Active Adaptive Noise Control: The First Phase

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Abstract

This report is a study of the development of an active, adaptive noise control system for a 4" I.D. duct. The study yielded the development of a series of circuits to be incorporated with a microprocessor in a continuous feedback loop. The four component circuits developed were: fixed and variable gain amplifiers, an AC/DC rectifier and a phase shift circuit. Each circuit was tested and operating characteristics and limitations for each component were determined. The complete system can attenuate signals between 90 hz and 1800 hz, with attenuations ranging from 28 to 34dB.

Acknowledgments

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I. Introduction

Over the past twenty years, attitudes towards man's environment have been expanded to include new parameters. One new aspect of the environment that has drawn much attention and research is that of noise pollution. Hearing loss has long been attributed to the natural process of aging. Recent studies have shown that, while some hearing loss is natural, a significant amount of hearing damage is the result of long-term exposure to high noise levels. As a result of this, the Occupational Safety and Health Act and the Noise Control Act have established noise level standards for various industries.

Noise initiates from many different types of sources. Machinery, traffic and aircraft all constitute major noise sources. Controlling noise at the source is often a difficult if not unfeasible solution to the noise problem. An alternative to this is trying to control the noise along the path between the source and the receiver. One common path that noise travels through is ductwork in buildings. Several methods have been developed to deal with this type of noise transmission.

The two most common forms of noise control in ducts are absorptive linings and reactive chambers. By lining interior sections of ducts with acoustically absorbing material, broadband noise (noise consisting of many different frequencies) can be partially attenuated. This system has three major drawbacks: first, the materials themselves are expensive; second, the linings degrade and have to be replaced; and third, the lining induces a pressure drop and consequently lowers the efficiency of the system. Reactive chambers are simply plenums which open into the duct. As a sound wave travels down the duct, it will actuate the air mass in the neck of the chamber (see Fig. 1). The air in the plenum acts as a mass and the air in the neck acts as a spring. When the noise wave is a certain frequency (the frequency being a function of the dimensions of the chamber), the spring-mass system will generate a pressure wave which will interfere with the noise wave traveling down the duct. This interference results in an attenuation of the noise wave. Although this is an inexpensive method of noise control, it is limited to noise of only one fixed frequency.

To combat the problems associated with the existing methods of noise control in ducts, the area of active noise control has been receiving much increased attention. This type of noise control involves actively imparting a destructive sound wave on the noise wave travelling down the duct. Variations in the frequency of noise that can be attenuated as well as low cost and eventual expansion into multiple frequency control are the key aspects to active noise control.

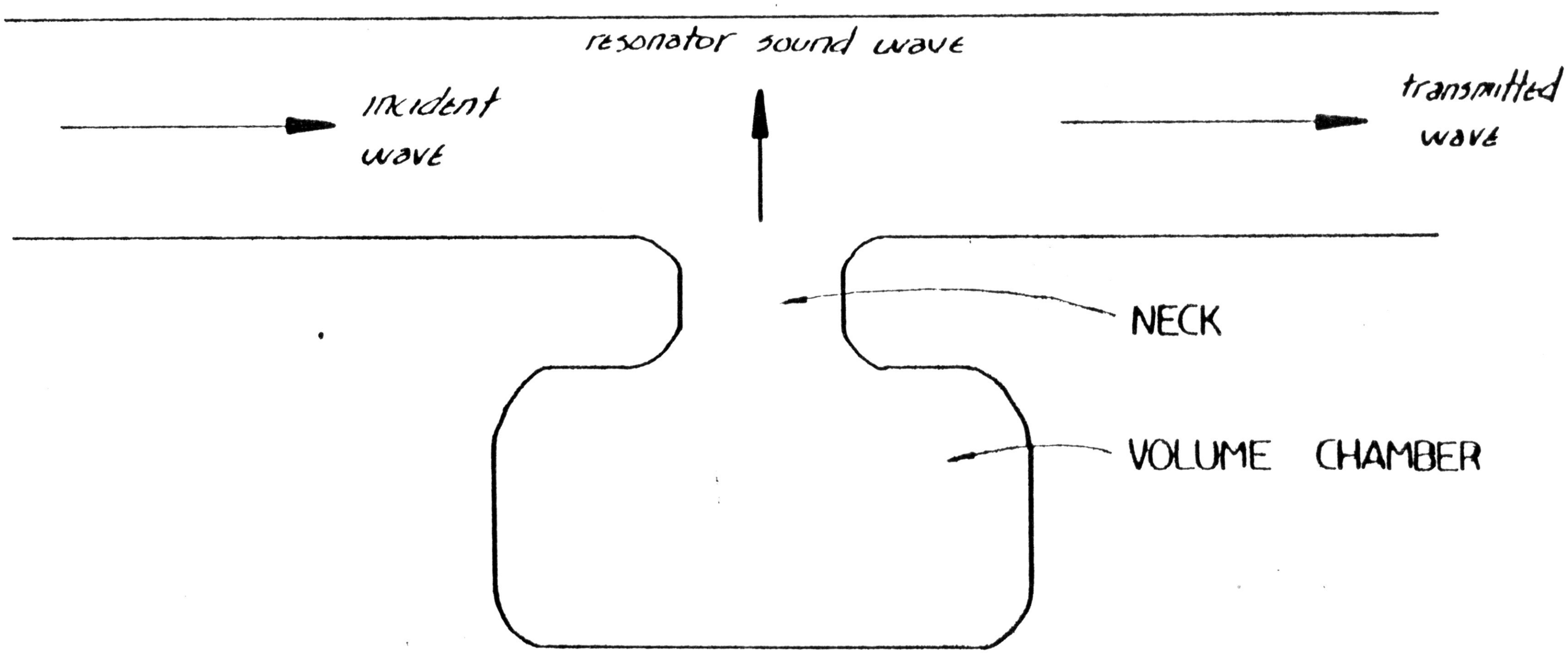


Figure 1 Hemholtz Resonator

II. Background

The theory of destructive interference is the basis of active noise control. This principal shows that if two planar waves that are of equal magnitude and 180° out of phase with one another, they completely interfere with each other. Theoretically, this results in total attenuation of both waves. Experimentally, it has been demonstrated that significant attenuation can be achieved with this approach [1,2,3,4].

