is bound to be a certain amount of envelope modification.

Simultaneous compression before and expansion after a noisy device is helpful in suppressing the noise while leaving the envelope unchanged. The signal which is applied to the input of the device is compressed by a control voltage derived from the signal. The same voltage operates an expander connected to the output of the device which restores the original envelope (see Fig. 9).

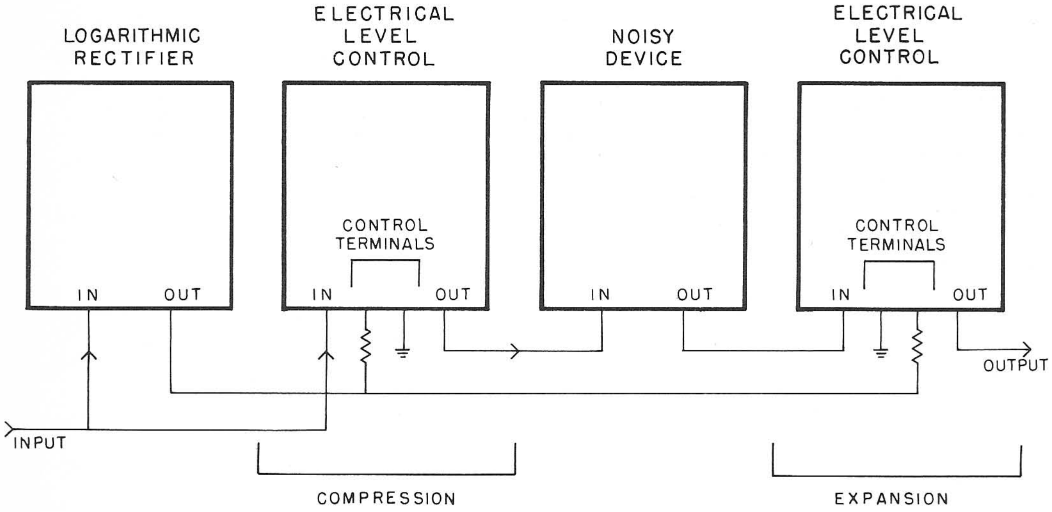


Fig. 9. Simultaneous compression and expansion before and after a noisy device.

The "Hamograph" of Myron Schaeffer is a device in which a hand-drawn envelope is imposed upon sound material[4]. Schaeffer used loops of clear 35 mm. motion-picture film upon which masks cut from black "Mystik" tape were stuck. 16 mm. single perforation film may be used; it gives half an inch of useable width. Paper charts may be marked with india ink or pencils. A photoresistor reader suitable for use in any of these arrangements is shown in Fig. 10. Half the case has been removed to show the six photovoltaic cells used with an inch-long aperture (for 35 mm. film). The resistors arranged as shown in Fig. 10 give a nearly linear relation between the length of slot exposed and the voltage developed. This leads to a nearly linear relation between the level in dB and the length of slot exposed.

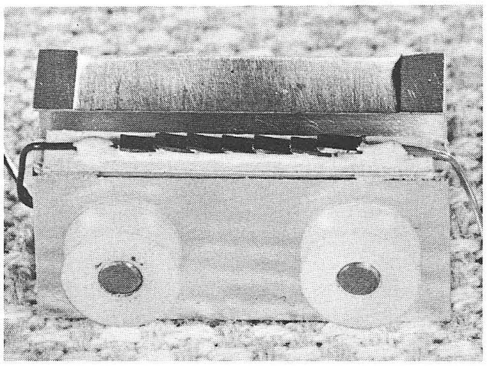


Fig. 10. Optical envelope shaper for 35 mm. motion picture film used in Myron Schaeffer's Hamograph.

In a modified form of the Hamograph developed by Ciamaga, magnetic recording on control tapes is used instead of photoelectric reading from motion picture film. This system has the disadvantage that the envelope cannot be identified by eye. However, envelopes are easily duplicated on an ordinary tape recorder and the apparatus is much easier to set up, most of the necessary components being standard tape equipment. Electrical level controls in this device are operated by the output from the control tapes.

The two channel alternator (Fig. 11) is a device suggested by Ciamaga which differs from the well-known electronic switch in that the gain in one channel is <u>gradually</u> decreased while the gain in a second channel is gradually increased at the same rate. This device is most interesting as a producer of small modifications of tonal material. It can produce a "complementary tremolo" consisting of a periodic gradual increase and decrease of level in one musical part accompanied by a decrease and increase of level in another part. A range of "alternation tremolos" where the tonal material is gradually replaced by an altered version at normal tremolo rates can be produced.

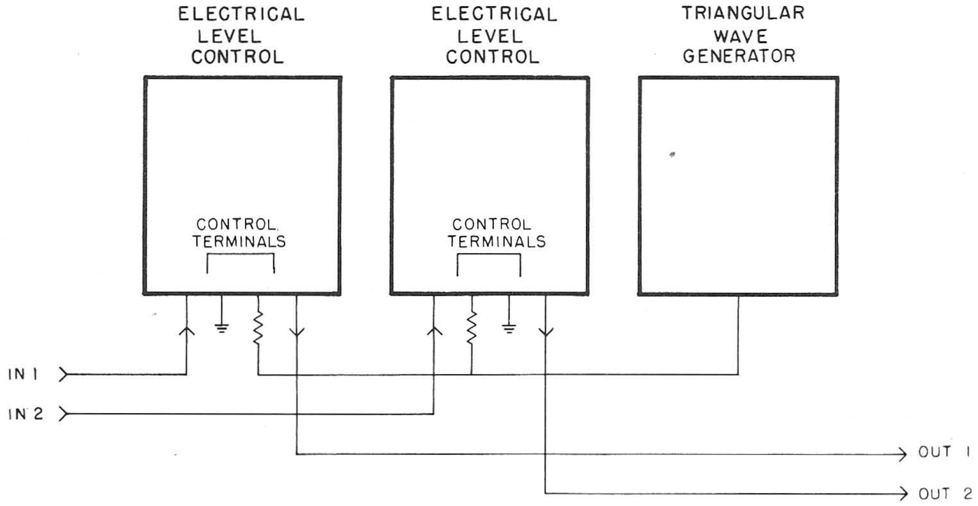


Fig. 11. Two channel alternator.

The envelope shaper or "gate" (Fig. 12) converts continuous sound material into a "note"; that is, it determines the rise and fall of intensity from the beginning to the end. The design objectives were:

(1) The attack and decay rates should be independently adjustable over a wide range.
(2) Peak intensity should be independent of the setting of the attack and decay rates.
(3) Tones consisting predominantly of "steady state" sections and those consisting of an attack period followed immediately by a decay period should be producible.
(4) Action should be initiated either by a key or in response to timing signals which can be recorded on tape or obtained from rhythm instruments such as the Rhythmicon[5].

To satisfy the first two requirements the circuit has been designed to terminate attack and decay periods with reference to a fixed voltage[6]. To satisfy the third requirement, two divisions have been incorporated, each having its own level control amplifier. The "sustained" division

