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INVESTIGATION OF THE TONE-BURST TUBE FOR DUCT LINING ATTENUATION MEASUREMENT

by

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Prepared by

ADVANCED TECHNOLOGY CENTER, INC.
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ABBREVIATIONS AND SYMBOLS

A	attenuation on insertion loss of absorber
A*	normalized attenuation
C	capacitor
C_n	Amplitude of nth harmonic
d	depth of absorber, inches
dB	decibel = $10 \log_{10} \frac{I_1}{I_o}$, $20 \log_{10} \frac{P_1}{P_o}$, $20 \log_{10} \frac{E_1}{E_o}$
E	electrical potential, voltage
f	frequency in Hz
f_{bc}	frequency at octave-band center
f_p	frequency of attenuation peak
FILT	electrical wave filter
FT/SEC, ft/sec	feet per second
GEN	generator
H	henries
Hz	Hertz, cycles per second
h	height of duct in inches
I	acoustic intensity
K	thousand, thousand ohm
KHz	thousand Hertz
L	inductor
l	length of absorber, inches
M	Mach number
m	meter length
mf	microfarad

mh	microhenry
N	Newton, 1 x 10^5 dynes
n	number, exponent
Osc.	oscillator
P	acoustic pressure
P-P	peak-to-peak
PPL	peak pressure level in dB re: 2 x 10^{-5} N/m^2
psi	pounds per square inch
psig	pounds per square inch gauge
Q	figure of merit of a resonator
R	resistor, pipe radius
Rayl	unit of specific acoustic impedance in cgs system
RMS, rms	root-mean-square
scfm	standard cubic feet per minute
Scope	oscilloscope
SPL	sound pressure level in dB re: 2 x 10^{-5} N/m^2
SQR-INT	squarer-integrator
U	gas velocity at Y = R
U_R	maximum gas velocity
u	local gas velocity
V	volts
VTVM	vacuum-tube voltmeter
y	radial distance from pipe circumference
δ	value of y where u = -.99 U_R

INVESTIGATION OF THE TONE-BURST TUBE FOR DUCT LINING ATTENUATION MEASUREMENT

By Arthur R. Soffel and Paul F. Morrow
Advanced Technology Center, Inc.

SUMMARY

The tone-burst technique makes practical the laboratory evaluation of potential inlet and discharge duct treatments. Tone-burst apparatus requires only simple machined parts and standard off-the-shelf components. Small, simply made, lining samples are quickly and easily installed in the system. Two small electromagnetic loudspeaker drivers produce peak sound pressure levels of over 166 dB in the 3-square-inch sample duct. Air pumps available in most laboratories can produce air flows of over plus and minus Mach 0.3 in the sample duct.

The technique uses short, shaped, pulses of sound propagated down a progressive-wave tube containing the sample duct. The peak-pressure level output of the treated duct is compared with the peak-pressure level output of a substituted reference duct. The difference between the levels is the attenuation or insertion loss of the treated duct.

Evaluations of resonant absorber linings by the tone burst technique check attenuation values predicted by empirical formulae based on full-scale ducts.

INTRODUCTION

The evaluation of sound absorbent structures for turbofan inlet and air discharge ducting has been carried out mostly at full scale.[1,2,3] For rapid measurements on numbers of samples an impedance tube[4] is used. The full-scale method uses two large reverberant enclosures connected by a section of a full-scale duct. Sound is generated in one chamber and picked up in the other. Air can be blown in both directions to simulate inlet and discharge. Results from such tests approximate noise reductions obtained on engine installations. Measurements on an impedance tube are convenient, fast, and require only a small sample. Results of these measurements are not directly applicable to noise reduction determinations. The direction of sound incidence is normal instead of grazing, as in service, and the inclusion of air flow is impractical. Some convenient method of combining the realistic results of the full-scale tests with the convenience of the impedance tube would be a worthwhile tool for the airframe and engine designer.