During last year, an experimental set-up was built at Texas A&M to test this principal (see Fig. 2). The feasibility of this system was studied by Meline [5]. By using the system shown in Fig. 3, the phase and amplitude of the destructive wave could be manually adjusted and significant levels of attenuation were achieved (see Table 1). Because planar waves are a function of the geometry of the system and the frequency of the wave, only a certain range of frequencies could be studied in this particular geometry. This experimental range is from 90 Hz to 1800 Hz.

The next step in this process was to find a way to automatically adjust the amplitude and the phase of the destructive wave, thus making the system adaptive. Fluctuations in the noise wave of a given frequency could then be compensated for without human interference.

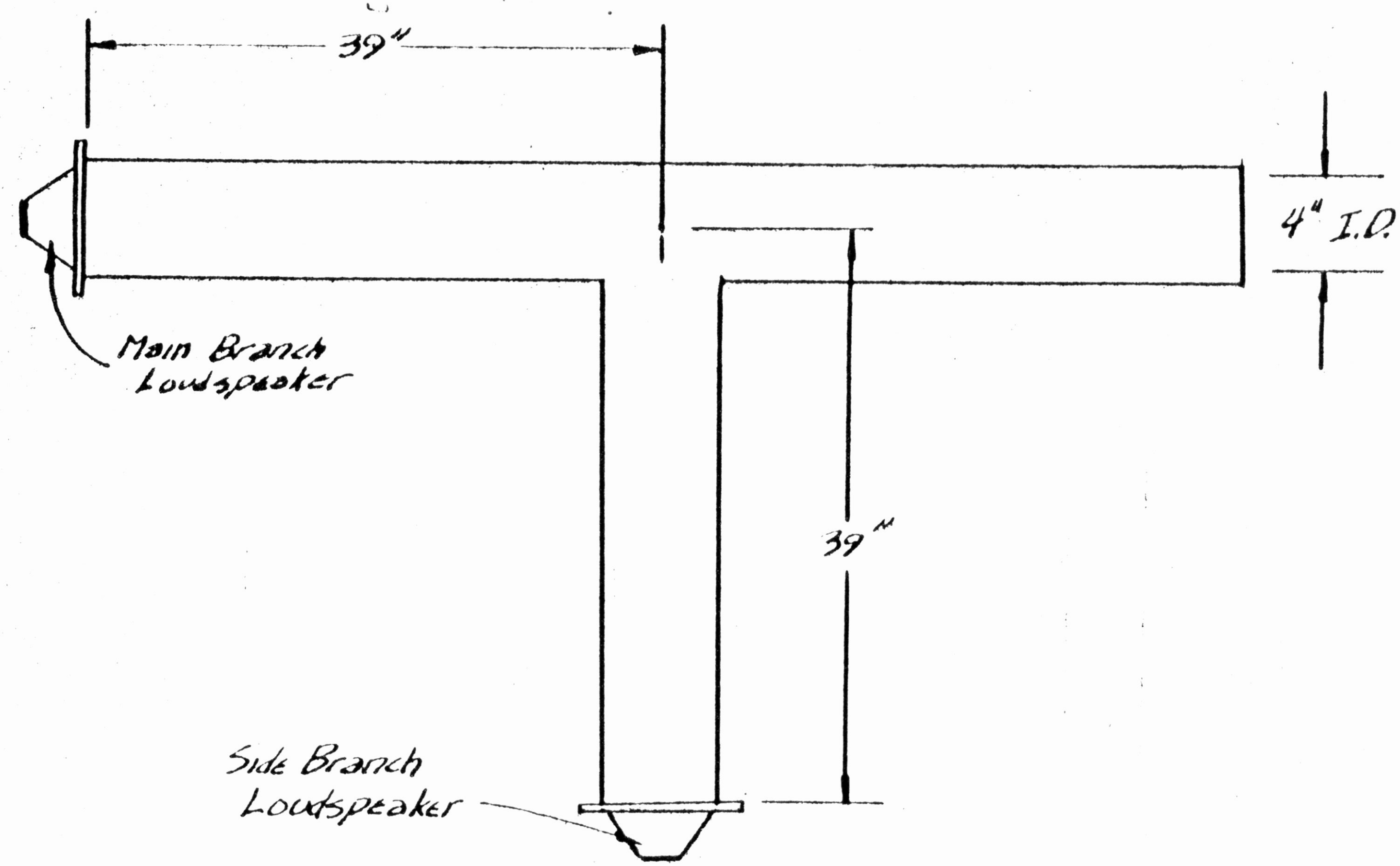


Figure 2 Initial System

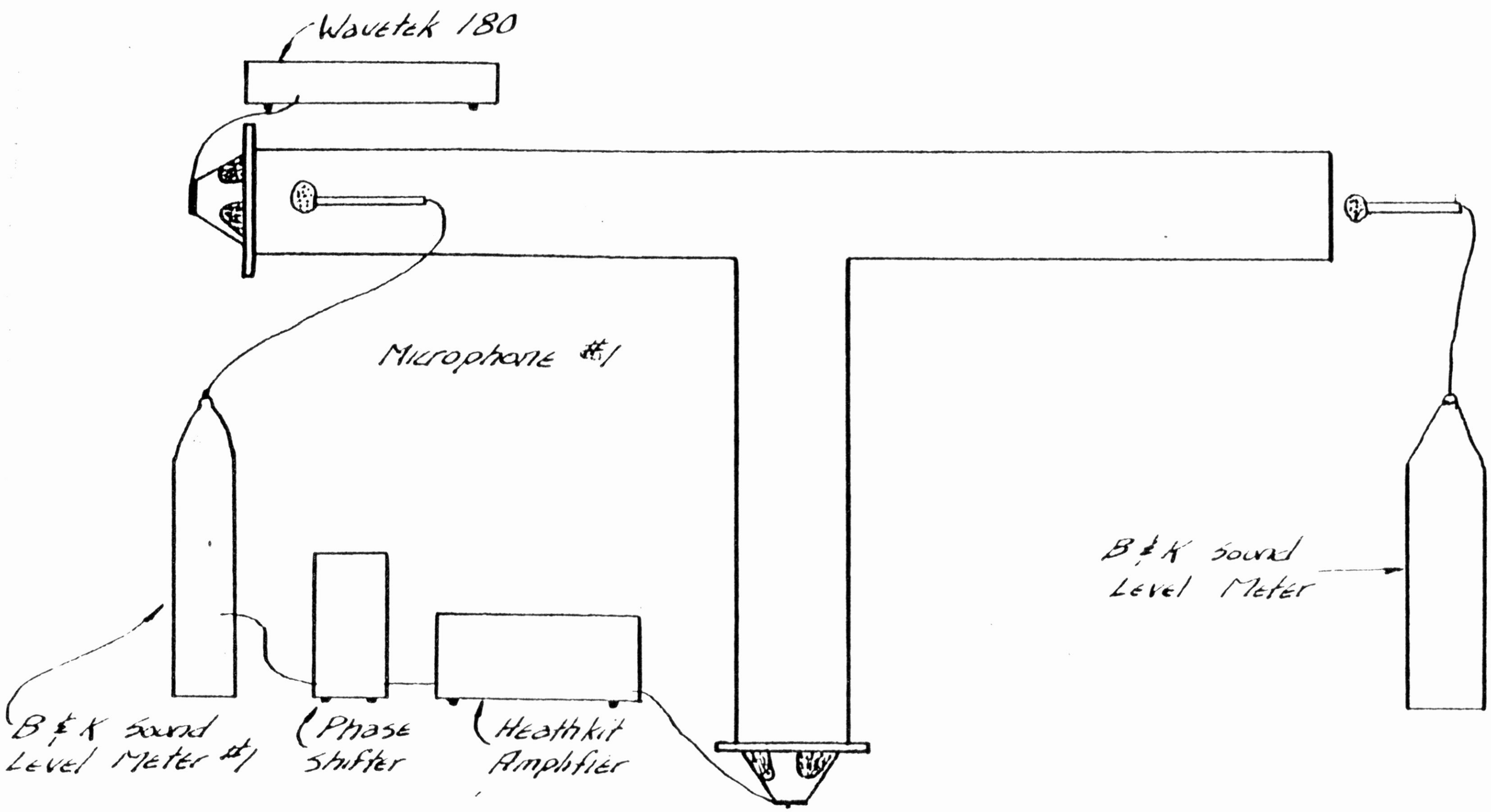


Figure 3 Previous Test System

Table 1

Results of 1982 Study

Frequency (Hz)	Attenuation (dB)
50	8
80	8
100	7
500	14
1000	24

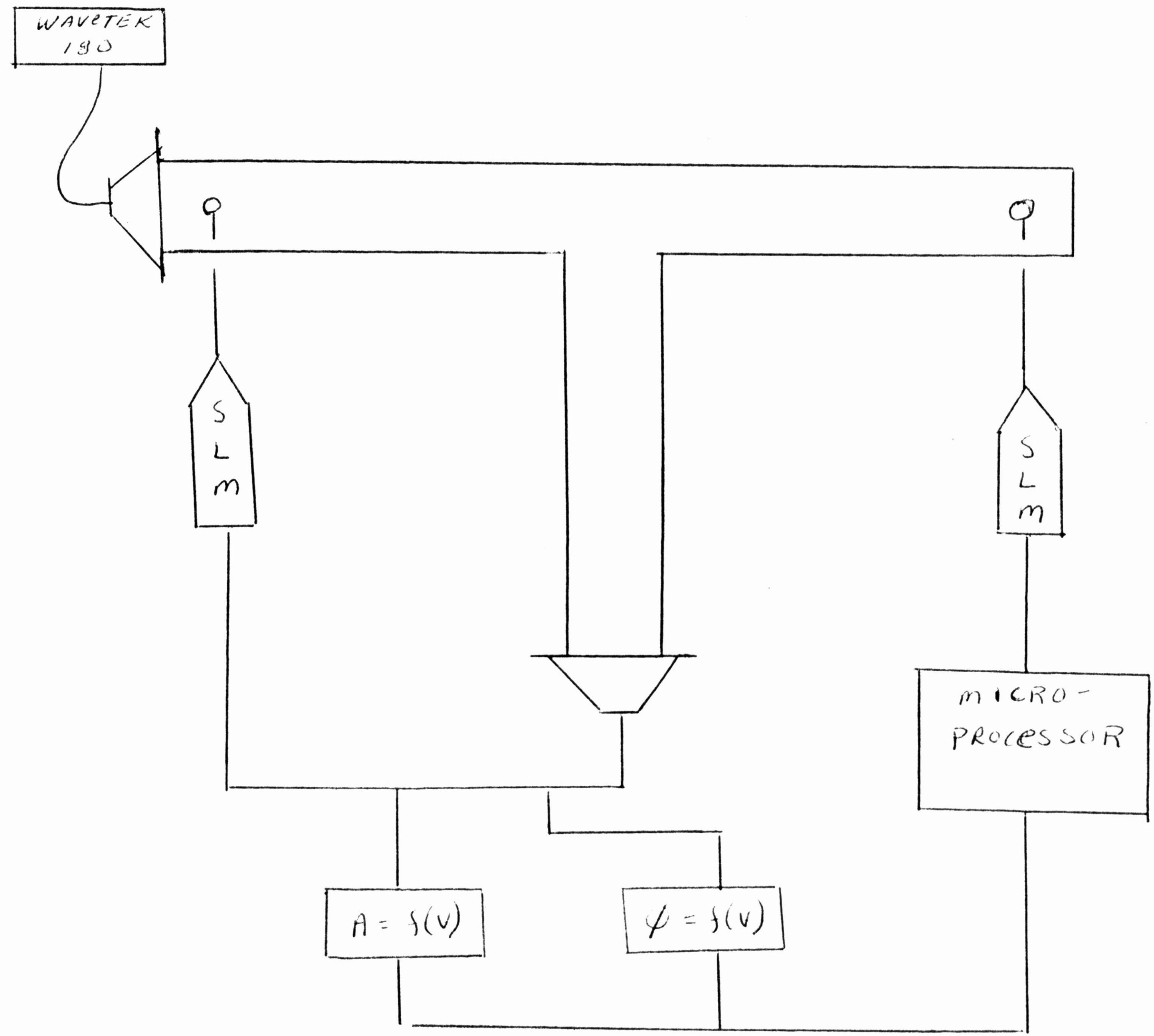
III. Conceptual Development

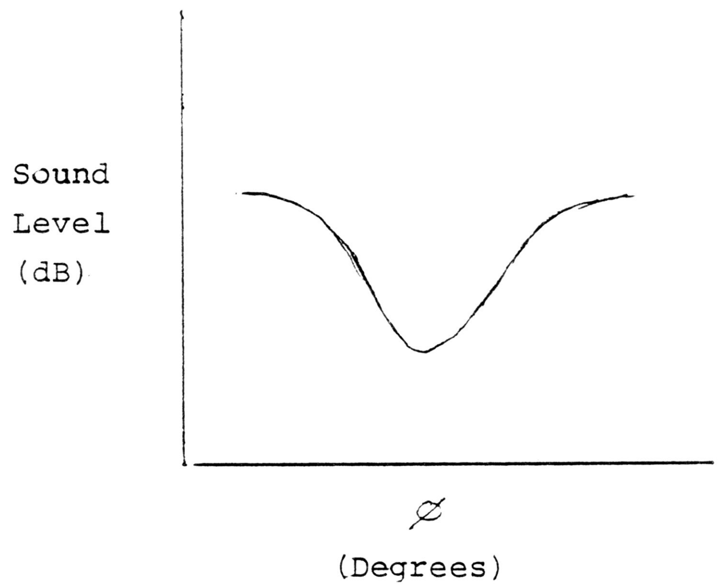
For high-speed adaptations to the noise signal, the incorporation of a microprocessor in a feedback loop was desired. The system in Fig. 4 was believed to the adequate for this process. In this set-up, the signal is fed directly into the microprocessor. The phase of the signal would be varied until a minimum was achieved (see Fig. 5). Then the amplitude would be varied until the lowest overall value of attenuation was achieved (see Fig. 6). These two adjustments are made by continuously sampling the transmitted sound level and adjusting the phase or gain.

The microprocessor chosen for this experimental set-up was the Miniature Instrument Computer (MINC) by Digital. This particular computer receives data in the form of voltage signals at its input ports. It then converts the input data to a digital signal and processes it. After processing, the signal is converted back to an analog signal at the output ports.

The microprocessor chosen had limitations that required alterations in the original conceptual set-up. The sampling rate, or the number of times the input voltage is read, is approximately 3600 samples per second. The greater the degree of processing of the signal that is desired, the lower the sampling rate that can be achieved. Because the sound wave is a time varying signal, this sampling rate

