MICROPHONE PLACEMENT EXPERIMENTS

The purpose of the microphone placement experiments was to arrive at a suitable microphone balance for amplification and recording of various extended techniques for the B-flat Tenor Saxophone. The extended techniques were:

1) The Throat Purr
2) Singing in the Horn
3) Key Slaps
4) Multiphonics(Split tones)
5) Double notes

1) The throat purr is the inconsistant oscillation of the vocal cords that takes place before the full amplitude of the sound to be made is achieved. It sounds very much like the purr of a cat. The problem with this very colorful sound is the very low amplitude of its performance. The saxophone acts as a resonator in this case. It colors the sound in various ways. The performer controls the pitch of the throat purr by alternating the number and location of depressed keys. For example, if all the keys are in a closed position, the purr will have a very low frequency quality, also the actual sound will be coming from the bell of the saxophone. This can be captured very easily with a microphone balance of one dynamic cardiod approximately 3 to 5 inches form the bell of the horn. As the keys are lifted, from lowest B-flat upward chromatically, the frequency or pitch of the throat purr gets higher. Due to the design of the saxophone it is possible to create a stereo pan effect with the throat purr and the proper microhpone balance. Using three locations of sound, on the saxophone, the bell, the right stack, and the left stack, a variety of effects are possible. On the demonstration tape that is provided with this paper, the microphone balance is as follows:

Left Channel -- Sennheiser -- 3" from the left stack
Right Channel-- Sennheiser -- 3" from the right stack
Composite -- AKG-D1000E -- 3-5" from the bell

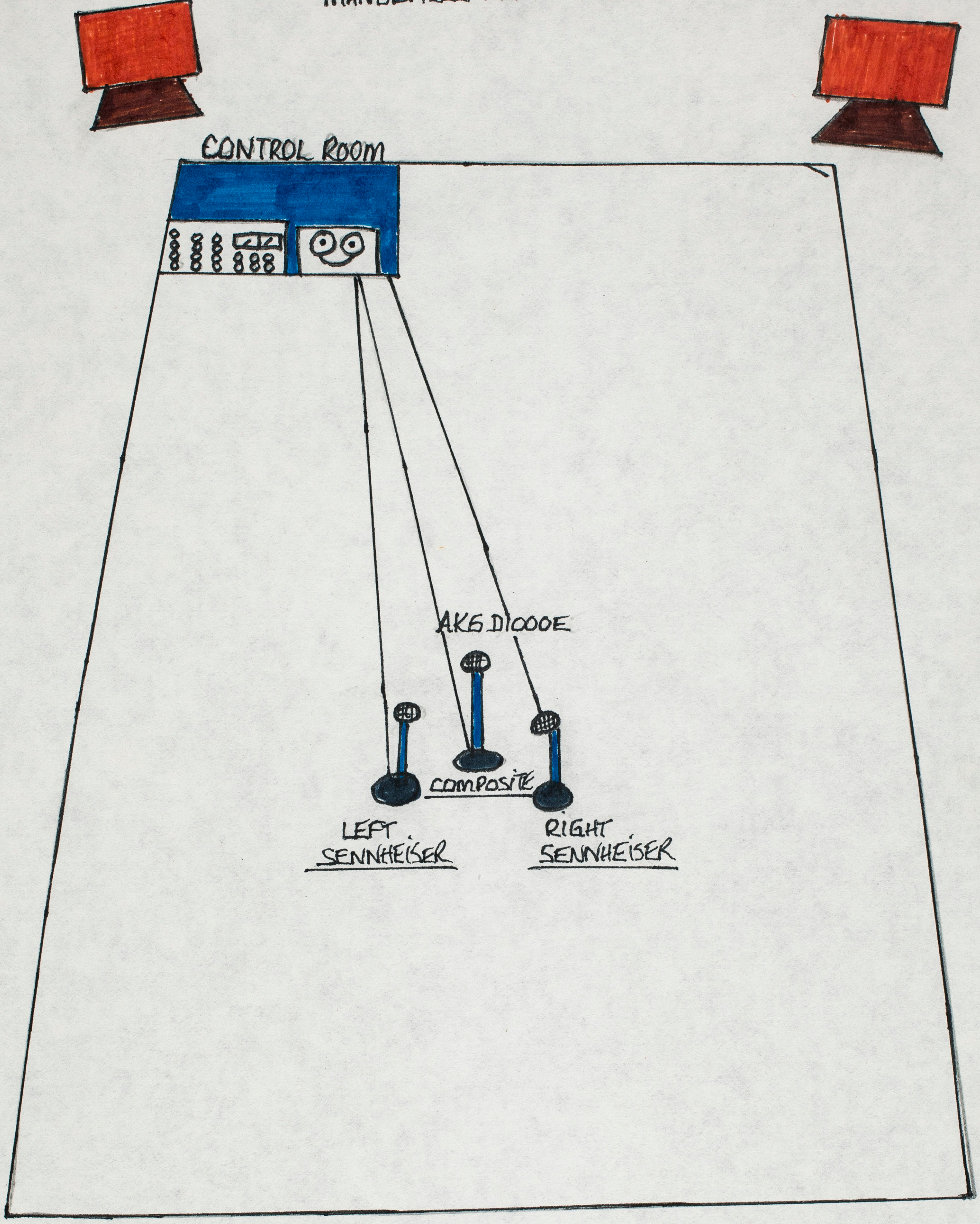
With this microphone balance the stereo pan was realized. Actually with the AKG feeding both channels the purr is consistantly in both channels but depending on which stack was open the emphasis switches from the left channel to the right channel. (Example I Demo-Tape) In this example the pan effect is created by lifting the B key of the left stack, thus emphasis in the left channel as well as a pitch change,and by lifting the D key of the right stack ,thus emphasis in the right channel.

(2) The same stereo pan effect can be created by singing a steady tone into the saxophone and lifting the same keys, However, the pan is not as noticeable as in example I. (Example II Demo-Tape)

(3) Key slaps are standard practice in many contemporary compositions. Using key slaps with the B-flat tenor saxophone is very colorful due to the large chamber and bell of the horn. It is possible to play a chromatic scale using the key slaps,however,like the throat purr,amplitude is low enough to be a performance problem. Microphone balance for the key slaps was approximately 3 to 5 inches from the bell of the horn with the AkG-D 1000E. With this balance the resonance of each key slap tone is made audible. (Example III Demo Tape)

(4) Example four is a short composition utilizing the afore mentioned extended technuques as well as multiphonics,or split tones,and double notes. Double notes are just what the name implies. Two different fingerings for one note,producing two different timbers. The double notes also created an interesting effect due to the change of key stacks . A stereo pan is almost acheived but the resultant sound is slightly different.

MANDEVILLE RECITAL HALL



ELECTRONIC MUSIC PROJECT:

TONE DETECTING SWITCHING SYSTEM

PROGRESS REPORT

by DAVID POYOUROW

for Seminar on Electronic
Music, Winter 1977,
Pauline Oliveros.

ELECTRONIC MUSIC PROJECT:

TONE DETECTING SWITCHING SYSTEM

PROGRESS REPORT

The purpose of the Tone Detecting Switching System is to turn on/off from one to sixteen signals. The idea for this system originated in a desire to have a box that would switch different colored lights with rhythmic accuracy, so that these lights could be used as a "conductor" for a performing ensembe, giving complex multiple tempi and cues. In a piece that I had made (in collaboration with Tom Nunn), information given to the performers over headphones while they performed proved unsuccessful because the performers were unable to hear the sounds they produced. Also, the headphones inadvertently leaked information to the audience, which I had not intended. Around this time, winter 1975, I asked Lou Prince, the department technitian, about the feasibility of building a device that would turn on signals with a prerecorded audio signal. He said it would probably cost $2000 to build. In the following spring I made a version of one of the sections of Warren Burt's Nighthawk in which my voice rotated, changing on each syllable, in a quadraphonic speaker system. To do this, I had to record my voice monaurally on four tracks and then selectively erase parts of each track; however, a device such as that which I had in mind would have made this project less time consum~~e~~ing and could have made live performance possible.

Also, at this time, I was investigating other situations where a rapid succession of sound elements occured. Some of these involved tape plicing techniques in which as many as twenty sounds per second would occur. One, a setting of the Gertrude Stein short story, As a Wife Has a Cow, A Love Story, the text was divided so that each 2- or 3-word phrase was given to one of several voices. The tape splicing process proved too time consuming and inflexible for a satisfactory realization of the piece.

In spring of '76, while casually talking with George Ritscher, I mentioned by idea of a box that would be able to turn on/off a number of circuits--not just lights, but line or mike level audio signals as well. He said such a device was possible to build without far-reaching technical difficulties or economic expense. During the following months, we discussed what capabilities this device should have and various means by which they could be achieved.

Although the capabilities desired of this device have changed in all stages of its developement, four goals appear to be basic: first, each controlled circuit must have a fastest on/off cycling rate of at least four per second; second, the control signal must be in the audio frequency range; third, each controlled circuit must be independently tunable (or controllable) form the other controlled circuits; and last, that the cost to build be less than $100.

Three designs were considered. First, a more "classic" of variable band-pass filters followed by Schmidt triggers was rejected because it was less effecient than other designs and was not particularly inexpensive. Cost ruled out using a transistor switch--it also called for a complex design as it affects the impedance of the circuit, therefor affecting the frequency response of the circuit to the input signal.

The design arrived at was one which uses a relay to swithh the input signal and a "tone decoding" integrated circuit to operate the relay. This design is the least expensive of the three designs. Additionally, it is more efficient than the filter-Schmidt trigger design, and much simpler than the transistor switch.

The Design of the Tone Decoder Switching System

The design for this device is built around an NE 567 integrated circuit, and is based on designs found in "Cue-tone Decoder that Actuates An External Circuit,"[1] by Kurt Blackburn, Integrated Circuit Projects,[2] by Charles D. Rakes, and Signetics data sheet on the NE 567.[3] From the last source

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1. BM/E (Broadcast Management/Engineering), Nov.'76, Vol.12, N.11.
2. Howard W. Sams & Co., Indianapolis, Kansas City, N.Y., 1975.
3. The Signetics Data Book, Sygnetics Corp, Sunnyvale, 1977.

comes the following description:

> "The NE 567 tone and frequency decoder is a highly stable phase-locked loop with synchronous AM lock detection and power output circuitry. Its primary function is to drive a load whenever a sustained frequency within its detection band is present at the self-biased input. The bandwidth, center frequency, and output delay are independently determined by means of four external components."

In the design that we used, the load that is driven by the output of the NE 567 is the coil of a DPDT relay. The center frequency is adjusted by means of a 20K ohm, 15 turn trimpot. The bandwidth is less than $\pm$ 6% of the center frequency, less than a semi-tone. Appendix A is a schematic of the power supply and one tone decoding switch. In the device that George and I built, the tone decoding switch is duplicated eight times. They share a common control input, a phone jack on the right side of the front panel. Eight vertical pairs of signal input phone jacks connect to the relay armature. The signal output jacks on the rear panel connect to the relay contacts which remain open where there is no power applied to the relay coil. Power may be applied to the relay coils manually by either momentary push-button or locking toggle switches located on the top panel of the box, or by the NE 567 when it senses a tone at the control input that is within the frequency band to which it is tuned.

Operating Characteristics Of The Switching System

Although this project has been assembled, not all of the decoding circuits are functioning correctly. Presently, all eight relays can be operated manually, but ONLY five of the eight decoding circuits are functioning. All my experiments and tests involving the operation of the decoding circuits have been with these five circuits.

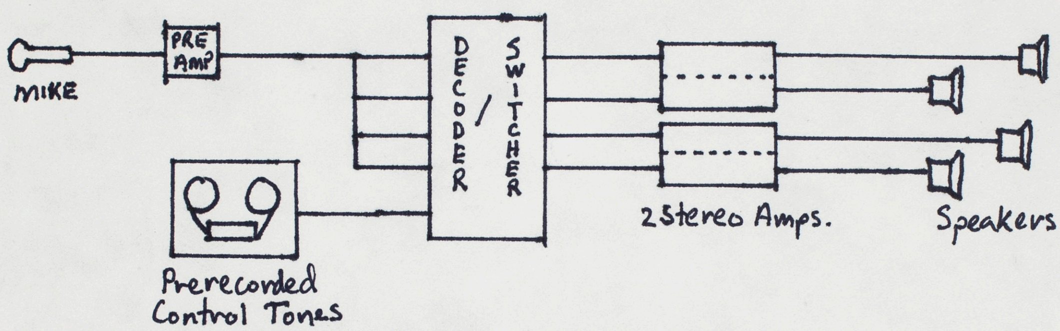
The most disturbing weakness in the device's operation is that an audible click at the output occurs whenever the relay is switched on or off. According to George, additional minor components will be added later to eliminate this defect. Because the clicks are so loud, I have only used line level signals at the signal inputs. Also, when the output is not switched on, there is some leakage of the input signal to the output. However, this leakage is -30 db or less than the signal when it is on, and I consider this flaw minor.

I am very satisfied with the device's performance on all other functions. My three basic goals were all met. Each decoding circuit is independently tunable over a range of three and one-third octaves or more, g (392hz) to b^3 (3950hz). The tuning also remains stable, and has not needed retuning to pitches for which it was set for several days, now. The above frequency range falls very well within the audio spectrum, and to test the "musicality" of this range, I used whistling, electric bass, and tape recordings of Handel arias and Purcell Gamba Fantasies as control signals; all worked effectively. I was able to determine with the electric bass as control input, that the harmonic partials were capable of being detected by the NE 567 when they are of sufficient amplitude. When the control input used was an oscillator, I found that triangle or sine waves caused a more reliable decoding function than pulse or sawtooth waves.

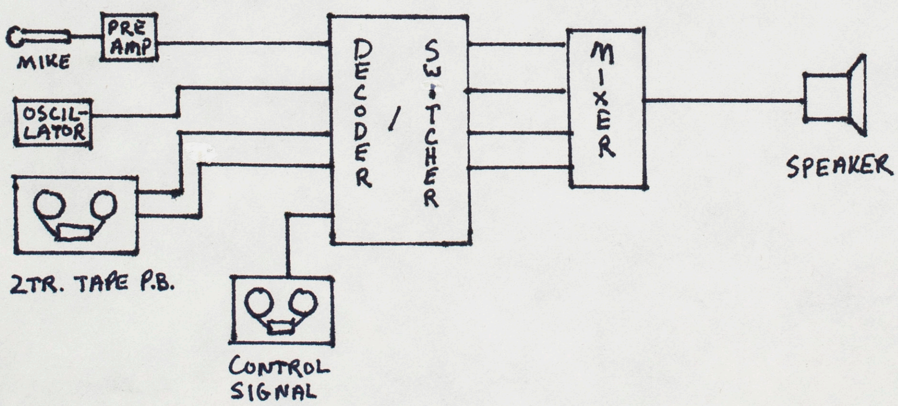
Another basic concern, that the on-off cycle should be at least four per second, was also satisfied. Of the five correctly functioning circuits, all met or surpassed this rate. With one the rate exceeded eight on-off cycles per second. When pushing the limit, the circuits would malfunction in one of two ways. They would falter, and ultimately stop in the off-position; or the relay would be held constantly in the on-position. Although the first malfunction seemed to be from fatigue of the relay or decoding circuit, the second seemed to occur because the output to the relay coil lagged behind the control signal. To test this, I recorded an input signal and a control tone on SEPARATE TRACKS OF a two-track tape recorder, and then measured the distances between the beginnings and the endings of the two signals. These measurements varied from circuit to circuit. The lag between "on" times was from 20 to 37 ms. The lag between "off" times was from 83 to 133 ms. Because of the off-time lag, the input signal will overlap the beginning of the next control signal at on-off cycle rates of 5.3/sec to 8/sec or faster.

Mock-up Applications

Basically, the Tone Decoding Switch System can be used in two ways. First, it may be used to distribute a single input signal to a number of locations. Multiples of the signal are patched to separate signal inputs. The signal outputs are then patched to different amplifiers/speakers. This arrangement will facilitate the distribution of one sound amongst the spatial positioning of the speakers. If the speakers are replaced by transduced objects as in the manner of Rainforest, this distribution can be timbral as well as spatial. Auditioning the system with transduced objects worked very well; beside making melodies of timbre, these objects easily forgave the inherent switching clicks.



The second basic use of the box is to select from a number of different input signals. The outputs of the box can then be mixed. Here, the end result is the melodic concatenation of different elements--but without the tedium of tape splicing.



The preceding uses can be combined in several ways, to form a switching system that both selects and distributes signals. A number of the switching circuits can be used to select from an equal number of inputs. These are mixed at the output, then multipled and patched to the remaining unused inputs. The outputs of the second group of switching circuits can lead to sound reproducers of varying spatial or timbral qualities.

Another combined selecting/distributing arrangement involves the addition of a tape delay. Signals are patched into the <u>upper</u> row of inputs to the box and the outputs are mixed as in the preceding example. This mix is recorded on tape and played back with a delay by another tape machine. The playback is multipled and patched to the <u>bottom</u> row of signal inputs. The outputs of the bottom row can be distributed to sound reproducers. By using the tape delay, the number of combinations of selected inputs to distributed outputs becomes the square of the number of switching circuits used-- as many as sixty-four combinations, if all eight switching circuits are used.

Planned Additions and Modifications

The inherent click caused by relay switching will be eliminated by the addition of minor components, click filters.

The NE 567 has been designed such that the center frequency of the bandwidth to which it is sensitive is present as an output as well. These oscillator outputs of the eight IC's will be controlled by an additional row of push-button switches and mixed to one common output. From this output, control signals can be encoded on tape for later decoding use.

To give the box more flexibility, the bottom signal outputs of two of the switching circuits will be changed from the opposite pole same throw as the upper signal outputs to the same pole opposite throw of the upper signal outputs. The result will enable the user to gate between two input siganl using one switching circuit rather than two.

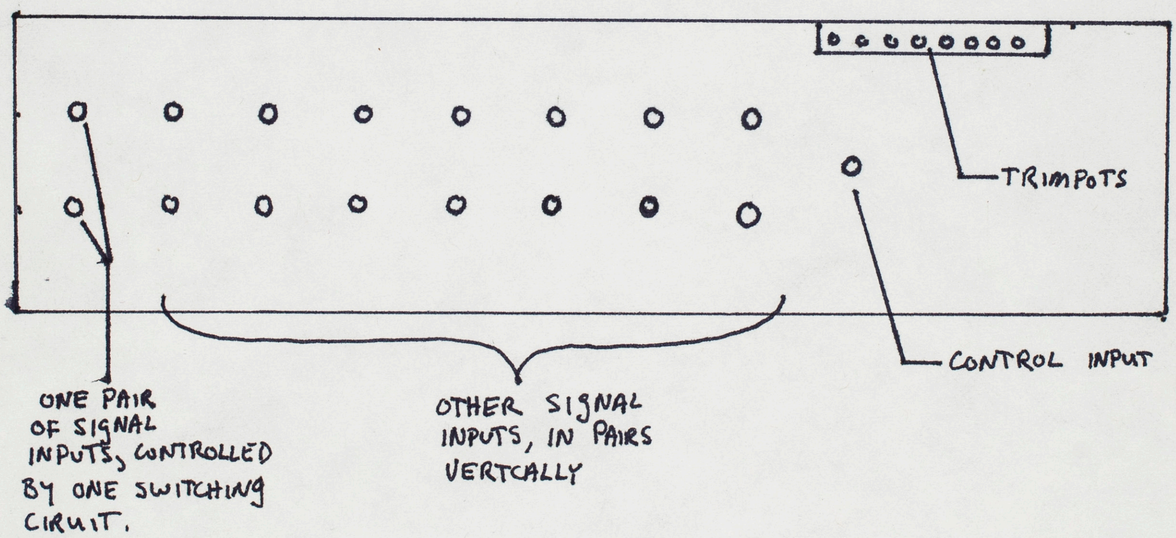
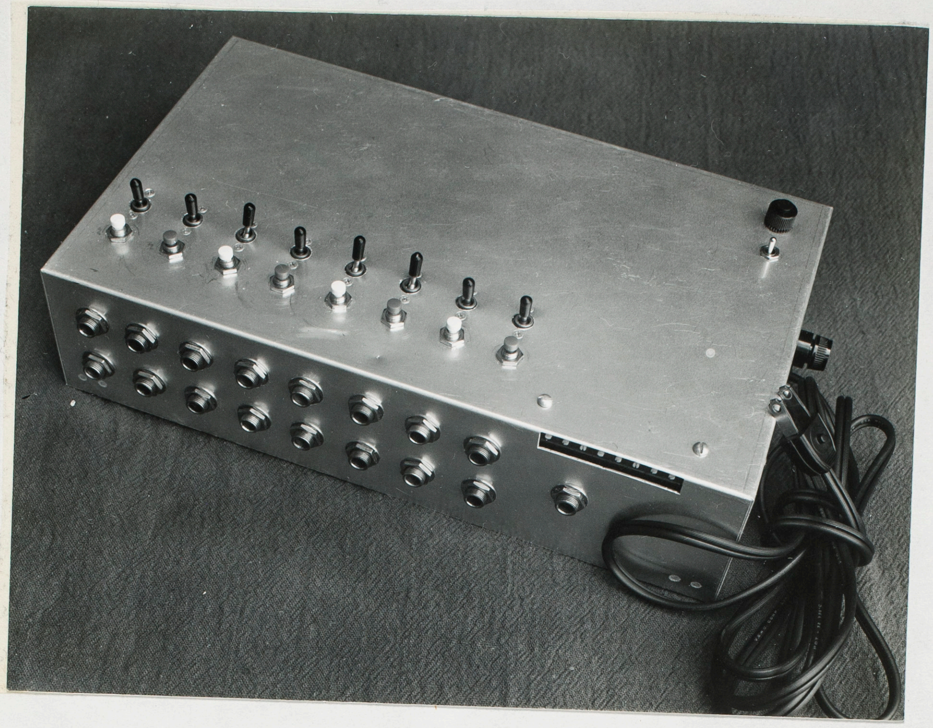
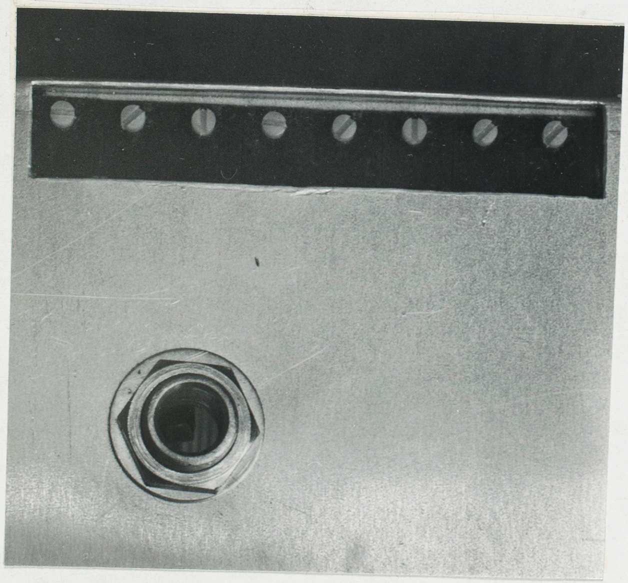


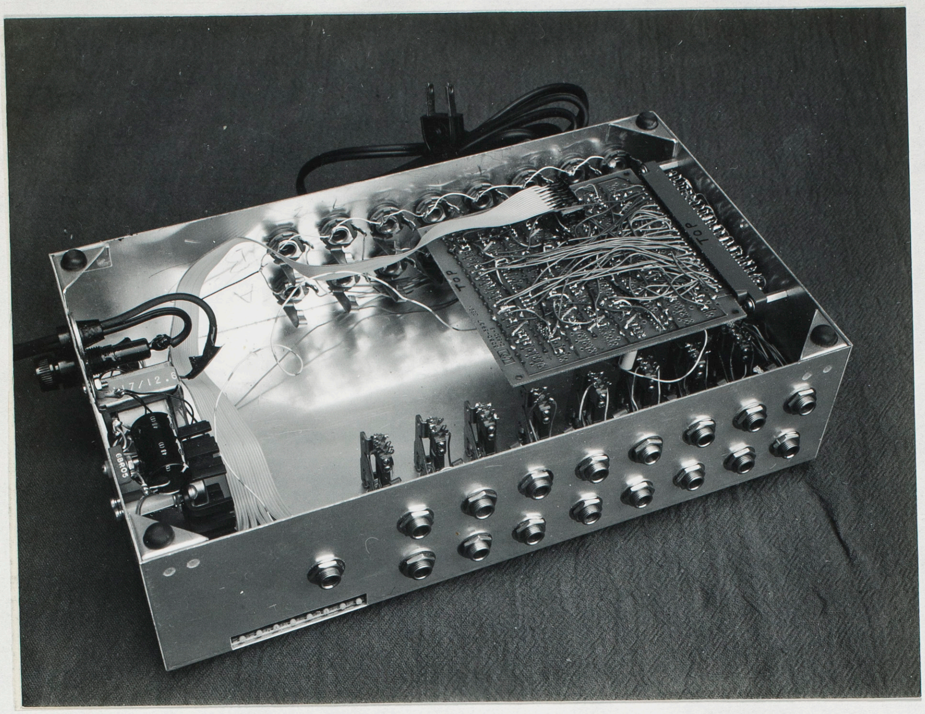
DIAGRAM OF FRONT PANEL



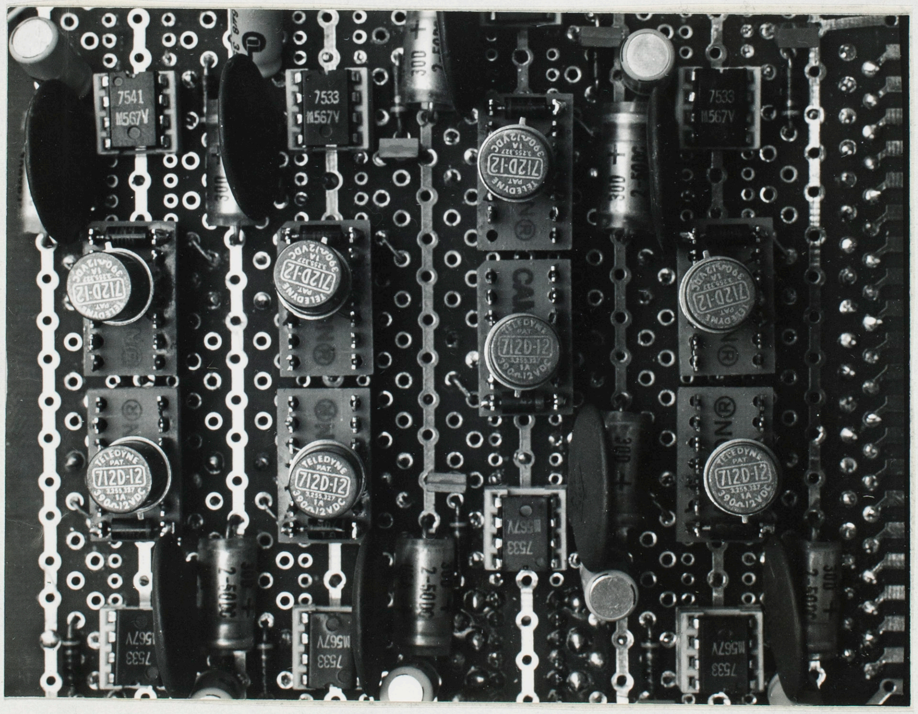
TONE DECODING SWITCHING SYSTEM: VIEW OF TOP & FRONT PANELS.



