An Investigation into Active Noise Control in a Duct

by

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Approved by:

(Faculty Advisor) Carl D. Gerhold
Dr. Carl Gerhold

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ABSTRACT

This report is an investigative study of the feasibility and limitations of active noise control in a 4" I.D. duct. The study yeilded results comparable to those which were predicted theoretically. The limitations of the active noise control were defined by the dimensions of our duct. The frequencies that were discovered to be useful with our system ranged from a lower frequency of 90 Hertz to an upper frequency of 1800 Hertz. The final system, which used a microphone as the signal transducer, was capable of attenuations between 10 and 29 decibels within our frequency range.

ACKNOWLEDGMENTS

For the opportunity to participate in this program, I wish to thank Dr. Mel Friedman. For their unending patience with me, I would especially like to thank both Dr. Carl Gerhold and Mr. Dana Stanton.

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INTRODUCTION

Noise is a form of pollution which can cause not only physical damage but also does psychological harm. The control of noise is essential if the quality of life is to be preserved in an industrialized society. To combat this noise pollution, a method must be devised to systematically and significantly reduce the sound level on a wide range of frequencies. Existing methods of attenuating sound in ducts include absorptive silencers and lined ducts, but these are good for medium to high frequencies. Low frequency noise reduction is accomplished by sound cancellation using reactive techniques. Such a silencer, referred to as a resonator, is effective only over a narrow range of frequencies; and once tuned, operates only at one center frequency. One method believed to be useful in reducing sound levels on a wide range of frequencies in ducts or tubes is active noise control. Previous work by students at Texas A&M University [2,3] and others elswhere [4,6] indicate that the concept of active noise control is valid. Active noise control uses sound cancellation in which the noise signal is electronically enhanced and fed back through the side branch. Such a feedback system can, thus, adapt to variation in the main duct noise signal. Development of suitable electronics

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can expand the range of frequencies over which the noise cancellation occurs. Our research starts with a very basic system and progresses to an active, adaptive system.

BACKGROUND

A method commonly employed to control noise in ducts is destructive interference between the incident sound wave propagating down the duct and the sound wave emitted from a side branch resonator, called a Helmholtz resonator; a method first introduced forty years ago. The resonator, as shown in Figure 1, consists of a volume chamber connected to the main duct by a neck. The air trapped in the chamber acts as a spring and the slug of air in the neck acts as a mass. The pressure fluctuations due to the sound wave travelling down the duct cause the mass of air in the neck to vibrate, producing a pressure wave which is fed back into the duct. When the frequency of the sound wave in the duct is the same as the resonant frequency of the resonator, the pressure wave from the mass of air in the neck is a maximum and out of phase with the pressure wave in the duct. At this frequency, the two pressure waves add destructively and the sound pressure transmitted down the duct is a minimum. This form of noise control is considered passive because the frequency at which destructive interference occurs is fixed by the volume of the chamber and the dimensions of the neck. For low frequencies, the size of the resonating chamber can be prohibitively large. For this reason, the resonating chamber is replaced by a duct with a sound source as shown in Figure 2. This is the distinguishing characteristic between passive and active noise control. Two waves, a "noise" wave in the main duct, and a "destructive" wave in the side branch are generated and combine at the intersection. The phase of the destructive wave is adjusted so that the waves are 180 degrees out of phase when they combine. As shown by the mathematical proof in Appendix A, the transmitted wave should be a minimum when the noise and destructive waves are 180 degrees out of phase. Likewise, the transmitted wave should be a maximum when the phase shift is 0 degrees. Also, from the proof, it is evident that both the noise wave and the destructive wave must have the same magnitude. Previous recent work by M.A. Swinbanks [4] suggests that not only does this theory work, but is a practical method of reducing sound levels in ducts. In Swinbanks' work, sound sources (speakers) were placed directly on the duct wall to produce the destructive wave. Other works [2,3,6,7] have also suggested that active noise control is possible and can be made into a practical sound reduction system.

RECORD OF PROCEDURES

The first task was to investigate the feasibility of the active noise control concept and the frequency limitations of active noise control in our particular system. Theoretically, there should be a lower frequency below which, our system would not be useful, and an upper frequency above which, our system would not be useful. These cut-off frequencies exist because all mathematical properties derived for our system have been based on planar wave theory; i.e. planar sound waves are the basic assumption of the mathematical model. Intuitively, a plane wave may be described as a wave such that it is far enough away from its point source that the wave front may be considered planar. The reflection of sound from the side walls of the duct tend to channel the sound into a plane wave configuration at a distance down the duct from the source. A plane wave can exist in a duct provided that the wavelength of the sound is: 1) short in comparison to the duct length (lower cut-off frequency) and 2) long in comparison to the duct diameter (upper cut-off frequency). The lower cut-off frequency exists when the $\frac{1}{4}$ wavelength of the sound wave equals the duct length. This lower cut-off frequency is calculated to be 90 Hertz. When the $\frac{1}{2}$ wavelength is less than the duct diameter, the wave in the tube becomes non-planar. This upper cut-off frequency is calculated to be 1800 Hertz. A graph of expected results is shown in Figure 3.

THEORY TEST

To test our theory, we used the Wavetek 180 and the Wavetek 186 with it as shown in Figure 4. The wavetek 180 was used to drive the main speaker and also as a phase reference for the Wavetek 186. The Wavetek 186 has a built-in variable phase shift capability and was used to drive the side branch speaker. Thus, essentially the same signal was used to drive both loudspeakers. Such a system has no practical application, and is used only to test the feasibility of noise cancellation in a duct and the frequency limitations imposed by the duct dimensions. To record the sound level output, a B&K sound level meter was used with the microphone set just outside the end of the tube. Due to this exterior position of the microphone, noise emission radiated outside the duct work through the backside of the speaker cones greatly affected the sound level measurements. To alleviate this problem, the speakers were encased in insulated boxes. Continuing with the investigation, frequencies outside our designated useful range (90-1800 Hz) were explored for their noise control characteristics. Frequencies below the lower cut-off limit at 50 Hz and 80 Hz were explored. By exploring the frequency we mean that both function generators were turned on and the phase of the side branch pressure wave was changed at 15 degree increments. Plots of the sound level output versus the phase shift (as read iff the function generator) are shown in Appendix B. Frequencies above the upper cut-off frequency at 2000, 2500, 3500, and 4000 Hz were also explored for their characteristics. The plots of these are also in

Appendix B. Along with these are plots of frequencies within the useful range. Three frequencies within the useful range at 100, 500, and 1000 Hz were explored to use for comparison.

RESULTS OF FIRST TEST

From the plots generated from the data, it can be seen that for all frequencies, regardless of whether it is in the useful range or not, the minimum and maximum sound level outputs occur 180 degrees apart. These maximum reductions (maximum S.L.-minimum S.L.) were tabulated and a plot of the maximum reduction versus frequency is shown in Figure 5. While sound attenuation is measured both inside and outside the useful range, the sound reduction is greater within the range. These two facts do not prove the theory, but strongly suggest its validity. The data in Figure 5 show some scatter which was not expected. As can be seen, the graph looks somewhat like a damped sinusoid. In Swinbanks' works 4 he produces a downstream response as shown in Figure 6. While the two curves are not the same, they show enough resemblance to lend credibility to our results. The damping effect evident in the graph is difficult to explain, but it may be attributed to unequal magnitudes of the two signals or some inherent non-linearities in our system.

THE FINAL EXPERIMENTAL SYSTEM

After establishing the feasibility of active noise control, we modified the system to make it adaptive. An adaptive noise control system must be able to pick up an unknown sound, process it, and send that processed signal down the side branch to destructively combine with the main branch wave. An ideal adaptive system is shown in Figure 7. To accomplish this, we used a microphone in the duct, as shown in Figure 7, as our transducing device. The microphone signal was passed through a sound level meter (#1) for preamplification and to filter out extraneous noise. The signal from the sound level meter was phase shifted, amplified, and sent to the side branch loudspeaker. Our system is shown in Figure 8. We used the Wavetek 180 function generator to drive the main speaker. A seperate B&K sound level meter was used to monitor the sound level output. For this system we still expected a useful frequency range of 90-1800 Hz and a similar graph to that in Figure 3. When we began the experimental procedure, the system seemed to have an inherent instability in it. With all components turned on, and the microphone #1 inserted into the tube, the system resonated. We felt, at first, that it was due to a standing wave caused by the finite length of the duct. At the end of the tube where the measurements are taken, an impedance mismatch occurs, therefore sending a reflected wave back down the tube producing the resonance. To alleviate what we felt was the problem, we structured a cone out of sound absorbant material. The purpose of this cone was to produce an anechoic ending to the tube. This did not help the situation any. One other possibility was that the feedback signal was unstable. In the original system configuration, little attention was paid to the sound level meter #1 except for its filtering mechanism. Thus, no consideration was given to the decibel setting on the meter. The microphone signal was assumed to go straight through the sound level meter without regard for the decibel setting. It was discovered that by a low enough dB setting on the meter, we could generate tones of all frequencies available on the filter system, simply by changing the filter frequency. Conversely, there was a minimum decibel setting for each filter frequency which would eliminate resonance. Thus, it was found that when the sound level meter was set to read too low a value, the microphone actually measured the ambient sound level. This extraneous noise signal was then amplified and sent back into the side branch loudspeaker. Thus, the system was overly sensitive and became unstable. Adjusting the sound level meter attenuator to a value on the order of the main duct noise signal, and well above the ambient sound level eliminated this instability. The purpose of the experiments which followed was to establish the possibility of sound attenuation in our system, determine its boundaries, and, hopefully, lay the groundwork for future study. Using the same procedures as before, we were able to generate a graph of the maximum attenuation versus frequency for our system (Figure 9). The maximum attenuation for this system is the sound level with just the main speaker on, minus the sound level when the two waves are 180 degrees out of phase. This will give a true reduction in noise, rather than a maximum to minimum reading. The results show trends up at, or around 90 Hz and a down trend at 1800 Hz. This, as was discussed earlier, was exactly what we expected. To get an idea of how it compared, we generated another curve for the "2-Wavetek" system from before. The results are shown in Figure 10.

DISCUSSION OF RESULTS

The fact that there was an upward trend where we expected, and a downward trend where we expected, tells us that the system will work for planar waves only, and the frequency range is determined by the dimensions of the duct. From Figure 10 of the "2-Wavetek" system, one wonders why there is a negative attenuation around 500 Hz. This can be explained by the fact that one tone, either the main branch or side branch, was louder than the other and therefore, even though they are destructively combining, one was masking the other. As was shown in the proof, the magnitudes of both signal must be the same in order for positive attenuation to occur. Both the adaptive system and the "2-Wavetek" system show the definite cut-off frequencies. This is furthur evidence that our theory is valid. The graph of the adaptive system shows hills and valleys between our cut-off points. A simple explanation would be resonating frequencies at 200, 400, 800, and 1600 Hz, but there is no apparent resonance at 400 Hz. We can only attribute these irregularities to standing wave phenomena and possibly some non-linearity in the speaker response. The system may have better response if more volume control was allowed. As it was, our Heathkit amplifier afforded very little volume control.

CONCLUSIONS

Through these experiments, the validity of an active noise control system was established. Our adaptive system provided attenuations of from 10 dB to 29 dB. We feel, however, that if a better amplifier, with better volume control, is used, our attenuation levels would be much higher, or the variance would not be so great.