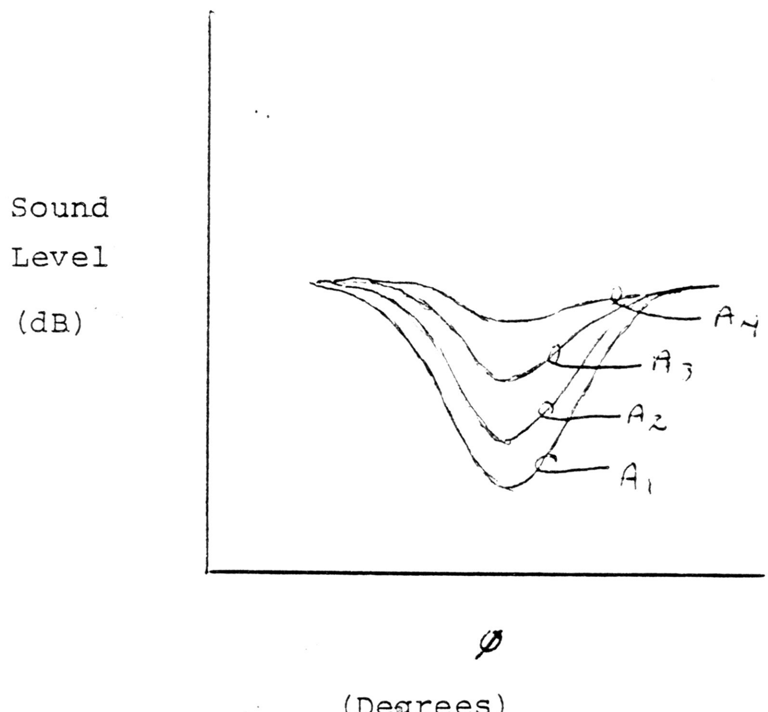
Figure 4 Conceptual Design





Sound Level as a Function of Phase

Figure 5



Note: A_1, A_2, A_3, A_4 represent specific values of amplitude

Sound Level as a Function of Phase and Amplitude

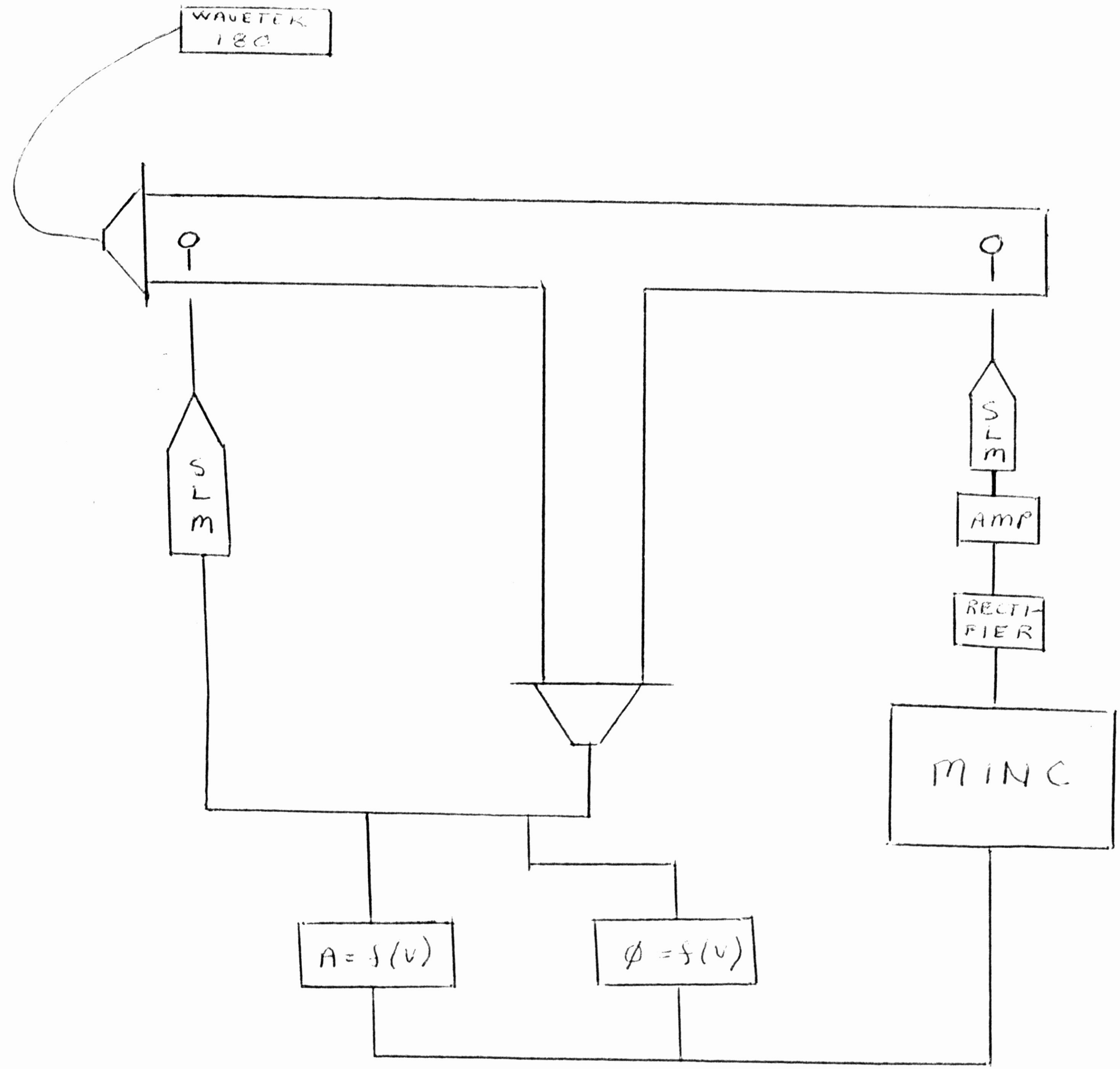
Figure 6

would have allowed accurate testing of signals of up to only a few hundred hertz. As a result, the signal needed processing before it entered the MINC.

This pre-processing was achieved through two circuits. The first circuit needed was an amplifier. The MINC needs input voltages in the range of 0 to 10 volts and the the measuring microphone puts out only a few millivolts. Therefore, a fixed gain amplifier was inserted in the feed-back loop. The second circuit needed to compensate for the MINC limitations was an AC/DC rectifier. This circuit would change the time varying signal to one of constant voltage. A schematic of the final system is shown in Fig.7. One of the benefits of this type of system is that only the relative magnitudes of the transmitted signal are important. The actual values of the amplitude and the phase shift that produce a minimum sound transmission are of little importance. The only objective of the procedure is to produce the minimum sound transmission.

To complete the circuitry necessary for the feedback loop, the existing phase shifter and variable gain amplifier had to be replace because they were designed to be manually adjusted. Both of these devices needed to be controlled by variations in voltage coming from the output ports of the MINC.

Figure 7 Final System



IV. Hardware Development

A. Voltage Driven Phase Shift Circuit

A phase shifter controlled by two variable resistance pots was built last year. The first idea investigated for this area of the system was to use the existing phase shifter and control the pots by using servo-motors. This idea was rejected for several reasons. First, the motors were expensive and their response was slower than desired. Second, upon testing the existing phase shifter, the circuit distortion was found to be so severe that the output signal barely resembled a sine wave.

The next idea investigated involved digitally controlling the signal. By dividing the total number of degrees of phase shift into discrete steps, a code system within the microprocessor would be developed that would assign each discrete step of phase shift to a code. Each code would activate an analog switch which would, in turn, select the correct resistor to provide the necessary phase shift. This system was ruled out due to its complexity, expense, and the amount of time necessary to build the system.

There appeared to be an existing integrated circuit that produced voltage controlled phase shift. After further investigation, it was discovered that the chip, made by National Semiconductor, did not have the capabilities necessary for this system.