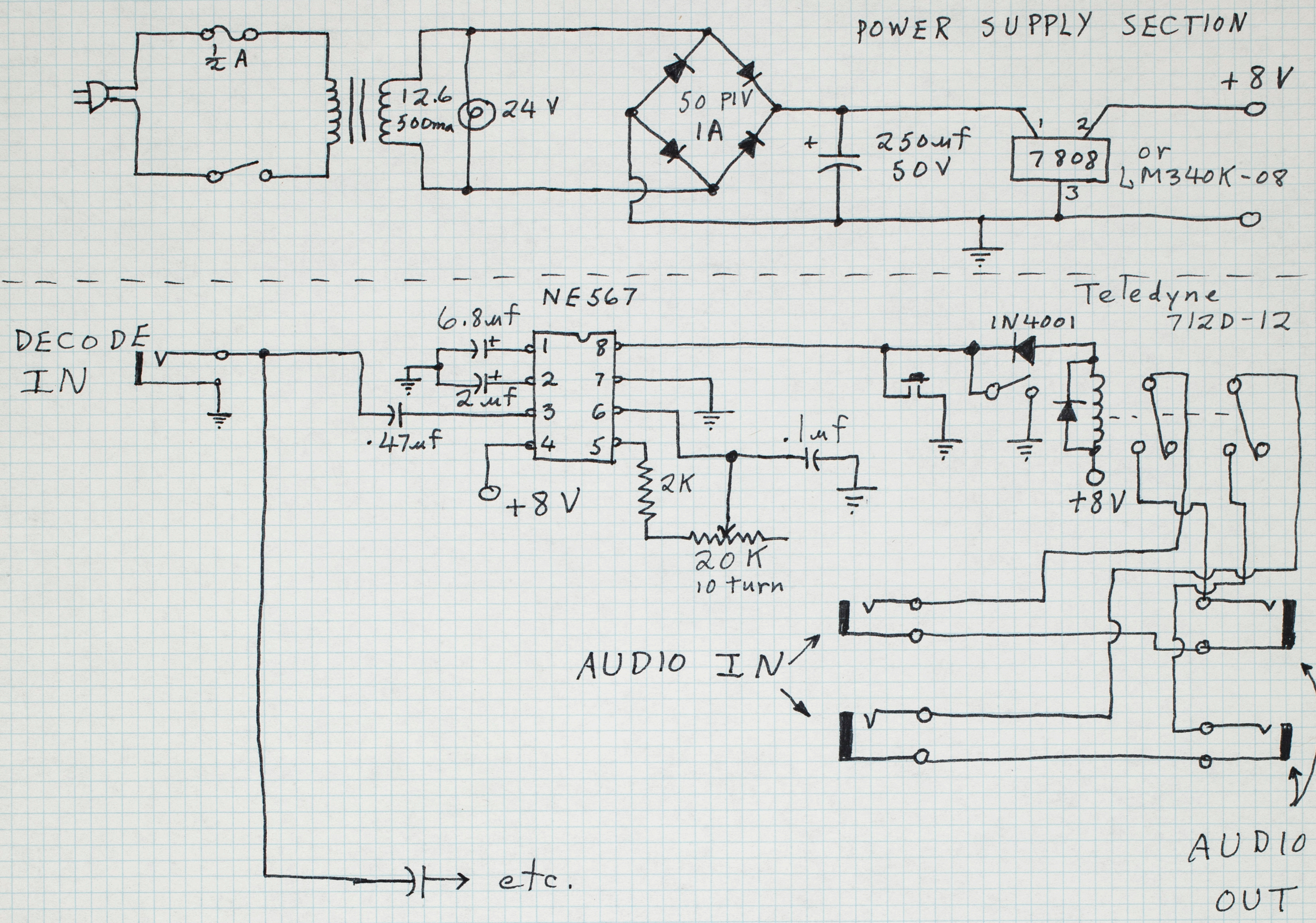
DETAIL OF CONTROL INPUT AND 15 TURN TRIMPOTS FOR ADJUSTING FREQUENCY.



VIEW OF BOTTOM & FRONT PANEL.



DETAIL OF HIDDEN SIDE OF CIRCUIT BOARD, Relays, IC's, and external components.



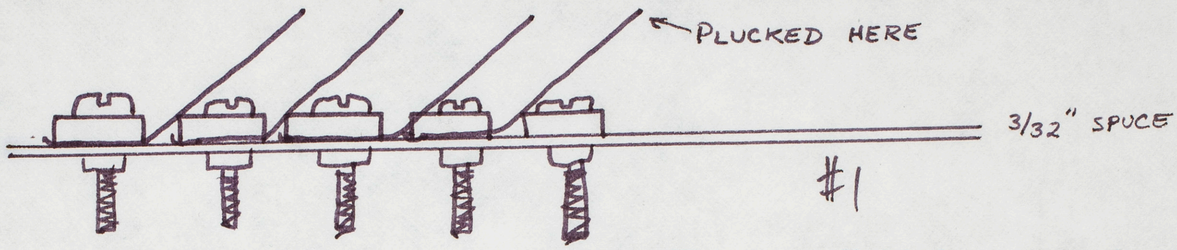
THIS CIRCUIT TIMES 8

TONE DETECTOR

Electro-mechanical and Electromagnetic
Transduction of Spring Steel Finger Piano
Reeds and Related Research for Music 205

Prent Rodgers
Music 20~~5~~2
Pauline Oliveros
May 4, 1977

The research I have undertaken in the 205 seminar has involved improving the sound quality of an electronic finger piano that I built during the fall. This instrument utilizes strips of spring steel which are rigidly mounted at one end and plucked with the fingers at the other, as can be seen in diagram #1 below.



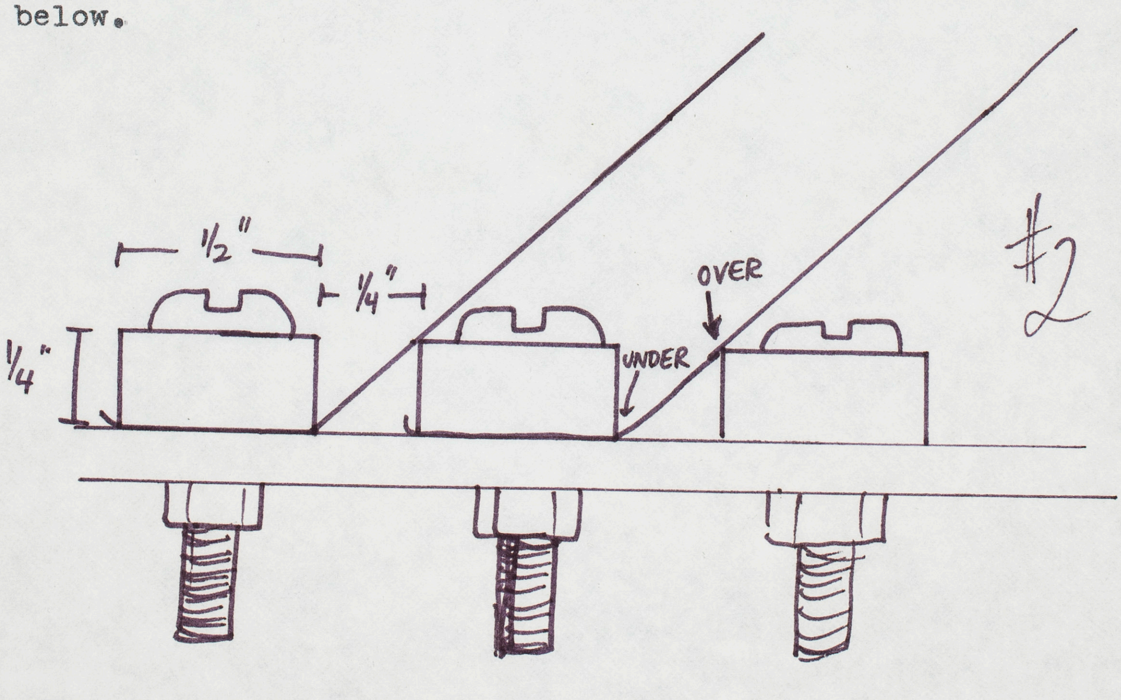
It is a similiar principal to the African Mbira or Kalimba, except for the amplification and the placement and mounting system of the spring steel reeds.

The instrument I built during the fall has 125 reeds, four octaves, with 31 steps to each octave. The reeds are tuned by shortening to raise the pitch and lengthening the reeds to lower it. I chose to utilize 31 tones to each octave (equally tempered) because of the greater flexibility of intonation and the similarity of 31 tone tuning to certain intervals in just intonation. 31 tone is an extension of mean tone temperment, which has just major thirds in the ratio of 5:3, and has slightly flat fifths. These characteristics are present in 31 tone. In addition, since the 31 steps are equal, complete modulation to any degree of the scale is possible, without disturbing intervalic consistantcy. This potential is exciting to me.

The main problem with demanding so many pitches to each octave is that for percussion and keyboard instruments, there must be a separate key or vibrating object for each of the 31 tones. Space conservation and bulk become a severe problem. One of the few existing 31 tone marimbas is the one owned by Erv Wilson, a tuning theorist in Los Angeles. The instrument has three octaves and is ten feet long. Complex figures on the instrument are difficult.

The present finger piano overcomes this problem by using very small reeds for each pitch. In a space 20"x9"x3" I have mounted 125 reeds, laid out in the standard Bosanquet Generalized Keyboard first uncovered in the twentieth century by Erv Wilson.[1] This keyboard pattern specifies where each of the 31 tones are to be placed on the keyboard. It was first advanced in the sixteenth century as an alternative to the split keys in use on many harpsichords for the extended mean tone temperments that were being used at the time. In many cases the split keyboards gave the performer the option of both a D♭ and a C# or a G# and an A♭. Bosanquet and others[2] calculated that if the mean tone temperment were to allow total modulation to any degree of the scale, 31 equal steps were necessary. The idea was droped at the time as too impractical in the face of twelve tone equal temperment, which was beginning to be used.[3] Recently, a group in the Netherlands, the Huygens-Fokker Foundation, and isolated groups around the United States, have been carrying on study of the intonation system, and in some cases writting music for the limited musicians and musical instruments available at the time. It is my hope to increase the resources available to composers and performers. A copy of Erv Wilson's design and my adaption of the original Bosanquet keyboard can be found in the Appendix.

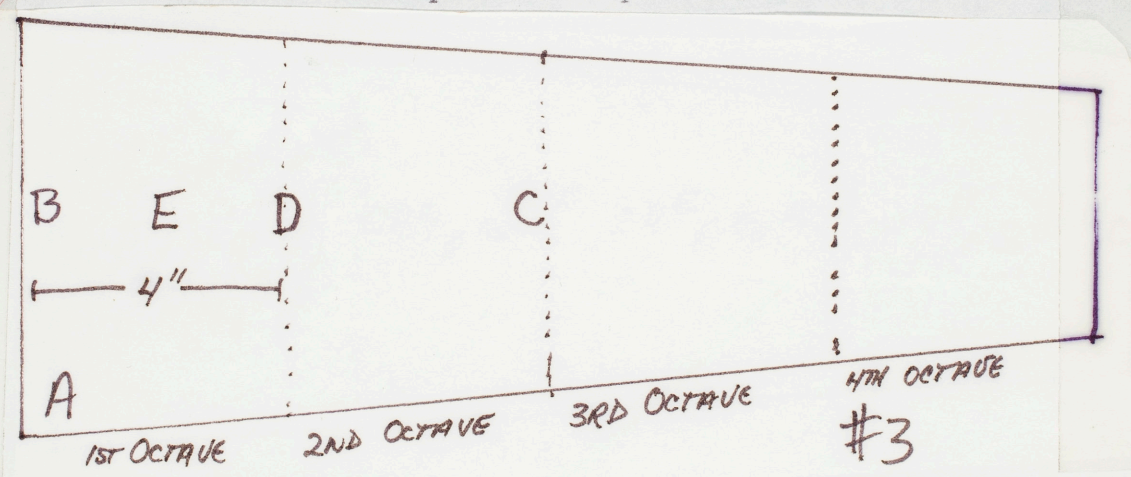
The reeds in my instrument are mounted at one end by passing over one ¼"x½" brass bar and under another, as shown in the diagram #2 below.



The reeds vibrate at their resonant frequency when plucked. The vibrations are transfered through the spruce plyboard base to the contact microphone that is glued to the underside. The electronic output of the contact microphone is fed into an amplifier and from the amplifier to a loudspeaker.

Because each reed is a different distance away from the pickup transducer, each has a different amplitude, and to a certain extent frequency response, which is heard as a change in timbre. The bass reeds are especially influenced by position. Some low notes are inaudible or have loud enharmonic partials that make them sound out of tune or weak. This problem was the reason for the research undertaken in the 205 seminar.

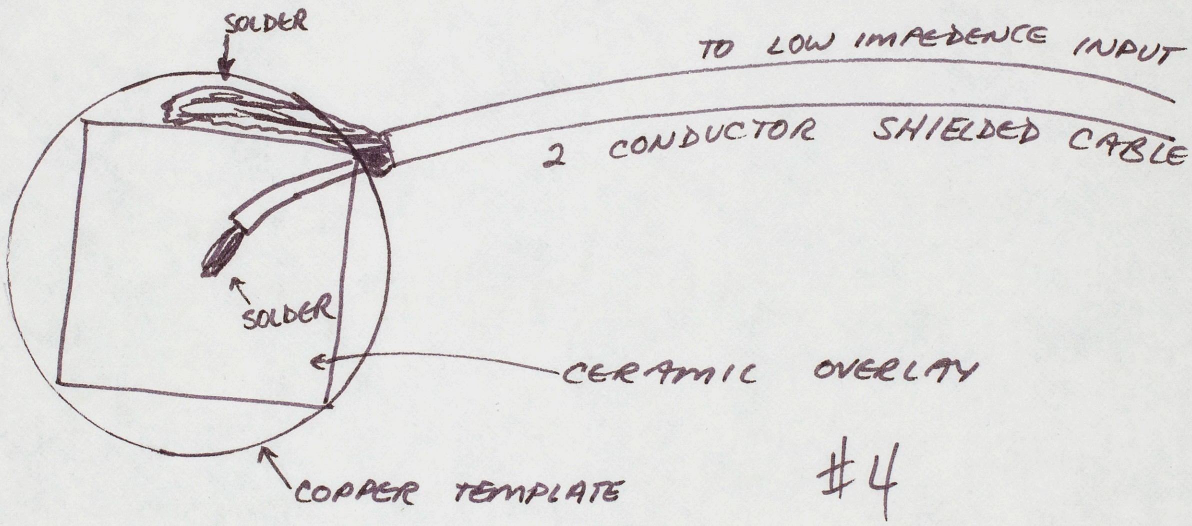
The first experiment I performed was designed to determine if microphone affected the sound quality. I glued the copper template of the ceramic contact microphone (which is described later in the paper) to several different places and played the instrument to determine the effect. The accompanying diagram #3 shows where on the instrument the microphone was placed.



The letters A through E indicate the placements. In each position the reeds directly above the microphone were transduced efficiently but those that were 2" or more away from the pickup were inferior.

The microphone position I finally settled on was letter D, dirctly under the pitch C, one octave above the lowest CC, since most of my playing was in the key of C. This pitch is also the vertical center of the instrument (see appendix diagrams.) This was an unsatisfactory compromise, so I continued my research.

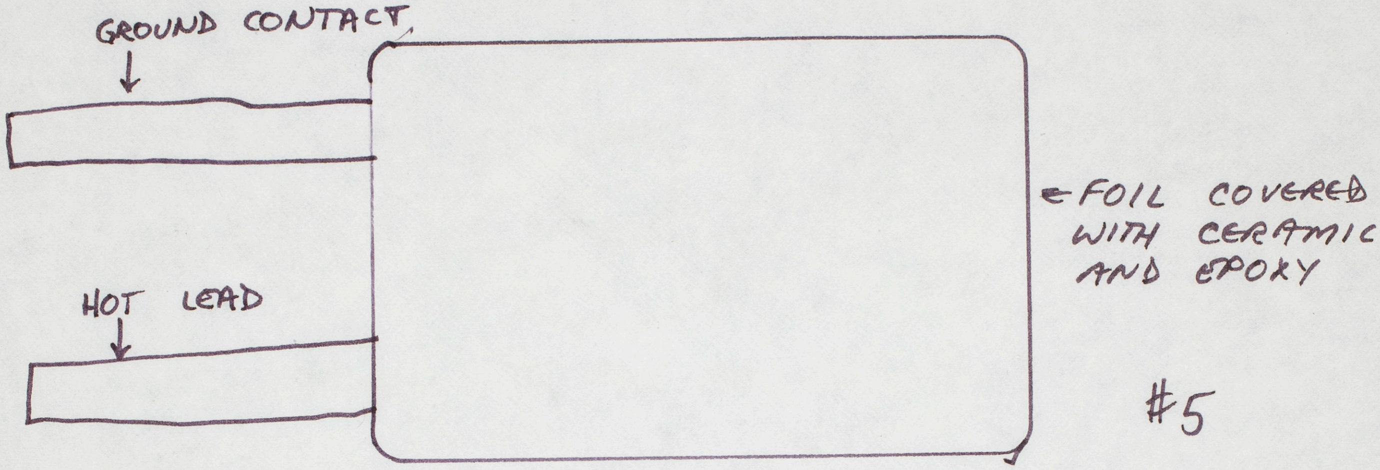
The next step was to try different brands of contact microphones to determine which had the best tone quality for this application. The first microphone I experimented with was one of the $12 copper template ceramic contact microphones that the Kiva ensemble and I purchased from Abco Sound Company in New York. The person I bought the mike from said that they were underwater throat microphones for frogmen, which he bought at a government surplus auction. I bought the last ones he had and he knows of no other source. They are essentially a ceramic transducer element that is glued to the surface of a copper template. Electrical contact is taken off the ceramic for the positive lead and the copper is for ground. The diagram #4 is a representation of this microphone.



#4

This one has since proved to be the best mechanical vibration transducer I have tried. In every contact microphone instrument I have built, this one has the most even frequency response, both on the high and low end of the audio spectrum. But because of the limited supply, I have tried other models too.

I also tested Varco ceramic phono cartridges, of which I have a virtually unlimited supply from Olson Radio at $1.49 each. I disassemble the phono cartridge and remove the ceramic element, which is a piece of metal foil with ceramic on both top and bottom, and coated with an epoxy overlay. Two foil contacts extend from one end, as shown in diagram 5.



These microphones sound good, but their low frequency response is weak. Since the supply of these microphones is excellent at the moment, I am using them a great deal in related research. The next microphone I tested was a dynamic contact microphone from Universal*. These need ferrous vibrating surfaces in order to function properly, although they work to a lesser extent on any surface that is vibrating. Mounted on the spruce soundboard of the finger piano, their sound was similiar to the ceramic phono cartridge, with a slightly improved high frequency response. However, when the microphone was placed near the vibrating tip of a spring steel reed, the frequency response was the best yet. The steel disturbed the magnetic field around the dynamic element in the contact mike and induced voltage changes in the output of the microphone. This is magnetic transduction. The sound quality caused me next to investigate magnetic transducers for the finger piano reeds.

* Universal Tape Corporation, Pembroke Park, Florida. MODEL RE.

I researched magnetic transducers in The Electronic Musical Instrument Manual, a Guide to Theory and Design by Alan Douglas. This book is concerned primarily with electronic organs, a few of which utilize magnetic transduction in their sound synthesis. Notable among these is the Hammond organ, which employs toothed metal wheels that rotate in proximity to a magnetic transducer. The diagram #6 below shows one such wheel.

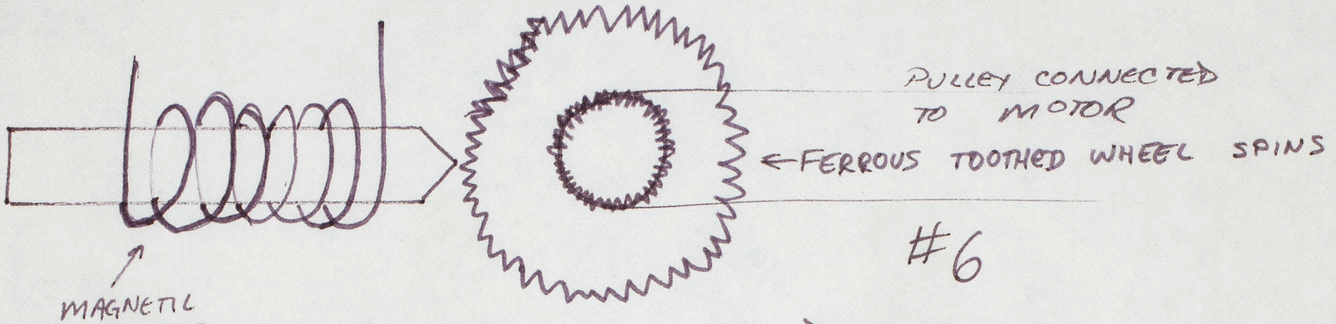
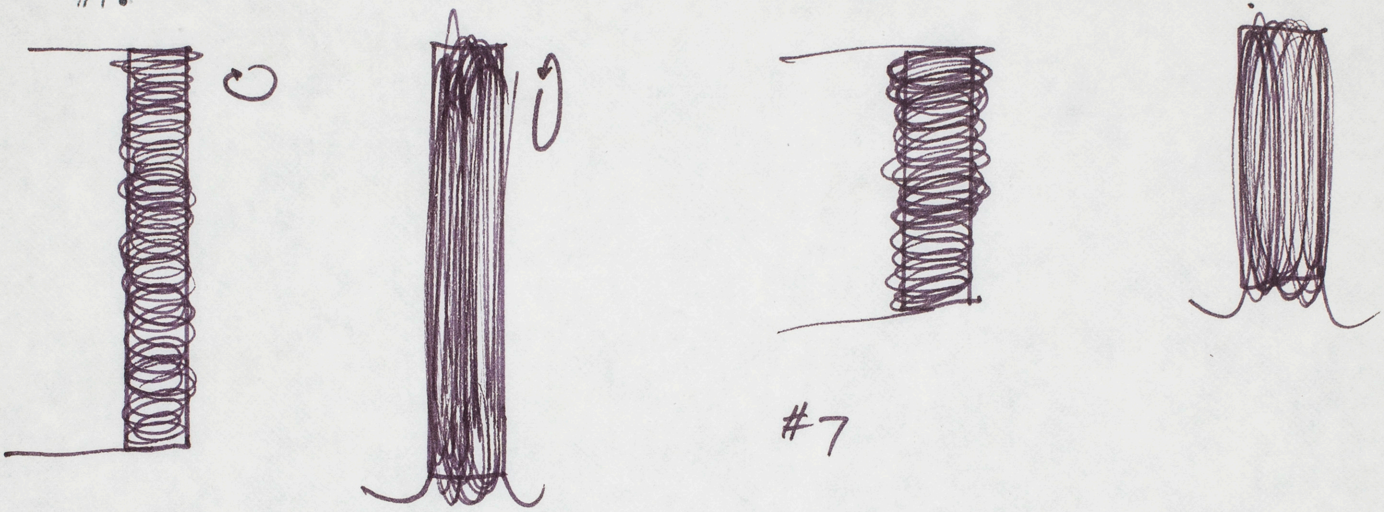


FIGURE 4.84., PAGE 102, DOUGLAS.

The teeth are alternately close to and away from the transducer. This change in the magnetic field around the transducer is in a repeated pattern that produces an electronic waveform at the output of the transducer.[4] The output may vary from square waves to spike to sine wave depending on the shape of the gear. Numerous waveform possibilities can be found in the Appendix III, which is a Xerox copy of a publication of ElectroCorp, A magnetic transducer manufacturer.[5] It is included here because of the experimental possibilities of waveform synthesis using toothed gears. This was not explored in the Hammond organs, which utilized the toothed gears to produce only sine waves, which were then added to other sine waves of integral multiple relationships to synthesize overtones.[6]

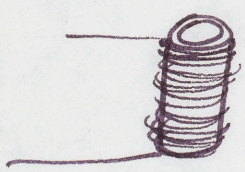
In addition to tone wheel magnetic transduction synthesis, Douglas describes wind driven free reed synthesis using transducers. Some organ manufacturers utilized brass harmonium reeds driven by an electric bellows wind source as tone generators.[7] The tips of the reeds apparently contained some ferrous material, since magnetic transduction will not take place with brass bars as the vibrating element. Later on in the book Douglas describes a sustain device for electronic organs that would utilize pitch exclusive sympatheticly vibrating spring steel reeds with magnetic transducers as pickup elements.[8]

The information on how to construct the transducer element itself was incomplete in the Douglas book. He merely showed a magnet with coils of wire around it. Electrical connection was unclear.[9] George Ritscher advised me to buy some magnets and magnet wire and experiment myself, which is what I did. The wire I purchased was #48 AWG magnet wire. This is extremely fine steel wire that is coated with a thin plastic film. I puchased several magnets including some inexpensive flexible ceramic strip magnets from Earl's Supply in Clairmont. I wound coils of magnet wire around a variety of different magnet axes, as shown in the diagram #7.