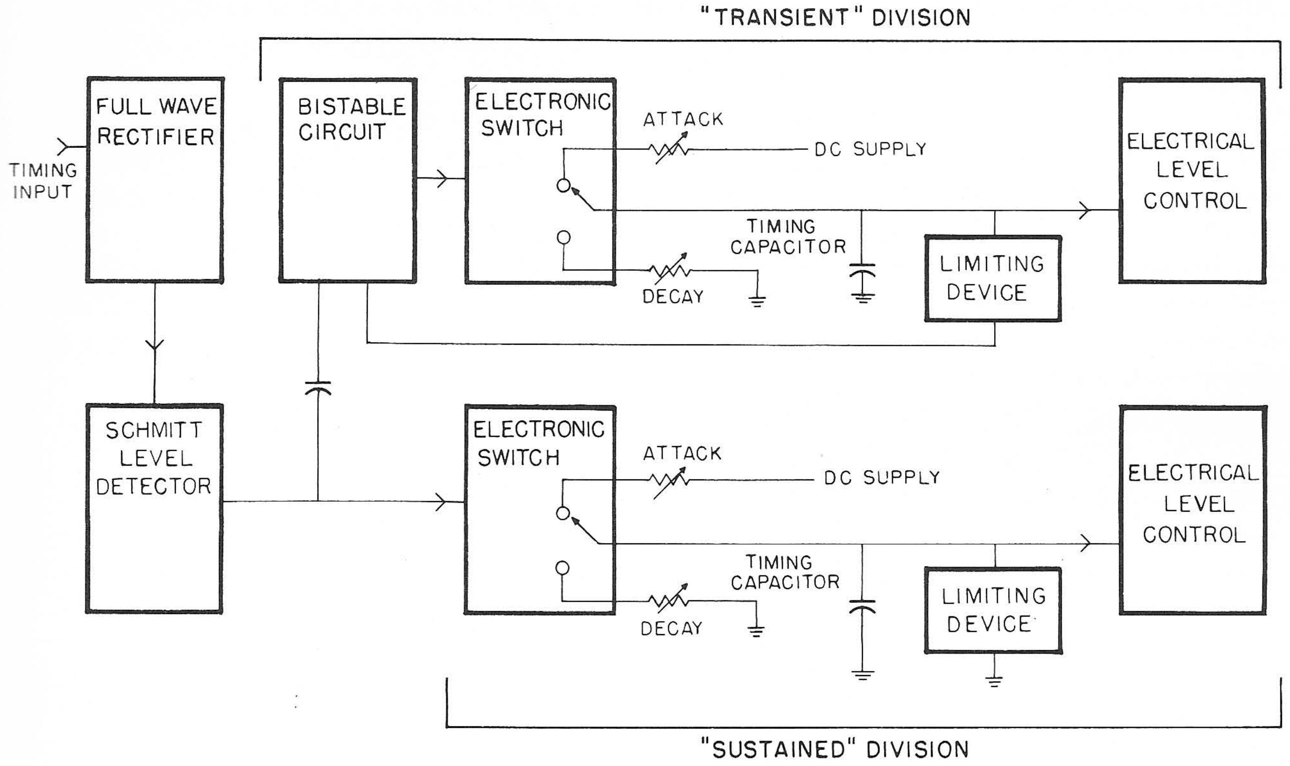


Fig. 12. Envelope shaper.

begins the attack when the timing signal reaches the threshold level. When the standard voltage has been reached, the level remains at the maximum value as long as the timing signal continues. When the timing signal stops, the decay period begins and continues until the sound is inaudible. In the "transient" division the attack period begins when the timing signal reaches the threshold level. When the standard voltage has been reached, however, the decay period begins whether the timing signal is still present or not.

The DC output of the Schmitt level detector in Fig. 12 is used to operate the "sustained" division. When the timing signal starts, the electronic switch connects the "sustained" timing capacitor to the DC supply through the adjustable "attack" resistor. When the voltage has reached the predetermined value, a limiting device prevents any further rise. At this point the output level has reached a standard maximum value. When the timing signal stops the electronic switch connects the timing capacitor to the decay resistor and the decay period begins.

When the timing signal starts, a pulse from the Schmitt level detector moves the bistable circuit into the "attack" state. In this state a separate electronic switch in the "transient" division connects a timing capacitor to the DC supply through an adjustable "attack" resistor. When the voltage has reached the predetermined value, a limiting device moves the bistable circuit into the "decay" state and the decay period begins. Action in the "transient" division thus depends upon the start of the timing signal and not upon its duration.

The sound from both divisions can be mixed in any proportion to give a variety of envelopes. The two divisions have separate inputs and outputs as well as mixed inputs and outputs so that

filters, ring modulators, and other signal modifying devices can be used to produce related but different tonal material for the transient and sustained divisions. Several transient divisions can be used together, each carrying different material.

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2. Used on the "Electronic Sackbut"; see: H. Le Caine, "Electronic Music", Proceedings of the Institute of Radio Engineers, XLIV, 4, April 1956, 457.
3. Alfredo Lietti, "Soppressore di disturbi a selezione d'ampiezza". Elettronica, 5, Sept.-Oct. 1955, 1.
4. Myron S. Schaeffer, "The Hamograph", Institute of Radio Engineers Transactions on Audio, AU-10, 2, Jan.-Feb. 1962, 22.
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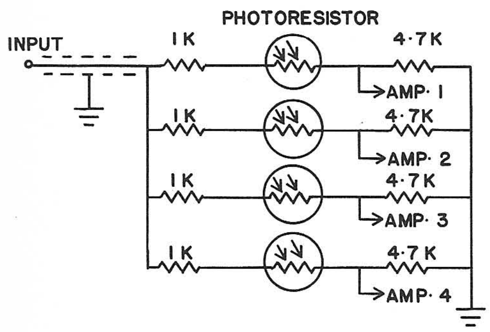
6. Auxiliary apparatus with Canadian patent 587,741.

A Photoresistor Mixer for Live Performance

Frederic Rzewski

The photoresistor mixer is an extremely simple device that I designed and constructed for use in a specific composition of mine, Impersonation. In this composition the effect of movement of sound in space is called for; furthermore, each channel (there are eight) is required to "move" independently of the others, so that one channel may move clockwise around the room, another counter-clockwise, while a third darts rapidly from point to point, a fourth meanders in a random fashion, and so forth. The solution that I found to this problem is, I am sure, by no means the best one, but it satisfied the internal musical requirements that I had set up, and remained within the external limitations imposed upon me by lack of funds, technical experience, and assistance.

Each input signal is split into four parts, the level of each part controlled by illuminating the photoresistor with a penlight (see Fig. 1). The four outputs go to four amplifiers and loudspeakers. The loudspeakers may be set up in the four corners of the room. The photoresistors I used (Philips B8-73105) are very inexpensive. For each channel the four photoresistors are mounted in a circle, 4 cm. in diameter, on a bakelite panel (see Fig. 2).



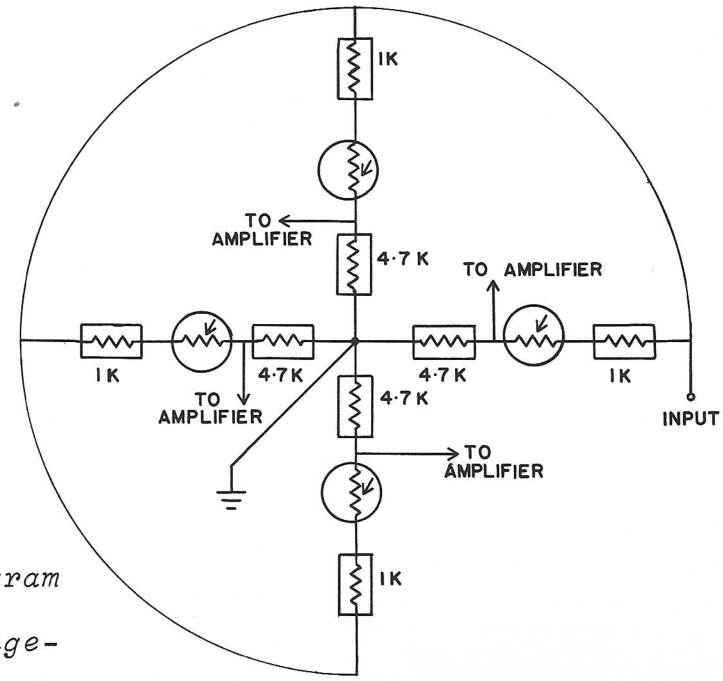


Fig. 1 (above). Schematic diagram of one channel.
Fig. 2 (right). Physical arrangement of one channel.

I mounted four such circular groups on one large panel to form a four-channel mixer. Over each group I mounted a cylinder, 5 cm. long and 5 cm. in diameter, painted black (for this I used tin cans). Inside each cylinder, resting firmly over the photoresistors, I placed a disc of dark transparent plexiglass, with a rubber sleeve fitted around it, to protect the resistors from stray light. (Later I thought of using two discs of polarized glass, one of which could be rotated, so as to find the proper degree of transparency for any ambient light situation.)