The purpose of this program has been to investigate the practicability of evaluating potential duct treatments at very high sound levels in the presence of air flow by the tone-burst technique. The tone-burst technique uses short, shaped, pulses of sound propagated down a small diameter progressive wave tube, through a section of sample-lined duct, and discharged into the air via an acoustic termination. Air can be blown through the sample-lined duct in either direction.

The work has been performed under the sponsorship* of the NASA Langley Research Center in Hampton, Virginia. It is a direct outgrowth of a previous contract** with the same agency for development of a tone-burst technique for measuring absorption by the reflection of a pulse from a sample at the end of the tube.[5]

The work was performed at the Anaheim, California laboratory of the LTV Research Center and later at the Dallas, Texas laboratory of the Advanced Technology Center, Inc.***

The program was divided into three tasks:

Task I - System Development

This task required the optimization of the system and components, including: acoustic driver, air-sound mixing section, flow tube, test section, pressure transducer, tube termination, and instrumentation.

Task II - Test Procedure Development

This task required the optimization of operating procedures from sample preparation to determination of final results. The following items were to be optimized:

1. Sample size and location

2. Pulse characteristics

3. Instrumentation

4. Pressure transducer location

5. Readout procedures

6. Data reduction procedures

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***Name was changed in 1971 from LTV Research Center to Advanced Technology Center, Inc.

Task III - System Demonstration

This task required evaluating typical lining materials and comparing the results with available data on similar duct treatment.

The technique could be used by engine and airplane designers in conjunction with existing impedance tube and full-scale methods of absorber evaluation.

TECHNICAL DISCUSSION

Overall Considerations

Concepts. - The tone-burst tube system for duct lining attenuation measurement involves three concepts. The first concept is that a valid attenuation measurement can be made on a relatively small sample of material when it forms a short section of the internal wall of a progressive wave tube. The second concept is that an insertion loss measurement best represents the improvement that can be made in an engine duct when part of its reflective internal wall is treated with the absorptive material being measured. The third is that measurements made with short tone bursts are equivalent to those made with steady-state sinusoidal signals.

The first two concepts result in a simple and compact arrangement that can be set up in a laboratory. They also make it easy to have air or gas flow in the tube with or against the direction of sound propagation.

The third concept mades it possible to generate very high acoustic intensity with low-power electromagnetic drivers.

System operating principles. - The physical system is shown as a block diagram in Figure 1. The outputs of the Acoustic Source or Driver and the pneumatic sources are combined in the Air-Sound Mixing Section. The pneumatic source can be either the Compressor or the Exhaust Pump for positive or negative air flow. The combined outputs pass through the Flow Settling Section to develop turbulent flow. The Reference Duct is first inserted. The sound and air pass through to the Acoustic Termination where they are both exhausted to atmosphere. Sound pressure levels are measured by the Sound Detector at the input and output of the Reference Duct. The Sample Duct is substituted in the system and the sound pressure levels are again measured. The difference in dB between the output sound pressure levels of the two ducts is the insertion loss of the sample duct at the prevailing air flow and acoustic input level (measured at input to Reference Duct).

The tone burst. - The primary purpose of using a tone burst is to obtain very high peak acoustic power while maintaining a low rms power. This is done by using a small ratio of "on" time to "off" time. A pulsed sinusoidal signal which starts and ends at a zero crossing and is a fixed number of cycles long has stable and reproducible characteristics. It is, however, a broad-band signal. Bu passing this "square" pulse through a band-pass filter centered at the sinusoidal signal frequency, a smoothly rising and falling pulse with a narrow band results. A convenient pulse is one produced by passing a "square" pulse of 8 cycles duration through a one-third octave filter. Figure 2 shows 8-cycle pulses before and after passing through the one-third octave filter. For a repetition rate of 2 to 3 pulses per second, at 1000 Hz, for the same peak power, the average power for the pulses is 1/40th the average power of the sine wave. This factor permits one to increase the peak input to an electromagnetic driver without causing overheating. Presumably the peak pressure output is limited by the mechanical limits to diaphragm and coil excursion. Fatigue life is increased at the same time.