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6.) J.C. Burgess; 1981; Journal of the Acoustical Society of America; Vol. 70(3); pp.715-726; "Active adaptive sound control in a duct: A computer Simulation"

7.) S.E. Keith and H.S.B. Scholaert; 1980; UTIAS Technical Note No. 228; "A Study of an Olson-Type Active Noise Controller and the possibility of the reduction of cabin Noise"

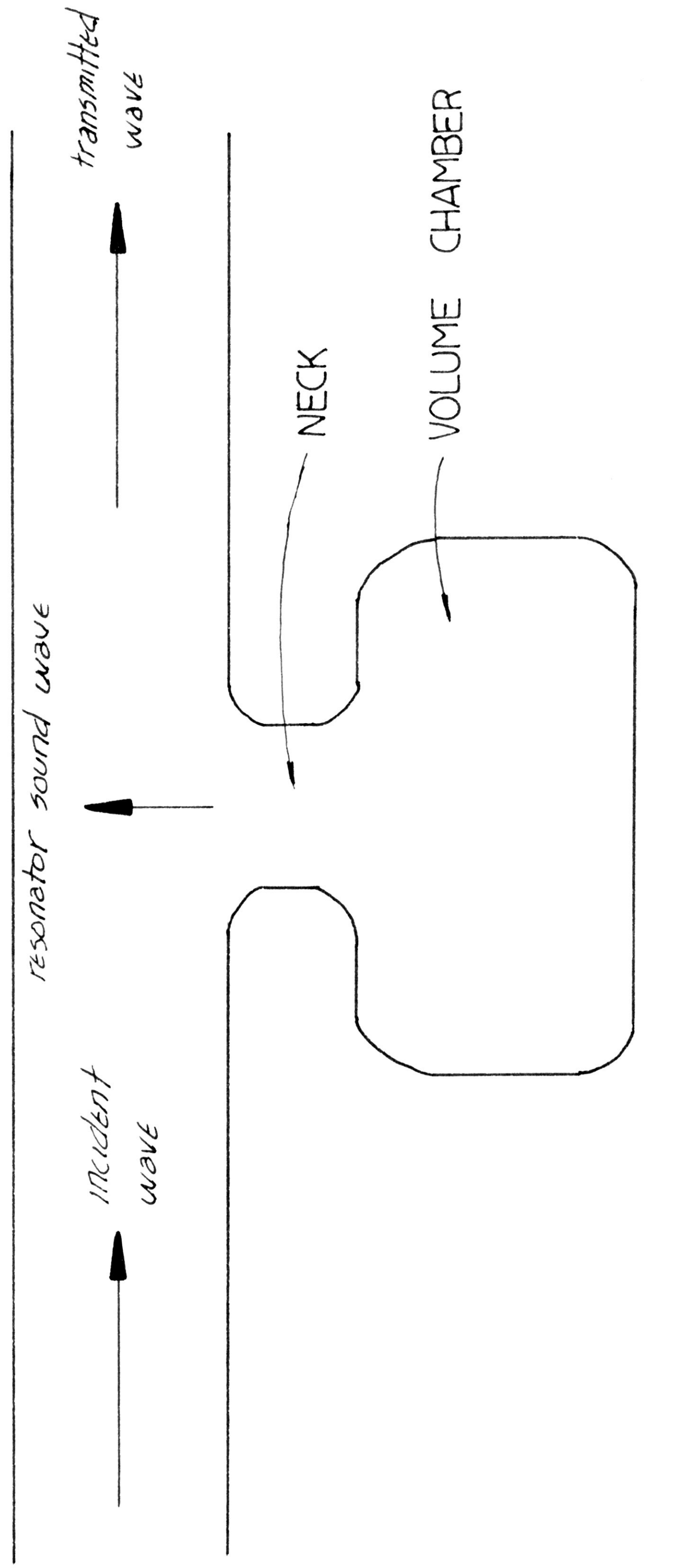


Figure 1   Helmholtz Resonator

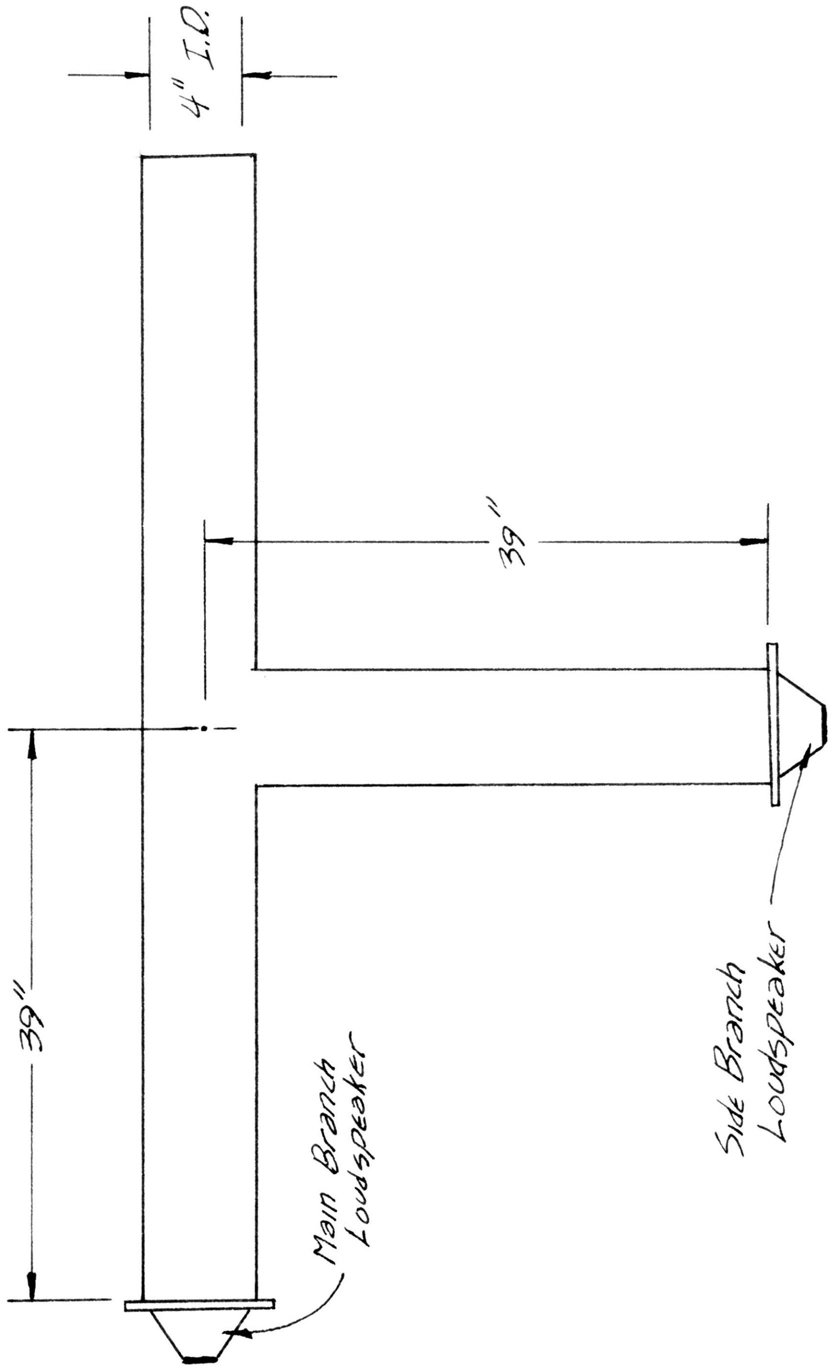


Figure 2 Initial System

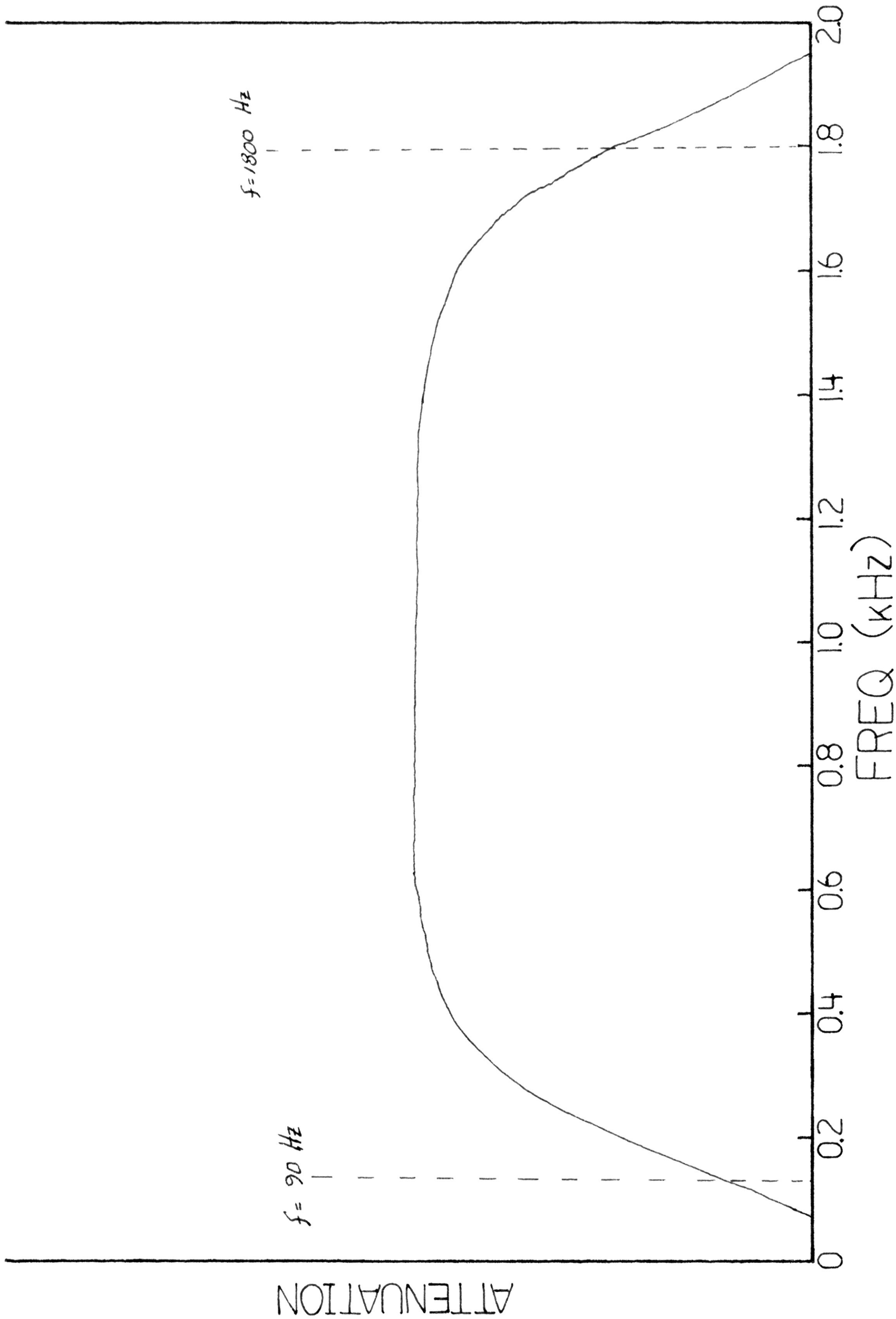


Figure 3 Expected Attenuation Graph

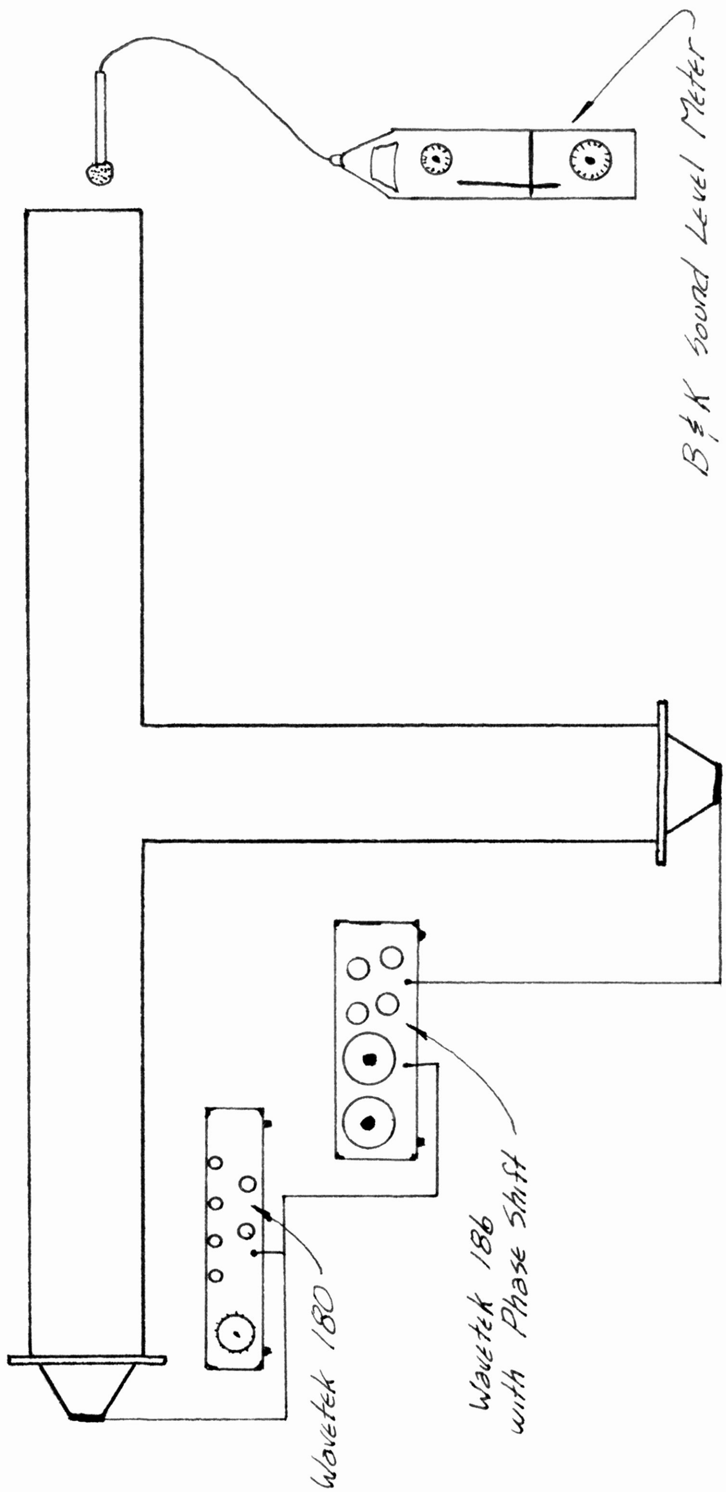


Figure 4 Schematic of "2-Wavetek" System

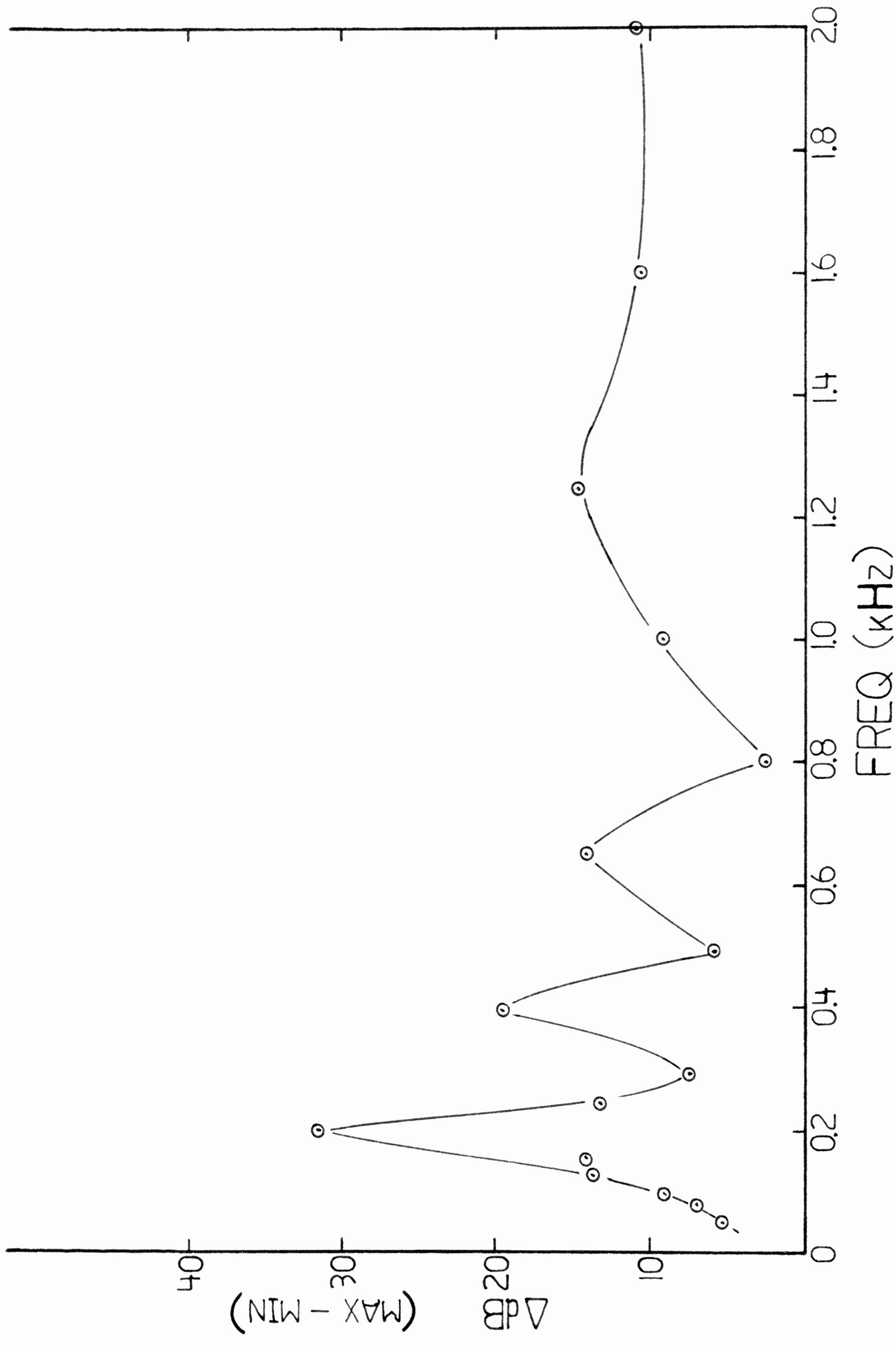


Figure 5 Max to Min Attenuation vs. Frequency for the "2-Wavetek" System

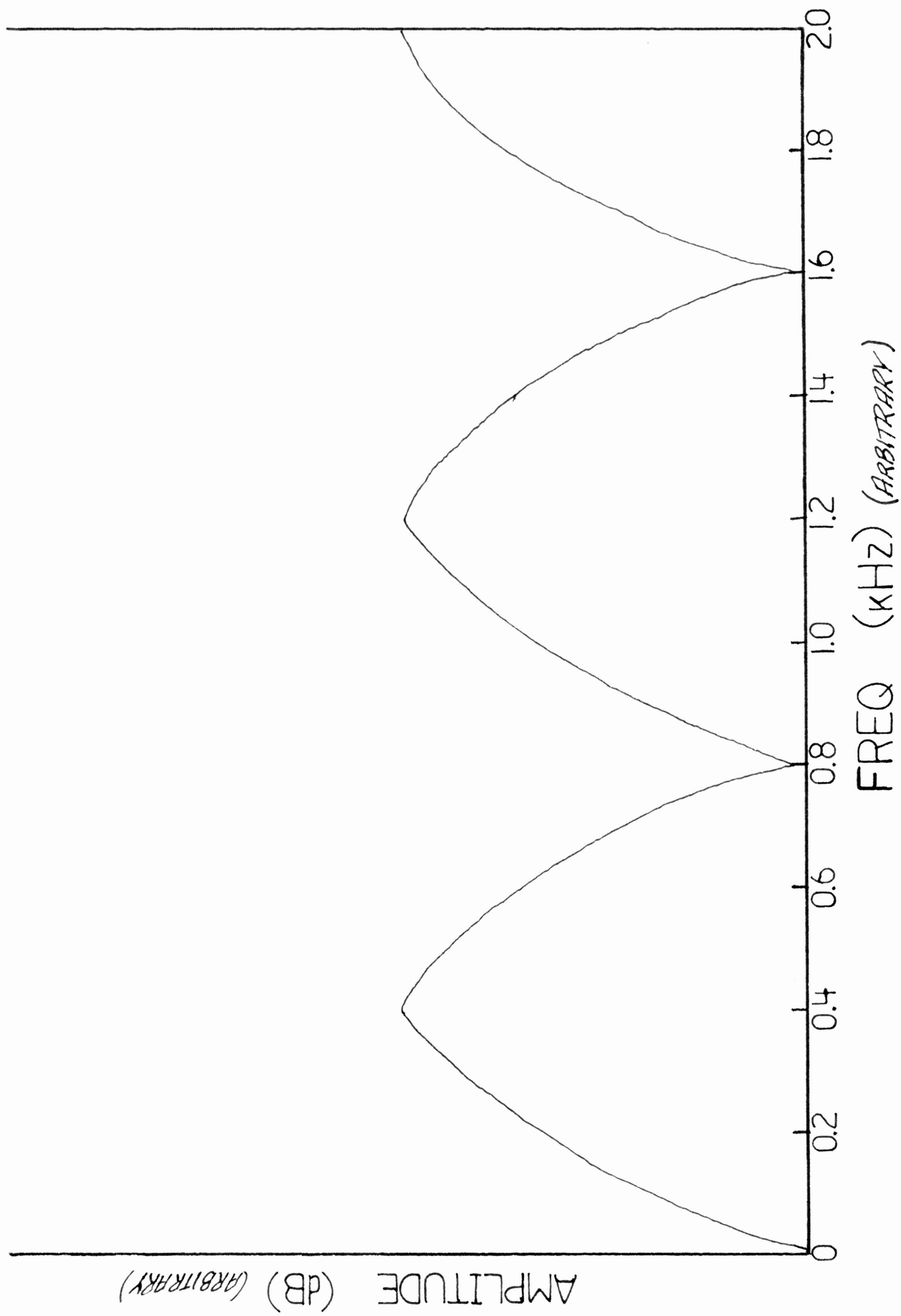


Figure 6 Graph of Swinbanks' Downstream Amplitude

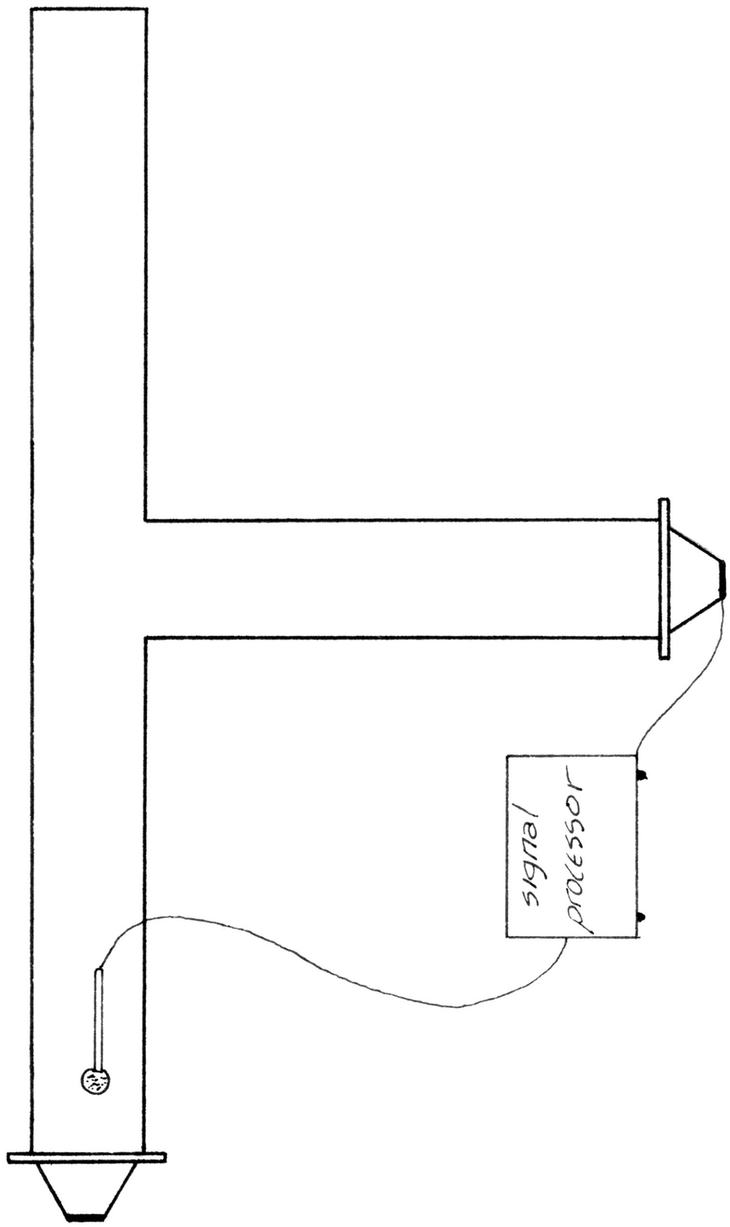