Finally I mounted the panel inside a metal box, 20 cm. x 20 cm. x 10 cm., with four circular holes cut in the top that fit snugly over the tin cans. On the sides of the box are jacks for the four inputs and sixteen outputs. The shields and grounds are all connected together and soldered to the box at one point. The sixteen outputs are divided into four groups of four, labeled "north", "south", "east", and "west". The photoresistors on the "north" pole of each circle go to the "north" group, those on the "south" pole to the "south" group, and so on. The position of the photoresistor in the circle is therefore equivalent to that of the corresponding loudspeaker in the room; ideally, for each channel, the panel presents a two-dimensional abstraction of that space, with the audience in the middle. By moving the penlight horizontally and vertically over each cylinder, variations in luminance are created that should correspond to like variations in the intensity of the sound from each loudspeaker. Two people can control the levels of four channels, each player with a penlight in each hand. Or one person, using one or more light sources, can control all four channels.

Although a mixer of this sort presents obvious disadvantages from the studio viewpoint (inaccuracy, difficulty of controlling the distribution of light, etc.), it acquires a certain interest if one considers it as a performing instrument. For example, by exploiting the inherent characteristics (and limitations) of light-sensitive resistors, certain unusual effects are obtainable. By sudden illumination of the photoresistors (with a stroboscope, say), very sharp, explosive attacks can be obtained (which are, of course, noise-free); and, by altering the values of the resistors, the characteristic "tail" which follows can be lengthened or shortened. In general, very radical variations in level can be created that are awkward, at best, and often accompanied by noise when attempted with conventional potentiometers. When mixing four different tapes, for example, extremely rapid passages from one channel to another can be effected by moving the light manually across the various cylinders. Why sixteen outputs instead of four? In situations where more than four loudspeakers are used, the channels can be combined to move in different ways.

This mixer was very inexpensive and easy to build. Technically, it is unsophisticated, but musically it is effective: for it satisfies my first precept, which is to make music with whatever means I have at my disposal.

The Sound Level Photoprogrammer

Fernando von Reichenbach

This experimental device substantially improves stereophonic sound reproduction in an auditorium. Its main features are a very wide dynamic range and displacement of the sound configuration around the audience with no restrictions on the number of channels or speakers used. The device is automatically triggered by signals recorded on the same magnetic tape that reproduces the sound.

Music, voices, or sound effects commonly used in theaters require high and low volumes alternately. The adjustment of volume controls for the loud passages causes tape hiss to become audible when sound level decreases. Different ways to remedy the problem have been suggested. Use of tapes of the low-noise type and recording on a wider track are the obvious solutions, although the latter sometimes involves sacrificing the number of tracks to be recorded. Use of volume compressor and expander units, variations of bandwidth according to the volume reproduced, or improvement of special recorder equalizations are more costly approaches.

Another solution is to record the whole program at almost the same level and, on reproducing, to fade down to the proper levels the passages which so require. But a very skilled operator is needed for this, and it becomes impossible when the volumes of many speakers must be controlled in varying proportions for the achievement of changing sound patterns. By means of the sound level photoprogrammer, this complicated manipulation becomes programmed for automatic operation.

Six speakers are located around the auditorium. Each has its own power amplifier and the volume of each amplifier is controlled by means of two photoresistors per amplifier which connect to both outputs of a two-channel tape recorder. Flashlight bulbs with lenses illuminate the photoresistors through a transparent film, twelve inches wide, on which the program is prepared with segments of plastic tape. The different degrees of opacity accorded to the film determine the exact amplitude supplied to the speaker. A transport mechanism (synchronous motor with magnetic clutch) moves the film at approximately six mm. per second. Longitudinal tracks on the transparent film correspond to six different speakers: each track controls the sound level from either one or both of the recorder channels. Another lateral track takes care of synchronization and is controlled by its own photoresistor. The apparatus may be seen in Figs. 1 and 2.

Speech, music, or sound effects are recorded on two tracks of the magnetic tape. On a third track, a pilot tone is recorded which triggers each change of volume. The tone closes the relay, which sets the film in motion. As the film advances, a transparent section passes over the photoresistors; each new section of film contains the information for the volumes of the speakers for a corresponding section of tape. This motion continues until a black bar on the synchronization track appears and stops the film. Upon receiving another signal, the film advances again. The length of time during which the film advances is determined by the distance between black bars; the signal indicates only at what moment movement must begin. The volume of sound is-

suing from each speaker depends on the quantity of light going through the film over the programming unit. For black tape, the attenuation is better than 70 dB. Tapes of various colors in single and double layers allow adjustment of the volume in each passage. The sound is recorded near 0 dB. The reproduction levels are so adjusted that tolerable maximum volume coincides with direct passage of light.

This inexpensive gadget, built in April 1966 for a theatrical performance, still operates well. In some respects it can be compared to the experiences of Pierre Henry with coils. Once the initial temptation to use the photoprogrammer merely for the sake of effect is overcome, a new field of experience in sound perception will remain open to art production.

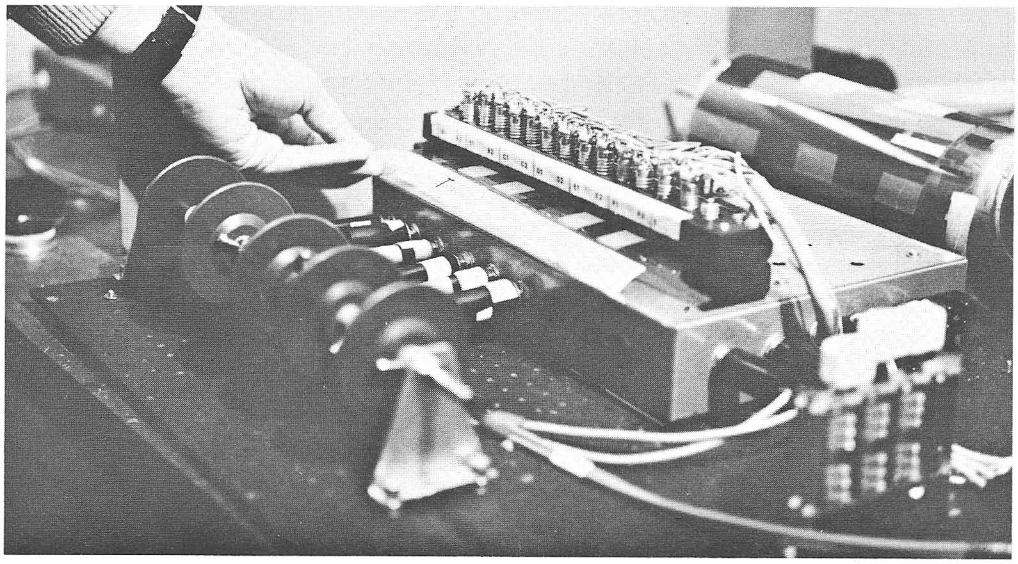


Fig. 1. A static stereophonic sound configuration is inserted on transparent film. It is equivalent to the volume adjustment by 12 independent potentiometers.

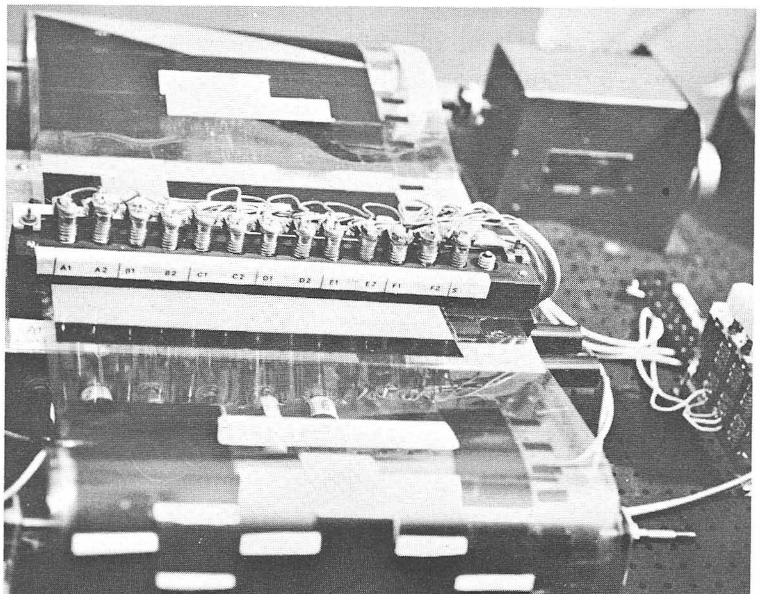


Fig. 2. A monaural program is fed to the photoprogrammer. Diagonal bar on the film (top of picture) allows sound to circle the room.