#7

I also wound some coils around a plastic tube, as in diagram #8.



#8

I then mounted each of the pickups in proximity to a spring steel reed and noted their outputs when the reed was plucked.

The highest output was transduced by the ¼" round magnets 1¼" long wound latitudinally as in diagram 7a. A weaker signeal was found with the longitudinally wrapped magnet (#7b.) Coils with no interior magnet that were wound around the plastic tubes had an even weaker output. For these pickups, the reed had to be magnetized, which was accomplished simply by touching a magnet to each of the reeds.

Later I purchased two Al-Nico rectangular bar magnets from Magnet Sales and Manufacturing Company in Santa Monica, California. They were each 1/8"x3/8"x5", stressed across the 3/8". The ceramic magnets I had been using were not alligned on any axis, but were instead composed of thousands of tiny magnets, each with its own axis of allignment within the flexible ceramic base material. The rectangular bar magnet is therefore much more efficient for the applications for which I was intending to use it. They cost $7.50 apiece as opposed to the 45¢ for the flexible magnets. I wound approximately 1500 turns of wire around one magnet on the wrong axis, which made it inefficient and bulky. In order to wind on the correct axis, I must first construct a plastic frame on which to wind the coils. I have not been able to construct a strong enough frame at this time. They are either extruded or welded ultrasonically when they are made commercially, and I do not have access to either of these processes and am therefore attempting to glue the frame together. On this I have yet to be successful. Possibly in the next month or two I may be able to finish a strong enough frame.

Although temporarily held up in my research, I have continued working on the non-electronic portion of the instrument that I am building to utilize the pickups I have been unable to build. The design for the instrument is in appendix IV.

This design utilizes 41 reeds in two rows, similiar to the two rows in a twelve tone keyboard: one for the naturals and one for the sharps or flats. The new instrument has four octaves like the previous design, but with only ten tones to each octave, instead of thirty-one. The ten notes are taken from among the thirty one, and so the scales are compatible in many cases. The notes of the scale in thirty-one tone notation are C,D,D#,E,F,F^{+},G,A,A#, and B$\flat$. This scale is called Helix Song Tuning, and was developed by Erv Wilson as an aid for teaching some of Harry Partch's ideas.[10] Helix Song is derived from Partch's tonality diamond, which is made of the first six prime number overtones of the first six prime number subharmonic partials of the tone G.[11] Helix Song utilizes the six overtones of only two tones: C and F. The overtones of C are C,D,E,F†,G,and A#, while those of F are F,G,A,B-,C, and D#. C and G appear in both series but occur once in the scale.

The pickups I intend to build will fit end to end between the two rows of reeds and transduce the vibrations of both. They are 1/8th inch wide and should fit , but the final test will have to wait until the pickups are built.

With the magnetic transducer technology I learned in my research for the finger piano, David Poyourow and I carried on some related experiments this term. After listening to Alvin Lucier describe his experiments with long strings, David and I constructed some extremely long stringed instruments with music wire, screw eyes, the cement of Mandeville Center, and some tachometer pickups from Electro Corp. The latter are used to determine the speed of rotating toothed gears in automobile engines. They work with the same principle as the Hammond organ tone wheel. Using several guages of music wire, we assembled cement electric guitars of 18, 26, and 160 feet in length; the latter was the most interesting. It was constructed outside the Music Department offices in Mandeville Center with .080" and .011" music wire. We placed the tachometer pickups, which are essentially small magnets with coils of wire wrapped around them, in proximity to the wire at one of its ends. The amplified sound of the wire when

It was excited by various means was similiar to a heartbeat, a jet plane buzzing overhead, M-1 rifles at forty miles, and Star Trek phasor sounds. For a more objective description, I will wait until the concert in May when we will be constructing two, three and four hundred foot models in the Muir dormatory quad. This is for the May 3 Atomic Cafe, and is entitled Son of Tower Music. For this concert we have four different guages of music wire and some nylon fishing line which will be transduced mechanically with a contact microphone.

In addition to functioning as input transducers, the tachometer pickups also work as output transducers, if they are in proximity to a ferrous surface, such as a length of music wire. With this, we were able to excite the wire with the output of a ten watt amplifier. George Ritcher calculated the output impedence to be in the neighborhood of 300-400 ohms. We have had no problem with the transducers being overdriven.

Using the string technology gained in my experiments with David Poyourow, I made a version of Fontana Mix by John Cage. My source material was multiphonics on an aeromembranophone. (One of my own inventions, the aeromembranophone is a membrane instrument that is excited by a high pressure air stream from the performers mouth. It is constructed from the lid of a peanut butter jar and a balloon. The lid is the frame for the balloon membrane.) The source material was processed by being transduced through six different objects, including a long string. The final mix was played on David Poyorow's concert on April 9, simultaneously with the Cage Piano Concert.

8. Keyboarding Genus 31 (ref diagram 4)

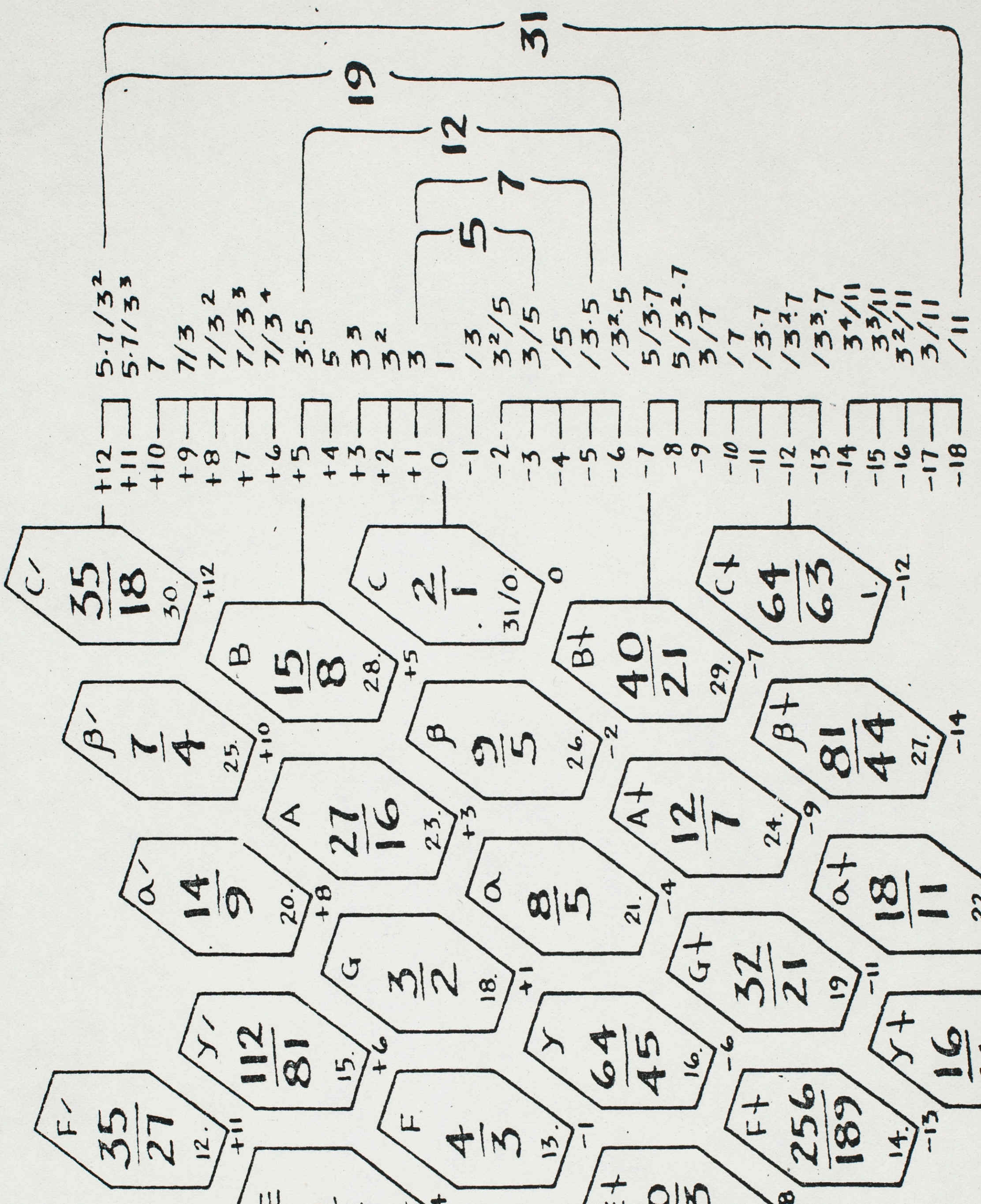
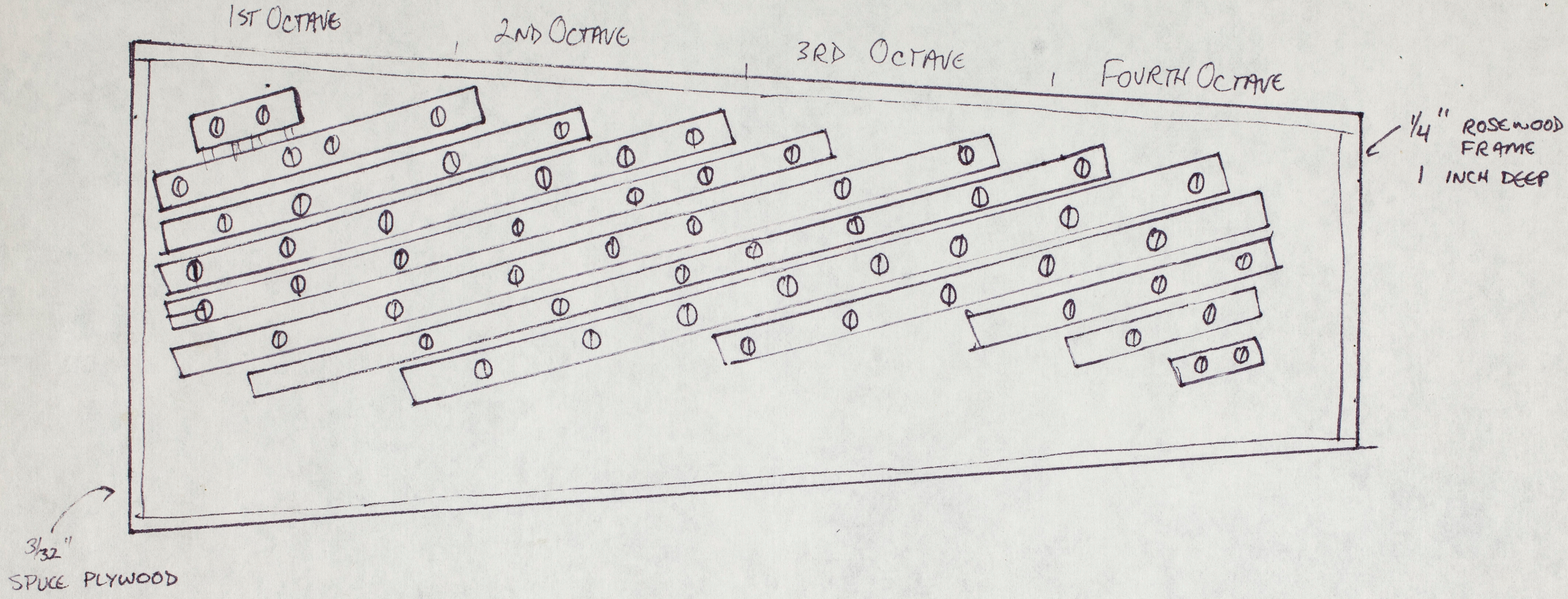


Diagram 7

APPENDIX I



DESIGN FOR 31-TONE FINGER PIANO

APPENDIX II

Appendix III

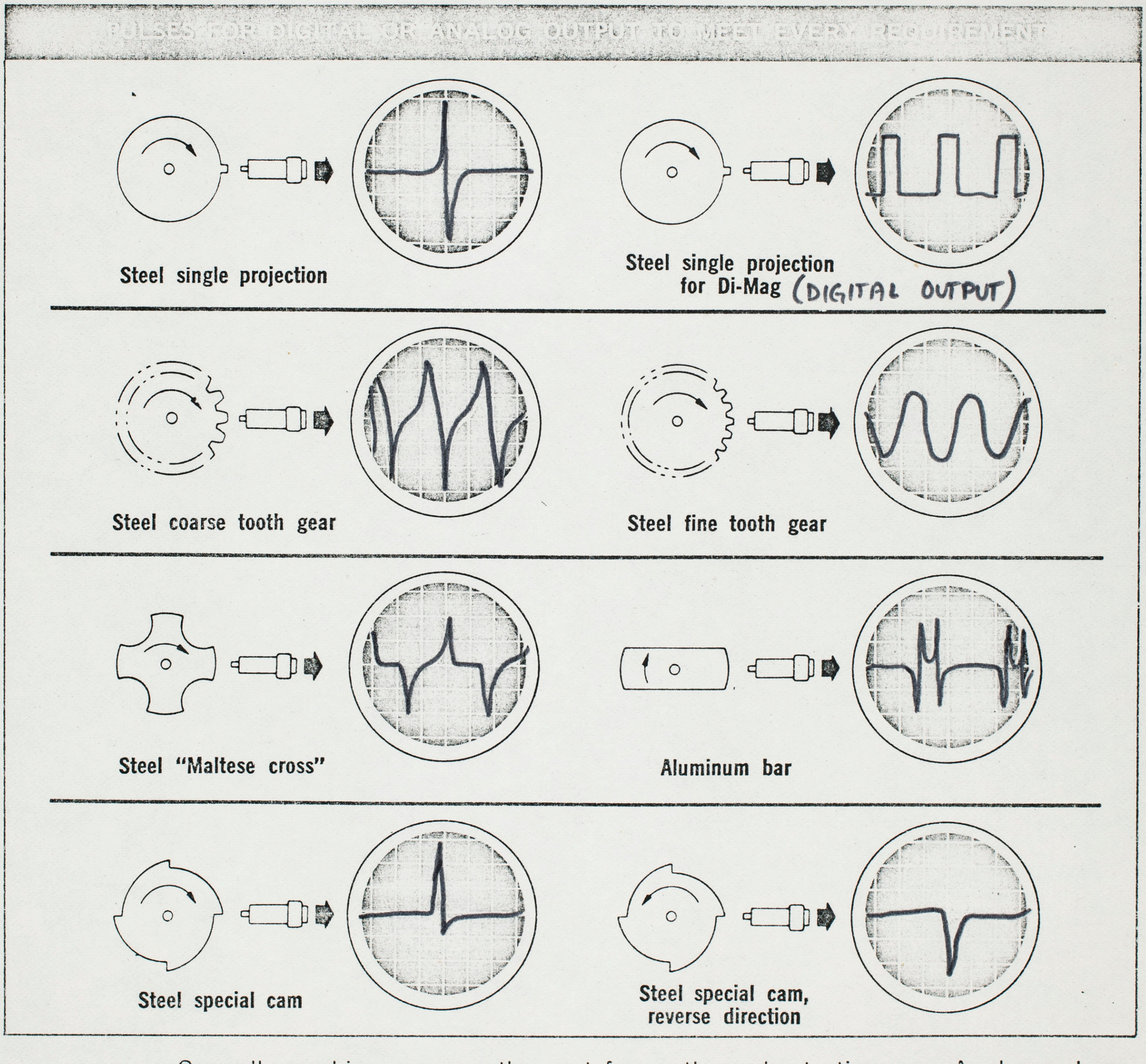
ELECTRO | Magnetic Sensors

OUTPUT VOLTAGE

Magnetic sensors are non-contact transducers whose output voltage reflects the rate of change that is taking place in their magnetic field. The output voltage that results is a measure of this change in terms of electrical energy.

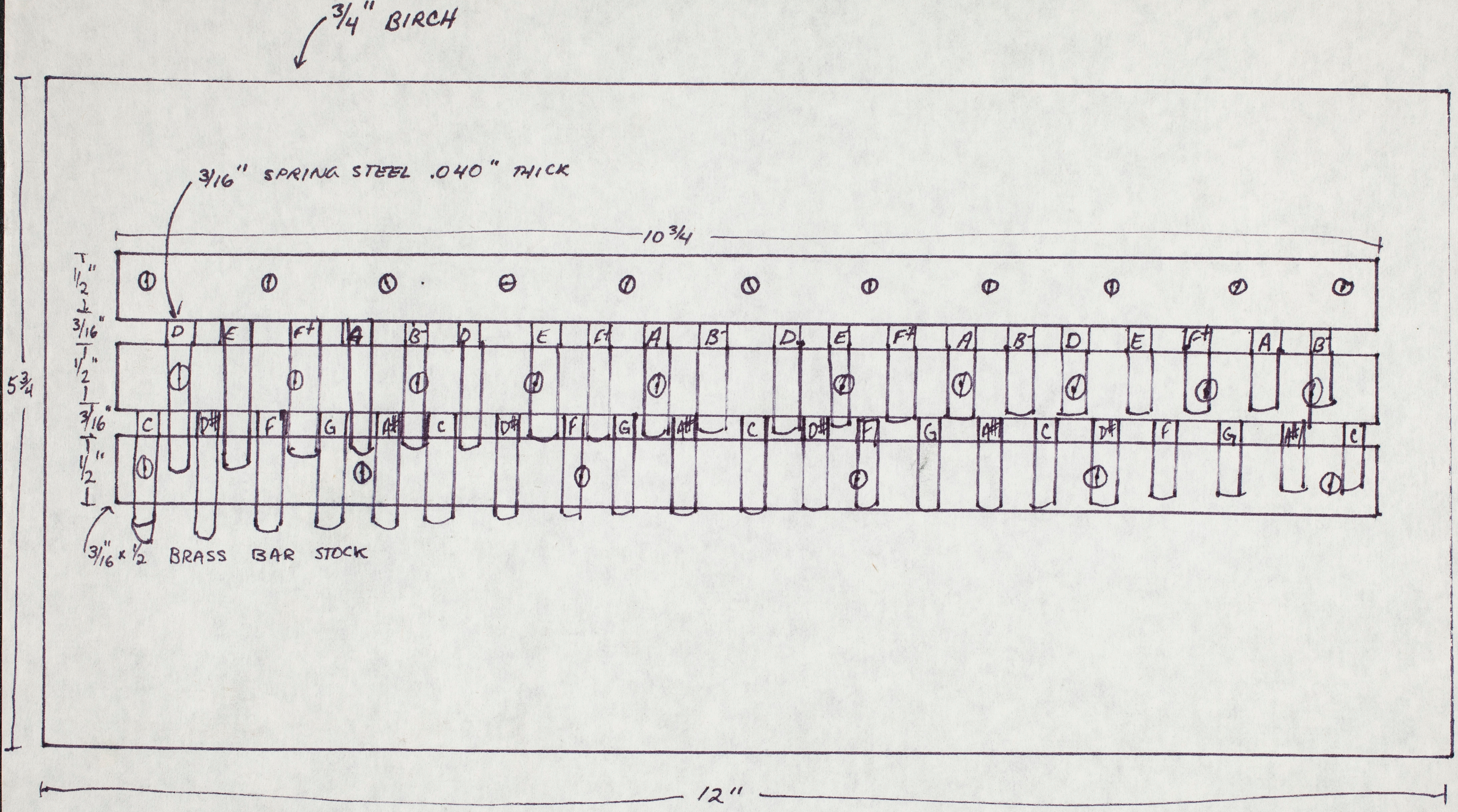
Three factors effect the output voltage, and they are:

1. The peripheral speed of the actuating mass.
2. The size and shape of the actuator, or gear teeth.
3. The pole piece clearance.



Generally speaking, gears are the most frequently used actuating mass. As shown above, actuators of varying sizes and shapes produce different waveforms. Fine tooth gears generate a near sine wave output, while coarse tooth gears produce a sharply peaked pulse. However, the same results can be obtained by utilizing wheel spokes, a vibrating surface, a moving bar, crank, or even a steel screwhead mounted on a moving surface.

APPENDIX IV



DESIGN FOR HELIXSONG FINGER PIANO

Footnotes

1. Erv Wilson,"On Linear Notations and the Bosanquet Keyboard," Xenharmonicon 3 (Spring 1975), p. 1.
2. "31-Tone Music: a Unique Experience", pamphlet distributed by the Huygens-Fokker Foundation, P.O. Box 142, Montebello, California, 90640.
3. James Murray Barbour, Tuning and Temperment, a Historical Survey, 2d ed.,(East Lansing; Michigan State College Press, 1953), page 127.
4. Alan Douglas, The Electronic Musical Instrument Manual: A Guide to Theory and Design, 5th ed., (London: Sir Isaac Pitman and Sons Ltd., 1968), p. 102.
5. Magnetic Sensors, pamphlet distributed by Electro Corporation, 1845 57th Street, Sarasota, Florida, 33580.
6. Op.Cit., Douglas, p. 281.
7. Ibid., p. 106.
8. Ibid., p. 319.
9. Ibid., p.102.
10. Erv Wilson,"Helix Song Musical Instrument", copywrite, 1976.
11. Harry Partch Genesis of a Music, 2d ed., enl.,(New York: Da Capo Press, 1974.

Chuck O'Neill

Discussion of Project

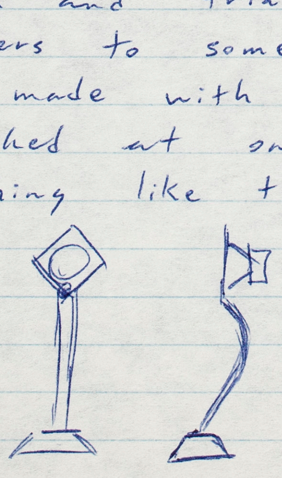
In origionally taking this course I had expected to do some form of "experimental" work with the standard equipment we all think we know ... amplifiers, speakers, microphones, synthesizers, tape recorders, etc. To me this is not so much experimenting as it is investigating. In the begining of the course it was made clear to me that the emphasis was to be on the experimental and that I was to choose a topic. It seemed to me then that for my topic to teach me the most, I should choose one I know little of ... and that this would be conducive to experimentation. So I choose underwater sound. It seemed a facinating topic, ripe for experimentation and capable of providing one with a large number of interesting sounds. The problem then became simple: to figure out experiments of intrest in a field I knew nothing of ... while problems are usually simple, answers are more often not.

I began by investigating the basic principles of underwater acoustics in hopes of finding some property or propenties which I could exploit. At this point an old friend Dr. Gerry Wick, an oceanographer at Scripts, and Dr. Vick Anderson of Scripts - Point Loma helped me very much by answering numerous questions and suggesting many readings. A good part of my formal acedemic investigation work was done at this stage and has solidified in the paper which follows this one. While my investigation continued to reveal the many interesting and unusuall properties of sound in fluids, I still had much trouble figuring practical experiments to exploit them. This has continued to daunt me even unto today, ~~finding~~ feeling that there must be a vast area of untapped potentials lying just beyond my ken; but be it intellegence or fate the door for the most part still remains closed to me. Dispite my inborn pestimism I am very thick headed... so I continued and tryed to gather material for experiments.

The two basic catagories to underwater sound work are of course reception and transmission. Thus a hydrophone and an underwater transducer (speaker) became my hearts desire. Believing that I might possibly pull away a bit of the class budget I started investigating purchases. Helle Engineering in Curney Mesa proved the most reasonable on a hydrophone with a good one running about $60. The cheapest underwater speaker that is truely usable is University Sound's U.W. 30 which can be purchased locally through Shanks and Wright Inc. for $92.40. My hopes were high... I planned to transmit signals through U.C.S.D.'s swiming pool for my first experiment. Well as they say "not all things come to pass". I recieved two $5 hydrophones from Edmund Scientific and two plastic coned waterproof outdoor speakers (4"). With my tail tucked between my legs I proceeded to borrow an aquarium from a friend of mine and to set myself up in B-108.

By now I had several major ideas to try. ~~This~~ The first was to broadcast a signal underwater on a carrier wave whose wavelength was comparible to the displacement of the surface; the signal would be bounced off the surface and demodulated with the hopes that it would exhibt the dobbler effect. George did not particularly like this idea since it would be mainly his personal gear which would be borrowed to do this project. My second idea was to set two speakers in an aquarium 180° out of phase such total cancelation occurs at a central node into which a hydrophone is placed, then introduce bubbles into the field of one of the speakers. Bubbles you see have a resonate frequency and will attenuate the passing of this frequency quite well. This was found when certain sonar units proved inoperative because they operated at the resonate frequency of the wake bubbles produced by the ship they were on. My idea was that since a frequency was attenuated on one side of the field there would no ~~lager~~ longer be phase cancellation of that frequency, thus it would pass. This then would be a very unusuall type of band-pass filter whose frequency was controlled by the size of the bubbles introduced into the system. I talked to Dr. Nick Anderson who assured me my idea was sound and so I proceeded with it.

My first problem was with mounting the speakers. I had origionally intended to use a matched set of sterio cabnet speakers and mount them to the exterior walls roughly like this: , but since the small plastic coned speakers were given to me I decided to use them. After much search and trial and error I affixed the speakers to some aluminum L-brackets I had made with screw mounted suction cups attached at one end. The result was something like this:



Now when I put these in the same water as the hydrophone the only signal that was effectively passed ~~was~~

a very complex hum. It took me nearly two and a half weeks to get rid of that hum. Durring this time most of my experiments were confined to sticking sound producing objects into the water and listening with the hydrephone. Most objects tend to sound normal unless effected by the impedance of water (excepting of course those which are destroyed or disabled by being in water). The most interesting sounds that were produced durring this period were bubble sounds. I had gathered togather several simple objects for bubble production in order to do my experiment, these consisted of 15 feet of polyflo tubing, a small air pump (I also used my breath as an air source and found some very nice sounds that way), and a group of three different size air stones. Air stones are porous stones through which air passes in order to more effectively aerate water in aquariums. Different sizes (actually different grades of porousness) have different average bubble sizes and thus different noise bands. The starting and stopping of the air flow as well as the erratic actions of the bubbles themselves produce an unusally complex noise spectra, location in relationship to the hydrophon effects phase and the bubbles exploding upon the surface produce a metallic ring which lasts throughout the event.