Figure 7 Ideal Adaptive Noise Controller

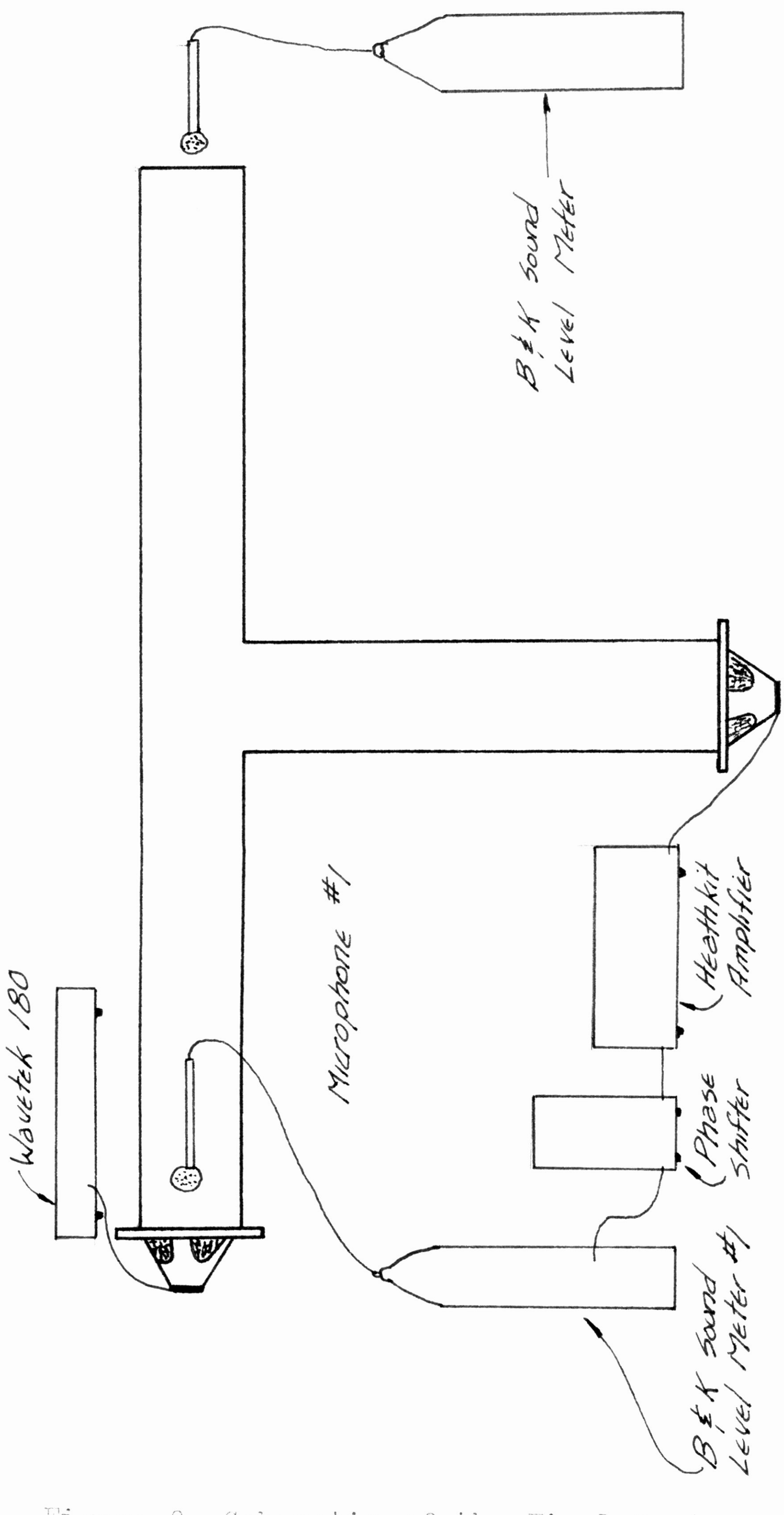


Figure 8 Schematic of the Final System

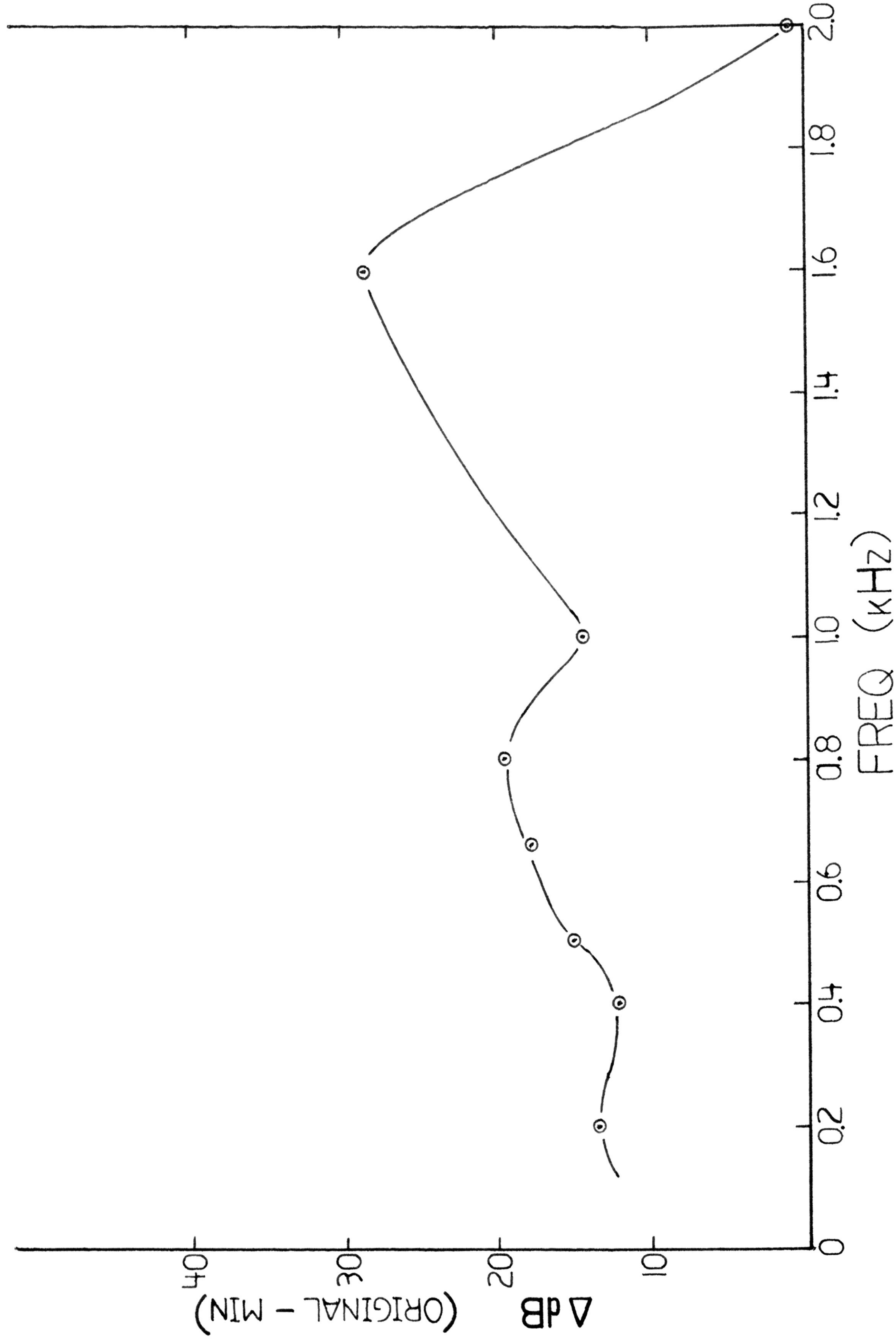


Figure 9 Attenustion vs. Frequency for Adaptive System

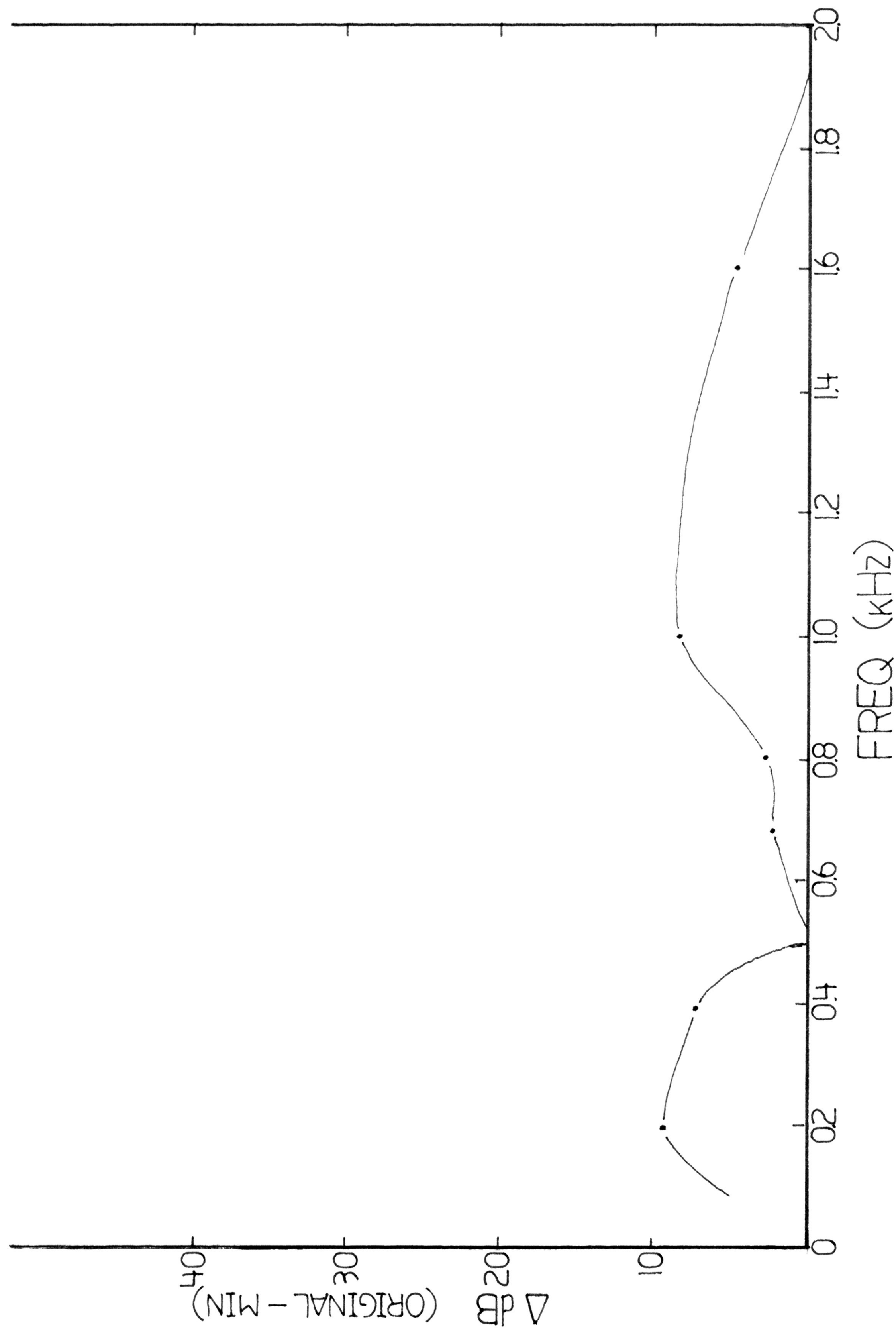
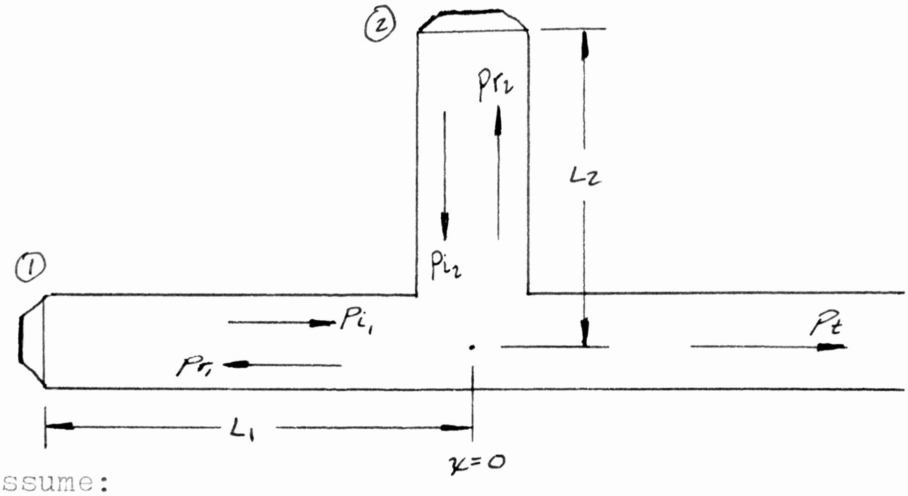


Figure 10 Attenuation vs. Frequency for "2-Wavetek" System

APPENDIX A

MATHEMATICAL PROOF



Assume:

1) Both tubes of equal cross-sectional area $S = \frac{\pi}{4} D^2$

2) Plane waves exist in pipe $\lambda > D$, $(\lambda = \frac{c}{F})$

Loudspeaker at ① is driven to produce input pressure:

$$P_{i_1} = \underset{\sim}{A}_1 e^{j(\omega t - kx)}$$

ω = freq (rad/sec)

K = wavenumber

and volume velocity:

$$U_{i_1} = \frac{P_{i_1}}{Z_0} = \frac{\underset{\sim}{A}_1 e^{j(\omega t - kx)}}{Z_0}$$

where:
$\underset{\sim}{A}$ is complex $= A_1 e^{j\theta}$
$j = \sqrt{-1}$
Z_0 = characteristic impedance
p_0 = air density