Symposium: Mixers and Level Controls

Construction of a Simple Mixer

Robert A. Moog

This mixer fulfills the basic requirements of a small studio or a modest performance setup. There are two identical channels; each has one microphone input, four line inputs, a master gain control, and a VU meter. A segue pot mixes the two outputs in any proportion. The entire mixer is powered by two 9-volt batteries.

Two variations of a simple operational amplifier circuit perform the amplifying and mixing functions. Operational amplifier A1 is connected to provide high input impedance for the microphone transformer (see Fig. 1). The gain of the microphone preamplifier portion is determined by the switched resistor network between the output and the inverting (-) input of A1. Operational amplifier A2 is connected as an analog adder to perform the mixing function. The gain of the adder portion, like the gain of the microphone preamplifier, is determined by the particular resistor that is switched in between the output and the inverting input of A2.

The complete schematic diagram of one channel is shown in Fig. 2. The microphone impedance switch selects the tap on the input transformer for the appropriate input impedance or, in "HI" position, bypasses the input transformer. The operational amplifier circuit of the mic preamp consists of a balanced input stage Q1 and Q2, and second stage Q3. This particular configuration gives high open loop gain without coupling capacitors, resulting in excellent low frequency response and overload recovery. The gain is roughly equal to the ratio between the feedback resistor switched by SW 2 and R5. The operational amplifier circuit of the adder is identical in configuration to that of the mic preamp, except for the inclusion of one transistor (Q6) to provide greater current gain. The adder gain is equal to the ratio between the feedback resistor switched by SW3, and any of the input resistors R18-R22. The VU meter is driven by emitter follower Q8 so that the non-linear impedance of the meter does not introduce distortion at the output.

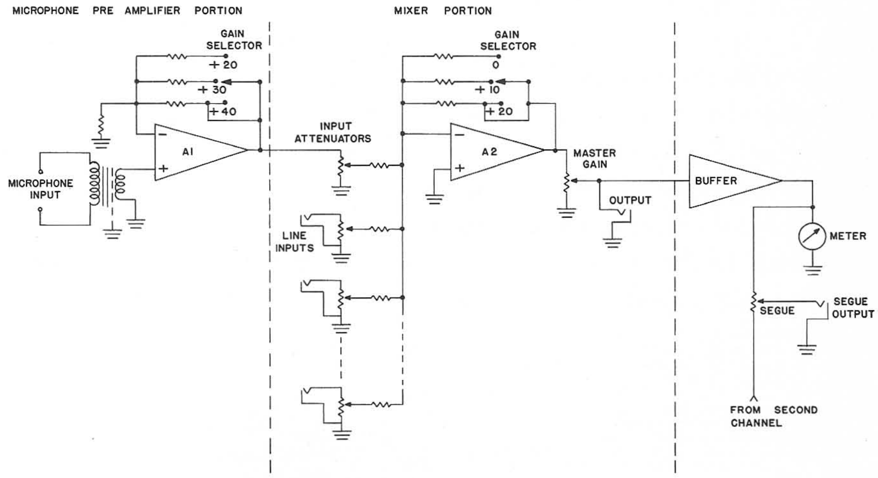


Fig. 1. Simplified diagram of one mixer channel.

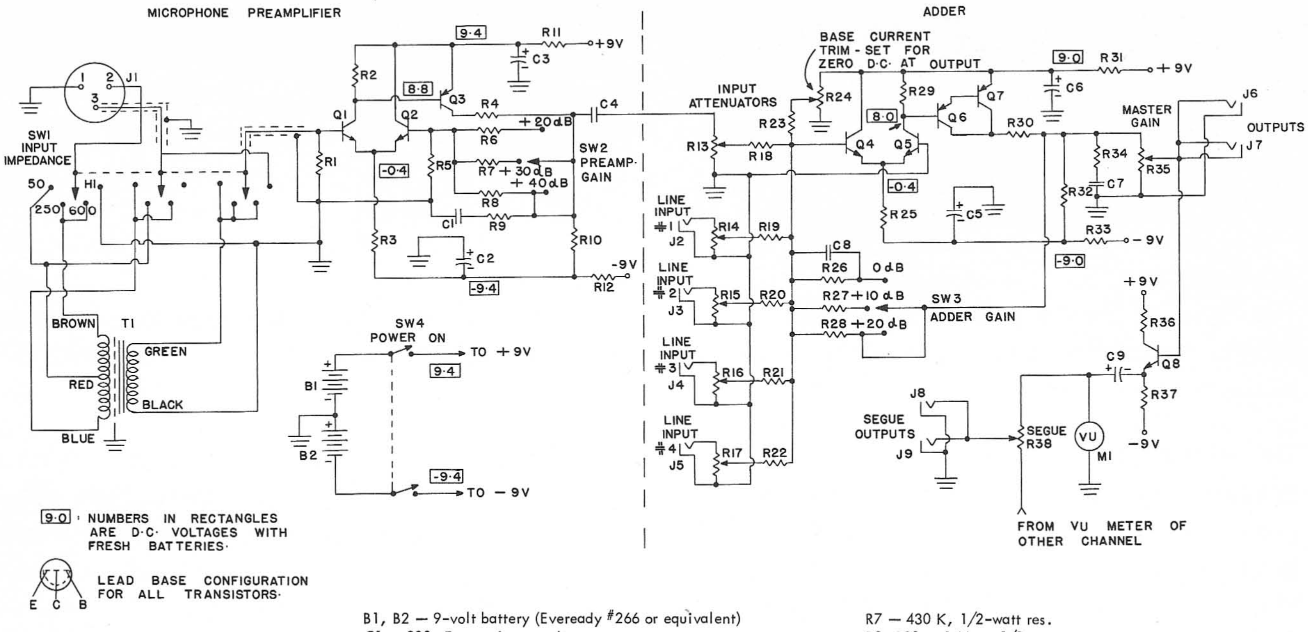


Fig. 2. Schematic diagram and parts list for one channel. Except for the batteries, power switch, segue pot, and segue output jacks, the entire circuit is repeated for the other channel.

B1, B2 — 9-volt battery (Eveready #266 or equivalent)
C1 — 220 pF ceramic capacitor
C2, C3, C5, C6 — 80 mF, 16-volt elec. capacitor
C4 — 0.2 mF, 100-volt mylar capacitor
C7 — .0033 mF ceramic capacitor
C8 — 4.7 pF ceramic capacitor
C9 — 1 mF, 16-volt elec. capacitor
J1 — Three-pin female audio connector (Cannon XLR 3-31 or equiv.)
J2-J9 — Two-conductor phone jack
M1 — Small VU meter (Simpson 10472 or equiv.)
Q1, Q2, Q4, Q5 — High gain, low noise silicon NPN transistor (2N3391A or equiv.)
Q3, Q6, Q7 — High gain, low noise silicon PNP transistor (2N4058 or equiv.)
Q8 — Medium gain silicon NPN transistor (2N3392 or equiv.)
R1*, R6, R18, R19, R20, R21, R22 — 100 K, 1/2-watt res.
R2* — 47 K, 1/2-watt res.
R3*, R25 — 180 K, 1/2-watt res.
R4, R9 — 1 K, 1/2-watt res.
R5 — 10 K, 1/2-watt res.
R7 — 430 K, 1/2-watt res.
R8, R28 — 1 Meg, 1/2-watt res.
R10* — 22 K, 1/2-watt res.
R11, R12, R30, R31, R33, R36 — 100-ohm, 1/2-watt res.
R13, R14, R15, R16, R17 — 100 K audio taper pot
R23 — 22 Meg audio taper pot
R24 — 100 K screwdriver-adjustment trimmer pot
R26 — 110 K screwdriver-adjustment trimmer pot
R27 — 470 K screwdriver-adjustment trimmer pot
R29* — 82 K screwdriver-adjustment trimmer pot
R32* — 2.2 K screwdriver-adjustment trimmer pot
R34 — 220-ohm screwdriver-adjustment trimmer pot
R35, R38 — 5 K linear taper pot
R37 — 3.3 K, 1/2-watt res.
SW1 — 4-position, 3-pole, shorting-type rotary switch
SW2, SW3 — 3-position, 1-pole, shorting-type rotary switch
SW4 — 2-pole, single-throw toggle switch
T1 — 50-, 250-, and 600-ohm primary to 85-Kilohm secondary microphone input transformer (Triad A9J or equiv.)