These experiments though made me worry

over the sound of the bubbles masking the frequencyies passed by my system. I was also getting a lot of heavy doubt vibes from various people as to whether or not I would be able to achieve phase cancellation. Some people thought that I would only get phase cancellation upon a certain frequency, whose wave-length was equal to the length of the tank, and its overtones. Again ~~dimayed~~ dismayed I continued work on the hum untill one day I accedently grounded the speaker cables to the power supply and lo and behold there was silence (or at least the hum got a damn sight quieter). At that point with an aligator clip, some wire and a soldering iron a grounding method was materilized. After some initial testing of the speakers in water I began to search for the cancellation node. Tho I tried the complete reproducible frequency range (under both phase conditions), used white noise as well as sine tones, and searched diligently each time I found no cancellation node. I did find that the white noise passed by the system had an interesting garbled quality. This I exploited by passing various frequency bands and by movement of the hydrophone with respect to the speaker, tapeing the results. My next question was why the garble when I had come to expect fairly accurate reproduction. I tried speach though the system finding it also very distorted so I pulled the speakers placeing them in air to check their reproduction ... they had became water legged and practically useless. It is also my present belief that with the equipment avaiable I was not able to generate sufficient power to create a nodal situation. By this time the end of the quarter was approaching and my friend asked me for the aquarium back..., while I am sure he would have let me keep it untill the class completed itself I decided I really had no more formal use for it so I completed a few short experiments (on very distorted sounds), photographed the setup and plan to return it to him today (March 25th) or tomorrow,

Although this may sound like an ending actually another line of research had been slowly becoming dominate. While the origional idea of using fluids as a sound processor was loosing its foothold as I studied its properties, I was becoming increasingly interested in marine animal noises. My interest was origionally inspired when Dr. Gerry Wick loaned me a tape of his and again when I ran across a small disk recording archive at Scripts Library. I checked out two books:

<u>Marine Bio-Acoustics</u>
ed W.N. Tavolga, Pergamon Press (1964)
and
<u>Animal Sounds and Communication</u>
ed W.E. Lanyon and W.N. Tavolga
American Institute of Biological Sciences (1960)

and began to search out who was doing recording work in the field. I decided to gather the recordings of others rather than to make my own since I was already engaged in one research project but would have considered the field work to have been interesting and practical (with Scripts and Sea World being so close) otherwise. my most valuable source finds were:

Dr. John S. Pearse and Dr. Kennth S. Norris
of U.C. Santa Cruz

Dr. Howard Winn of the University
of Rode Island

Dr. Lou Herman of the University
of Hawaii (Kewalo Basin Lab)

Mrs. Samy Holmes of Public Affairs
Naval Undersea Center

Dr. Tomas C. Poulter of the Stanford
Research Institute

Dr. Norris and Samy Holmes were particularly helpfull in my contacks with the others and in getting me recordings, much thanks is due them. Dr. Poulter is perhaps the best single source I found, he is now preparing an anthology of underwater sounds for the Library of Congress and the National Archives. A priced listing of tapes avaible to date follows this. Having at present collected some six hours of recordings I have now called a quit to my my own collection due to lack of funds.

Reviewing what I had done to date and that I had two weeks left untill the project became due I proceeded to document and utilize my work in several ways. First I wrote the article which follows this, It is written in non-technical language and should familiarize the reader with the major aspects of underwater acoustics. Secendly I decided to photograph the equipment with which I was working. Third I wrote this summary of my efforts and fourth I began work on a tape piece utilizing both sounds I had collected and produced, thus hopeing to show some musical application of the type work I was doing.

If it is of any help the following list is provided for the services of anyone proceeding further

Bidwell Sales (~~State~~ distributor for Universal Sound)
(213) 770-0300 (talk to George Perry)
Helle Engineering 278-7050
International Transducer Corp. (I.T.C.)
(805) 967-0491 (Ted Madison)
Channel Industries
(805) 967-0171 (Bob Carlson)
Shanks an Wright Inc.
239-0176 (Ray Brubacker)
U.C. Santa Cruz (tie line) 121-2 + 129 + 0111
+ indicates wait for dail tone
Dr. John S. Pearse ex 4026
Dr. Kenneth S. Norris ex 2003
Public Affairs, Naval Undersea Center
225-7225
Dr. Tomas Poulter, Stanford Research Institute
333 Ravenswood Ave, Menlo Park, Ca. 94025
Dr. Victor Anderson 452-2304

PRICE LIST OF TAPES AVAILABLE TO DATE

MARING MAMMALS

P-2	Steller Sea Lion	$ 4.00
P-3	California Sea Lion	$ 4.00
P-3, C-42	California Sea Lion as Compared with Bottle Nosed Dolphin	$ 4.00
P-3	California Sea Lion Dialects	$ 4.00
P-3	Whiskers Echolocating	$ 4.00
P-4	Australian or White Capped Sea Lion (On 0 only)	
P-9	Guadalupe Fur Seal	$ 4.00
P-10	South African Fur Seal	$ 4.00
P-12	Northern Fur Seal	$ 4.00
P-13	Walrus	$ 4.00
P-14	Harbor Seal	$ 4.00
P-15	Ringed Seal (On 0 only)	
P-18	Gray Seal, P-19 Ribbon Seal, P-20 Harp Seal	$ 4.00
P-19	Baby Ribbon Seal	$ 4.00
P-21	Bearded Seal	$ 4.00
P-21, C-71	Bearded Seal and Bowhead Whale	$ 4.00
P-27	Leopard Seal	$ 4.00
P-27	Leopard Seal side one and Adelie Penguins side two	$ 4.00
P-28	Weddell Seal	$ 4.00
P-31	Northern Elephant Seal	$ 4.00
P-31	Northern Elephant Seal (5" reel recorded on both sides)	$ 6.00
S-3, C-2, C-6	Manatee, Inia, and Nonwhale	$ 4.00

C-1	Ganges Dolphin (Susa)	$ 4.00
C-2	Amazon Dolphin (Inia)	$ 4.00
C-5	Beluga or White Whale	$ 4.00
C-6	Nonwhale	$ 4.00
C-7	Harbor Porpoise	(On 0 only)
C-12	Dall Porpoise	(On 0 only)
C-18	White Beaked Dolphin	(On 0 only)
C-19	Atlantic White Sided Dolphin	(On 0 only)
C-20	Pacific White Sided Dolphin	$ 4.00
C-24	Common or Saddle-back Dolphin	$ 4.00
C-28	Blue, or Blue-White, or Euphrosyne Dolphin	$ 4.00
C-29	Spotted Dolphin	$ 4.00
C-42	Bottle Nosed Dolphin	$ 4.00
C-44	Grompus, or Gray Grompus, or Rosso Dolphin (On 0 only)	
C-45	Common or Pilot Whale	$ 4.00
C-46	Short Finned Pilot Whale or Short Finned Blackfish	(On 0 only)
C-47	North Pacific Pilot Whale or North Pacific Blackfish	(On 0 only)
C-49	Slender Blackfish	(On 0 only)
C-50	False Killer Whale	$ 4.00
C-51	Killer Whale	$ 4.00
C-51	Two Baby Killer Whales 1 male and 1 female	$ 4.00
C-68	Sperm Whale	(On 0 only)
C-71	North Atlantic Black Right Whale	(On 0 only)
C-71	Bowhead Whale	$ 4.00

C-73	Gray Whale		$ 4.00
C-73	Baby Gray Whale		$ 4.00
C-75	Sei Whale	(On 0 only)	
C-77	Fin or Finback Whale	(On 0 only)	
C-78	Blue Whale	(On 0 only)	
C-79	Humpback Whale		$ 4.00

RELATED AND UNRELATED TAPES

A	Under Arctic Ice	$ 4.00
B	Under Arctic Ice (5 hrs playing time 15/16 IPS	$ 8.00
C	Whistlers from Outer Space	$ 4.00
D	Ice Noise	
E	Unusual Sounds Made by Marine Mammals Both Sides P-13, 21, 27, 28, 31. C-7, 51, 71	$ 6.00
F	Gibbon Monkey	$ 4.00
G	Many Species, Can You Identify Them?	$ 4.00
H	Siamong Monkey	$ 4.00
I	A Walk in the Cold and Under Arctic Ice	$ 4.00
J	Some Unusual Recordings	$ 4.00
K	Whistling Language (3" reel)	$ 1.00
L	1965 Tape (7" reel)	$ 6.00
M	1966 Tape (7" reel)	$ 6.00
N1 thru N18	Veterinary Working Conference, U.C. Davis Feb. 8-9, 1969 (9 7" reels)	$50.00

O1, O2	42 Species of Marine Mammals on 7" reels both sides (about 3 hours)	$25.00
P	Asian River Otter, Marmot, Siamong, Gibbon	$ 4.00
Q1 thru Q16	Tape Recordings of 5th Annual Conference on Biological Sonar and Diving Mammals	$15.00
R	Bradycardia in Steller Sea Lion Pups	$ 4.00
S	Recording of the 1959 Volcanic Eruption in Hawaii	$ 4.00
T	Unidentified Animal Sounds at Cape Crozier Antarctic	$ 4.00
U	Unidentified Species of Baleen Whale in Antarctica	$ 4.00
V	42 Species of Marine Mammals on 5" reels Recorded on both sides - Reel 1 of 2	$10.00
	Reel 2 of 2	$10.00
W	Unidentified Animal Sounds in the Arctic	$ 4.00
X	Marine Mammal Vocalizations	$ 4.00
Y	Leopard Seals and Unidentified Sounds	$ 4.00
Z	Unidentified Sounds in Scammons Lagoon and the Pacific	$ 4.00

PROCEEDINGS FOR THE ANNUAL CONFERENCES ON BIOLOGICAL SONAR AND DIVING MAMMALS AND PROGRESS REPORTS PRICE LIST

There were no proceedings prepared for the First Conference in 1964.

Progress Report, 1965, Echo Ranging Signals of Sea Lions and Seals, 85 pages, $1.00.

The Proceedings for the Second Annual Conference held in 1965 are available in limited numbers at $3.00 per copy so long as the supply lasts (100 pages).

The Proceedings for the Third Annual Conference held in 1966 are available in limited numbers at $5.00 per copy so long as the supply lasts (215 pages).

The Proceedings for the Fourth Annual Conference was published in book form under the title, The Behavior and Physiology of Pinnipeds, by Appleton-Century-Crofts, New York, 1968 and is available from them at $12.00 per copy (411 pages).

There were no proceedings published for the Fifth Annual Conference but a complete set of magnetic tape recordings are available at $15.00 per set.

The Proceedings of the Sixth Annual Conference is now available for distribution at $12.50 per copy, 1969.

The Proceedings of the Seventh Annual Conference held in 1970 are available at $12.50 per set.

The Proceedings of the Eighth Annual Conference held in 1971 are available at $15.00 per copy.

The Proceedings of the Ninth Annual Conference held in 1972 are available at $15.00 per copy.

No proceedings was published for the Tenth Annual Conference held in 1973.

These items are available through my office at prices listed above plus postage. Check should be made payable to Thomas C. Poulter.

Thos. C. Poulter

Thomas C. Poulter
Senior Scientific Advisor
Stanford Research Institute

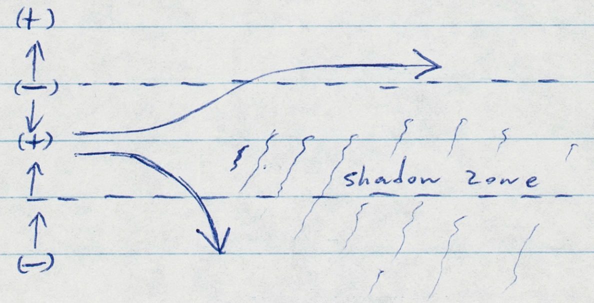
Basic Underwater Acoustics

Let us begin by stateing that acoustic energy (sound) is transmitted more efficently in water than any other form of energy (for example heat, light or radio waves). Due to this fact the reference standard for volume when useing logarithic measurements (dB) is quite often different than that used for air. The standard for air, 0.0002 daynes per centimenter squared, is sometimes used but more often one dyne per centimeter squared (one microbar) is the excepted standard, upon occation the M.KS system of 10^{-6} newtons per meter squared ($1 \mu N/m^2$) is also used. In general I will not deal this specificly with the relations between math and physics but at times when thought nessary some points may be clarified so that the reader will have less problems if he continues his research elsewhere.

The speed of sound in water increases with an increase in depth, tempature or salinity. The equations involving these parameters need not be gone into but the reader is refered to the graphs contained at the end of this ~~articlle~~ article for a general perspective. It is primarily do to these factors that most of the transmission anomalies in water occure. The standard speed of sound in open water is usually given as 1500 meters per second (4,900 feet per second) and is for surface water at 13° centigrade at a salanity level of 35 parts per thousand. When compared to air at 331.6 meters per second it can be seen that sound travels roughly five times faster in water. Water also has a standard acoustic impedance of 1026.4 kilograms per meter cubed or about sixty times that of air.

In regards to transmission it may be seen that the higher the frequency the greater the signal loss when compared to distance traveled. This is primarily due to thermal absorbtion which has practically no effect upon frequencies lower than 2,000 cps but does increase proportionetely with a frequency increase above that point. Lower frequencies, obviously, tend to transmit much better over longer distances, this becomes particularly observable at distances of 100 to 3,000 meters. Frequencies below 500 kc. are attenuated much more in salt water than in fresh water. Two additional comments should be made. First both absorbtion and dispersion tend to occure at rates larger than math predicts. Secondly, even so, the transmission of sound underwater is an extreamly efficent process as compared to air and thus sounds may often travel seemingly incredible distances.

The main reason for deviation of a sound wave in water from propogating in straight lines is the effect of ~~tempature~~ temperature. Sound tends to bend in the direction of lower velocity, thus a refraction of 1% for a temperature drop of 10°F is the major source of deviation far outweighing all others. Roughly at a distance of 6,000 meters an unhampered wave will have descended 1,000 meters. Such refraction will continue untill the wave strikes bottom at which point its reflection will travel the inverse of its path. An isothermal layer is a region in which temperature increases with depth for a distance before resuming it natural course. Thus in an isothermal layer the speed of sound increases. Again, sence sound is bent towards a lower velocity region this means that a sound occuring in an isothermal region tends to be refracted upwards. Strong boundries of isothermal layers may cause "shadow zones" or areas where very little sound is propogated. It is hoped that the following diagram may make this clearer



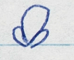
(+) positive velocity gradiant, warmer water, faster sound
(−) negative velocity gradient, colder water, slower sound
↑ indicates direction of change
--- isothermal boundry

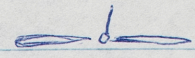
In general, there is a heavy transmission loss within particular zones underwater and these are caused by refraction when passing from one velocity structure to another (thermocline). Roughly speaking one encounters two isothermal layers in normal water conditions. The first is usually between 50 and 200 feet deep and the second is at a depth of approximately 2,000 feet. After 2,000 feet (800 meters) tempature tends to stabilize and increasing pressure brings an increase to the speed of sound. whenever a minimum velocity occures (ie. an isothermal layer) a portion of any sound traveling through this region will tend to be "trapped" propogating parallel to the temperature layer (see top arrow of previous diagram). This occurs again because sound tends to bend towards a lower velocity region thus under such condition it has "no place it would prefer to be." Such an effect is termed a Sound Channel and has several interesting properties. First sound here experiences the minimum possible absorbtion and it is not uncommon, espacilly of lower frequencies, to travel several thousand miles. Thus these regions have been used for very long transmission. Secondly all of the signal does not travel at the same speed, instead some portion tends to spend more time in slightly faster regions (due to wavelength, frequency and angle of entry). Sounds recieved in this region therefore tend to crescendo as the earlier arrivals gradually give way to the main body of sound which after having passed is followed adbruptly by silence.

Shallow water transmission depends very heavily upon surface and bottom conditions. Reflections from either or both will tend to reinforce or cancle the direct wave according to the general laws of acoustics. This affect becomes more pronounced when the wavelength is large when compared to the surface variation. Bottom conditions such as silt, mud or stone all posses unique absorbtion and reflection characterics. Surface condition is largely determined by the prevailing weather.

Noise incured in the recording of any underwater source fits broadly into two catagories (1) internal and (2) exturnal. Internal or self generated noise is produced by the recording system itself and is generally higher in underwater systems than their air counterparts due to high impedance hydrophones which are most often coupled with very long cables. Wind, rain, ice, and the sound of waves produce a constant backround of noise in the ocean. Other soures of exturnal or ambient noise are man made sounds such as boats or propellers which may be heard for very long distances and marine animal noises which become very prominent at certain locations as for example a shrimp bed.(1)

Transducers for underwater applications must produce large forces at small displacements in order to match impedance with the medium. They must also be capable of operating at widely varying pressure levels, usually up to 15,000 psi. Transducers are classified as (1) projectors (2) recievers and (3) recipitical, capable of doing both. Detection of sound underwater is done by pressure sensitivity rather than particle velocity as is done in air, this is due largely to the impedance of the medium.

Only two basic beam patterns exist in underwater work. The first is omni-directional and the second is directional. In the directional pattern the shape is approximately  with reception-transmission at the minor lobes being 15 dB lower than the main lobe. If the minor lobes are reduced they tend to move forward and the main lobe gets wider forming a pattern somewhat like this: . Very narrow beam transducers are termed search light transducers and are generally recipitical.

The first type of transducer I would like to discuss is the MAGNETOSTRITIVE. It consist of a metalic core (usually nickle) placed in a magnetic field. Fluctuations in the field will cause the metal to contract and expand or conversly fluctuations in pressure upon the metal will create corresponding changes in the magnetic field. This type is alway therefore recipitical. It may be constructed in such a way as to produce any desired pattern, the most common being the "ring stack transducer" which is omni-directional along its horizonal axis but directional along its vertical axis (ie. ). Another common approach is the PIEZOELECTRIC. Here two electrodes are applied to an electro-lydic crystal such as quartz, rochelle-salt, ammonium dihydrogen phosphate or lithum sulfate. Changes in the crystal size correspond to changes in the E.M.F. Such transducers are of very high impedance and are almost always omni-directional. ELECTROSTRICTIVE forms are made by connecting electrodes to such metals as barium titanate or lead zirconate titanate. these usually take the form of disks or cyclinders and have an impedance greater than magnetostritive but less than piezoelectric. This form is recipitical and may be used at great depths and in high power applications.

Other common transducers are: EXPLOSIVES such as T.N.T. or hydrogen-oxygen. These are commonly used for topographic and geographic research, they produce frequencies in a band of 10 to 100 Hz and may transmit thousands of miles. PINGGERS & sonar boomers are electrical condensers producing between 1,000 and 13,000 watts. These are designed to operate at their resonate frequency which is usually 24 Kc though frequencies as low as 200 cps are used, lower frequencies have a wider beam and travel further. Durration of the pulse is approximately 0.5 milliseconds and a hydrophone reception of the first echo triggers the next ping. The UNDERWATER SPARK is an electrode pair producing very short P-waves, succesive pulses very considerably. The HYDRO-DYNAMIC or FESSENDEN OSCILLATOR was the first source to be invented (1912), origionaly being designed for morse code on ships. It is good for producing continuous high level pure tones at low frequencies (500-1,000 cps). I would like to mention one other device before going on even though it is not technically a transducer. That is the underwater ACOUSTIC LENS. Fluid lenses may be constructed in sphere or tube shape allowing for control over the focus and dispersion of sound.(2)

The bandwidth of a transducer in marine applications is defined as where the frequency curve when compared to power is greater than 0.01 of the first resonate frequency of the device. In more elaborate systems filters are coupled to limit devices to this range.

The older projectors tend to be magno-restrictive but these lack high power handleing abilities, performance varies with depth and efficiency is low, 30-35%. Piezoelectric materials have largely replaced them having a 50 to 70% efficiency, no varience with depth and no power problems. Please note that the efficiency of an underwater projector must be much greater than that of an air type due to the power requirements of impedance matching with the medium. Liquid filled, pressure equalizing projectors have been developed and have the advantage of requiring much less carriage assembly. As directivity increases and frequency range decreases (reaching optimum at the first resonate mode of the device) the projector becomes more efficient.

Hydrophones are quite often recipitical transducers. Generally the first resonate mode is placed outside of the intended reception band so that a flat response is obtained. As bandwidth decreases and directivity increases, the efficiency of the hydrophone will increase. For example, a reduction of bandwidth by 20° will give a 20 dB rise in signal. Note; receptivity may be confined either horizonatally or virtically or both, but that the overall pattern of any device may be affected by thermal gradients.

Notes

(1) Most recordings of marine animals tend to have trouble determining their source. The most common approaches to this problem are (1) capativity (2) small submarines having recording gear and lights and (3) Lighting an underwater area which is monitored by both T.V. and sound recording equipment.

(2) For more information see: Boyles, C.A.

"Theory of Focusing Plane Waves by Spherical Liquid Lenses" Journal of the Acoustical Society of America, 38: 393-405 (1965)

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). It is possible to derive equa-
: changes in transmission loss
l factors for certain idealized
ariable in their characteristics
tributions of the above factors
nsmission anomaly. The total
by

$r + A$ (15.7)

ons on our ability to transmit
: noted that sound is immensely
agnetic waves as a means for
· example, the lowest frequency
re attenuated by one decibel in
ncies are attenuated even more
ring of a beam of light passing
um is for all practical purposes
the most penetrating γ-rays are
ath. In comparison with other
r transmitting energy through
t suffers only when contrasted
radio and light waves through

important phenomenon that
nd straight line propagation of
n resulting from variations in
the principal factors influenc-
mperature, salinity, and depth.
ne mouths of large rivers, where
the sea, in the vicinity of large
d in water close to the surface
aximum effect. Variations in
ed pressure, are quite regular,
ase in depth. The resulting
dest for moderate depths and
variation in temperature is
city at a depth of 100 m as
ut 0.1 per cent. By contrast,
in temperature are normally
m, especially near the surface
, time of the day, cloudiness,

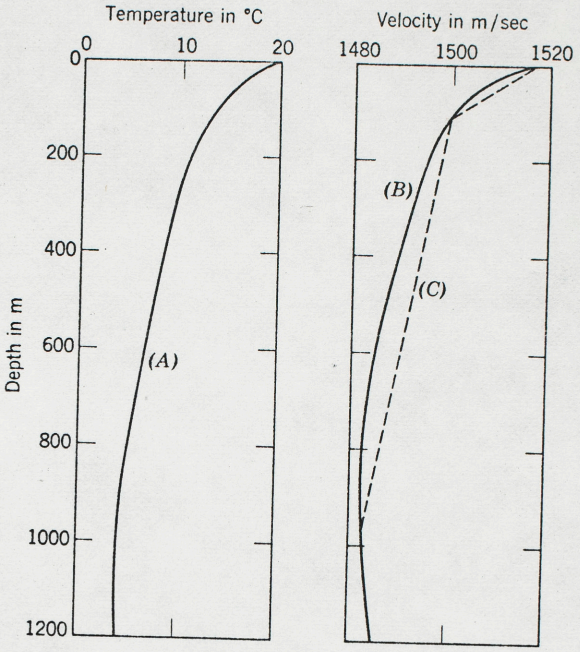

Fig. 15.2. (*A*) Typical thermocline. (*B*) Velocity of sound vs. depth curve. (*C*) Simplified velocity of sound vs. depth curve.

wind velocity, and sea state, all have their effect on the temperature gradient. Differentials of more than 5°C are common in the first 100 m, and since the change in velocity with temperature is about 0.2 per cent per °C for a temperature in the vicinity of 15°C, this effect makes the prediction of the exact path of a sound beam quite difficult. The influence of the resulting refraction on the transmission of sound waves is in many respects similar to that of heated air on the transmission of light rays.

Below 100 m in deep water the temperature in general decreases more regularly until at depths ranging from 500 to 1500 m it reaches a temperature near 4°C and then decreases very slowly until the bottom is reached. Curve *A* of Fig. 15.2 is a typical *thermocline* illustrating the manner in which the temperature of sea water varies with depth, and curve *B* of this figure shows the corresponding variation in velocity. It is to be noted that when the near constant temperature water is reached at a depth of 1000 m, the velocity then increases in accordance with the pressure effect rate of 0.017 m/sec per meter.