c = speed of sound in air
S = Tube area

Because of impedance mismatch at pipe junction, a wave is reflected back down the tube:

$$P_{r_1} = \underset{\sim}{B}_1 e^{j(\omega t + kx)}$$

$$U_{r_1} = \frac{-P_{r_1}}{Z_0} = \frac{-\underset{\sim}{B}_1 e^{j(\omega t + kx)}}{Z_0}$$

Similarly, for the waves in tube ②:

$$P_{i_2} = \underset{\sim}{A}_2 e^{j(\omega t - kx)} \qquad P_{r_2} = \underset{\sim}{B}_2 e^{j(\omega t + kx)}$$

$$U_{i_2} = \frac{\underset{\sim}{A}_2 e^{j(\omega t - kx)}}{Z_0} \qquad U_{r_2} = -\frac{\underset{\sim}{B}_2 e^{j(\omega t + kx)}}{Z_0}$$

The two waves combine at the junction (x=0) where the continuity equations hold:

$$(p_{i_1} + p_{r_1}) + (p_{i_2} + p_{r_2}) = p_t$$

$$(u_{i_1} + u_{r_1}) + (u_{i_2} + u_{r_2}) = u_t$$

Adding:
$$2\underset{\sim}{A_1} + 2\underset{\sim}{A_2}\,\epsilon^{-jk\Delta L} = 2\underset{\sim}{A_t}$$

Assume that signal A has relative phase of zero:

$$\underset{\sim}{A_1} = A_1$$