* These resistors should be deposited carbon or other low noise type.

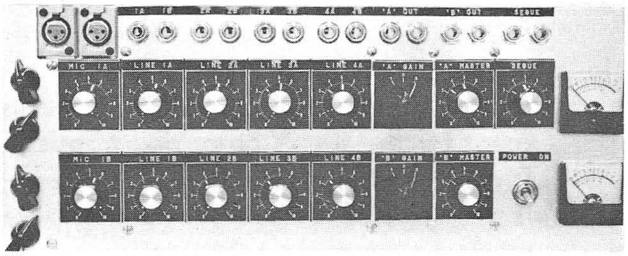


Fig. 3. Front view of mixer.

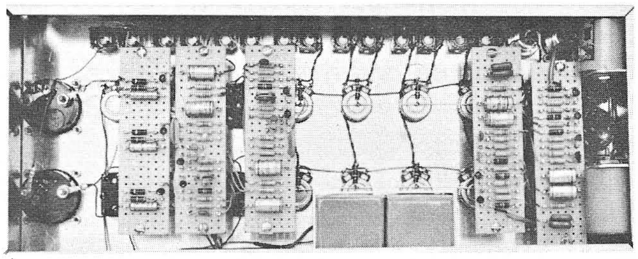


Fig. 4. Rear view of mixer.

The entire mixer maybe housed in a standard 17"x7"x3" aluminum chassis. The leads associated with the inputs and the gain switches of the mixer should be kept as short as possible. The input lead to the mic preamp should be shielded. The ground symbols in Fig. 2 are actual connections to the chassis. Each operational amplifier is built on a separate circuit board and is positioned for shortest lead length. Fig. 3 shows the arrangement of the panel while Fig. 4 reveals the placement of the circuits. The construction and testing of the mixer will present no problems to anyone with a modest amount of experience in building electronic equipment. Modifications in the overall design, such as changing the number of channels or inputs, or adding signal routing switches, should be similarly straightforward.

Performance specifications based on measurements made on a completed mixer are listed below. The preamp gain was set at +40 dB and the adder gain at +10 dB. Except where noted, controls were turned up for maximum gain, and the mic impedance was set at 250 ohms.
Impedance of microphone input: 50 ohms, 250 ohms, 600 ohms, or 80 Kilohms, selectable by SW1
Impedance of a line input: 50 Kilohms
Impedance of an output: 0-1250 ohms, depending upon setting of master gain control
Equivalent noise source referred to line input: 8 microvolts
Equivalent noise source referred to mic input: 0.5 microvolts
Frequency response of line input: ±1 dB 0-30 kHz
Frequency response of mic input: ±1 dB 40 Hz-13 kHz; ±2 dB 20 Hz-19 kHz
Total harmonic distortion at output level of 0 dB (1 kHz test signal): less than 0.2%

The Synket

Paul Ketoff

The Synket (Synthesizer and Ketoff) is an electronic system that both generates and controls sound. It can be considered a bona fide musical instrument because its flexibility permits the performance of electronic music compositions without pre-recording (something which has not been possible with traditional studio systems).

It would be useful, before discussing the Synket, to clarify general concepts of sound. The parameters of sound can be summed up as frequency, amplitude, waveform, and time. Frequency, amplitude, and waveform can be varied in time. These variations are called modulations. Modulations occur separately or several together (i.e., frequency, amplitude, and waveform can all be modulated simultaneously), and these can have different shapes during the same time span. From this derives the infinite variety of sound. Perhaps the most common and illustrative example of this phenomenon is the human voice, where the larynx can be considered a complex sound generator that varies amplitude and frequency. The sound generated in the larynx passes through the mouth and nose where the oral cavity, by changing shape and size, modulates the timbre, and at the same time the mouth, by opening and closing, modulates the amplitude.

The Synket is conceived to generate and modulate sounds in terms of frequency, timbral spectrum, and amplitude, where the modulations can be controlled individually or several together. The process suggests the word "sound-combiner", and there are three parallel systems so called, racked one above the other. A detailed block diagram of a sound-combiner appears in Fig. 1.

Each of the sound-combiners includes: (1) a square wave generator, with a range of 5-20,000 Hz, controllable continuously with a knob, or in discrete steps using a keyboard; (2) a chain of push-button dividers that divide the frequency of the generator by 2, 4, 8, 3, or 5, permitting an enriching of the harmonic spectrum as the dividers are depressed; (3) a selective (variable

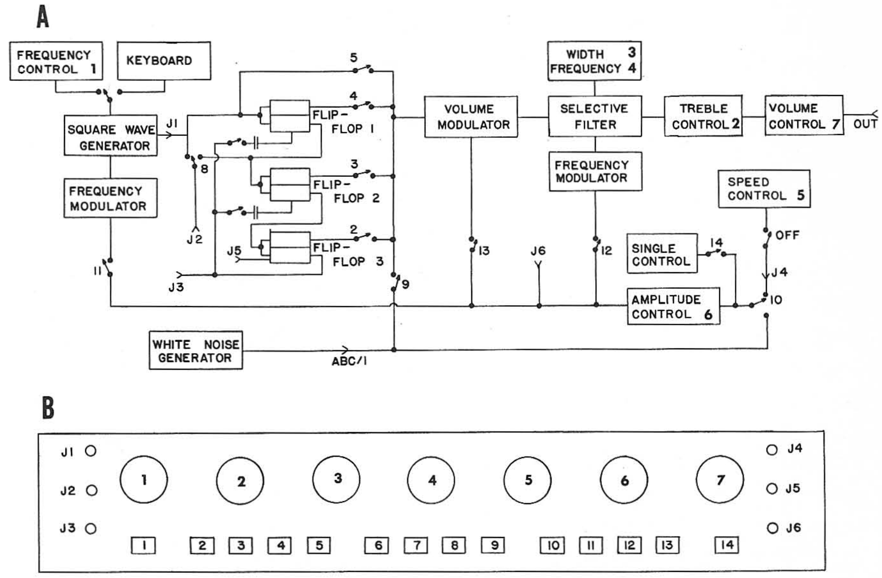


Fig. 1. Functional block diagram (A) and panel arrangement (B) of a sound-combiner. The numbers within the blocks in A are the continuously variable controls (large circles) in B. Similarly, the numbered switches are the rectangular push-buttons, and the numbered terminals are the small circular jacks.

bandpass) filter with continuous action from 40-20,000 Hz, making possible profound timbral changes; (4) control of volume to allow balancing with the other combiners; and (5) modulators that can modulate the frequency of the generator, the frequency of the filter, and the amplitude of the signal.

The square wave generators, with their dividers, can be linked together in different ways, so that different harmonic combinations can be obtained with one single generator. In this way one can obtain sounds that spread ten octaves or sounds that, derived from the same tone, with divisions in 6ths, 5ths, and 3rds, produce unusual beats. Further, when a very low frequency is used, the square wave is perceived as pulses, giving various rhythmic combinations.