The path of a ray of sound through a medium in which the velocity varies with depth can be calculated by the application of Snell's law. In frequency in kilohertz, and f_T is the temperature-dependent relaxation frequency given by

$$f_T = 21.9 \times 10^{6-1,520/(T+273)} \qquad \text{kHz}$$

where T is the temperature in degrees centigrade. At low frequencies ($f \ll f_T$), the absorption is dominated by the first term and becomes

$$\alpha = \frac{AS}{f_T} f^2$$

and is thus proportional to the square of the frequency. At high fre-

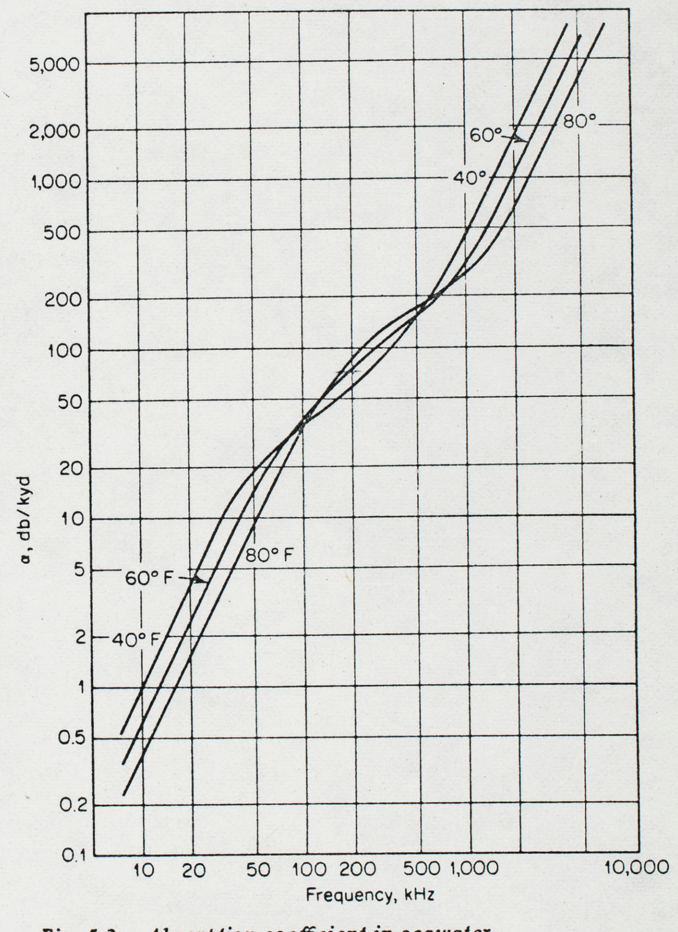

Fig. 5.3. Absorption coefficient in seawater of salinity 35 ppt as a function of frequency at three temperatures.

time, in which the
of the sound wave.
weight of the total
than the principal
nard, Combs, and
uent of seawater.
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ity, should yield a
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nents made at sea
result was
and B are constants
spectively, f is the

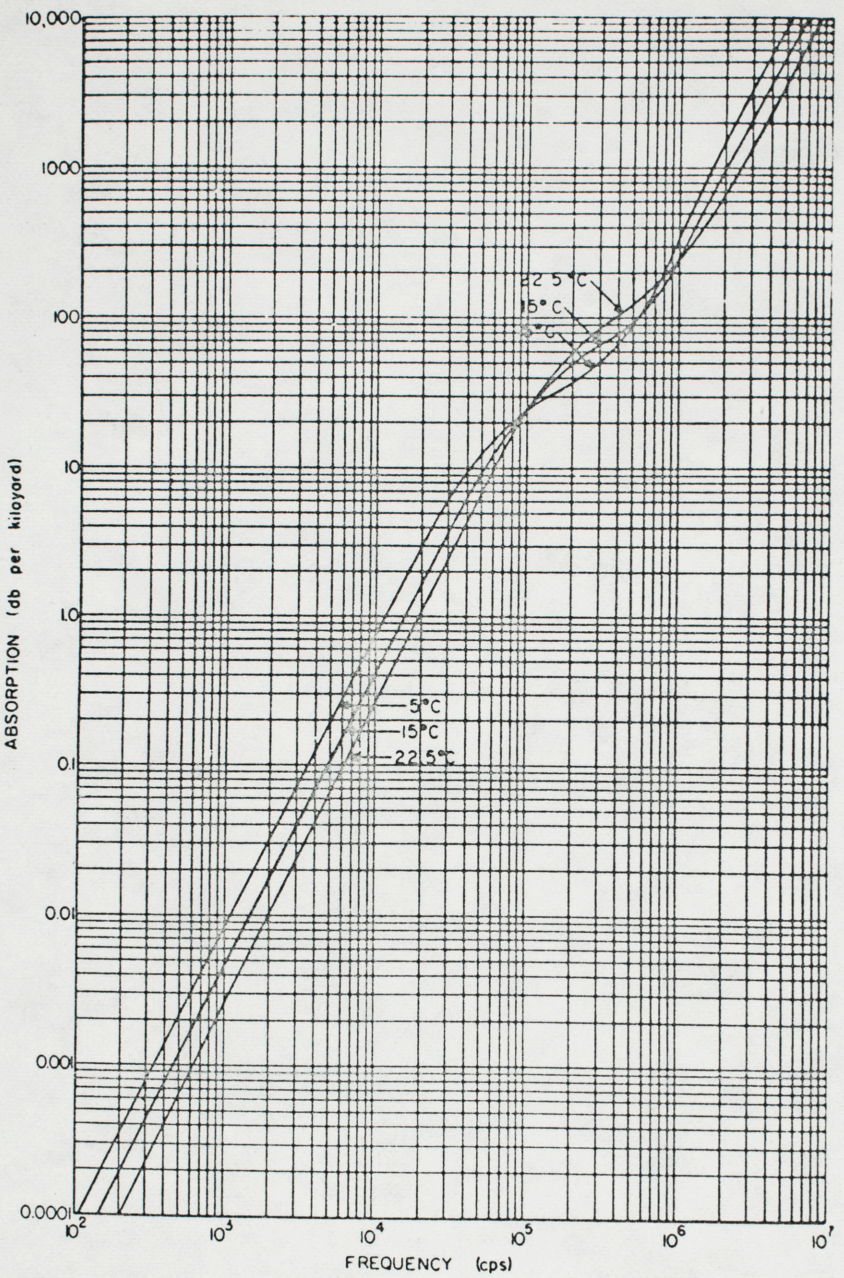


Figure 3. Absorption of sound in the ocean as a function of frequency, with temperature as a parameter.

If L is the value of I expressed on
10 log I_r and L_1 is equal to 10 log I_1

$$L_r - L_1 = -20$$

or

$$L_r = L_1 - 20$$

Equation (34) is the expression of the i
scale and shows the characteristic 6 dB
the distance is doubled.

The absorption in the medium, at an
function of range, so we can modify Eq
attenuation:

$$L_r = L_1 - 20 \log r -$$

where α is the attenuation in decibels
normally large compared to 1, Equation

$$L_r = L_1 - 20 \log$$

If W is the acoustic power in watts:

$$L_1 = 71.6 + 10 \log$$

5.3 Reflection

When a body with dimensions that are
wavelength of sound at the operating
sound field, it will intercept a certain am
intercepted sound power is re-radiated as

The amount of sound power intercepted
its area which is designated as σ' and det

$$\sigma' = \pi d^2/4$$

for a spherical body. If the body is not
orientation in the sound field. If F is
unit area at the target and W is the t
target:

$$W = F\sigma'$$

If the target is a perfect reflector, all of
flected as sound. If, however, it is not a
intercepted power will be absorbed and
reflected as sound. The re-radiated or
therefore:

$$W_s = \mu W - F\mu\sigma$$

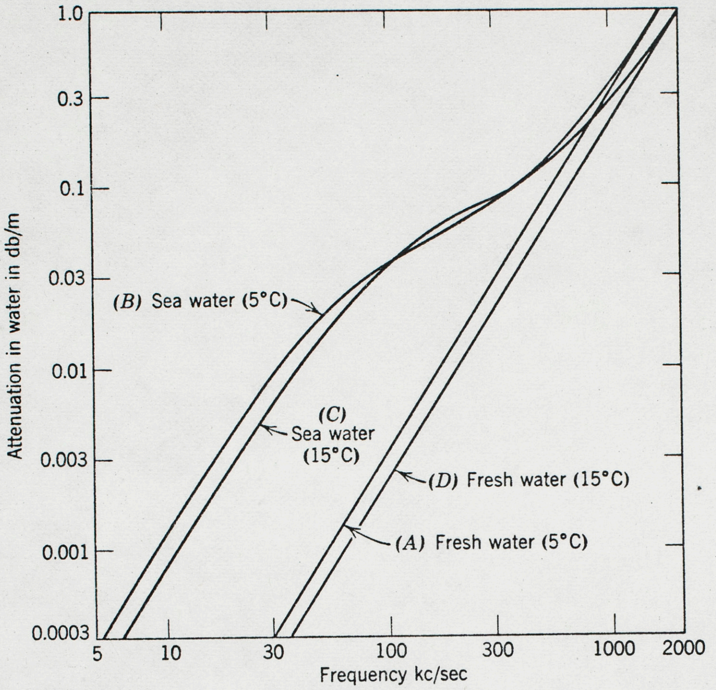


Fig. 9.7. Attenuation at ultrasonic frequencies in (*A*) and (*D*) fresh water; (*B*) and (*C*) sea water.

Plotted in curve *A* of Fig. 9.7 is a line representing the measured attenuation of acoustic waves in fresh water at 5°C expressed in db/meter. Curve *B* of this figure is a plot of measured values of the attenuation in sea water at 5°C. The pronounced difference between these two curves at frequencies below 500 kilocycles/sec makes evident the presence of yet an additional relaxational process and attendant excess absorption in sea water beyond those present in fresh water. It is natural to attribute this additional absorption to the presence of dissolved salts in sea water and to refer to it as a type of *chemical relaxation*. Laboratory measurements by Leonard[12] and co-workers have shown that the excess acoustic absorption of sea water as compared to fresh water is caused almost entirely by the presence of dissolved $MgSO_4$. For instance, as the concentration of $MgSO_4$ is increased in water, the measured excess absorption increases until at a concentration of 0.014 mole per liter it is essentially the same as that of sea water as measured directly in the ocean. The relaxation time of the process may be computed from a measurement of the frequency at which the absorption

[12] Wilson and Leonard, *J. Acoust. Soc. Am.*, **26**, 223 (1954).

ater. If the water composing
ogeneous, only divergence and
pressure level as a sound beam
or instance, a wave diverging
sented by the equation

$_{2}-r_{1})$ (15.2)

ures measured respectively at
of origin of the wave and α is
nepers/meter. Upon applying
tion 15.2, the latter becomes

$_1/r_2) - 8.7\alpha(r_2 - r_1)$

y a, where a is the absorption

$(r_2/r_1) + a\,(r_2 - r_1)$

presents the *decrease* in sound
m r_1 to r_2 and may be replaced
transmission loss in decibels.

$a\,(r_2 - r_1)$ (15.3)

es to a given distance r as being
at one meter from the effective
done, the sound transmission
ter to any distance r is given by

$+\ ar$ (15.4)

waves in sea water has already
e on frequency and temperature
cal values of its magnitude as a
C may be obtained either from
. For instance, in sea water at
0.001 db/m at 10 kc/sec, and

from equation 15.4, showing the
e r for each of the above three
of this figure shows that at low
transmission loss is caused by
However, as the frequency and
the absorption loss becomes of

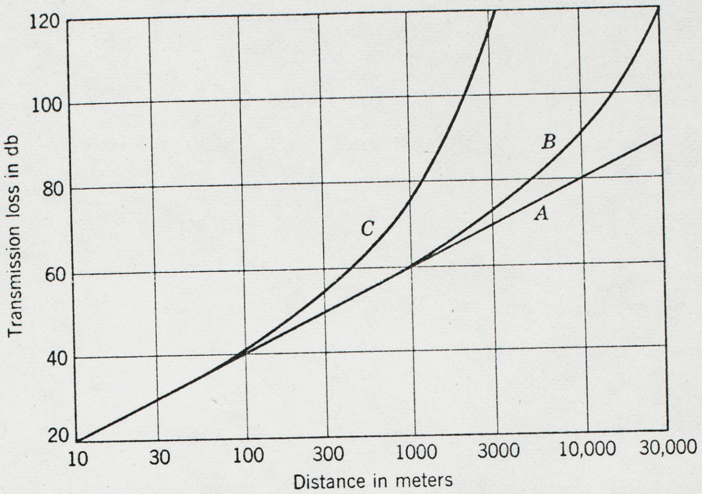


Fig. 15.1. Curves showing dependence of transmission loss on distance and frequency. Curve A at 1 kc/sec. Curve B at 10 kc/sec. Curve C at 50 kc/sec.

greater and greater significance. If we differentiate equation 15.4 with respect to r, then

$$\frac{dH}{dr} = \frac{20}{2.3} \cdot \frac{d(\ln r)}{dr} + a = \frac{8.7}{r} + a \qquad (15.5)$$

is obtained for the spatial rate of transmission loss in decibels/meter. Let us now define a *crossover range* r_c as one within which the rate of attenuation caused by divergence is greater than that caused by absorption and vice versa for ranges greater than r_c. Upon equating $(8.7/r)$ to a, this crossover range is given as

$$r_c = \frac{8.7}{a} \qquad (15.6)$$

For 10 kc/sec, the crossover range is 8700 m while at 50 kc/sec it is reduced to 600 m. From the above discussion it is evident that low frequencies must be used if sound energy is to be transmitted through sea water to great distances with a minimum of transmission loss.

When sound transmission loss measurements are made in the ocean, they are frequently observed to be at a considerable variance (usually larger) with those predicted by equation 15.4. Factors contributing to this variance include additional divergence or partial convergence caused by *refraction*, destructive and constructive *interference* associated with multipath types of propagation including reflections from the surface and bottom of the sea, *diffraction* and *scattering* caused by the presence of

Finally, upon substitution into equation 15.34

$$E = 124 + 20 - 2(94 + A) = -44 - 2A$$

is obtained for the echo level. It is evident that even if no transmission anomaly losses are present, i.e., if $A = 0$, the echo level being returned under the above assumed conditions corresponds to a sound pressure of less than 0.01 microbar. Factors influencing the detectability of such a weak signal will be discussed in the following two sections.

15.10 Masking by Noise. Two basic types of noise tend to mask echoes returned from underwater targets. One type consists of those noises present even when no sound is being radiated by the transducer. These include such noises as ambient sea noise, machinery and propulsion noises generated by the echo-ranging ship, and turbulence noises generated in the vicinity of the sonar transducer. The second type consists of a multiplicity of weak echoes returned from small scatterers located in the sound beam and near the target. This second type of masking noise is referred to as *reverberation* and will be discussed in Sect. 15.11.

Ambient sea noise results from the summation of all noises generated within the body of the ocean. However, its principal source is associated with wind and wave action at the surface of the sea.[4] Figure 15.13 is a plot of average values of the pressure spectrum level of ambient sea noise as a

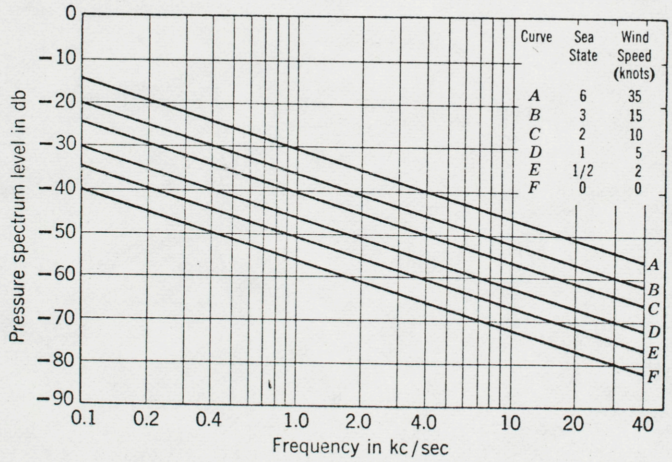


Fig. 15.13. Pressure spectrum level of ambient sea noise for various sea states and wind speeds expressed in decibels relative to 1 microbar. (After Knudsen.)

[4] Knudsen, Alford, and Emling, *J. Marine Research*, 7, 410 (1948).

function of frequency for six
state. It is to be emphasized
averages since the standard de
5 db. It is interesting to note t
to have any systematic effect
show the same uniform decreas
5 db per octave. This behavio

$N_f = N$

where N_f is the spectrum level
spectrum level at a frequency
be essentially isotropic at depth
equally from all directions and

Ambient sea noise, which is
detectability of wanted sonar
moving at appreciable speeds, i
ambient sea noise. This self-no
It may increase by more than 2
in the case of destroyers. It al
transducer, the type of ship, an
received. In essence, it differs s
typical values cannot be given.

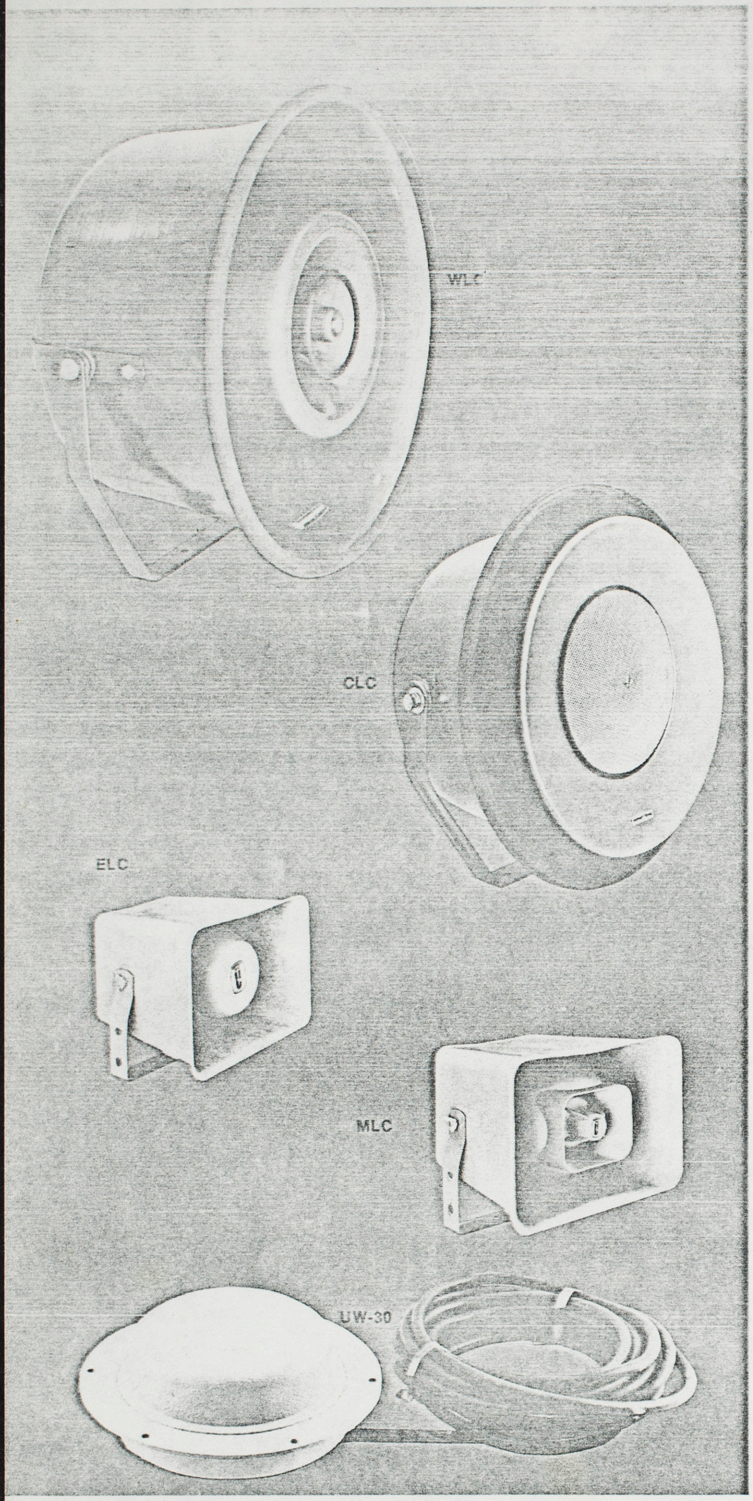
In all echo-ranging systems
This may be either the same tr
transducer. Because of the dir
effective pressure level of the n
ducer's receiving directivity ind
effective level of the masking n
of the receiving system. A fini
the short echo-pulse to attain fu
the echo frequency by the doppl
noise L_N in decibels is therefor

L_N

where the term 10 log w conve
accordance with equation 13.6.

For each particular sonar de
presented audibly or visually,
probability of detection may b
the equation

E



MODEL WLC, world renowned as the "Standard of Excellence" for outdoor, weatherproof high fidelity sound installations. Rugged, heavy duty dual range reproducer for public installations in arenas, stadiums, church towers, concert halls, etc. High efficiency and wide, flat frequency response covers the entire musical spectrum.

MODEL CLC, a full range weatherproof speaker, is the ultimate in natural sound reproduction. The all-metal CLC is ideal for outdoor high fidelity music systems on patios, lawns, terraces—wherever high fidelity sound quality with maximum efficiency is the primary consideration.

MODEL ELC, weatherproof high fidelity wide-range speaker system, is especially well suited for use on patios, lawns, and terraces. Fiberglass reinforced polyester housing provides a scratch-proof indestructible finish.

MODEL MLC, a weatherproof, high fidelity, 2-speaker system, is especially well suited for use on patios, on lawns, around swimming pools, and for all other outdoor high fidelity applications. The light gray color of the horn allows it to blend with outdoor surroundings and is molded into the fiberglass reinforced polyester housing to provide a scratch-proof, indestructible finish.

MODEL UW-30, a departure in the design of underwater sound sources, its unique design (pat. app. for) uses the case structural enclosure as the sound transducer. This development makes possible a speaker that has no metal parts exposed to the outside, thus, eliminating rust and corrosion.

A single model UW-30 will deliver uniform sound throughout moderate size pools, up to 30 by 30 feet. For home stereo, two speakers may be used but very little separation will be noticed. For larger pools up to 30 by 60 feet, two speakers should be used. If a high level of turbulence is to be overcome, such as swimming and diving instruction classes, several speakers should be used.

SPECIFICATIONS

Model	Power Capacity (I.P.M.)	Frequency Response	Impedance	Dispersions	Sound Pressure Level*	Dimensions	Shipping Weight
WLC	50 watts	50-15,000 Hz	8 ohms	120°	120 dB	$33\frac{1}{2}$" Dia. x 20" D	72 lbs.
CLC	30 watts	55-14,000 Hz	8 ohms	120°	118 dB	$22\frac{3}{4}$" Dia. x $12\frac{11}{16}$" D	20 lbs.
MLC	15 watts	150-15,000 Hz	8 ohms	120°	117 dB	$12\frac{3}{4}$" W x $9\frac{1}{8}$" H x $10\frac{5}{8}$" D	$10\frac{3}{4}$ lbs.
ELC	15 watts	150-12,000 Hz	8 ohms	120°	114 dB	$12\frac{3}{4}$" W x $9\frac{1}{8}$" H x $10\frac{5}{8}$" D	7 lbs.
UW-30	30 watts	100-10,000 Hz	8 ohms	Omni		$7\frac{3}{16}$" Dia. x $2\frac{5}{8}$" D	4 lbs.

*SPL reading taken 4 ft. on axis with "Full-Range" power input. Reduce by 6dB each time distance is doubled. Reduce by 3dB if power input is halved; increase by 3dB when power input is doubled.

Omar A. Ramirez
Music 202

Potentials of Transduced Guitar in Live Performance

It is the purpose of this endeavor to compile a sound catalog using the guitar in a variety of potential live performance situations, in an effort to expand the technical limits of the guitar through applied technology. It is based on phenomena observed through the use of an audio transducer mounted to the body of the guitar. The guitar functions as both the principle sound source and as a timbral processor.

This project could be of value to guitarists seeking to broaden their horizons and apply the instrument creatively in other mediums, and to composers who might wish to write for the guitar; it would be beneficial to observe how it might be used and what it is capable of.

A guitar has been prepared for experimentation and a transducer mounted to the sound board to maximize energy transfer. It was placed behind the bridge so that it will not interfere with playing position. A microphone is then placed in front of the guitar, amplified, and fed into the guitar body by way of the transducer.

The major effect of transduction is one of sound reinforcement, as the vibration of the sound board is reinforced by induced vibration of the transducer, making the guitar sound louder (signal plus altered signal). The increased vibration supplied by the transducer caused the body of the guitar to become hypersensitive to sympathetic vibration. (Ex. 1)

The effect of this phenomenon can be applied in several ways. If an open string (low "E" for example) is played staccato, all other strings containing common partials to the "E" harmonic series will be excited producing a sympathetic "glow", approximating a reverberation colored by several natural harmonics. (Ex. 2)

A variety of percussive sounds are also available, as tapping the body of the guitar in different areas produces sounds of different timbres. (Ex. 3) The open strings, set in motion by the impact, can be allowed to ring, or can be damped so that purely the percussive sound will result. (Ex. 4) A wide range of coloration becomes possible when a chord is fingered in the left hand, and the body of the guitar is struck, so that the impact is colored by the notes of the chord. (Ex. 5)

The use of amplification via transducer enhances other effects inherent in the guitar. Increased vibration on the sound board allows the "multiphonic harmonics" to be reinforced, thereby making them practically accessible. Multiphonic harmonics are natural harmonics, inherent in the properties of open strings. They are produced by touching the strings at positions other than the harmonic nodes and are plucked in the same manner as natural harmonics. A wide array of pitch combinations becomes possible, determined by where the string is stopped and the pitch of the string chosen. (Ex. 6)

The transducer, when used in conjunction with an air mike, can produce feedback. This effect can be exploited to sustain a note for any length of time. It can be controlled by varying the proximity of the microphone to the sound hole. If a note is stopped, and plucked and the guitar moved close to the mike (by leaning forward), the feedback will begin as a crescendo, building up to the maximum level that the string and resonating cavity can produce. (Ex. 7) A long sustain can be produced at a lower dynamic level by never allowing the feedback to reach its maximum level. This is done by leaning backward and forward as necessary to keep the string vibrating. (Ex. 8)

If the performer choses he can allow two notes to feed back by the same method described above. The two pitches can then be sharpened or flattened to produce frequency beats of varying speeds approximating a tremolo effect (Ex. 9) The remaining two fingers of the left hand may be used to play a melody above the sustaining strings if so desired. (Ex. 10)

The guitar was the major factor in determining timbre when used with an air mike, in the preceeding experiment. In the following experiment, the contact mike placement becomes the principle factor in selecting a tone color.

The interior acoustical design of the guitar is such that different areas of the sound board are "tuned" to reinforce certain harmonics. This can be observed by placing the contact mike in different positions on the sound board, plucking a string, and listening for the higher partials of its harmonic series.

By using the contact mike as a "stethoscope" one can determine readily the pitch areas on the sound board, thereby choosing the coloration effect desired. (Ex. 11)

Depending on the desired effect, the composer/performer must choose between air mikes and contact mikes, as each has its advantages and disadvantages.