A secondary advantage in using a tone burst is that the measurement of input and output is made as the pulse passes through the sample duct. The length of the terminating section can be long enough so that echoes from the end of the tube arrive back at the sample late enough not to interfere with the measurement.

PPL. - Measurements in a tone-burst system are made in terms of PPL or peak pressure level. PPL is equal to 20 $\log_{10}$ of the zero-to-peak pressure at the maximum value of the filtered tone-burst envelope, relative to 2×10^{-5} N/m^2. The term is also used in this report as 20 $\log_{10}$ times the zero-to-peak pressure at any designated point of time in a tone pulse.

Significance of tone burst measurements. - The measurement of the insertion loss of a sample duct is a transmission measurement. Normal transmission measurements are made with sinusoidal signals. Input signals are either discreet frequencies or frequency-swept sinusoidal signals of constant level. The output signals are measured as discreet points or as a continuously varying level. The resulting db-versus-frequency curves are the same for both methods.

Another method of measuring transmission is to use white noise as the input. The output level-versus-frequency curve is obtained by using a very narrow-band swept filter at the output and recording the results. If the filter bandwidth (constant number of cycles) is sufficiently narrow and the sweep is slow with a long averaging time, the transmission curve will match the sinusoidally measured curve. If the white noise input is used with a one-third octave stepping filter, the transmission curve will not match the sinusoidally measured curve. One-third octave bandwidths are a constant percentage of the filter center frequency. A transmission curve made in this way on a "flat" circuit will yield a curve that rises at a rate of 3 dB per octave with increasing frequency. This discrepancy can be eliminated by measuring the input signal with the same filter and subtracting the output curve from the input curve. The result will be a true transmission curve.

Discrepancies can arise, however, when measurements are made with a one-third or other relatively broad-band filter set. A transmission system may have sharp peaks or dips that have bandwidths comparable to or less than the one-third octave filters. Here, a transmission curve measured with random noise and one-third octave filters will have lost much detail.

The tone burst used in this program is not a discreet frequency signal. It has a one-third octave bandwidth. Unlike an equal band of random noise, energy within the band is not constant but varies with frequency. Maximum energy is at the fundamental frequency of the original sine wave. For "flat" transmission paths or those with only gradual slopes, PPL measurements will give results comparable to sinusoidal measurements. For paths with sharp peaks or dips sinusoidal measurements will not be matched. This phenomenon was investigated both analytically and experimentally during the program.

One of the features of the program is that very high sound intensities can be generated using tone bursts. The technique being used assumes that a tone burst of a given PPL is equivalent as far as level is concerned to a sine wave of that same peak amplitude. Exactly how valid the assumption is has not been determined.

When sinusoidal acoustic signals of high level are propagated through tubes, the non-linearity of the air causes the sine waves to develop into sawtooth waves. A sawtooth wave contains energy at harmonic frequencies of the sine wave. If rms determinations are made, the results are for a group of frequencies. If peak determinations are made there will be even more uncertainty in measurement of transmission because of the sine-to-sawtooth conversion. This conversion envolves generation of harmonics. This problem exists in tone-burst measurements. The problem has been considered but no easy way of compensating for the uncertainties was worked out.

Although the tone-burst technique is subject to some errors they are relatively small being in the order of 1 or 2 dB. Any other method of evaluating acoustic absorbers at high sound levels in the presence of high air flow is subject to errors of the same or greater magnitude. Precise comparisons among methods would not be expected. Agreement within a few dB would be considered good.

System Development

Development program. - The development of the tone-burst flow system proceeded with parallel development of the hardware and procedures. In reporting, the two phases have been separated to a great extent to follow the Task concept specified in the work statement. In the course of the work, a preliminary system was developed and evaluated. A final system was then developed in which the performance of the components was improved to meet the original objectives.

Preliminary system. - The components of the tone-burst flow system are: Acoustic Source (Driver); Air-Sound Mixing Section, pneumatic sources; Flow Settling Section; Reference-Sample Duct; Sound Detector; and Acoustic Termination. Each component was chosen after careful analysis and experimentation.