The method chosen for this circuit incorporates the use of a photocell and the circuit diagram, along with descriptive notes on the circuit, can be found in Appendix A. In a photocell, as the voltage to the cell is increased, the intensity of the cell increases, causing a decrease in the resistance across the top of the cell. As the resistance goes down, the voltage across the capacitor increases causing the phase to change. There are many advantages to this system. Photocells are inexpensive and stable. They also allow for continuously variable phase shift as opposed to only discrete steps. Also, by incorporating operational amplifiers as opposed to transitors, essentially all of the circuit distortion has been eliminated within the voltage range being used.

B. Voltage Driven Amplifier Circuit

The amplifier used in previous experimentation was a Heathkit which offered little volume control, limiting the quality of response of the system. Like the old phase shifter, it was controlled by pots which could have been controlled by servo-motors. However, it was felt that a more responsive and less costly method of boltage controlled amplification could be found.

The concept of digitally controlling the amplitude, similar to the system considered for the phase shift, was examined. In this system, the amplitude gain range would be divided into a series of discrete steps, and an analog switching system similar to one described for the phase shifter would be used. In addition to expense and complexity, this system would be limited in the range of gain possible.

The system selected was similar to that used in the final phase shifter circuit. Amplification is controlled by a variable resistor amplifier. The resistance is varied by changes in voltage through a photocell. This system has proved both stable and economic. There is some crossover distortion in the circuit, but it has had no noticeable impact on system performance. Information on the circuit and a circuit diagram are included in Appendix A.

C. Amplifier - Rectifier Circuit

A simple fixed gain amplifier was inserted after the downstream sound level meter that filters the transmitted signal. This amplifier drives a transistor circuit that half wave rectifies the input signal. This rectifying process produces a current boost, and this current charges the capacitor. A parallel resistor bleeds off charge from the capacitor. Because the time constant associated with the circuit is very small, the DC output of this circuit closely follows the magnitude of the AC input through a direct proportionality constant. The current boost through the circuit yields very low ripple up to 6 volts of input. A circuit diagram and specifications can be found in Appendix A.