The air mike affords the possibility of controlled feedback though it is limited in the choice of position. Also inherent in the air mike is the problem of proximity to the sound hole, as plucked instruments create strong transients as a result of the mode of attack. Though all microphones induce their own coloration effects on the sound source, their effects are quite subtle and not always readily perceptible. When used with an air mike, the guitar itself is the principle source of coloration.

The contact mike will not allow the use of controlled feedback, but is much less limited in the choice of position. The "proximity" problem becomes a matter of how close to the transducer it is mounted, and the operating level of the amplifier used. Unlike the air mike, the contact mike allows a wide range of coloration effects, as mike positioning allows the choice of timbral variation.

This project is centered on observation of internal sources of coloration, that is to say, those inherent or native to the guitar. Though sympathetic vibration, percussive sounds, and multiphonic harmonics are purely natural phenomena, (the transducer does not create these phenomena), the transducer serves to enhance these effects, making them louder and more easily recognized in live performance situations, by using the body of the guitar as a timbral processor.

The transduced guitar opens the door a world of external sources of coloration, limited only by the extent of the imagination. Among these are the use of synthesized sound, and prerecorded tape.

LIST OF EXAMPLES

note: Tape to be played during class Presentation.

PROJECT 202B
MICROPHONE PLACEMENT

HOLLIS GENTRY III
WINTER/SPRING 1977
PAULINE OLIVEROS

PURPOSE: To determine by experiment and observation, interesting and colorful results from various microphone placements when amplifying and recording extended techniques of the B-flat tenor saxophone.

THEORY: That standard correlations may be derived between microphone placement and achieving a particularly colorful effect in amplification and reproduction for recording of various extended techniques of my own invention, as well as some commonly used techniques, for example, flutter-tongue and singing/playing.

FURTHER ASPECTS: Some of the more particular considerations of this study include determining which microphone type(s) to use. This is the initial concern, in that individual microphone characteristics will bear individual results. These results will be obvious in characteristics such as tone color and timbre, and will therefore be of particular concern. The microphones should be compatible with the saxophone in regards to frequency response and boost areas in order to attain a smooth response and as natural a sound as possible. The extended techniques are colorful acoustically, and capturing their natural beauty is the main objective.

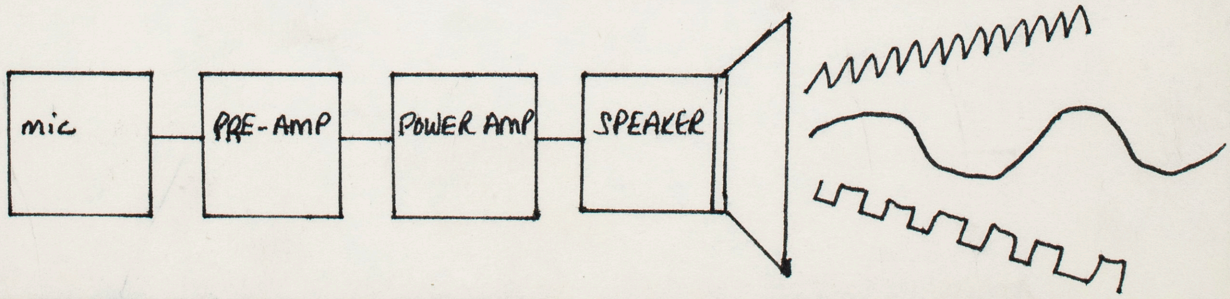
Multi-microphone placement is another consideration in this study. This is of prime importance. In conceiving this study I became faced with a question of whether a more natural sound would result from using more than one microphone in close proximity to the instrument. By nature of its design, the saxophone, like other woodwind instruments, emits sound from numerous places other than the bell. Sound escapes under the left stack, above the left stack, the right stack, along the bell and even behind the horn. The question of where to place the microphone then becomes obviously more difficult to answer. Now when the physical location of these various extended techniques are taken into consideration, one discovers that the sound escapes from a variety of locations on the saxophone. Where depends on the keys depressed and the unused areas. In essence it is my intention to determine if multi-microphone placement is beneficial in the electronic reproduction of these extended techniques,or not. This idea of multi-micing is an intented augmentation and possible improvement to the standard microphone placements such as omni-directional or the six inches in front of the bell proceedure.

Another aspect of this experiment is its function as a study of microphone characteristics themselves. This is important to me as a preformer in that understanding about microphones and microphone placement is beneficial for live preformances or in a recording situation. From past experience I have learned not to take technical considerations of this nature for granted. These are things that could,and very often do,make the difference between a good preformance,and a poor one,from the listeners perspective.

MICROPHONE TESTING

My first series of experiments were aimed at determining which microphone type was best suited for use with the B-flat tenor saxophone, the E-flat alto saxophone and the B-flat soprano. I have reduced the scope of the experiment by excluding the alto and soprano saxophones in order to concentrate on the B-flat tenor saxophone.

The initial effort took place in the Mandeville Recital Hall. Due to restrictions beyond my control the focus of this session was on the recording of aural information minus the amplification experiments. The recital hall was chosen partly because of it dimensions, 60x60x20, which are relatively similar to those of a small nightclub. This is important because this is the type of room I have worked in over the past few years and it is probable that I will continue to find myself in similar situations in the future. This aspect effects only the amplification experiments and makes little difference to the recording study. The recital hall does influence the recording studies in that the recording facilities in the hall are of the finest professional quality which implies such things as the use of low impedance microphones, the availability of faster tape speeds as well as high quality mixers, etc.. With equipment of this nature and quality, a good engineer is capable of reproducing or creating any sound he wants in the re-mix process. From this point of view the type of microphone used seems of less importance than this experiment implies. All sophisticated equipment aside, however, when the design of the <u>basic</u> sound reproduction system is taken into consideration, it is evident that the microphone is indeed an important element;



after the microphone, the remaining components serve mainly to realize the aural information from the microphone. In effect the actual sound depends on how the microphone recieves, and especially, transmits the sound. The microphones used in the first experiment include:

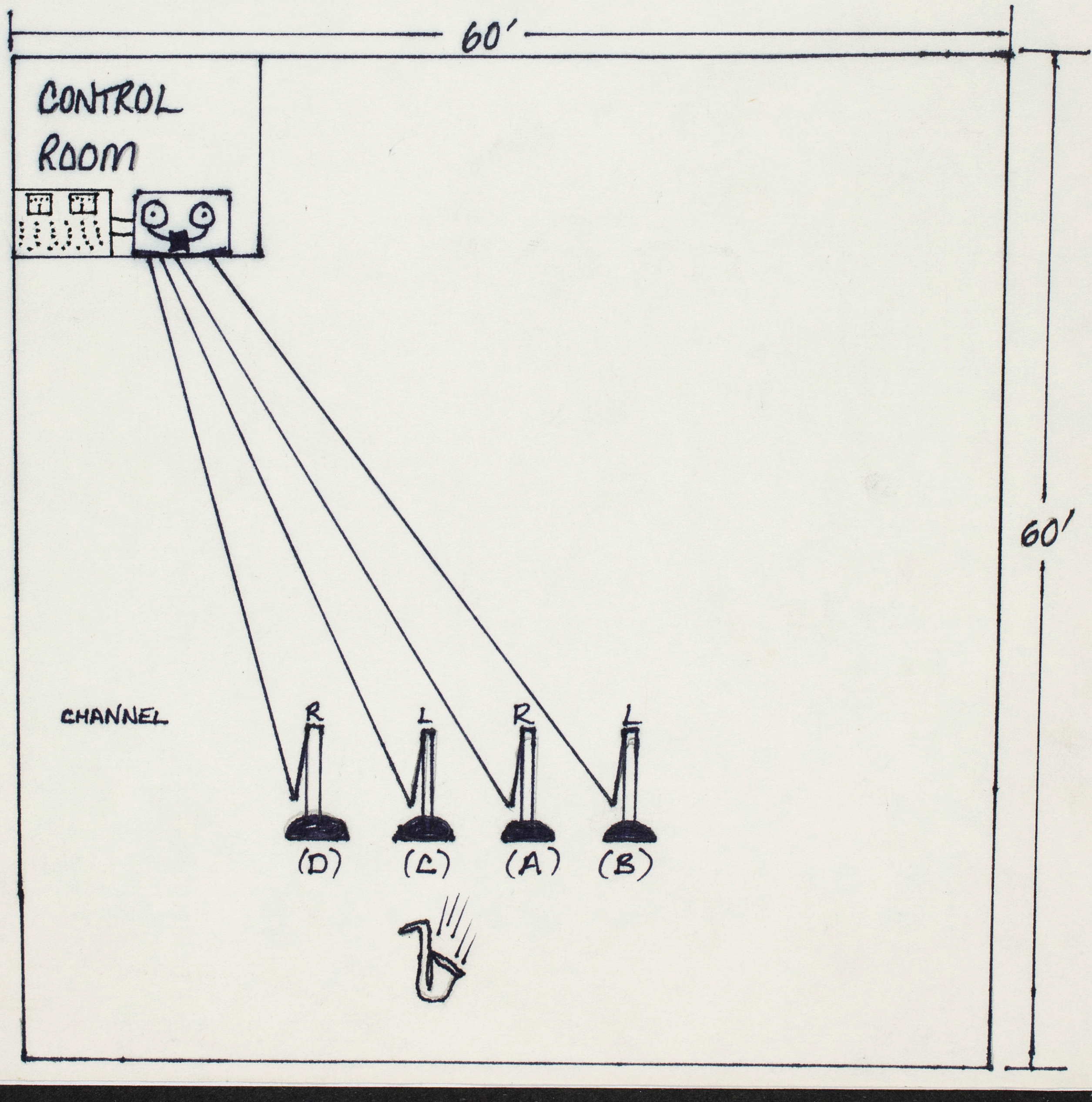
A) ELECTRET CONDENSER CARDIOD C-505E

B) AKG-SE5 DYNAMIC OMNI-DIRECTIONAL

C) AKG-D1000E DYNAMIC CARDIOD W/MODE SWITCH

D) SENNHEISER MD421-U-5 DYNAMIC CARDIOD STUDIO TYPE

These microphones were mounted on individual stands, side by side, with leads running directly into the mixing board a Taskam Model 10. See diagrahm for positioning in the room;



In order to reveal the aural effects of each microphone, a series of musical events was composed and subsequently preformed on each. These include;

1) A series of long tones situated in each register of the saxophone. Played once forte,and once piano.

2) A scalar passage played once with normal articulation once with slurred articulation,and once with staccato articulation.

3) A rapid passage,excerpt from the John Coltrane GIANT STEPS solo,involving the entire practical range of the saxophone.

Using two microphones at a time,one plugged into the left channel and one into the right channel,the information was recorded. Distance from the microphone was approximatly 3 to 5 feet. This stereo balance of the two microphones on the tape,$\frac{1}{4}$"-$\frac{1}{2}$ track-7$\frac{1}{2}$ips,makes comparison of each pair of microphones easier than it would be in the tail-to-head position. Tone settings at the mixing board remained in a <u>normal</u> position. Also due to the relatively close microphone placement,unwanted room noise is reduced to a minimum,almost undetectable,therefore the sound of each microphone is relatively close to being accurate.

EXPERIMENT RESULTS

Upon listening to the recorded material, the following results and conclusions were attained.

<u>MICROPHONE A</u>: Electret Condenser Omni-directional produced the brightest tone quality. This microphone is much too sensitive for recording the saxophone at close proximity. This is a good microphone for omni-directional recording of large ensembles such as orchestras,etc.. It seems to strip the tone of its full harmonic character,producing excessive emphasis on upper frequencies.

MICROPHONE B: The AKG-SE5 Dynamic Cardiod produced a darker tone quality than the electret condenser microphone. The overall result was acceptable. The perceptable difference being the presence of a more full-bodied saxophone sound from the AKG-SE5.

MICROPHONE C: The AKG-D1000E Dynamic Cardiod W/MODE SWITCH WAS not bad in overall sound quality, on the other hand, it was not outstanding in overall sound quality either. In general it seems to be a middle-of-the-road microphone for recording the saxophone. The tone quality is noticeably darker than both the Electret Condenser and the AKG-SE5.

MICROPHONE D : The SENNHEISER MD421-U-5 Dynamic Cardiod Studio Type displayed the smoothist frequency response of all four microphones tested. It stands apart from the others by being true to the tone in all registers. Like the AKG-SE5, the SENNHEISER produces a full-bodied saxophone sound.

As a result of these experiment results, the multi-microphone placement experiments will use the SENNHEISER and the ADG-D1000E. Further study of the AKG-D1000E is desired in order to fully explore the function of the Mode Switch which can be set on B-bass,M-medium,or S-sharp.

SENNHEISER

Studio Cardioid Microphone
MD 421 - U

FEATURES

- Dynamic cardioid studio type
- Directional pattern almost equally distributed over entire range
- Rejects low frequency undesirable noises
- Helps control feedback
- Built-in hum compensating coil
- Built-in continuously variable low frequency roll-off filter
- Professional broacast quality

DESCRIPTION

The MD 421–U is a dynamic microphone of the very highest calibre, designed to meet the most demanding requirements of the sound system installing, recording, film, and broadcast industries. This microphone enjoys an excellent reputation, and can be seen in the company of many famous performers and statesmen throughout the world today. The MD 421–U covers the spectrum of 30 to 17,000 cps with a "transparent" quality which captures every nuance in a vocal pianissimo as well as the loudest crescendos of the symphony orchestra. Its response curve is exceptionally flat, assuring perfect frequency linearity and tonal balance throughout the audio spectrum. Because of its excellent directional characteristics, which are virtually independent of frequency, the MD 421–U is ideal for use in "spot" recording, where surroundings may be acoustically unfavorable, or in high-fidelity studio public-address systems, where faithful reproduction must be combined with freedom from acoustic feedback. The MD 421–U has an especially high nominal front-to-back ratio of 16 db. It is equipped with a special comperisation coil, which protects against the effects of stray magnetic fields. A continuously variable bass filter allows adjustment for difficult acoustical environments.

The MD 421–U is a Dynamic Cardioid, low impedance, microphone.

ARCHITECTS AND ENGINEERS SPECIFICATIONS

The microphone shall be of the dynamic moving coil type. The frequency response shall be 30-17,000 cps. The pick-up pattern shall be cardioid with a front to back ratio of 16 db min. This microphone shall have an impedance of 200 ohms (150/250 classification) and shall be balanced in respect to ground. It shall also be equipped with a continuously variable acoustical bass filter for attenuating frequencies below 100 cps for speech purposes.

The microphone shall weigh not more than 15 oz. and shall be housed in the case illustrated, which shall have a non-reflecting finish.

The microphone shall have an output of —53 dbm (—53 db/10 dynes/cm²).

The performance specifications shall be listed under SPECIFICATIONS and shall be met or exceeded.

The microphone shall be SENNHEISER Model MD 421–U. MD 421-9N.

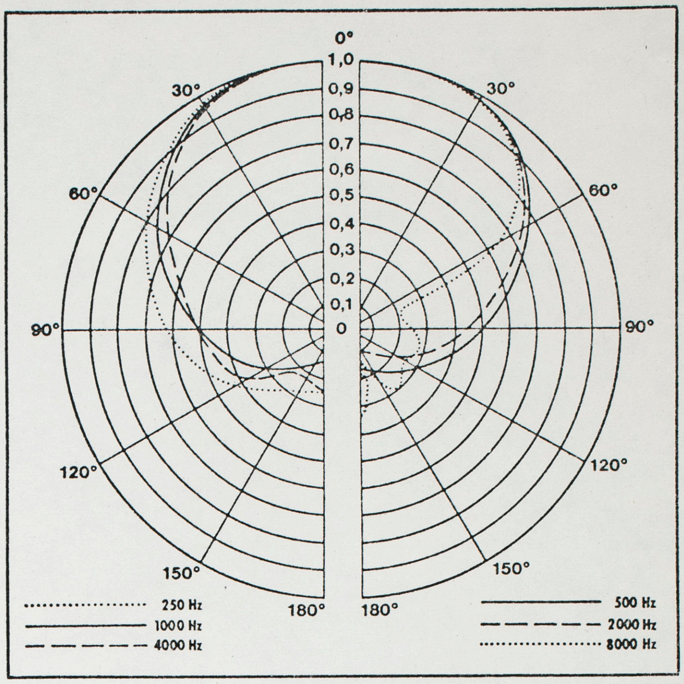
NO – 1

SENNHEISER ELECTRONIC CORPORATION

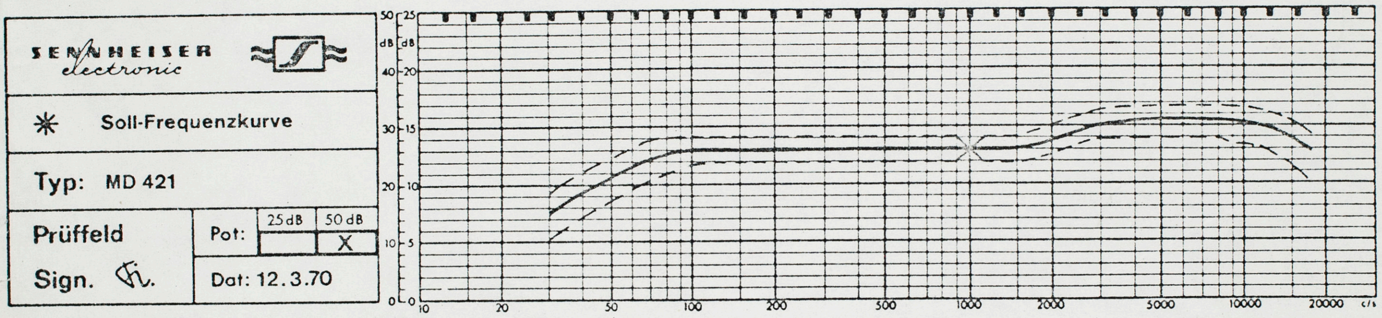
10 West 37th Street, New York, N.Y. 10018

Technical Data

Acoustical mode of operation	Pressure gradient
Frequency range	30 – 17,000 Hz
Tolerance limits	See frequency response curve above
Sensitivity (1 kHz)	0.2 mV/µbar ± 3 dB
Output level ref. 1 mW/10 dynes/cm²	– 53 dbm
EIA rating	– 145.8 db
Impedance (1 kHz)	200 Ω
Directional characteristics	Cardioid
Front to back ratio	18 dB – 2 dB
Sensitivity to magnetic fields	approx. 5 µV/50 mG
Dimensions	177 x 48 x 46 mm
Weight	14 oz.
Connector	XLR
Pin connections	2 + 3: signal, 1: ground



A smooth 5 db increase is provided in the range above 3000 Hz to add presence to the recording.



Nominal frequency response curve (with tolerance limits) MD 421

Every MD 421 is supplied with its own factory frequency response curve.

We reserve the right to alter the specifications especially with regards to technical improvements.

CONSUMER NET PRICE LIST

Effective date: January 15, 1972

All prices are F.O.B. New York Warehouse

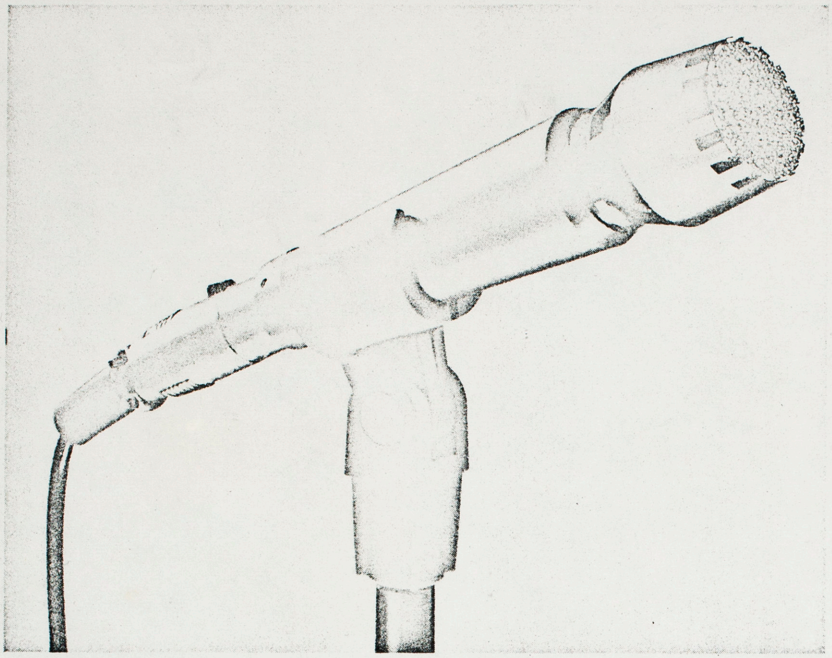
Prices are subject to change without prior notification

MD 421–U	dynamic microphone as described dark gray non reflecting finish	$ 129.00
MD 421–U2	dynamic microphone as described without roll–off filter dull black non reflecting finish	$ 125.00
	ACCESSORIES :	
MZW 22	fibre glass windscreen	$ 17.60
MZW 421	foam windscreen	$ 10.00
MZA 216	thread adapter for American stands	$ 1.40
MC 22	cable with XLR connectors on both ends, 15 feet long	$ 15.00
TM 514 U	transformer cable for high impedance inputs with standard phone plug	$ 24.20
MZT 421	desk stand	$ 11.00

MICROPHONES • HEADPHONES
DISTRIBUTED BY
NORTH AMERICAN PHILIPS COMPANY, INC.
100 EAST 42nd STREET, NEW YORK, NEW YORK 10017

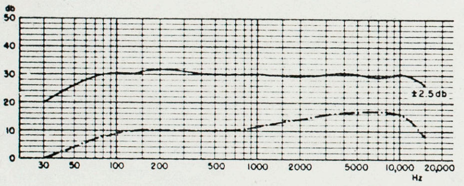
CARDIOID Dynamic Microphones

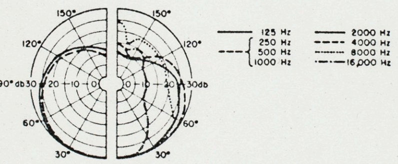
D-1000E/D-1000TS



TECHNICAL DATA

Frequency response	40-16,000 Hz ± 3 db
Sensitivity	-53 db (re 1 mW/10 dynes/cm^2)
Impedance	D-1000E, 200 ohms. D-1000TS, high
Directional characteristics	Cardioid
Mode selection switch	B = Bass, M = Medium, S = Sharp
Dimensions	6" x 1 7/16"
Weight	9 1/2 oz.

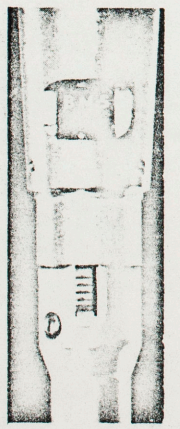




ACCESSORIES

MSH-58E	Flexible shaft
SA-10/3	Stand adapter (around connector)
MK	Cable series
AKG	Standard and anti-shock stands

*B-M-S MODE SWITCH POSITIONS



B = Bass: Normal, smooth frequency response. Emphasized low frequency response when close-talked.

M = Medium: Permits close-talking for normal sound without booming or bass accentuation. Emphasized middles and highs at greater working distance.

S = Sharp: High pitch sensation by added, upper mid-range brilliance and de-emphasized bass.

The combination of mode switch positions and varying pickup distances enables a variety of effects.

DESCRIPTION

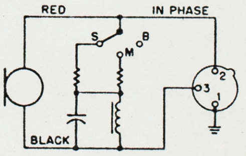
New AKG D-1000 Series microphones are specially designed for the performing artist. A built-in, exclusive AKG mode switch provides creative control at the microphone for professional projection of a personalized sound. Performing singles or groups, vocal or instrumental can adapt Series D-1000's special frequency response effects to any type of music from ballad to beat. Recessed for easy mode selection without accidental operation, the three position B-M-S*switch functions as a unique mood and environmental sound control.

The effective cardioid (one direction) sensitivity keeps sound personal and rejects unwanted rear noise from the audience. The close-working pick up pattern minimizes reamplification of instrument loudspeaker sound through the vocal reinforcement system. Feedback squeal is suppressed when the microphones must be used close to loudspeakers or in an acoustically unfavorable area.

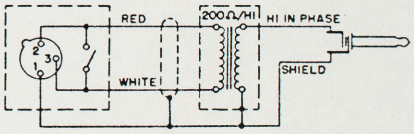
The special non-metallic MAKROFOL diaphragm of D-1000 Series withstands high sound pressures, and a sintered bronze cap filters breath noises and controls pop while protecting the system from dirt and moisture. AKG D-1000 Series are extremely rugged. The system is shock-mounted to minimize handling noise. The attractive and practical fit-to-the hand design is enhanced by a non-glare, silver-gray satin finish. They are supplied complete with 15' cable, necessary connector(s), quick-remove stand adapter and case.

CONNECTION DIAGRAMS

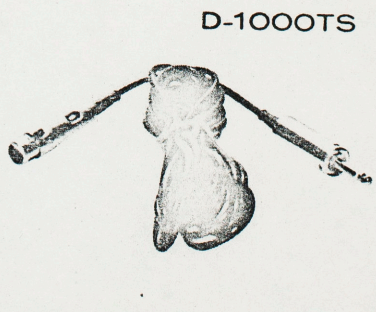
D-1000E



MK-7/TS-5



D-1000TS



The D-1000TS (Transformer/Switch) may be ordered for high impedance and on-off switch operation. Package includes D-1000E with stand adapter and case plus MK-7/TS-5 cable assembly consisting of 3-pin microphone mating connector with built-in on-off switch, 15' cable, high impedance transformer and phone plug for amplifier input.

SD1000X51-268-5M

PRINTED IN U.S.A.

B Musical Applications?

T.A. Setter
Music 202
Pauline Oliveros

MICING FOR COMPUTER MIXING

My objective was to supply transduced information which would provide significantly different impressions of a single acoustical event, and to acquire different "impressions" before any active circuitry. This was to be achieved by using three different means: first, by using reflected air mic signals; second, by allowing phase interaction which occurs when signals from various types, phases, and locations of mics are mixed without isolation resistors and summing amplifiers; and third, by using a helium atmosphere to alter the speed of sound propagation in a small enclosure. These varied harmonic spectra would be sent to the Sonic Research synthesizer at CME for computer-controlled mixing and processing (see Russell Lieblich's paper).