Acoustic Source: In keeping with the idea of using available components, "stock" electro-magnetic loudspeaker compression drivers were tested. The first driver designated 290-E* was a unit rated at 100 watts input and designed for high-powered outdoor public address systems. The second driver, 291-16A, was a comparable unit rated at 40 watts input and designed for

*Specifications for drivers are given in Appendix B.

theatre sound systems. The third driver, 802-D, was a smaller unit rated at 30 watts and designed for small theatre systems and high-fidelity equipment. The type 291-16A driver was found to produce the highest output. The limiting factor in driving this unit during these early tests was voltage breakdown in the voice coil. The particular units were chosen for convenience and are representative of high quality industrial units. Drivers of other manufacturers may work as well if specifications are similar. The 291-16A unit was chosen for the Acoustic Source.

The main flow sections were to be 2-inch pipe. A transition section to couple the 1.4-inch diameter of the driver throat to the 2-inch diameter of the mixing section was made.

Air-Sound Mixing Section: Preliminary experiments showed that simply mixing the air flow and sound at a "Y" connection between the driver output and a compressed-air line caused acoustic difficulties. A mixing system avoiding this problem was designed and built.

It is essential that constant cross-sectional area be maintained throughout the system to preclude generation of cross-modes and reflections. The propagation of sound upstream to the pneumatic source must be held to a minimum to prevent reduction of sound pressure in the flow tube. To meet these requirements a section of the flow tube was replaced by a sintered bronze cylinder of the same inside diameter as the tube. This cylinder was enclosed by a larger cylindrical plenum chamber through which air was introduced. The configuration is shown in Figure 3. The sintered element is 9 inches long, and 2 inches in diameter made of 1/16-inch thick 10-micron material. The acoustic impedance of the sintered element is very large compared to the characteristic impedance of the flow tube. Very low acoustic loss results. Figure 4 shows the insertion loss at no flow for an input level of 160 dB PPL. The loss decreases with frequency indicating a rising impedance in the sintered material.

If equal lengths of pipe are used to connect the pneumatic source to the two air ports on the plenum, the flow profile emerging at the outlet of the mixing tube is symmetrical with respect to the tube axis.

The configuration shown in Figure 3 was used in the preliminary system.

Flow Settling Section: The purpose of the Flow Settling Section is to permit the full development of turbulent air flow upstream of the sample.

A study was made to determine the length of 2-inch pipe required for turbulent flow to develop. The criterion adopted at this point in the program for fully-developed turbulent flow was that the flow profile across the pipe conform to the relationship:

$$\frac{u(y)}{U} = \frac{y}{R}^{1/n} \qquad (1)$$

Where: n = a function of Reynolds Number (7.3 for $R_e = 2 \times 10^5$)
y = radial distance measured for circumference toward center

$u(y)$ = local flow velocity
U = flow at $y = R$
R = pipe radius
R_e = Reynolds number

Flow profiles were measured downstream from the outlet of the mixing section at distances from 0 to 3.4 feet (21 diameters). The criterion was met at 3.4 feet. The measured curve showed a value of $n = 9.8$ as opposed to the theoretical value of 7.3, indicating well developed flow. The Flow Settling Section was made of 24 inches of 2-inch pipe followed by a 6-inch round-to-square transition (constant area) to match the sample section. The total length was 30 inches which is shorter than the experimental length of 40 inches. Thirty inches represents the distance at which the flow was almost fully developed.

Reference-Sample Section: The Reference-Sample Section serves a dual purpose. It is used to mount samples of absorptive wall material and when furnished with a hard reflecting sample becomes the basis for a "sample-out" or reference measurement.