V. Results

Each circuit was individually tested to insure its proper functioning. Although there was some cross-over distortion in the amplifier circuit, it had no significant impact on the function of the complete system.

Due to time limitations, the software for the microprocessor has not yet been written. Two notes should be made with regard to this procedure. First, the same minimum seeking algorithm can be used to alter both the phase and the gain. Second, the sampling rate should be carefully set so as not to exceed the time constant of the rectifying circuit. If this were to happen, for every adjustment to the output voltage made, the capacitor would not have time to fully charge or discharge (achieve steady state) before the input to the rectifier would be changed again. Thus, the system would never reach a steady minimum value.

Because of delays in circuit construction, a full range of system tests was not performed. However, a sample of input frequencies was introduced to insure that the range of attenuations achieved were comparable to the those achieved with the old system. With an approximately 95 dB noise signal, the resulting attenuation level for the old system was between 10 and 29 dB. For the sample frequencies tested, the attenuations ranged from approximately 28 dB to 34dB.

VI. Conclusions and Recommendations

The new system developed is compact, efficient and more responsive than the old system. Although the software is not complete, this phase should represent no significant problems since the input and output parameters for each circuit have been defined. This completed network will provide maximum attenuation automatically for any set frequency.

The next phase of this project would be to expand it into multiple frequency capacity. By using multiple filters and a much higher speed microprocessor, each frequency could be separated and treated, and then the resulting signals could be recombined and introduced in the side duct. This would allow several different noise sources that were all transmitting down the same duct to be controlled by one device.

Eventually, a system like the one shown in Fig. 8 is desired. By feeding a signal directly into the microprocessor where a frequency spectrum analysis could be performed, the dominant noise frequencies could be selected automatically, treated and reintroduced in the side duct at high speed. This would result in the ability to control

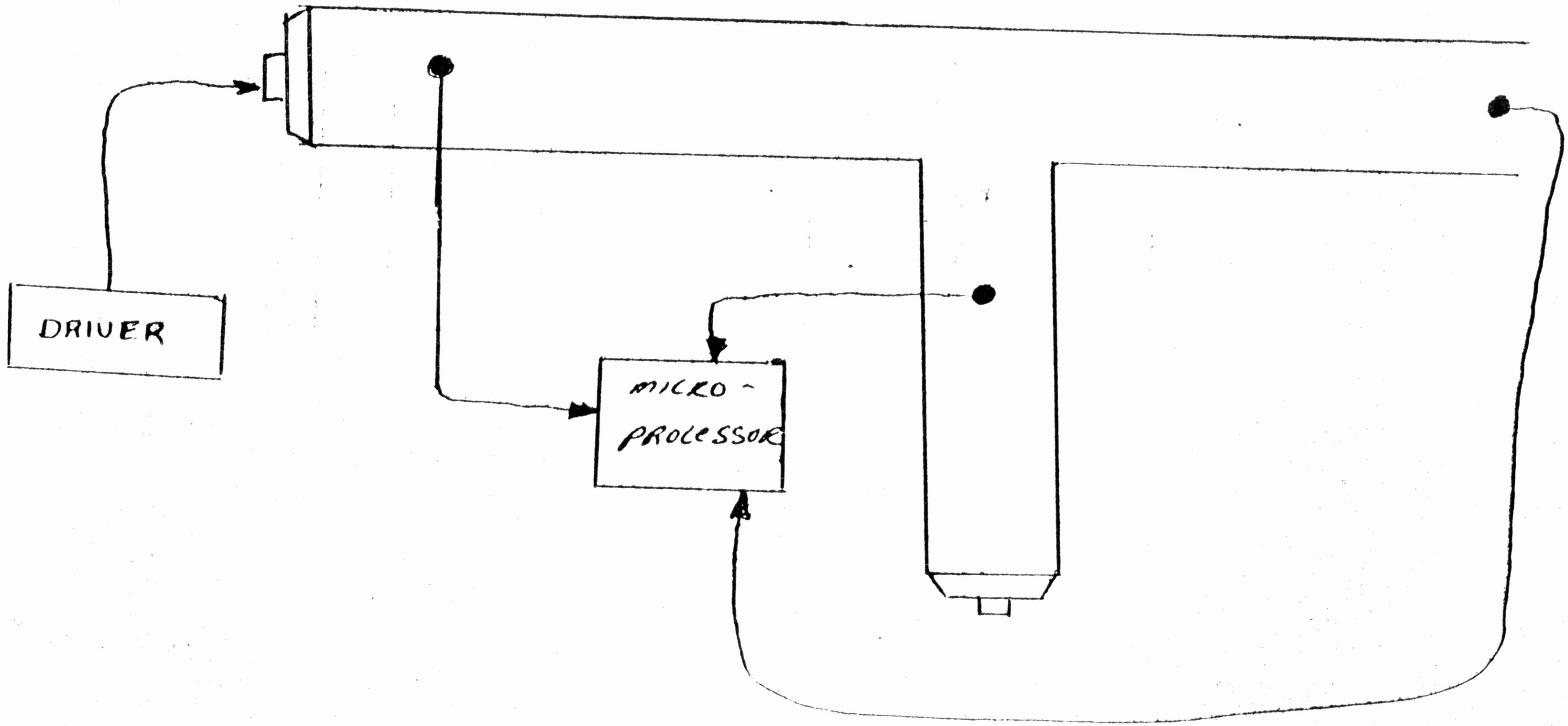


Figure 8 Future System

a significant number of transmitted noise signals in any combination or variation.

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[1]Lawrence E. Kinsler and Austin R. Frey; 1950, "Fundamentals of Acoustics", pp. 196-207.

[2]Robert Achgill, "Findings for an Experiment on Active Noise Control".

[3]Kevin Frank; 1980, "Noise Reduction Project".

[4]J.H. Poole and H.G. Leventhall; 1976, Journal of Sound and Vibration, Vol. 49; pp. 257-266; "An Experimental Study of Swinbanks' Method of Active Attenuation of Sound in Ducts".