My research into these three types of modification revealed some interesting, as well as some dull, results. The reflected mics were interesting in two applications (as far as altering the acoustic information). The parabolic dish mic, which provides its won reflector, when used in a reverberant space like CME would isolate direct sounds from reverberated sounds because of its tight focus, thus sounding "dryer" or "closer" despite the fact that it could be used at considerable distance due to the combination of its focal characteristics and the high gain factor caused by the dish. The second interesting reflected application was a cardioid mic (RE-16) placed in a sphere of music stands. A window was left in the sphere to allow sound to enter and the mic was faced away from the window. This acted in two ways: it was very directional due to the reflective quality of the outside of the sphere, and it acted as an acoustic resonant filter, the properties of which varied with the size of the sphere and placement of the mic within it.

The phase interaction experiments also proved quite interesting. I designed two simple circuits for mixing two pairs of LoZ balanced mics. The first pair were mixed in parallel and could be reversed in phase relative to each other and/or phase lagged ninety degrees. The second pair were mixed in series and had identical phase changing circuits. Of course, the type and position of the mics relative to the sound source played an equally important role in determining phase characteristics. These pairs of mics, however, produced very different sounds when the phase switches were changed---typically "thinner" or "fuller" sound in varying degrees.

Two of the most interesting discoveries I made were the difference between parallel and series mixing (using the same mics in the same positions), and the number of "tone types" possible when one pair of mics was mixed and then fed as one mic to the imput of the second module where it was mixed with a third mic. The result is sixteen possible sounds, some, of course, more divergent than others.

The actual set-up for the live demonstration at CME used the TASCAM model 5 as a pre-amplifier for the mics. A close air mic (ARG D-1000E) was mixed at the model 5 with a contact mic on the neck of the soprano saxophone. This was intended to be the most natural sounding channel. These were assigned to output channel #1. A parabolic mic (using an EV RE-16) focused on the saxophone and an identical mic in the same plane (12 foot radius from sax) were mixed in parallel through my special circuit and their phases set ninety degrees apart. The output from my curcuit ran to the model 5 and was assigned to output channel #2. A series mix in the second channel of my phase processor using two Shure SM 53 mics in the same plane (5 foot radius from sax) with their phases unaltered (inphase) was assigned to output channel #3 of the TASCAM board. Finally, the "Setter Helium Processor" (Ha-Ha!) which consists of a close air mic (AKG D-1000E) going to a Shure M 68 pre-amplifier/mixer and 15watt power amplifier driving an AR4ax speaker inside an 85 gallon aquarium filled with helium at the far end of which was an EV 655 mic. This mic was run to the model 5 and assigned to output channel #4. All mic attenuators were adjusted so that all mics provided identical signal level when Russell played and no equalization or reverb was used. The four outputs of the model 5 (hiZ, unbalanced) drove the four VCAs and four filters of the synthesizer used in conjunction with the computer. For the demonstration all air mics in the synthesizer room were highly directional in order to avoid as much ambient noise as possible.

The two "specially prepared" mics, the parabolic dish (EV RE-16), and the "Setter Helium Processor" (EV 655) are both represented in the attached spec sheets. However, their response in these configurations is altered quite a bit. The parabolic mic becomes much more directional, gives much more output, and rolls off more severely in the low frequency range. The "Setter Helium Processor" is at the mercy of the resonant frequency response of the aquarium and of the speaker. Some acoustic damping material was added to help flatten-out the response of the aquarium but the sound is still quite colored; i.e. another acoustic filter.

My expectations were that the channel #1 mics would sound close (distance-wise) and realistic, channels #2 and #3 would be medium-far sounding and quite variable as to tone color, and that channel #4 might experience a slight frequency shift which I could measure with a tuning strobe.

Channels #1 and #3 held no surprises. However, the "Setter Helium Processor" (channel #4) only added a reverb quality with no pitch change. To my surprise, the parabolic dish mic "subtracted" reverb, creating a usable effect I hadn't anticipated.

The selection of a micing room which was not very well suited to show some of these phenomenon was due to feedback and operator convenience considerations associated with live processing and amplification. More dramatic results could have been produced on a tape but I felt that our demonstration should be of possibilities for live electronic application.

T. S.

Also used was a ceramic contact (Hi Z) on the sax but I couldn't get spec-sheets on them.

Ch. 1 & 4

cardioid dynamic microphones

technical data

D-190ES with built-in on-off switch

D-190E/D-190ES

DESCRIPTION

AKG D-190E Series. These microphones are based on a cardioid dynamic transducer which distinguishes itself by a smooth, wide range response with exceptional front-to-back discrimination at all frequencies. These features ensure clear and effortless sound transmission without feedback, as well as exceptionally good recordings even under acoustically unfavorable conditions. An integral sintered bronze cap eliminates the disturbing effects of breath noise and air turbulence and also protects the microphone system from dust, iron particles and moisture.

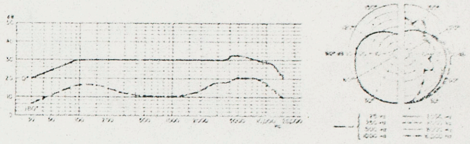
D-190TS is delivered with MC-20TS cable assembly for High impedance operation with on-off switch.

SPECIFICATIONS

Sensitivity: −50 dB (re 1mw/10 dynes/cm²)
Impedance: D-190E, D-190ES = 200 ohm
D-190TS = High Z
Dimensions: 6¼" lg x 1½" dia.
Weight: 6 oz.

ACCESSORIES

W-8 Windscreen
H-24 Suspension
MSH-58E Flexible Shaft
SA-11/1 Stand Adapter (Replacement)
AKG Stands

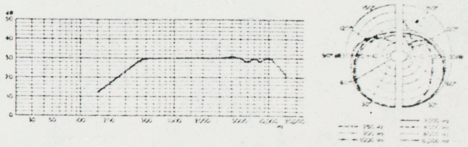


D-590

The D-590 is similar in performance characteristics to the D-190 except that the low end of the response range has been attenuated to defeat low frequency oscillation and table-born noise when mounted directly on surfaces. The unit comes equipped with a 9¾" long, non-detachable, flexible shaft plus mounting hardware and cable.

Sensitivity: −55 dB (re 1mw/10 dynes/cm²)
Impedance: 200 ohms
Dimensions: 11⅛" lg x 1⅝" dia.
Weight: 9 oz.

W-8 Windscreen
ST-4A Table stand, weighted
ST-41 Same as above but with one push button.
ST-43 With three push buttons.



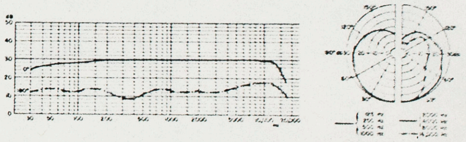
D-200E

Based on the revolutionary "Two-Way" concept (See page 13) this is truly a superb cardioid dynamic microphone. Objective, faithful reproduction of sound, without off-axis discoloration, is obtained with the D-200E. In public address systems it permits control of feedback and offers almost complete freedom of microphone and speaker placement. Smooth, full range response, linear off-axis acceptance and total lack of proximity effect are the main characteristics of the D-200.

D-200TS is delivered with MC-20TS cable assembly for High impedance operations with on-off switch.

Sensitivity: −55 dB (re 1 mw/10 dynes/cm²)
Impedance: D-200E = 200 ohm
D-200TS = High Z
Dimensions: 7-5/16" lg x 1⅝" dia.

W-4 Windscreen
MSH-58E Flexible Shaft
SA-20/1 Stand Adapter (Replacement)
AKG Stands



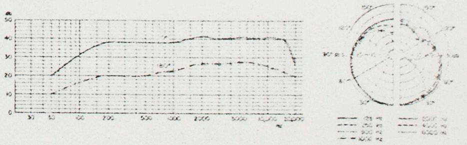
D-707E

The 707E is a multi-purpose quality microphone with professional performance characteristics and is suitable for recordings and public address applications. It is particularly capable of handling high sound pressure levels without overload or distortion. It features an integral windscreen, rugged construction and is resistant to rough handling.

D-707TS is delivered with MC-20TS cable assembly for High impedance operation with on-off switch.

Sensitivity: −52 dB (re mw/10 dynes/cm²)
Impedance: D-707E = 200 ohms
D-707TS = Hi Z
Dimension: 6" lg x 7/16" dia.
Weight: 5.7 oz.

H-24 Suspension
MSH-58E Flexible Shaft
SA-11/1 Stand adapter (Replacement)
AKG Stands



D-1000E

The D-1000E has been developed for the performing artist, record, broadcast and public address system applications. It features a mode selection switch to shape the microphones response characteristics.

B = Bass: Emphasizes smooth low frequency response.

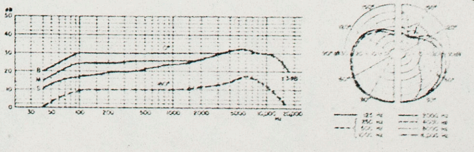
M = Medium: Attenuated bass; emphasizes middles and highs for greater working distance.

S = Sharp: Added upper mid-range brilliance and presence.

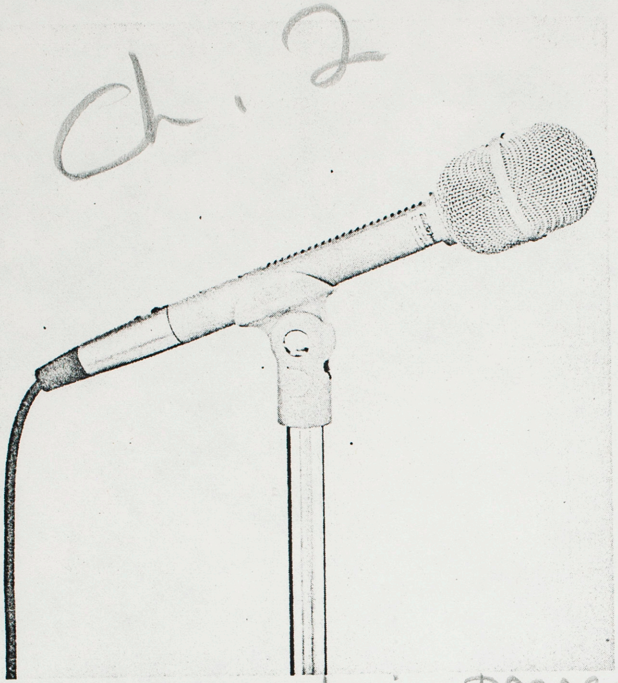
D-1000TS is being delivered with MC-20TS cable assembly for High impedance operation with on-off switch.

Sensitivity: −51 dB (re 1 mw/10 dynes/cm²)
Impedance: D-1000E = 200 ohm
D-1000TS = High Z
Dimension: 6" lg x 1-7/16" dia.
Weight: 9½ oz.

MSH-58E Flexible Shaft
W-4 Windscreen
SA-12/1 Stand Adapter (Replacement)
AKG Stands



RE16 DYNAMIC CARDIOID MICROPHONE



USED IN PARABOLIC DISH

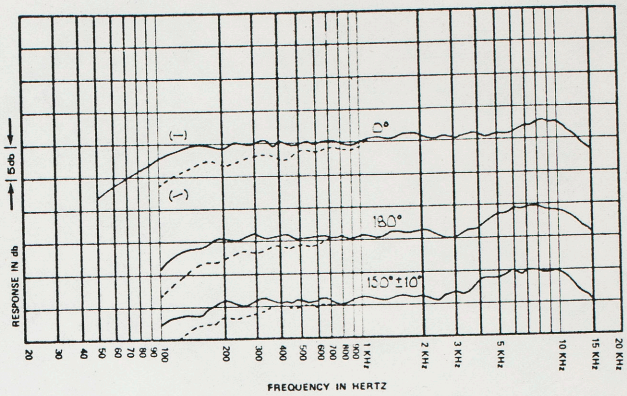


Figure 1 – Frequency Response

DESCRIPTION

The Electro-Voice Model RE16 is a dynamic cardioid microphone created especially for the most exacting professional use. It is much like the RE15 except that it uses a new and unique blast filter. The blast filter, an integral part of the RE16, makes possible hand-held use with lips almost touching the microphone and outdoor use without danger of "p-popping" or excessive wind noise. Emphasizing a major technological breakthrough, the RE16 features a degree of directional control so effective that frequency response is virtually independent of angular location of sound source. The result is a microphone that generates little or no off axis coloration, yet provides greatest possible rejection of unwanted sounds. A super cardioid, the RE16 provides its greatest rejection at 150° off axis. (Typical cardioids provide greatest rejection at 180°.) This assures greatest rejection in the horizontal plane when the microphone is tilted in its most natural position – 30° from horizontal (as on a boom or floor stand). An easily operated "bass-tilt" switch corrects spectrum balance for boom use and other longer reach situations.

A "hum buck" coil has been added (See Figure 4) to the RE16, in addition to its screw-machined steel outer casing, to insure rejection of hum under all conditions. When a dynamic microphone is subjected to extremely heavy magnetic fields caused by AC currents due to heavy stage lighting, proximity to power transformers, or due to many other conditions that can occur in remote operation, it has a tendency to pick up hum. The hum buck coil in the RE16 gives an additional 25 db of hum rejection. Hum pickup level is –125 dbm (re: .001 gauss field).

Using the mechanical nesting concept of design, by means of which the internal transducer parts are nested one within another, the RE16 transducer is a nearly solid mechanical structure that is highly resistant to damage from mechanical shock. The esclusive non-metallic Electro-Voice Acoustalloy diaphragm is virtually unaffected by extremes of atmospheric conditions. A carefully designed steel outer case provides additional mechanical protection. Finish is nonreflecting fawn beige Micomatte.

SPECIFICATIONS

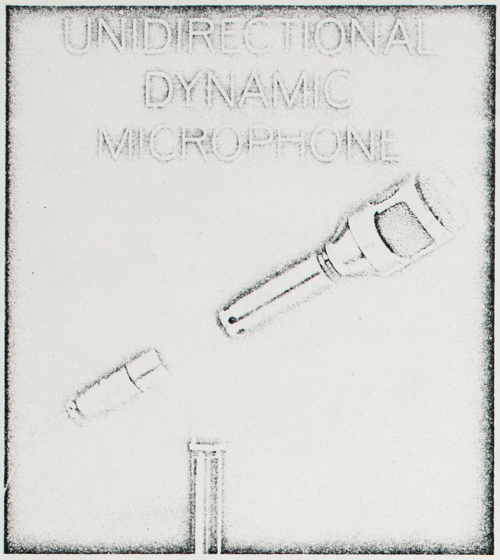
Element:	Dynamic
Frequency Response:	80 – 15,000 Hz
Polar Pattern:	Super Cardioid
Impedance:	Lo-Z (150 ohms nominal)
Output Level:	–56 db (0 db = 1 mw/10 dynes/cm²)
EIA Sensitivity Rating:	–150 db
Hum Pickup Level:	–125 dbm (re: .001 gauss field)
Diaphragm:	Electro-Voice Acoustalloy®
Case Material:	Steel
Dimensions:	7-3/8" long, 1-25/32" dia. (¾" shank diameter)
Finish:	Fawn beige Micomatte
Net Weight:	8 oz., not including cable
Cable:	18' 2-conductor shielded, broadcast type cable, synthetic rubber jacketed with Switchcraft A3F connector
Accessories Furnished:	Protective metal carrying case, Model 312A stand adapter
Optional Accessories:	Model 314 windscreen, Model 307 suspension mount, Model 421 or 422 desk stand.

ARCHITECTS' AND ENGINEERS' SPECIFICATIONS

The microphone shall be a super cardioid type with integral blast filter. It shall have a wide-range uniform frequency response from 80 to 15,000 Hz. Response at any angular position away from the major axis shall be essentially similar to the response on the major axis, attenuated uniformly at all frequencies by an amount appropriate to that angular position. Attenuation at

ENGINEERING DATA RE16 DYNAMIC CARDIOID MICROPHONE

SHURE SM53



PROFESSIONAL UNIDIRECTIONAL DYNAMIC MICROPHONE THE MOST VERSATILE MICROPHONE IN ITS CLASS

You'll find the Shure SM53 has far more uses, in far more situations—because it has been functionally engineered. It is the result of careful analysis of the varied applications a studio microphone must handle in today's across-the-board programming.

SMOOTH NATURAL RESPONSE

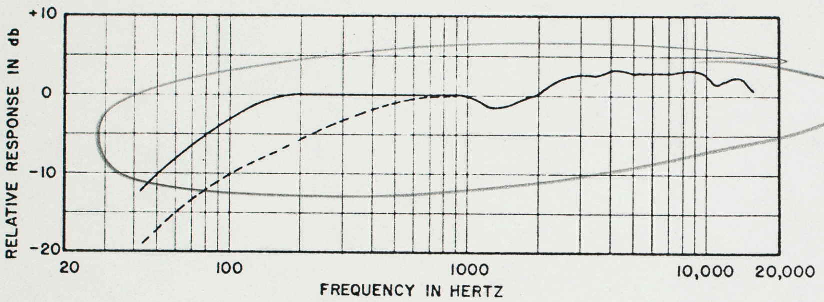
The frequency response of the SM53 is essentially flat across its broad frontal pickup area to the top end of the audible spectrum. Its response is natural, without strident peaks, without false coloration. (See charts below.)

SOME ADDITIONAL FEATURES

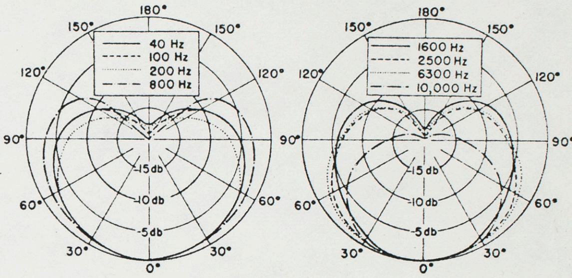
Changes in tonal quality due to accidental blockage of the rear entry ports are virtually eliminated because of the radial distribution of the acoustic entry. Built-in low end roll-off filter switch. Soft, neutral glare-free finish is ideally suited for on-camera use.

Ch. 3

FREQUENCY RESPONSE CURVE



POLAR PATTERNS



SPECIFICATIONS

Type: Dynamic

Frequency Response: 70 to 16,000 Hz. (See Response Curve above)

Polar Pattern: Cardioid (Unidirectional)
Response — Uniform with frequency, symmetrical about axis (See Polar Patterns above.)

Impedance: 150 ohms to permit proper match with any input from 50 through 250 ohms

Output Level: 1,000 Hz
Open Circuit Voltage: −81 db (0db = 1 volt per microbar; .09 millivolt per microbar)
Power Level: −58.5 db (0db = 1 milliwatt per 10 microbars)
E.I.A. Microphone Rating Gm (Sensitivity): −151 db (E.I.A. Standard SE-105, August, 1949)

Hum Sensitivity: −144 dbm at 1 milligauss

Bass Roll Off Switch: Response selector switch
See Frequency Response Curve above for roll-off response (dotted line).

Cable: 20 foot (6.1 m.) two conductor shielded Broadcast type with Cannon XLR-3-11C connector attached on microphone end.

Connector: Cannon XLR-3-12 type in microphone

Swivel Adapter: Positive action swivel to fit ⅝" —27 stand threads

Case: Aluminum and stainless-steel mesh

Case Finish: Matte metallic

Net Weight: (less cable) 8 ounces (227 grams)

Shipping Weight: 2 pounds, 2 ounces (964 grams)

ARCHITECT'S SPECIFICATIONS

The microphone shall be a Shure Model SM53 or equivalent. It shall be a moving-coil microphone with a frequency range of 70 to 16,000 Hz. It shall have a cardioid directional characteristic, with cancellation at the sides being approximately 6 db, and the cancellation at the rear being 15 to 20 db. The microphone shall have an impedance of 150 ohms. The microphone output shall be −58.5 db where 0 db equals 1 milliwatt for 10 microbars of sound pressure.

The microphone rating Gm (sensitivity) at 1,000 Hz. shall be within ±2 db of −151 db (E.I.A. Standard SE-105, August, 1949).

The microphone shall have a Response Selector switch to provide gradual low frequency roll-off.

The microphone shall be provided with a swivel adapter and a 20-foot, 2 conductor shielded cable having a Cannon XLR-3-11C connector attached. The microphone swivel adapter shall mount on a stand having ⅝" −27 thread.

The overall dimensions shall be 7.164" (182.0 mm) in length and 1.5" (38.1 mm) in diameter.

Shure Brothers, Inc. / Professional Products
222 Hartrey Avenue, Evanston, Illinois 60204
Phone: 312 328-9000 • Cable: Shuremicro

PAT. PENDING

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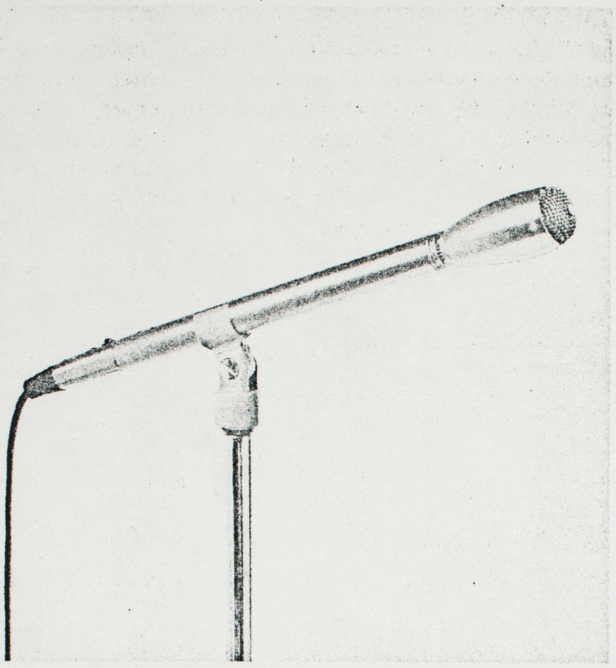
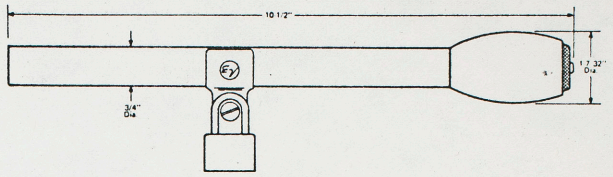
AL354

RE55 DYNAMIC OMNIDIRECTIONAL

ch. 4





SPECIFICATIONS

Generating Element:	Dynamic
Frequency Response:	40 to 20,000 Hz
Polar Pattern:	Omnidirectional
Impedance:	Lo-Z (150 ohms)
Output Level:	−55 db (0 db = 1 mw/10 dynes/cm^2)
EIA Sensitivity:	−149 db
Diaphragm:	Electro-Voice Acoustalloy®
Case Material:	Steel
Dimensions:	10-1/2" long (overall) 1-7/32" major diameter (shank 3/4" diameter)
Finish:	Non-reflecting matte satin nickel
Net Weight:	8-1/2 ounces, without cable
Cable:	18', 2-conductor, shielded, rubber jacketed, broadcast type cable with Switchcraft A3F connector.
Cable Connector:	Mates with Switchcraft A3F
Accessories Furnished:	Model 310A clamp and protective metal carrying case
Optional Accessories:	Model 311A "snap out" clamp

DESCRIPTION

The Electro-Voice Model RE55 is a dynamic omnidirectional microphone designed for the most demanding professional applications. Ideal for boom or stand mounting in recording and broadcast use, it is excellent also for close-up hand held use in stage and interview situations.

Smooth, wide range response, high output level, and low sensitivity to mechanical shock, combined with functional "easy to hold" styling, make the RE55 one of the most versatile professional microphones ever created. The non-reflecting, matte satin nickel finish eliminates hot spots for "on camera" use.

This microphone features the exclusive, non-metallic Electro-Voice Acoustalloy® diaphragm that assures very smooth frequency response and is impervious to damage from extremes of temperature and humidity.

APPLICATIONS

The RE55, an omnidirectional microphone, is outstanding in those situations where uniform sensitivity in all directions is desirable.

1. Frequency response, flat and uniform across the entire usable range makes the RE55 a good choice for:
   a. Symphony and other wide range recording situations where reverberation is not a problem.
   b. Calibration for use as a secondary standard for acoustic measurements.
2. Small diameter (¾-inch shank) makes the RE55 comfortable and convenient for hand held use by:
   a. Vocalists, for stage and television work.
   b. News reporters for interviews.

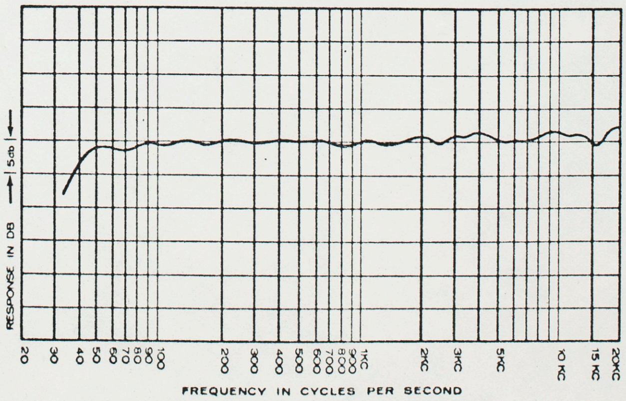


Figure 1 - Response Curve

ENGINEERING DATA RE55 DYNAMIC OMNIDIRECTIONAL MICROPHONE

Kevin Treuillian
202

The purpose of my project has been to try and investigate ways in which some of the acoustic characteristics of speech can be altered in a real time situation. My materials have been contact microphones, transducers, and different resonating surfaces, of the family of wood, metal, and membranes.