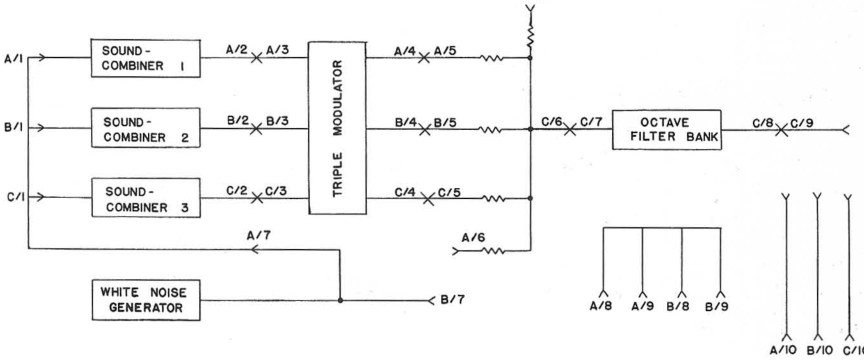
The filter of each combiner permits extreme variations in timbre; for example, the slow pulses can change continuously from a sound similar to a drum to the sound of a drop of water as the frequency of the filter is raised. It is worth noting, in passing, that each filter can function, in effect, as a sine wave generator.

The modulator on each combiner, as has already been said, makes it possible to vary the frequency of the generator, the frequency of the selective filter, and the volume of the signal. Each modulator can be controlled: (1) by a separate oscillator; (2) by the sound generators, giving the possibility of synchronizing rhythmic impulses and modulations; (3) from a keyboard; or (4) from an external sound source. In addition, the modulators of each combiner can be linked in such a way that they are synchronized with each other.

The other parts of the Synket are: (1) a white noise generator that can be put in each combiner and modulated as the square wave generator; (2) three amplitude modulators with intermittent action (with three different characteristics) that can modulate each or all the combiners; and (3) an octave filter bank that affects the signal output from any or all combiners. A block diagram of the complete Synket appears in Fig. 2, and the control panel in Fig. 3. A photograph of the complete Synket appears in Fig. 4.

While it is difficult to imagine all the rhythmic combinations and sounds one can produce, perhaps one can have an idea, from this description, what the possibilities of this instrument are. An important advantage of the Synket is that it permits the real-time composition of electronic music, eliminating many of the annoying interruptions of splicing, editing, and mixing tapes. What once made necessary many hours of work and a multitude of machines and processes can now be done in a brief time with a single apparatus. The problems of composition for and playing on the Synket become, more or less, similar to those of composing for and playing on traditional instruments and, with the possibility of live performance, one avoids the discomfort of the impersonality we have felt with electronic music until now, as well as the many mechani-

Fig. 2. Block diagram of the complete Synket. Letters and numbers refer to jacks that allow unusual or complex interconnections to be set up.



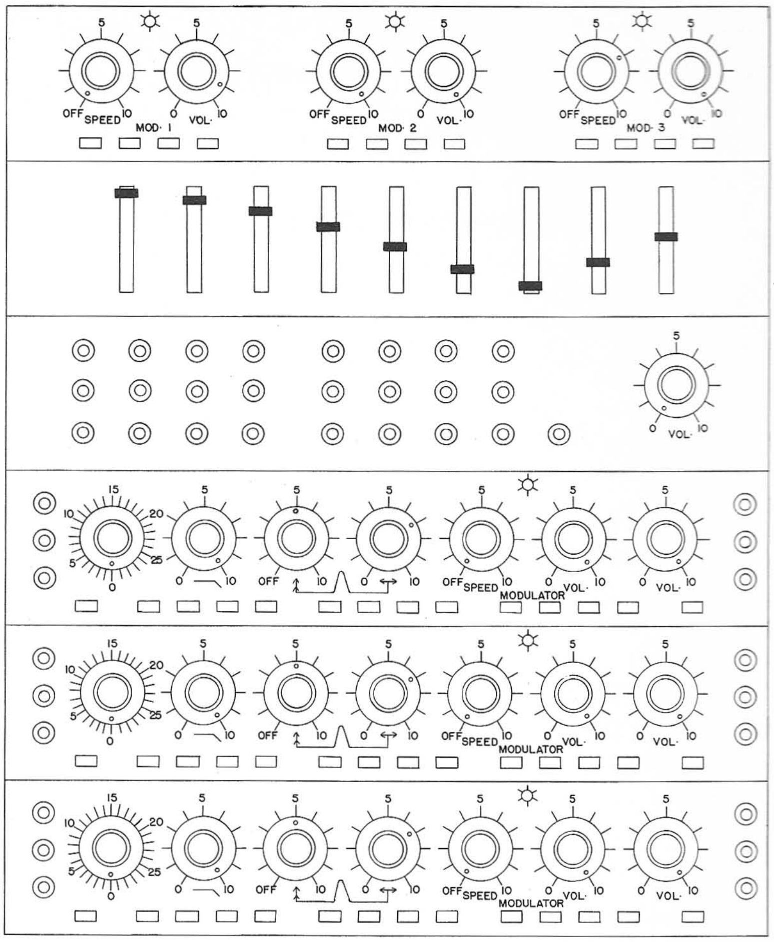


Fig. 3. Control panel of the Synket (top to bottom): modulators, octave filter bank, patch panel, three sound combiners.

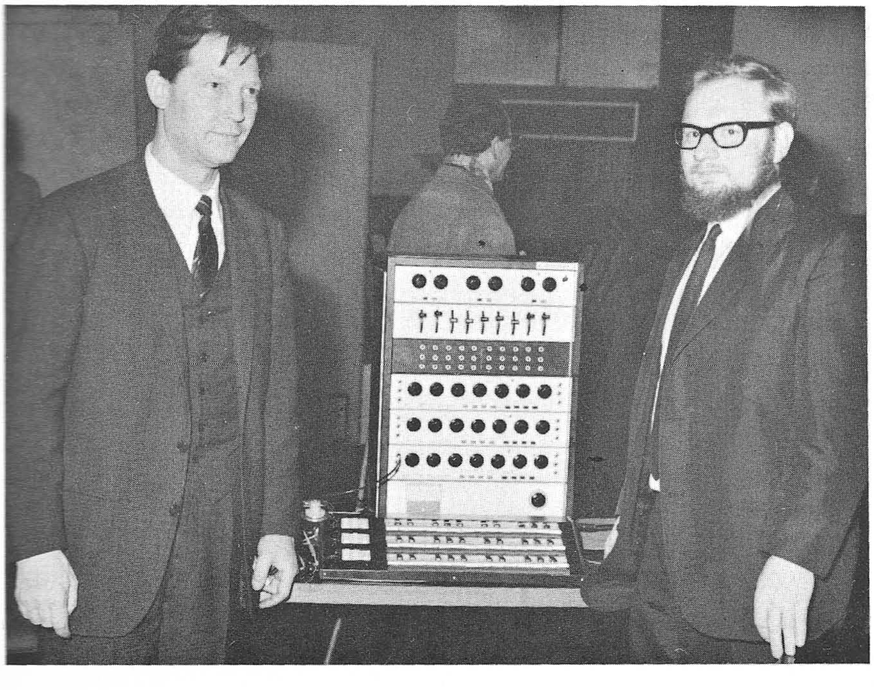


Fig. 4. Paul Ketoff (left) and composer-performer John Eaton with the Synket.

cal disadvantages the tape recorder presents when electronic music is combined with instrumental and/or vocal music.

(translated by Joel Chadabe)

Reviews

Organized Sound by Tod Dockstader; Owl Records, ORLP 6, 7, 8 (stereo only).

The following review restricts itself quite deliberately to the sounds that emanate from the speakers. No attempt has been made to delve into the underlying mechanics by which these sounds were produced. The review is exclusively a listener's reaction.

In the liner notes for these three records, Tod Dockstader emphasizes repeatedly that his creations are not music in the accepted sense, but "organized sound". He is an engineer, not a musician, and he likes it that way. On the liner for Quatermass he says: "I have the feeling that the training I've had (engineering) for what I'm doing, is the best training, and if I had musical training I'm not sure it would help me, and I suspect it might hinder me. You see, I deal in a sort of chaos of sound: in Apocalypse, for instance, there's one little movement that has in it a cat screaming, a dime store toy that moos when you turn it upside down, doors opening and slamming shut — of course, all this is very difficult to identify in the piece now, but those are the sources."

In the liner notes for the record containing Luna Park, Dockstader makes a statement that is more to the point and less likely to get him into hot water. Here he says that the pieces represent "simply an instinctive arrangement of alternating tension and release". With this more forthright explanation the listener can approach the recorded sounds without feeling an obligation to search in them for more than meets the ear.