and that signal into speaker 2 is out of phase with signal to 1 by a phase angle ϕ

$$\underset{\sim}{A_2} = A_2\, e^{j\phi}$$

Finally, the transmitted wave has some sort of phase relationship with respect to signal input to 1

$$\underset{\sim}{A_t} = A_t\, e^{j\lambda}$$

Then:

$$A_1 + A_2\,\epsilon^{j\phi}\epsilon^{-jk\Delta L} = A_t\,\epsilon^{j\lambda}$$

Now: $\epsilon^{j\theta} = \cos\theta + j\sin\theta$

So: $A_1 + A_2(\cos(\phi - k\Delta L) + j\sin(\phi - k\Delta L)) = A_t(\cos\lambda + j\sin\lambda)$

$\underset{\sim}{A_t}$ = Real + Imaginary $\qquad A_t = \sqrt{Re^2 + Im^2}$

So: $A_t = \{(A_1 + A_2\cos(\phi - k\Delta L))^2 + A_2\sin(\phi - k\Delta L)^2\}^{1/2}$

$\qquad = \{A_1^2 + A_2^2 + 2A_1A_2\cos(\phi - k\Delta L)\}^{1/2}$

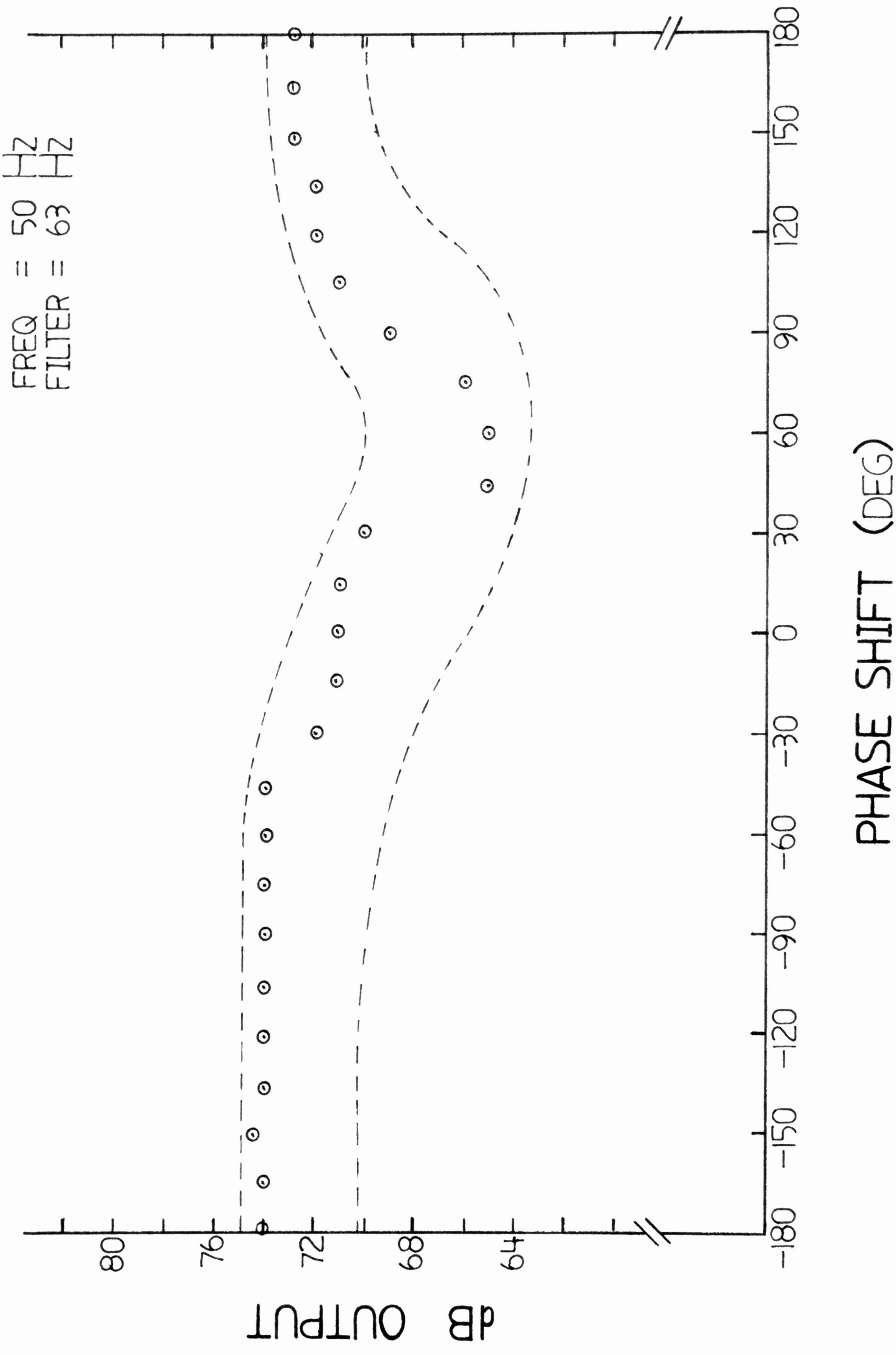
Now: A_t is a minimum when:

1) $\phi - k\Delta L = \pi = 180°$ $\qquad$ for our system $\Delta L = 0$
so ϕ must be $\underline{180°}$