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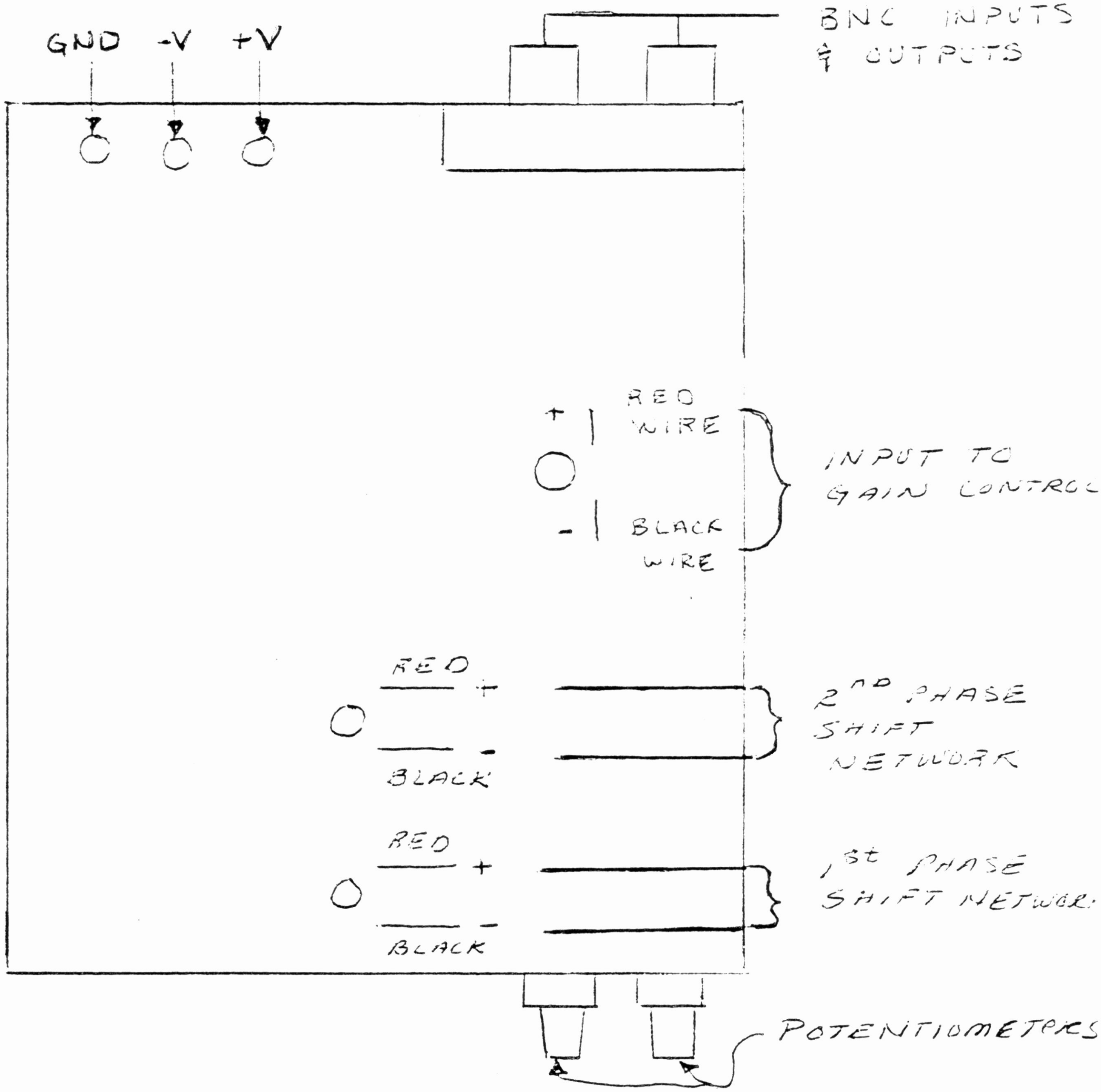
Appendix A

Overall System Controls



Power Supply Voltages - +10V, -10V. Adjust supply voltages before connecting circuit.

Use a bipolar supply, and connect the ground of the supply to the circuit ground.

Notes on the Phase Shift (PS) and Audio Amplifier (AA)

Input to PS/AA - Red wire

Ouput to PS/AA - Grey wire

Voltage Gain - Max $\cong$ 80, Gain control voltage $\leq$ 6.5V
Min $\cong$ 6, Gain control voltage = 8.5V

Power output - Maximum 3W continuous into 8

Maximum output voltage - 5V

Maximum input for rated output power = 65 mV

Input impedance $\cong$ 50 Ω

Distortion - Crossover distortion noticeable especially at low voltages.

Comments: Do not apply more than 10V to gain control.

Do not overdrive AA input (i.e. > 65mV).

Do not short output of AA.

Use signal source whose output impedance is 50 (i.e. Wavetek).

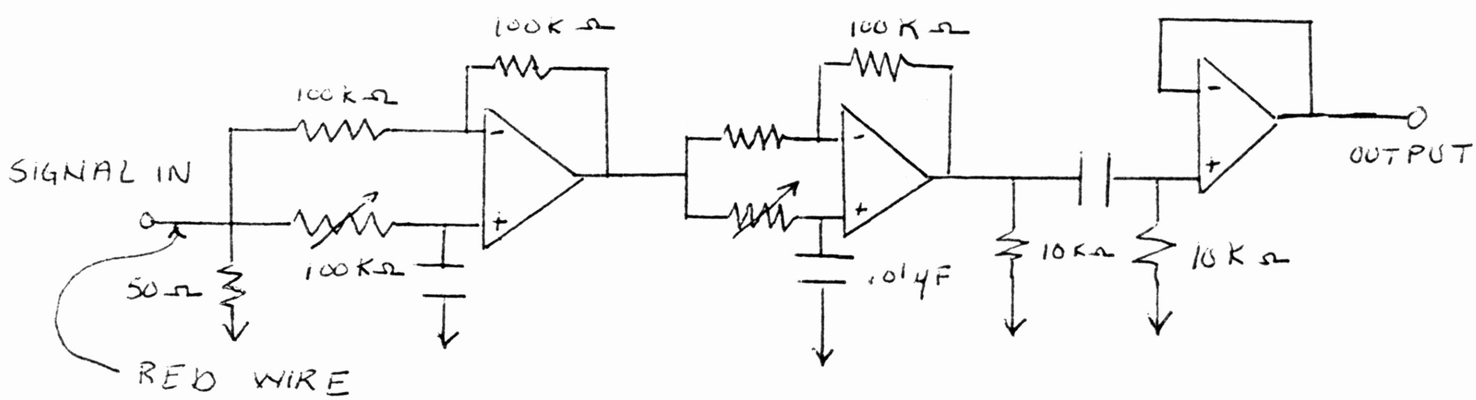
Phase Shift: Do not apply more than 10V to the phase shift circuit.