Record ORLP 6 presents three pieces. In the earliest of these, Traveling Music (1960), we hear clearly separated sounds: gong beats which initiate sustained, drone-like noises of indeterminate pitch. Soon these are opposed by sharply percussive, clear beats and pulsations which alternate stereophonically from speaker to speaker. Both ingredients — the sustained and the percussive — are now developed. The drones lose their gong-beat attack and become continuous bands of sound which overlap or pile up and change timbre, loudness, and other characteristics. At the same time, the pulsations also grow in timbral and dynamic variety, yet do not lose their metric regularity. They grow faster, however, and at their fastest they very nearly turn into pitch vibrations. At the same time, the drone entrances become more and more distinct and regular until they turn into regular, almost percussive beats. Thus the contrast of sustained sounds versus metric pulses is reversed.

Whereas the thematic ingredients presented in Luna Park (1961) are quite a bit more complex, they are nevertheless of equally unmistakable and impressive identity. The dominating sound pattern here consists of a kind of mechanical laughter. In the first section of the piece this pattern is manipulated through superimpositions, retards and accelerations, stereophonic alternations, echo effects, and the like. The second section counters these lively goings-on with sustained intervals which move upward in slow, dove-tailed glissandi. The third section returns to a fast and tumultuous design.

Apocalypse (1961) is somewhat more elusive in its sounds. There is a multitude of swellings, fadings, glissandi, and stereophonically alternated timbral contrasts, etc. Unfortunately, every-

thing is so undefined and chaotic that it is difficult, if not impossible, for the listener to find a focal point, a dominating (or at least germinal) source pattern. Throughout its four movements, Apocalypse seems to remain shapeless and aimless and thus grows more and more tedious in spite of all its activity.

One might interject at this point that Dockstader, throughout his explanations, mentions that his early works represent experiments with the medium of taped sound as such, the way primitive man may have enjoyed mere tappings on a hollow piece of wood, etc. Eventually, he says, these experiments began to lose some of their fascination for him and he felt he had to operate more planfully and to organize sounds more judiciously, just as primitive man eventually began to organize his taps and pitches into patterns and melodies. In Apocalypse Dockstader seems to be in a transitional stage; he has gone beyond experimentation but has not yet succeeded in harnessing his sounds convincingly.

The next record, ORLP 7, presents two fragments from Apocalypse which, in Dockstader's words, are chips that fell off the original Apocalypse in the process of reducing it in size from a form that runs "for hours" to its present duration of some 19 minutes. The first of these "chips" is a study in percussive tremolo effects ranging from the rapid "trrr" of a drum stick bouncing on a hard surface to something more akin to a pneumatic drill. Only the last sound in the piece is a single impact — it resembles (and may actually be) a timpani stroke. A multitude of rolls and tremolandi are manipulated in loudness and speed, timbre and pitch; there are glissando effects, sharp rattles and blurred whirrs, single patterns, overlapping ones, and simultaneous soundings of any number of variations of all of them; and after all the changes have been rung, the piece suddenly ends. Unfortunately, although it drums and chirps and makes a lot of entertaining noises, there is little evidence of an overall structural design.

The second fragment does more or less the same thing with a related basic sound pattern: flowing, vibrato-like pulsations rather than percussive ones. It seems less interesting and imaginative and has a somewhat disturbing ostinato pitch pattern of four chromatic steps of regular 6/8 meter which enters and dominates the action somewhere in midstream and seems peculiarly incongruous. This piece, too, runs around rather aimlessly and then ends abruptly.

In Drone (1962) Dockstader juxtaposes a guitar and electronically generated sounds. Through many different manipulations these sounds are made to oppose each other even more than they do normally. Conversely, they are altered to the point of near identity. This timbral meeting ground, however, is not constant but ranges from electronic imitations of "real" guitar sounds to guitar sounds whose timbral characteristics have been altered to simulate typical electronic sounds. It is quite a fascinating business.

Water Music (1963) is the most recent work on this record. It consists of six parts, each involving a different treatment of noises generated in one way or another in connection with water. To judge by the sounds emanating from the speakers, Dockstader avoided obvious splashings and gurglings. He concentrated instead on the hooting pitches produced by blowing over tops of bottles containing various amounts of (presumably) water, on percussive noises produced by dripping water taps, or on hitting pots, bowls, and tubs filled with different amounts of water. However, there also are noises that sound like the scraping of a fingernail along the teeth of a comb; there are strange, pitiful mewls and howls, and a multitude of other sounds. The individual movements are quite short and full of imaginative effects. Even so (and similar to the early Apocalypse) this work lacks direction, suffers from a paucity of potent germinal ideas, and thus remains aimless and shapeless.

The third record, ORLP 8, contains only one piece: Quatermass (1964). This work represents a major step forward from the earlier experimentations and artistic gropings toward genuine artistic communication. It is as inventive as Luna Park and Traveling Music but has in addition a strangely frightening, tensely driving air of mystery. Here, unlike in the preceding pieces, one senses a preconceived vision and a much more successfully controlled structure. The character of sounds and patterns is well defined in each of the five movements, and the movements themselves are well contrasted from one another, yet complement each other. Moreover, they are held together by certain ideas which go through the entire piece.

The first movement, Song and Lament (a rather misleading title since the movement consists of three, not two, parts and begins with the Lament), contrasts booming gong sounds, artificially lowered in pitch and prolonged in length, with loud, nasal, anguished wails. Added to these are synthetic sounds which, however, remain in the background. (The wails, incidentally, were generated by a toy balloon which was blown up, then the air let out slowly through its finger-pinched neck whose length was changed by pulling and stretching to alter the pitch, timbre, loudness, etc.) Once taped, the balloon sounds were treated to a remarkable variety of additional manipulations, and similar sounds were added synthetically. The total impression of this movement is strangely upsetting, almost frightening; a most unusual tour de force.

The second movement, Tango, sets percussive, motoric noises against siren-like glissandi, and shot-like reports against sustained and continuous sounds resembling some of the material of the first movement. It all begins mysteriously with muffled machine rhythms. These grow in complexity. Then, suddenly, the anguished wails of the first movement enter. The fabric now grows louder and ever more chaotic and aggressive. At its peak of menacing confusion there is a short burst of crossfire from speaker to speaker, followed by a stunned silence. A half-hearted attempt to resume the ghostly commotion is snuffed out by a final coup de grace.

Parade is the title of the third movement, but it is no parade in the ordinary sense. Dockstader explains that he wanted to celebrate a "sort of pompous, John Philip Sousa kind of crashing about". The entire piece is "of cymbals and white noise". The cymbal sounds are distorted and manipulated to create unusual effects, such as cymbal glissandi. And the white noise is the between-stations-hiss of an FM radio, filtered, altered, coaxed, and pummeled. Dockstader chose cymbals because they produce the closest "live" equivalent to electronic white noise. In the piece, the cymbals provide high metallic sizzlings and crashings, while the synthetic sound generator adds a turmoil of ramblings, swishings, and boomings, punctuated by roars and bangs. Eventually the low, electronic growls drive out the silvery cymbal glitter. Toward the end of this section the nasal balloon cries of the preceding movement enter the fray, but are quickly squelched and smothered by the powerful hissings and boomings. There is a strong atmosphere of tragic conflict, almost of catastrophy, over all of this. Considering the title Dockstader has given to this movement, one wonders whether he had intended it to elicit such a reaction from a listener or whether the piece just happened to turn out this way. But whatever the reasons for its character, this movement is an impressive feat and one not easily forgotten.

Flight is a long, reflective, rather quiet movement which presents a succession of episodes full of weird sounds and patterns. Unlike the preceding "sound organizations" it is not governed by a single idea or a dominating and unifying conflict. Rather, it operates along free association, one episode giving rise to the next, and so forth. This technique does not easily produce convincing forms, and in spite of an ostinato figure of two alternating low pitches at the interval of a fifth, which appears early in the piece and reappears at a higher pitch at the very end, the movement is in constant danger of falling apart. It is interesting to listen to, however, be-

cause of its great variety of sounds, textures, and patterns, and its atmosphere of portentous tension. Nevertheless, the listener is likely to lose track of its direction, let alone its goal.