2) Then if $\phi - k\Delta L = -1$ $\qquad A_t = A_1 - A_2$

and $\underline{if}$ $A_1 = A_2$ the $A_t = 0$

APPENDIX B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76
72
68
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−60
−30
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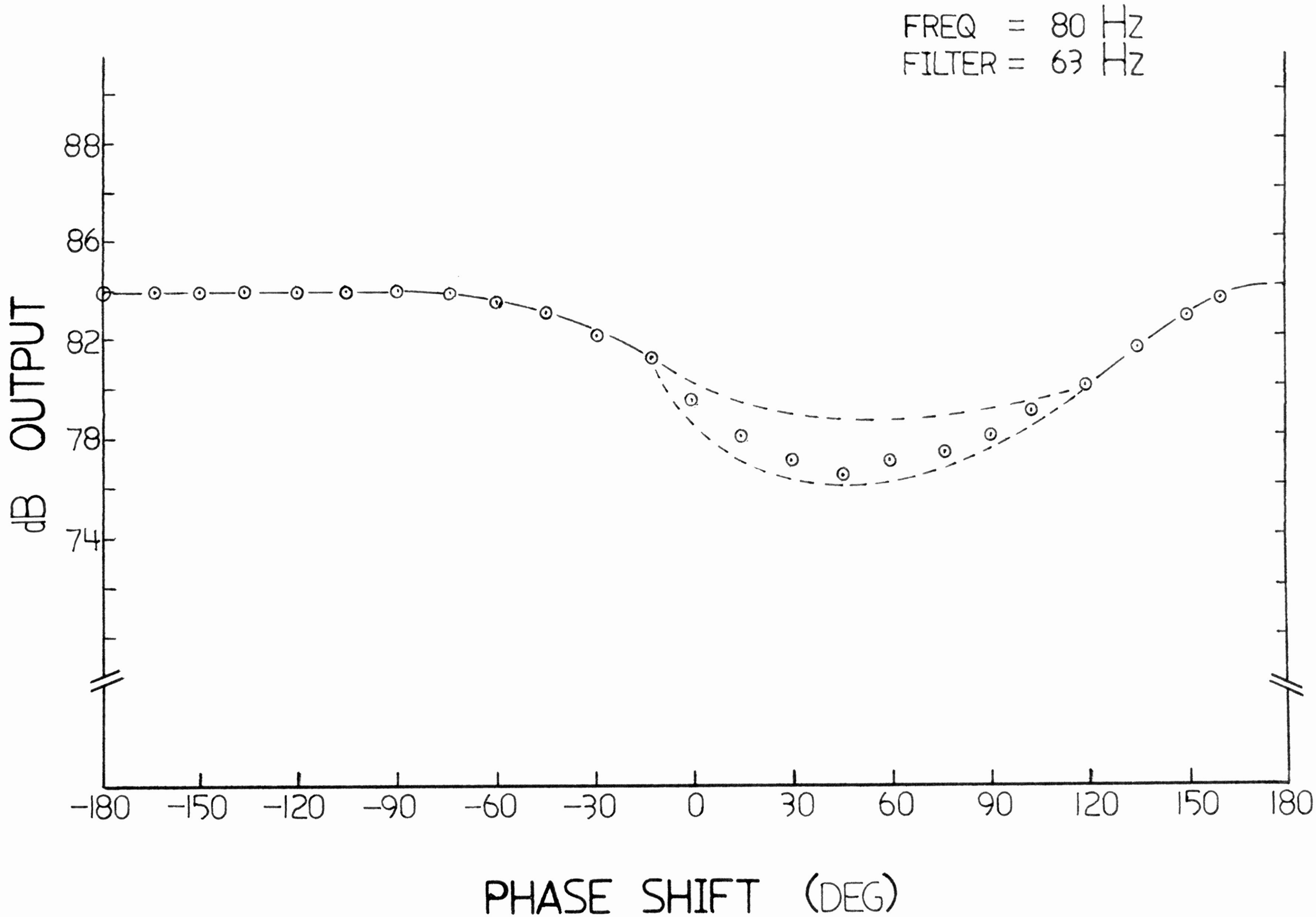
30
60
90
120
150
180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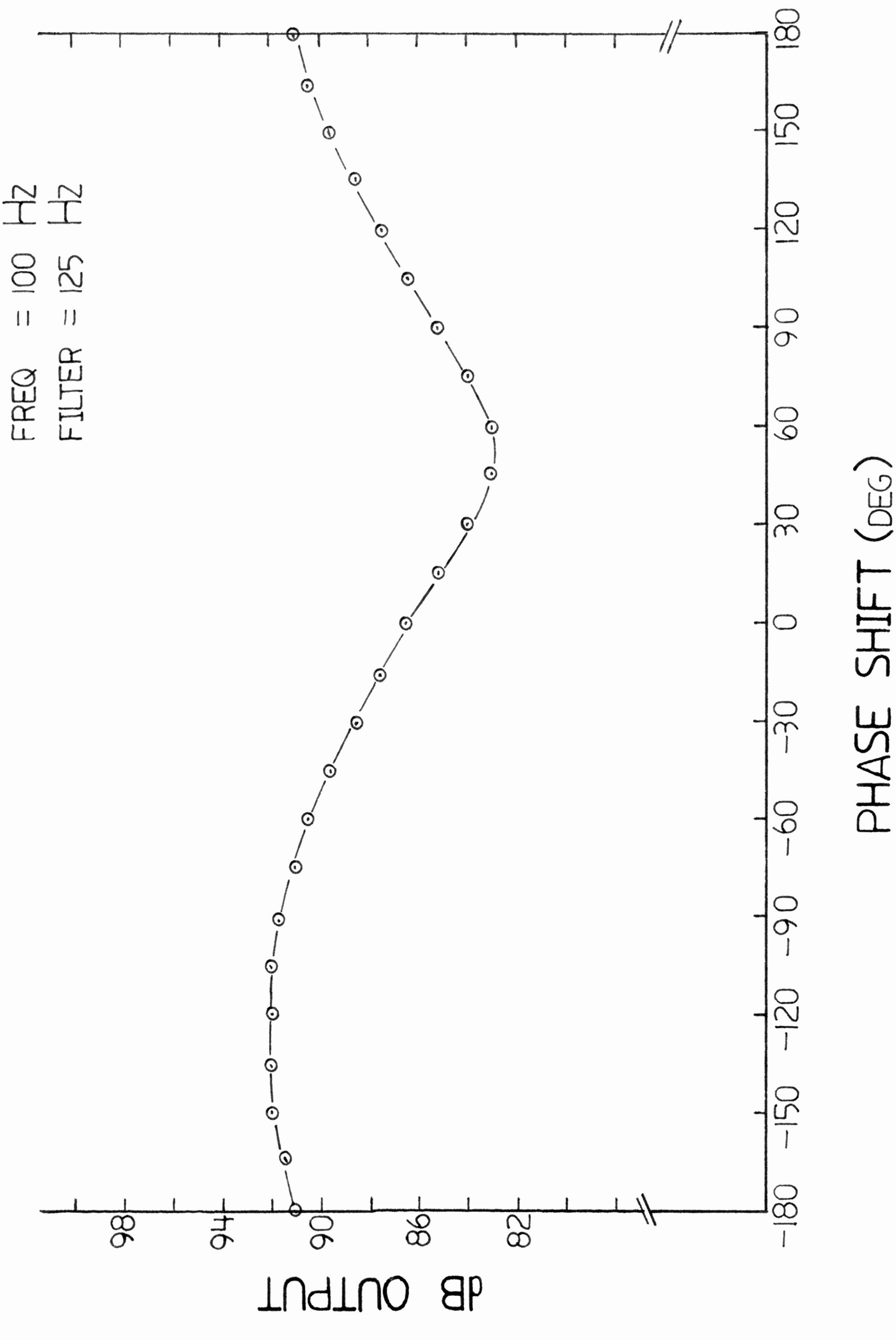
PHASE SHIFT (DEG)