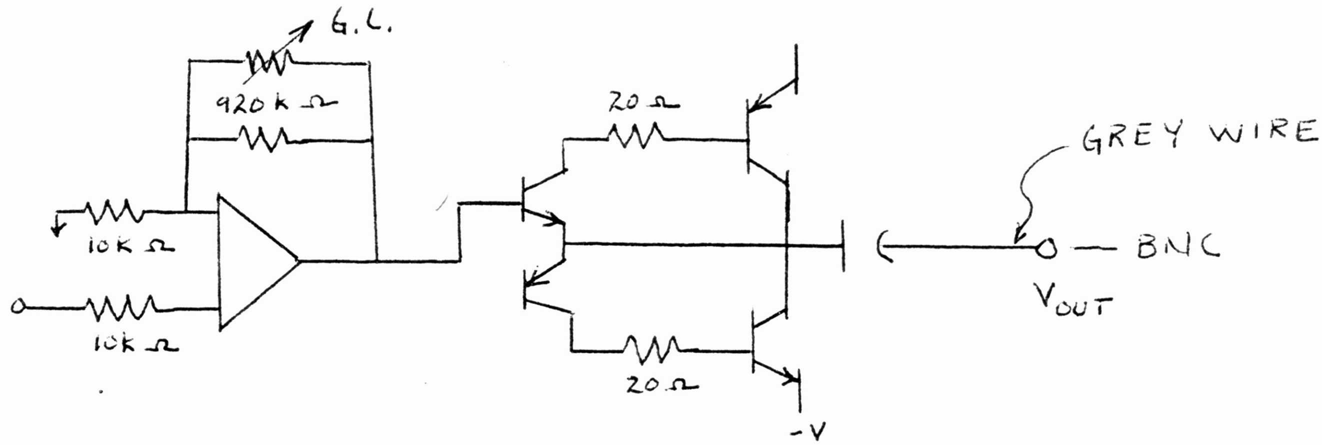
Phase Shift

1st Phase Shift Network: 0° Phase Shift = 10V
170° Phase Shift = 3V

2nd Phase Shift Network: 0° Phase Shift = 10V
140° Phase Shift = 3V

Total Phase Shift $\cong 310^{\circ}$





Audio Power Amplifier

Input to AA: Purple wire

Output to AA: Green wire

Voltage Gain $\cong$ 80V @ 1 khz

Power output = continuous 3W into 8 Ω

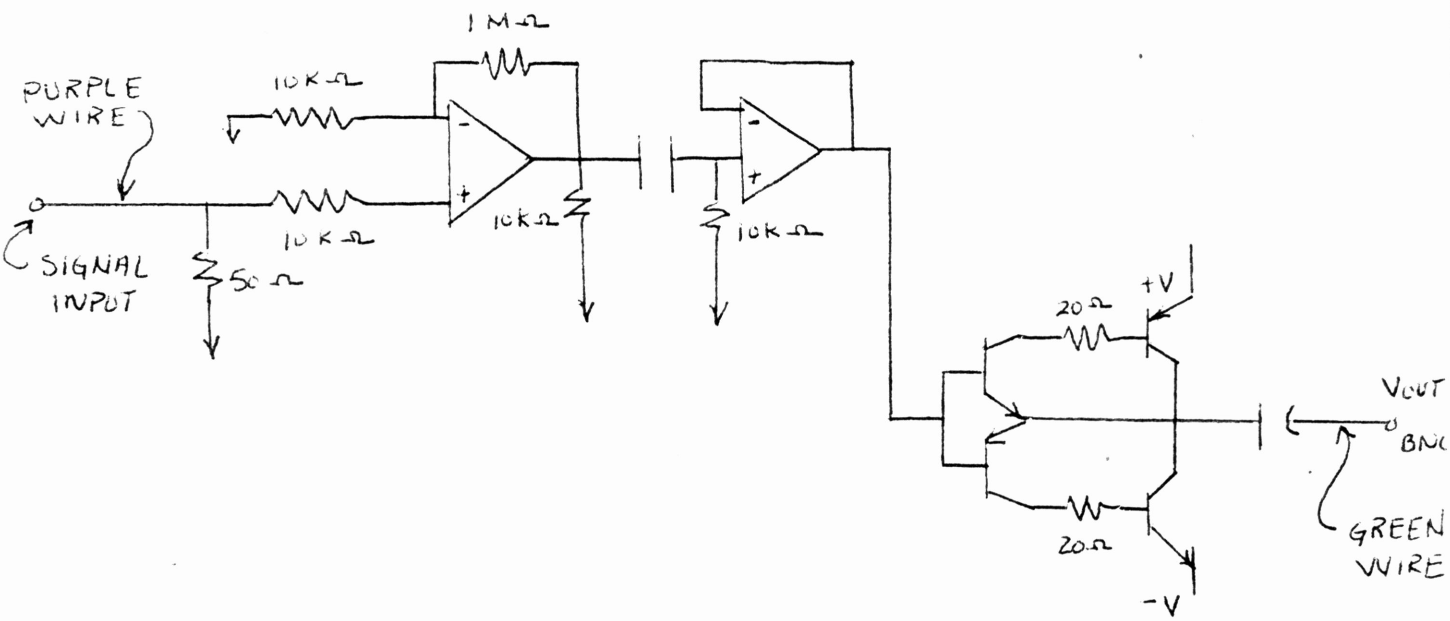
Maximum input for rated output power = 65mV

Input impedance $\cong$ 50 Ω

Distortion = Crossover distortion noticeable. Do not overdrive input of AA.

Comments: Do not short output as this will damage output amplifier.

Use a signal source hose output impedance is 50 Ω (Wavetek).



AC/DC Rectifier

Input to circuit: Blue wire

Ouptut to circuit: Yellow wire

Notes: Do not short output.

Ripple Voltage at 6V output = 100 mV

Minimum input voltage = 50mV