The reason that I am interested in the alteration of speech is that speech sounds, as opposed to meaningless vocal utterances, contain certain nuances of contour of frequency and rhythm that usually do not occur in vocal sounds that do not have corresponding conceptual meanings, even if these sounds are carefully composed. The reason for this would seem to be thatalanguage has had hundreds of years to evolve and develop, plus the fact that we as individuals use language fairly extensively almost every day of our lives, and hence attain a certain proficiency. The problem as I see it is that we are so strongly conditioned to percieve language as a means of communacation that we tend to overlook some of the more attractive acoustic aspects. Of course, poets, playwrites and authors have always utilized the musical aspects of language, but usually for the purpose of communacation of some conceptual, emotional or visual meaning that would not be fully implied by the purely acoustic aspects of speech. My intent then is to deprive speech sounds of some of the characterists that are necessary for word recognition, and by so doing, throwing into relief the purely surface aspects of the sound as heard through various kinds of transformation

For that purpose, I originaly constructed a box with three resonating surfaces as shown in accompanying diagram. Unfortunatly, since the "resonating surfaces" were attached solidly to the box, they lost a good deal of their ability to resonate, and thus neither the contact mike or the transducer were able to fulfill their relative functions very well.

Since that time, I have been able to do some experiments with some other surfaces and materials and have found that, for my purposes, free hanging metal objects such as pots, pans, hubcaps and so forth give a clear and audible ringing quality when activated by a transducer that is driven by an air mike.

The other diagram shows a system I have devised that can more efficiencntly aid my purpose. As can be seen, with this plan a speaker is used to activate the resonating surface with the contact mike (that is, a mechanical cone speaker)

The system is designed to be circular, so that after the vocal sounds have gone through processing through the gong with contact mike and hub-cap ~~of~~ or metal object with transducer, an air mike will pick up the processed sounds and feed them back into the system, until they gradually die away. The use of a tape delay of about two or three seconds avoids a situation of instant feedback.

For an improvised piece, the vocalist chooses a well known text and speaks into the microphone, improvising ~~imp~~ pauses of various lengths and different dynamics gradations. As the sound goes through the cycle then, there is a mixture and buildup of old sound in various states of transformation (depending on how many times it has gone through the cycle) and and new untransformed ~~so~~ vocal ~~sq~~ sounds from the vocalist. The overall sound from the audiences standpoint then consists of the vocalist, the speaker, the gong, and the transduced object.

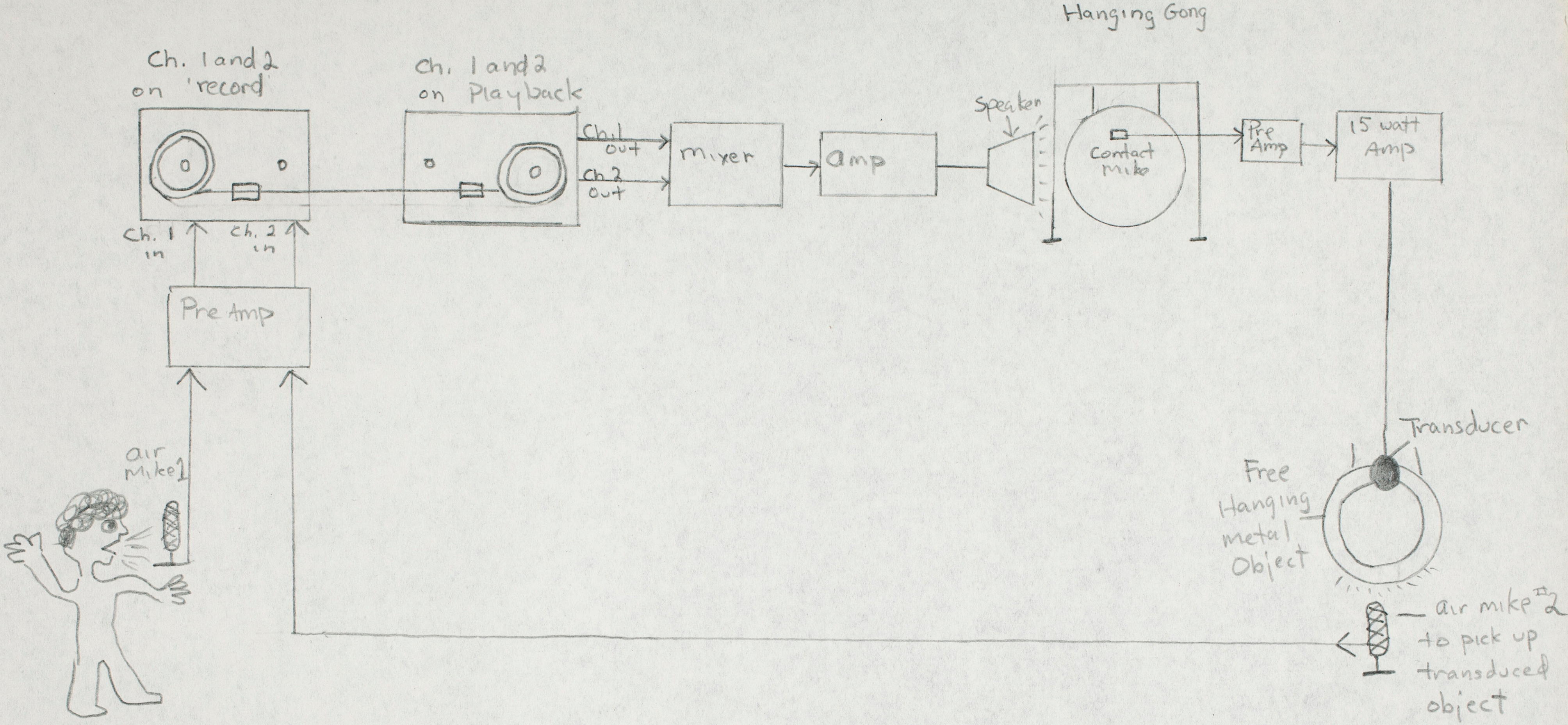

Hanging Gong
Ch. 1 and 2
on 'record'
Ch. 1 and 2
on Playback
Ch. 1
out
Ch 2
out
mixer
amp
Speaker
Contact
Mike
Pre
Amp
15 watt
Amp
Ch. 1
in
Ch. 2
in
Pre Amp
air
Mike 1
Transducer
Free
Hanging
metal
Object
— air mike #2
to pick up
transduced
object

Fig 1.

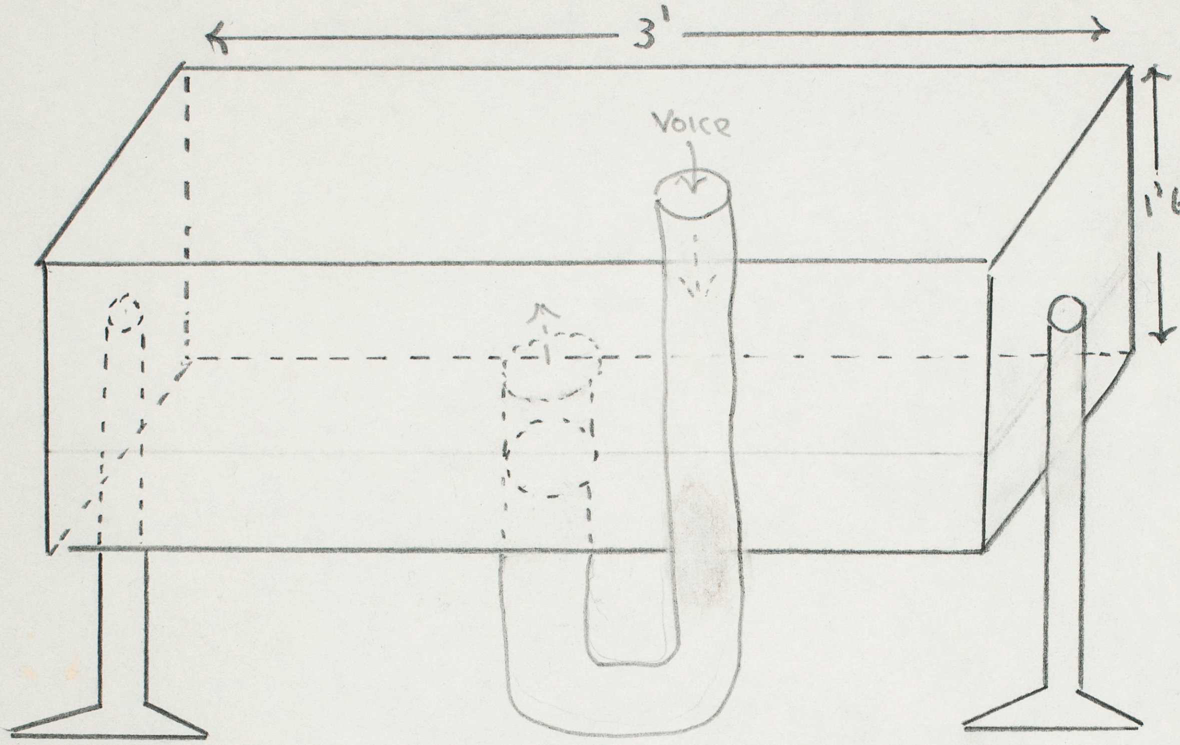
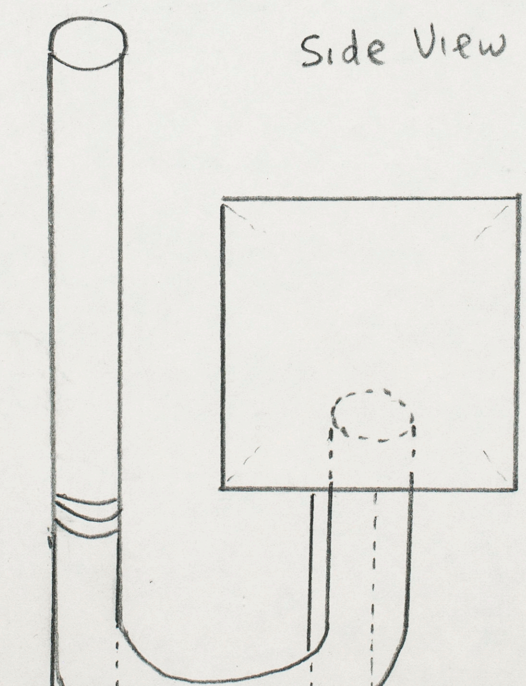


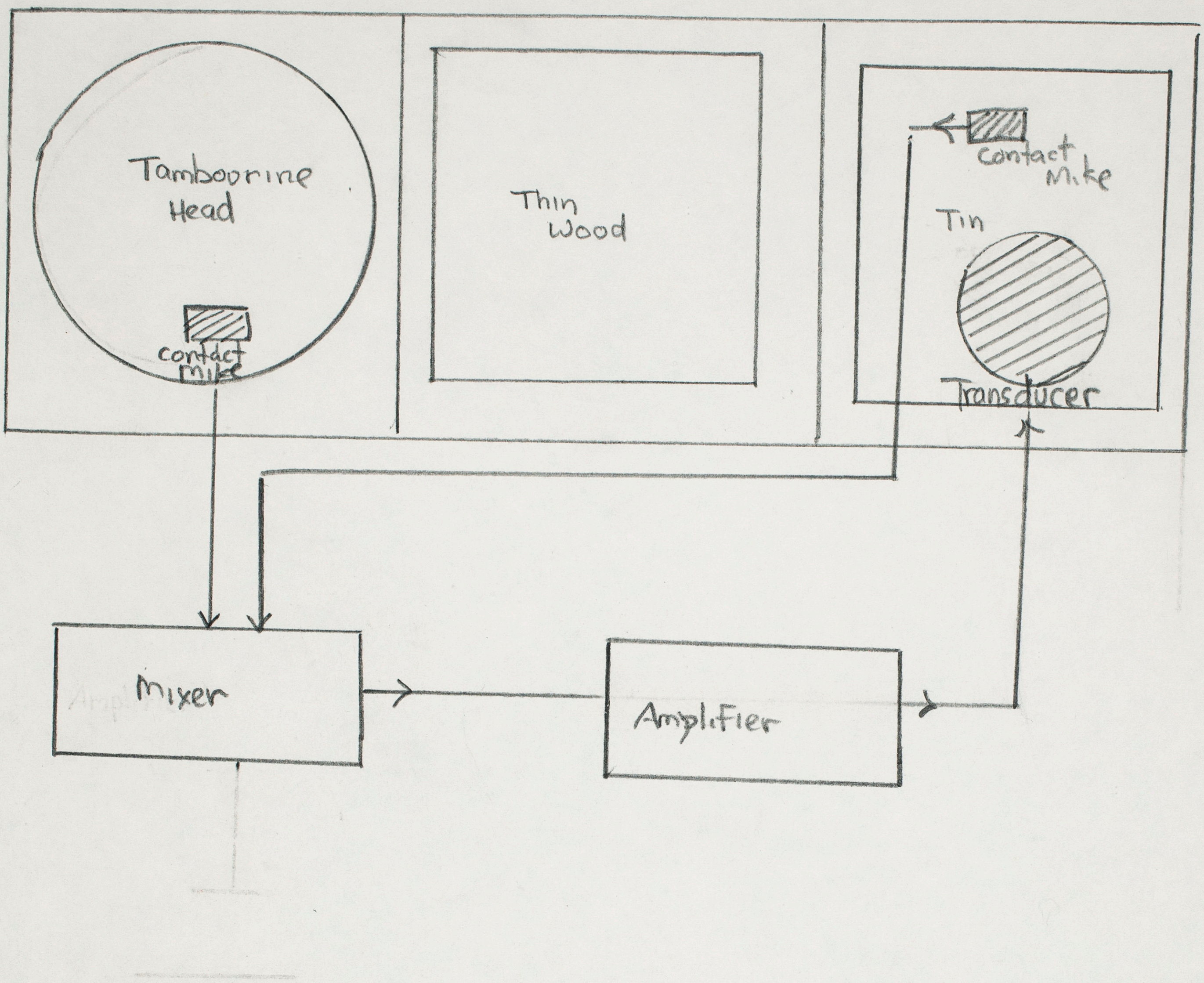
Fig. 2



Kevin Trevillian

Top View

Fig. 3



Thesis Proposal

Russell Lieblich

It is my intention in this paper to describe the research that I plan to undertake in the process of composing a work for symphony orchestra, tape and live electronic processing systems. The first area I shall discuss will be that of live processing of an orchestral ensemble. Refer to figure one while reading the following systems description.

Systems Description

1) The ensemble is being transduced within a sonic space defined by eight or more microphones(depending upon mixer availability). The microphone-ensemble configuration is as of yet undetermined and must awqit experimentation.

2)precision balance and color can be achieved by a high quality microphone to line level mixer. This mixer should have a matrix output facility, which is to say any given microphone input can be routed to any of the four outputs.

3)At this level the four output signals are routed to two different systems;

A) Interactive tape delay system

B) Analog signal processing network (APN)

These two subsystems do in fact communicate with each other,however for now I'll discuss them independently and discuss their inter-relationship later on.

Tape Delay System.

a) The tape delay network begins with a four channel delay(the time of which is controlled by the tape speed)

b) The output is both fedback into its initial input in channel one and routed to another stereo tape delay unit. The output of this unit is both fed back into the four channel tape machine at mixer two and into itself at mixer three.

c) May it also be noted that both of these tape delay units have direct access to the final audio output stage at mixer five.

Between mixer two which feeds the four channel tape machine and mixer three which feeds the stereo delay unit very interesting and complex signals can be created from the original input. A great deal of experimentation will have to be undertaken to fully understand the capabilities and mechanics of this process.

4) The analog processing modules will consist of voltage controlled filters, voltage controlled amplifiers, phase shifters and any other available electronic sound processing module.

5) The six x six matrix mixer preceeds these three independant circuits because it can provide an array of cross mixing which is not possible any other way. Literally every combination of outputs is possible.

6) The computer will behave in the capacity as master control for the analog system in real time. Control signals stored in the form of macros will control the filter settings (center band frequency and Q) define envelopes for the VCA's, control oscillator frequencies and operate a voltage controlled mixer.

The computer is drawn in the diagram in a dashed line because it is presently uncertain as to how the computer which is permanently housed at the Center for Music Experiment can directly communicate with peripherals in the Mandeville auditorium. I have thought of the following solutions:

1) Store the output of the computer on a low frequency response instrumentation tape recorder (ie brain wave tape recorder).

2) Transmit via radio the computer output

3) Do the performance at C.M.E.

7) The outputs of the analog processing circuits along with both the tape delay system outputs are fed into another matrix mixer(#5 on the schematic) which in turn is routed to a four channel panning facility (this too will be under computer control). These outputs are in turn mixed with the pre recorded tape and gated to the playback system.

Below is a brief outline of the general capabilities of the system;

a) Processing of direct signal from the orchestra

b) processing of delays plus unprocessed delays

c) processed delays plus direct signal

d) unprocessed delays plus processed direct signal

e) Direct processing plus delays plus direct signal plus processed delays

The combination of multilevel mixing, microphone placement, processing and delays will yield an enormous dimension to the orchestral sound pallete. One must also note that all of the above can be mixed with the pre recorded tape.

The next area I would like to address is that of creating the tape. The first step in dealing with this media is to pre record particular orchestral sonorities. These sonorities will include all twelve chromatic open fifths, each orchestrated in a slightly different manner. Additionally I will pre record some tone clusters. These particular sonorities were chosen because they will bear structural significance to the piece both harmonically and texturally.

The objective of this is to gather source material of an orchestral nature and to process it in a variety of ways such as filtering, phase shifting and tape manipulation. The difference between these proceedures and what I have described in the live processing is that in the studio I am going to work with long steady state sounds created by making a tape loop of a given orchestral sonority. The objective is to create a complex texture by multiple tracking. These techniques should yield sufficient material so that after a given amount of experimentation in mixing and panning the desired aesthetic will be realized on four channel tape.

Additionally I am interested in employing the timbre tuning system to generate sound that would both reinforce and complement the harmonic content of the source material. This can only be done after a spectral analysis was performed on the sound to in fact determine its respective harmonic content. The only advantage of using computer sound synthesis as opposed to analog means is to achieve the precise frequency control without drift which is not possible on analog equipment.

Conceptually the piece will deal on three dimensions; live ensemble, tape, live processing. The actual nature of the musical materials will be varied through out the course of the piece depending upon whether the orchestra is playing solo, in which case the music will be formed in closed episodes, or playing with the tape or as a source for the processing system. In the latter case virtually anything is feasible,however from prior experience in working with related systems it would compositionally suit my purposes to write both short high density gestures plus long sustain textures. Both of these extremes will yield the desired interplay between live sounds and processing that I am looking for.

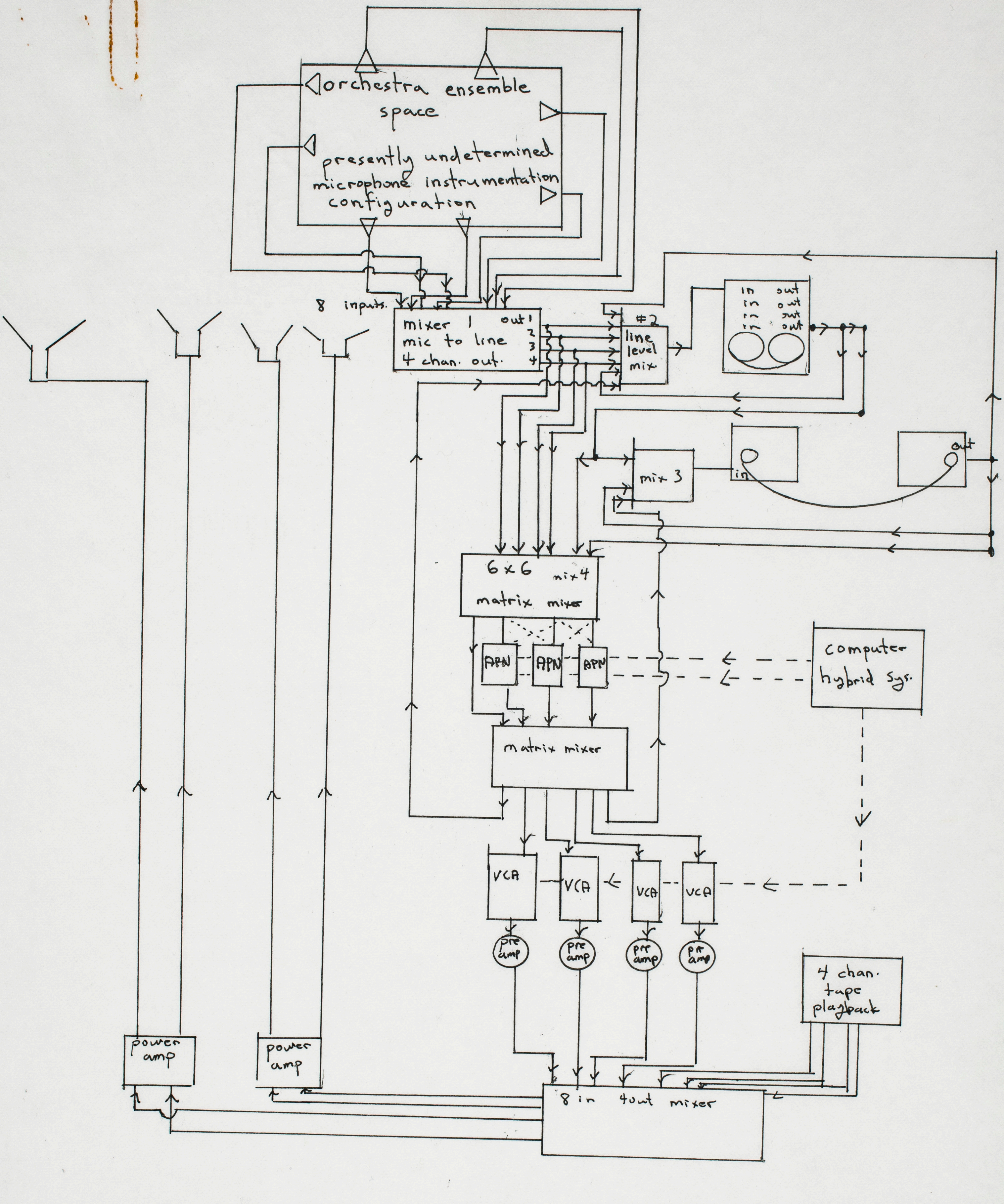


fig. 1

R. Lieblich

UNIVERSITY OF CALIFORNIA, SAN DIEGO

June 7

To Cherie

Date .... Time .... A.M. P.M.

WHILE YOU WERE OUT

Dr.
Mr. ....
Ms.

From ....

☐ Telephoned ☐ Will phone again ☐ Please phone
☐ Came to see you ☐ Will come again ☐ Rush

MESSAGE

David Casey
Hollis Gentry
Kevin
Terry Setter

Phone party at Where are their ?
papers for 202

Taken by P. O.

71475—136

Pauline,

here's the paper

I'll try to hit you

with some photos

come first part of

next week.

Chuck

Prelininary Report on Research for 202 seminar

Russell Lieblich

In this preliminary paper I would like to describe the research that I have presently involved with in relation to the 202 electronics seminar. My idea was initially to create an automatic preprogrammed microphone mixing facility by interfacing any given microphone configuration to voltage controlled amplifiers for additional processing by the C.M.E. hybrid system. In theory what was intended was to design an interactive system that would have the capabilities of effecting a timbre modulation of a sound source by selectively panning among the microphones. An obvious point that arises immediately is ""Why not control the VCA's with analog equipment?"

There are basically two answers, one pragmatic the other theoretical. On the practical side it just doesn't work due to leakage in the VCA'S when using an oscillator as a control signal. Additionally one can't obtain precise phase control between modules. This is of course crucial if one elects to have for example one device turning on while another is slowly turning off. For creating the timbre changes that I am speaking of this is fundamentally essential. Another critical limitation of analog equipment is the finite amount of waveforms that can be used as control signals in the kind of system I am describing. From experimentation it became clear that what was the most effective were slow linear waveforms with intricate phase relations as opposed to complex high frequency waveforms which in addition to creating leakage, create noise and unwanted sidebands. This might not be a disadvantage in analog sound synthesis, however for ~~for~~ sound processing of acoustic instruments it adds prominent enharmonic partials to the sound. The most effective control signals are sub-audio frequencies which hybrid can generate with unmatched precision. Furthermore hybrid offers the option to store a sequence of commands in the form of macros. It is this macro facility which can enable one to program a complex evolution of sonic events in time.

What I have thus far described is a model for a sound processing system which is fundamentally based on the interaction between disparate microphone properties and spatial location with respect to a sound source. The way in which these sound fields evolve in time is a function of both programming and the analog configuration.

When one actually approaches the equipment new realities unfold, generally in the domain of limitations. The system that I am now working with(which is the result of experimentation) incorporates an eight input high impedance output mixer. The high impedance output is necessary to be compatible with The SRA synthesizer. This fact eliminates the C.M.E. tascam board which doesn't deliver the appropriate signal level. There are four VCA's on the synthesizer to work with, hence four inputs. What I have been doing is using eight microphones and mixing two to each channel. So in effect the balance between each pair can be controlled manually on the mixing panel.

The most challenging problem in this project has been in writing an interesting program to control the system. Hybrid (when it is working) delivers sixteen channels of control signals, all of which can be used in the macro mode. One of the problems was in determing the voltage level at which the input microphone signal became audible. It turned out in the dynamic range of 0-7 the signal just becomes audible at the bottom of range 5. This was crucial in order to determine the temporal relationship between modules.

What adds an interesting dimension to the system is to split each signal which goes to the VCA's and run them in parallel with voltage controlled filters. The center frequency and Q of the filter can also be controlled by hybrid. The output of the VCA and filters are in turn mixed and appropriately balanced on the synthesizer mixer. This arrangement allows for vast possibilities in live performance. As indicated previously the bulk of the problem is in writing a program. Before this can even begin to happen one must experiment with each module to determine exactly how it operates on an input signal in conjunction with hybrid, the spatial location of the microphones and with respect to the other elements in the system. A final feature of this system which is most interesting is the four channel output capability. Considering the input can be anything from a solo instrument to a symphony orchestra the various processing networks can be routed to any of four channels in virtually any configuration.

Thus far I have finally figured out how it all basically works and interacts. However I have not been able to put the whole thing together to my satisfaction because the hybrid system has been inoperative for the past five weeks. This slight hindrance has virtually stopped all work on the programming. Hopefully all will be well again for future experiments.