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90
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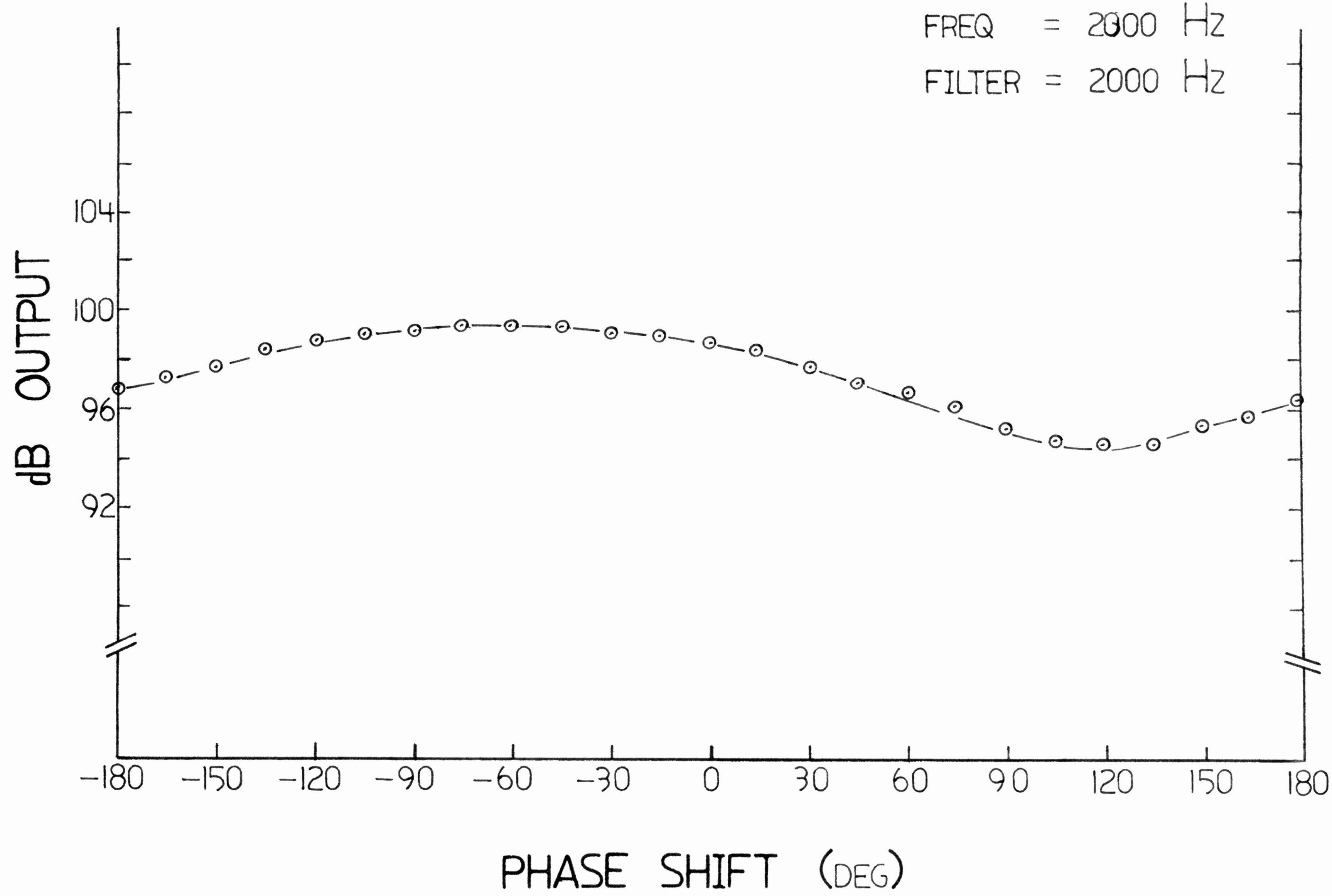
60
90
120
150
180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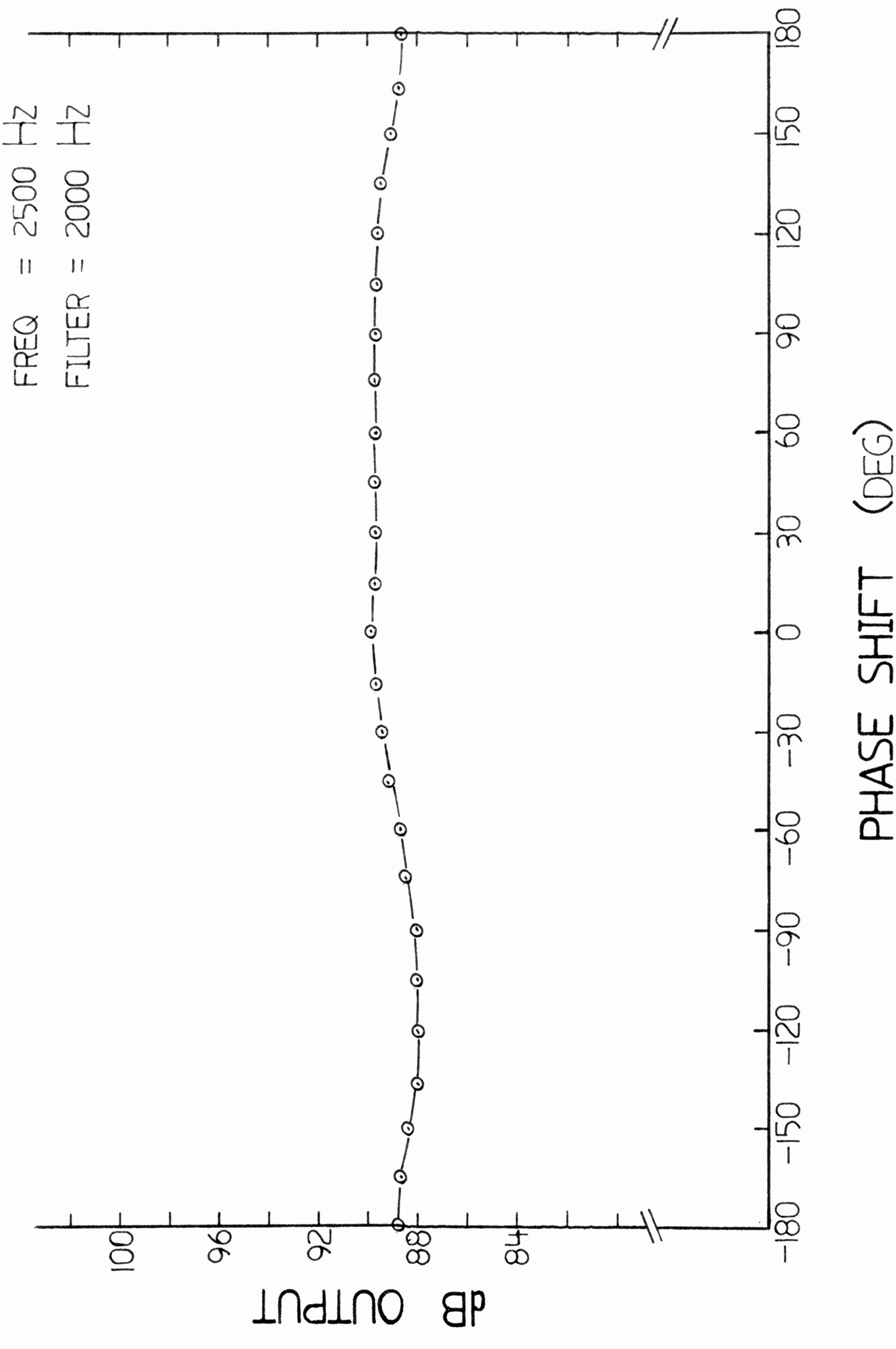
150
180
PHASE SHIFT (DEG)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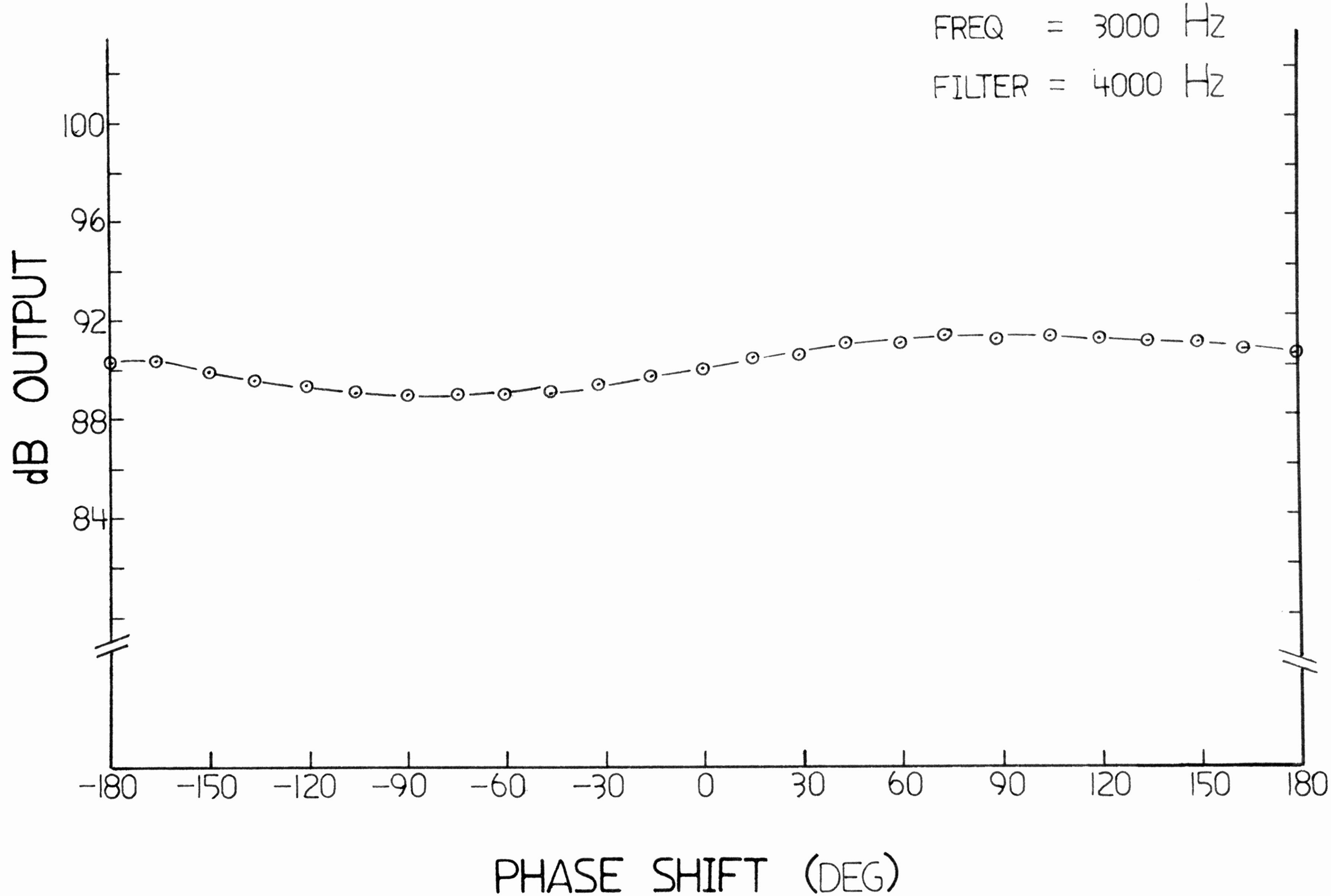
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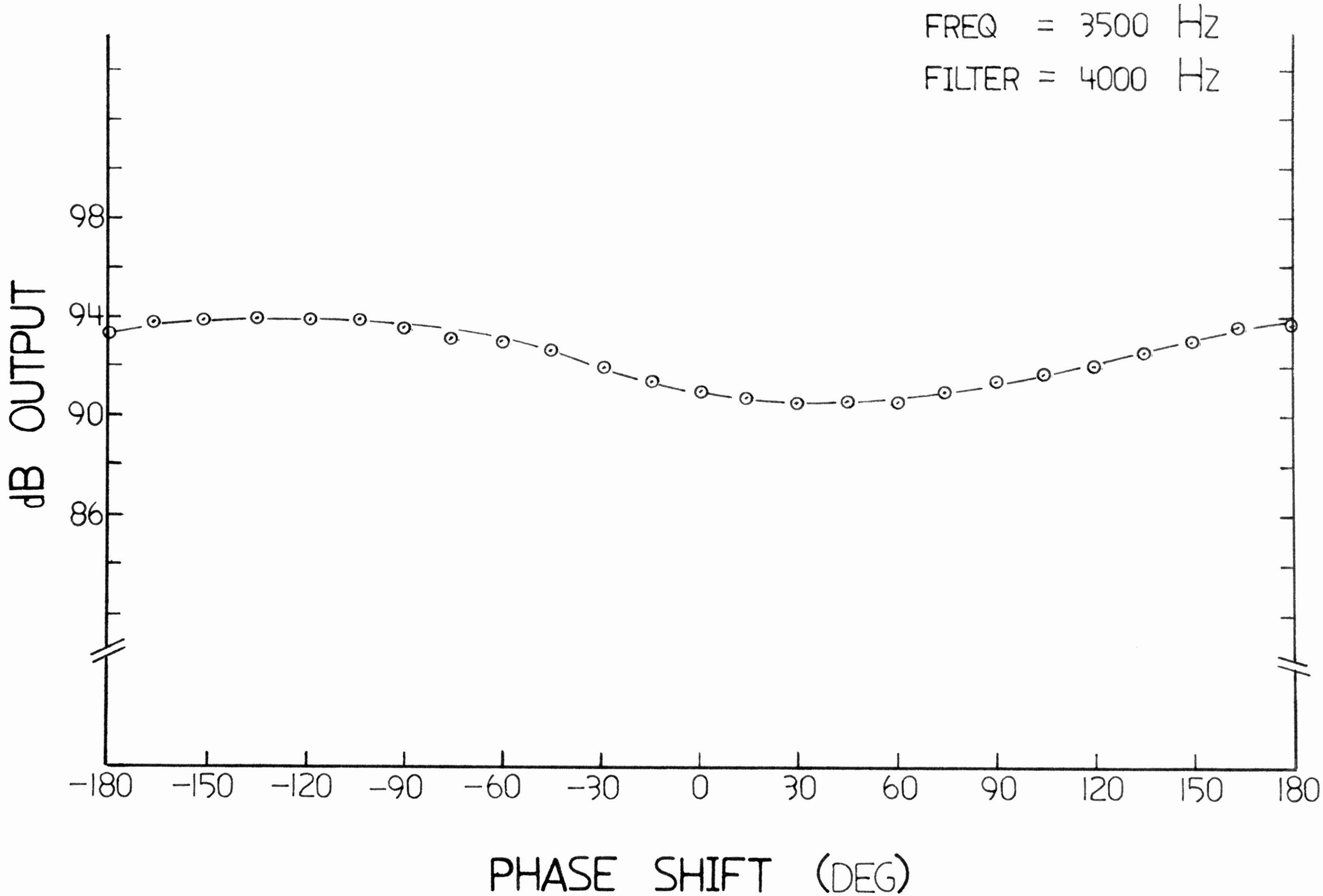