The last movement, Second Song, recapitulates and develops as its chief task the balloon wails, etc. of the first movement. In addition, it combines these ingredients with most of the sounds and textures of the other movements, separately as well as in various combinations and also brings to this new conglomerate of known materials a number of new sounds and patterns. It is thus the most ambitious and most complex of all the movements, and much of what it contains is remarkably imaginative and eloquent. But again, one encounters that certain weakness of structure: the movement fails to come to a focal point, a climactic culmination, and it ends unexpectedly and rather disappointingly — a real pity considering the wealth of ingenuity and imagination which has gone into it in every other respect.

* * *

In the beginning of this review I quoted Dockstader's acknowledgement that he has had no musical training and does therefore not consider his creations to be musical compositions in any conventional sense. He has simply experimented with random sound sources, manipulated the sounds in many ways, and arranged the results into pieces of some sort. Summing up his method he says: "Now, there's no [need for] musical training in making music out of that."

In the liner notes for the record which begins with Drone he writes: "In practice, music is an organization of abstract sounds that is acceptable; if unacceptable, it's called noise .... [My] pieces are full of such rejected sounds, as well as some of the more acceptable noise-makers of the orchestra. They are for people who listen — to sound; who listen to music as organized sound, enjoy sound, and follow and explore the organization."

After listening to six LP sides of these sound structures two impressions predominate: On the one hand, Dockstader is without doubt an enthusiastic and imaginative creator of memorable noises. No sound source is too lowly or too refined for him, none is too crude or too delicate for his purposes. On the other hand, his ever-alert sensitivity to sounds and his considerable resourcefulness in manipulating them is not equaled by a commensurate artistically creative imagination and by adequate knowledge of the craft of putting a musical composition (for want of a better term) together. Granted that some of the pieces on these records rise considerably above the level of mere experiments, even the best of them nevertheless fall short of being truly convincing artistic statements. This regrettable fact becomes increasingly apparent with repeated hearings: instead of growing more arresting and communicative, the pieces soon lose their initial fascination without revealing new facets. They simply become more and more obvious. No amount of Dockstader's rather awkward, pseudo-esthetic sophistry can gloss over this crucial shortcoming.

— Kurt Stone

Concert Piece for Synket and Symphony Orchestra, John Eaton

Performance: John Eaton, Synket, accompanied by the Berkshire Music Center Orchestra conducted by Gunther Schuller; Tanglewood, August 9, 1967.

It is difficult to discuss John Eaton's Concert Piece for Synket and Symphony Orchestra outside of the context of the growing interest in electronic performing instruments (and electronic compositional systems in general). It is as if the instrument should be reviewed as well as the piece, although it is in a certain sense unfair to the composer (the piece itself was excellent and in many ways compositionally innovative) to devote much time to his instrument. Yet to understand the composition one must understand the material from which it is composed.

The Synket is a performable and portable electronic sound system designed and built by Paul Ketoff in Rome, Italy. Basically, it consists of three sound systems (Eaton calls them "combiners") racked one above the other, each containing a square wave generator and various filters and control voltage generators which act upon the signal of that particular system; the systems can be patched into each other so that one signal generator can be controlled by any of the other systems. In addition there are various envelope generators, pulse generators, and control voltage generators that can be applied to any parameter of any system. (Ketoff's description of the Synket appears elsewhere in this issue; see also "A Portable Electronic Instrument" by John Eaton, in the October 1966 Music Journal.) Theoretically, the machine would seem extremely versatile within its particular limitations. But the most important criteria for judging a performing instrument are its controls.

The Synket is performed by pushing buttons, turning dials, playing keyboards, depressing a volume pedal, and every now and then patching. Although the complexity of type of control is understandable because of the instrument's history (it was originally designed as a compact studio instrument for the American Academy in Rome), and although Eaton's was a virtuoso performance, much exploration of the possibilities of this instrument still needs to be done. What are the capabilities of the instrument in terms of other esthetics or compositional ideas? To what extent does the difficulty of performing on the Synket lead one into Synket clichés? To what extent is electronic nuance (there is such a thing) possible, or how carefully can the sounds be controlled and predicted? And how can we now judge fully the use of the instrument without greater experience in listening to others' compositions for it?

The compositional problem of the piece is relating two worlds of sound in a way that they seem part of the same piece. One could unite them in procedure while giving them different material, or unite them in procedure and material, or make them totally different but simultaneous, or.... In this case, the orchestra was made to sound like a Super-Synket. The orchestra was divided into two groups tuned a quarter-tone apart, designed not to be heard stereophonically but as one mass of sound. Aside from Eaton's interest in quarter-tone music generally (his own words are worth repeating: "...this allows me to bathe rejuvenescently in the ancient but still pure springs of microtonal melody."), this tuning of the orchestra permits a meeting on common ground with the "tuning" of the Synket, which is, of course, not played diatonically (to imagine it so would be ridiculous). With the quarter-tone tuning of clusters, occasional legato phrases in the woodwinds and brass, strong shifts of register, and very sophisticated timbre changes, the orchestra enters the Synket sound world, which leaves Eaton free to mingle without fear of offending.

How does a Synket socialize with an orchestra without bringing to mind a strange piano concerto? The dramatic roles that the orchestra and soloist enacted in the Romantic piano concerto were basically quite simple, predictable, and formulated on the support-or-opposition drama between an instrument of one basic timbre (piano, say) and an instrument of many timbres (orchestra). The problem was how not to have a brilliant orchestra make a single piano seem dull. In Eaton's piece the problem is, in many ways, reversed. The Synket can be louder than the orchestra, it can change timbre (perhaps even more effectively than the orchestra), it can play higher, faster, and it has flashing lights. There is a necessary compromise in the Synket part, and the Synket must be brought into the world of the orchestra as well as vice versa. This was done very successfully. The Synket moved in and out of the texture, making statements of varying power, and of varying character, in relation to the orchestra, so that a kind of scale of difference was felt from similarity to dissimilarity. Some of the most effective moments had the Synket interjecting its own material against the orchestra playing waves of undulating phrases, and these strikingly effective moments were the moments best performed.

Performance of this piece is difficult in many ways. The Berkshire Music Center Orchestra is an excellent group, but playing quarter-tone chords is new enough, in this country at least, so that they are difficult to hear exactly. But then first performances of complex pieces are always difficult to hear exactly. I would say the performance was overall very effective. Gunther Schuller conducted, and the success of the orchestra's performance was due largely to that. Eaton is incredibly virtuoso on the Synket. But should his range of skills be always necessary, or should we consider the Synket a prototype of performing instruments to come that will give more possibilities for greater control with less effort?

It seems to me that deriving maximum potential from any performing instrument depends on two design characteristics: simplicity of control and diversity of output. I would suggest an instrument with three-dimensional controls, such as a lever that goes up and down, in and out, and sideways, each direction controlling a certain parameter defined by which of several foot pedals is depressed. Further, that the generators and modifiers are replaceable with other types that can be placed into the machine with a clasp. Thus, fewer controls can be handled more precisely and can be used to control a greater variety of components. I suggest this (and without a great deal of thought, because I have not been very involved with performing electronic music up until now) largely for purposes of comparison, because I feel that there is a great danger in thinking of the Synket as the instrument, and because the Synket is more, not less, interesting when thought of as a prototype that deals in an ingenious and sophisticated way with electronic performance problems. In portability, it will serve as a model. Perhaps Ketoff will redesign it in a different direction. Perhaps we'll have Synket ensembles.

The problem is that Eaton's piece raises so many interesting problems. I could not help but notice, again, that electronic sounds are so basically different from instrumental sounds even when they are united so skillfully. And yet, in a different way, they were closer than I've heard them before because performing the Synket is more unpredictable than hearing the same sounds on tape, and tape machines are notoriously indifferent to the mood of the moment. The piece raises questions at so many levels, and it is such a splendid composition besides, that it should be played in many places and soon. You know, the idea of performing electronic music is lively.

— Joel Chadabe

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