150
180
PHASE SHIFT (DEG)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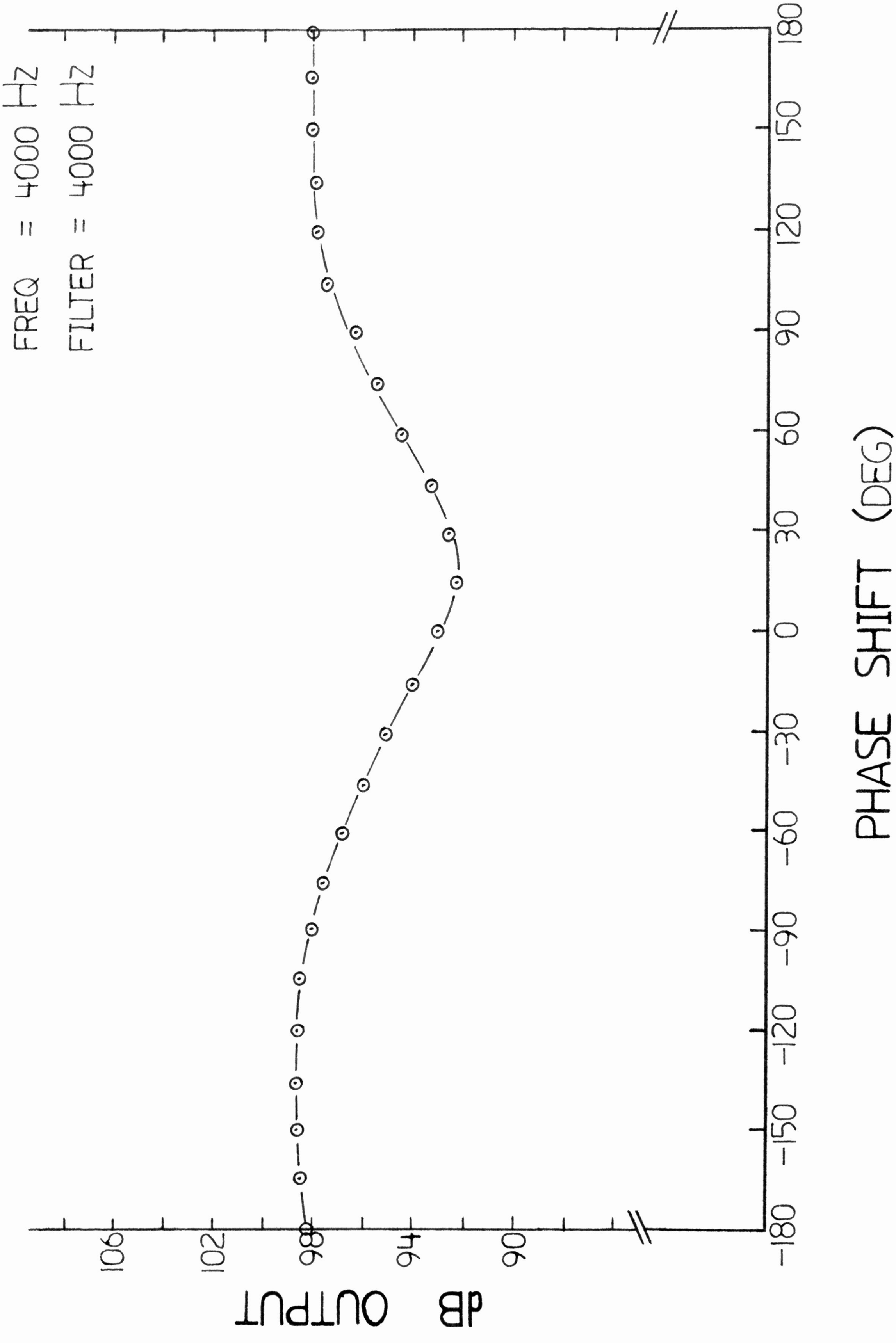
PHASE SHIFT (DEG)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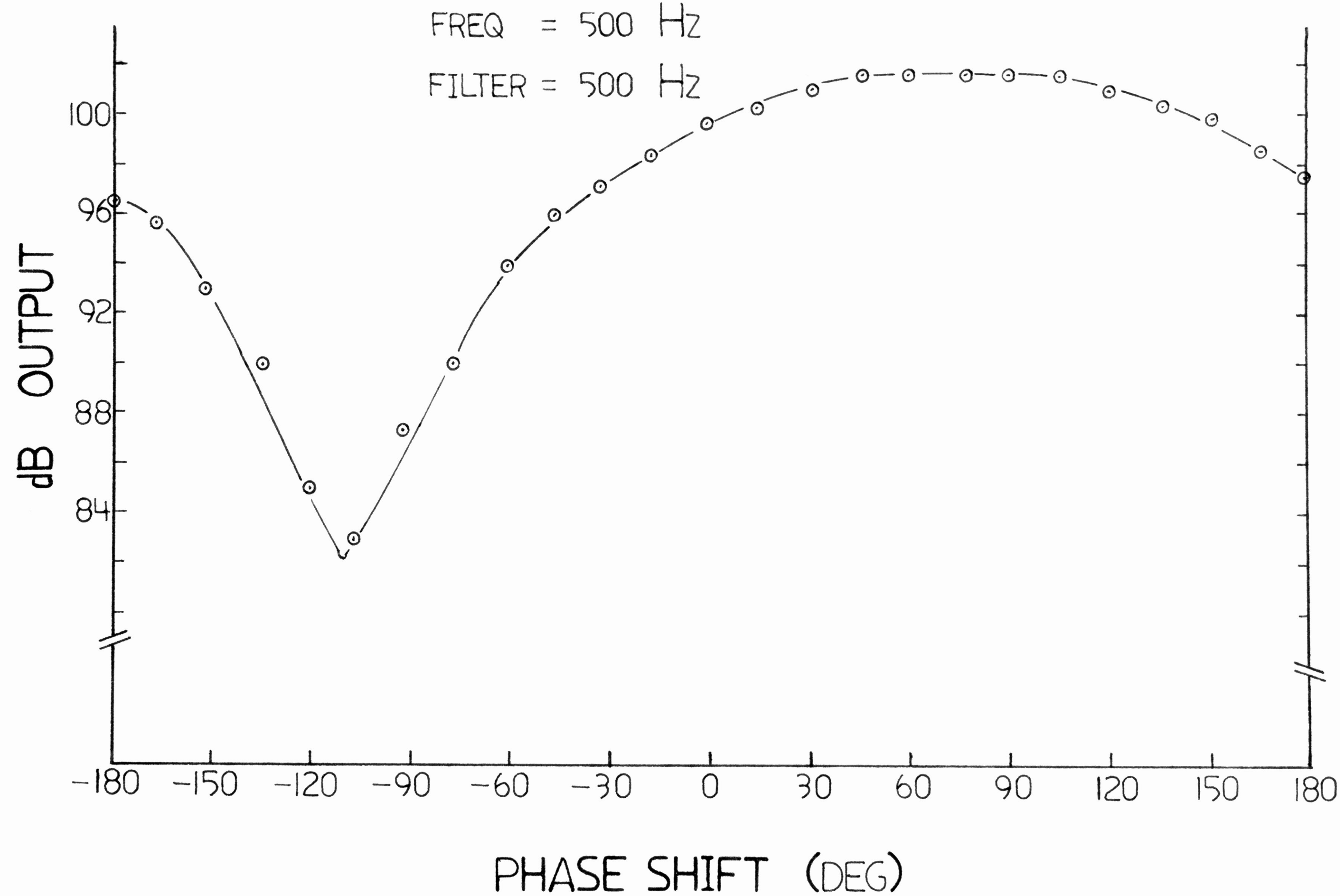
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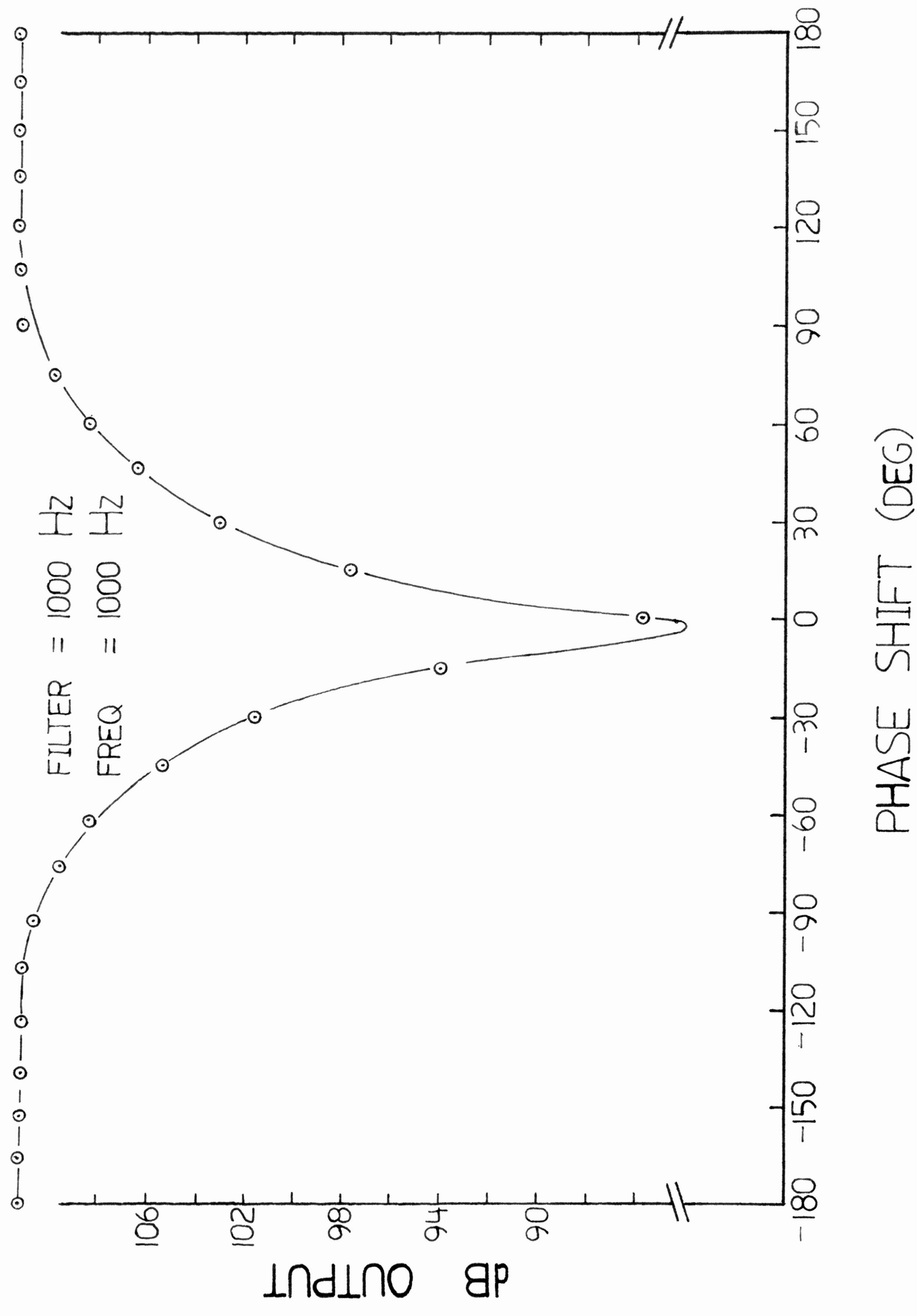
PHASE SHIFT (DEG)



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90
120
150
180
PHASE SHIFT (DEG)