A 2-inch diameter circle was chosen as the basic cross section of the flow system. This size was chosen because it is the largest cross section in which high sound levels can be conveniently generated with available drivers. It also keeps the volume flow of air at a reasonable value for obtaining the high velocity flows required. A flat sample is easiest to obtain and fabricate. Cylindrical samples of small radius would be difficult to make. Also, the 2-inch width for a sample appeared to be about minimum to avoid edge effects. A square test section of the same area as the 2-inch pipe was designed and built. It was 1.75 inches square and 14 inches long inside. The samples were 11.75 inches long. The section consisted of two opposite fixed sides of 1/8-inch aluminum between a pair of round flanges. Test samples or reference plates (actually the backs of the test samples) could be clamped into the other two sides.

Acoustic Termination: The purpose of the Acoustic Termination is to match the impedance at the outlet of the Sample Section. If the tube impedances were not matched at this point, reflections would occur and pulse measurements would be affected. The match can be provided in a number of ways. An absorptive termination, quite short in length, could be used. This type is impractical when air is flowing through the tube. A square-to-round transition and a very long length of 2-inch pipe would terminate the section well. The best compromise appeared to be the transition and 10 feet of 2-inch pipe, terminated in a 500 Hz exponential horn. The horn matches the 2-inch pipe impedance to free space within certain practical limits. Below the low-frequency cutoff of the horn the tube is not matched and reflections will occur. At the cutoof and for a range of 500 Hz the horn throat impedance fluctuates. In that frequency range reflections occur and the impedance facing the Sample Section outlet also fluctuates. The reason for the 10-foot length is to delay the reflections from the horn throat by 20 milliseconds. The pulse usually is 16 milliseconds long.

Pressure Transducer: The pressure transducer used to pick up the high-intensity tone bursts must be capable of flush mounting in the wall of a tube. It must be capable of working at sound pressures close to one psi, be immune to high static pressures, and be rugged enough to withstand the buffeting of high speed air flow. A Piezoelectric pressure transducer* designated LC-71A was found with the desired characteristics. The transducer was mounted where required by screwing it into a threaded hole in the pipe.

Instrumentation: The instrumentation used in the tone-burst system is relatively complex but all items are standard and readily available. There are two groups of instrumentation, signal generating and signal processing.

Signal generating instrumentation**: Figure 5 is a block diagram of the signal generating instrumentation. A sinusoidal signal is provided by the Beat Frequency Oscillator. Frequency is precisely indicated by the Digital Counter.

The Pulse Generator accepts a sinusoidal signal. In a predetermined (adjustable) sequence it will start a pulse at a zero crossing, count out an integral number of cycles and stop at a zero crossing.

The 1/3 Octave Audio Spectrometer provides a selection of filter center frequencies from 20 to 40,000 Hz in the standard steps 20, 25, 31.5, etc.

The Power Amplifier is a wide-band solid-state device with a maximum output of 1000 watts at a one ohm impedance. It is provided with a very fast-acting output current-voltage cutoff.

To accommodate a number of different transducers and to make fine adjustments in impedance for better power transfer a special 1000-watt, multitap transformer was procured. Transformer taps are provided at: 4, 5.8, 7.8, 10.2, 13, 16, 19, 23, 31, 41, 64, 88 and 126 ohms.

Driver input is measured by the Oscilloscope. Usually peak-to-peak voltage of a tone burst is measured by using a high-gain setting and off-setting the trace with a calibrated voltage and multiturn potentiometer alternately plus and minus to set the minus and plus peaks to the scope centerline. The voltage read from the potentiometer is the zero-to-peak voltage of the tone burst. PPL in dB relative to 1 volt is calculated by taking 20 $\log_{10}$ of the zero-to-peak voltage.

Signal processing instrumentation: Figure 6 is a block diagram of the signal processing instrumentation. The output of the Piezoelectric pressure transducer is amplified by the Pre-amplifier. It then passes through the Audio Frequency Analyzer. This instrument is a band-pass filter with

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*Specifications for pressure transducer are given in Appendix B.

**Specifications for all instruments are given in Appendix B.

adjustable bandwidth and center frequency. The bandwidth is set for 25 dB* and the frequency adjusted to that of the 1/3 Octave Audio Spectrometer. An audio spectrometer could be substituted for the Audio Frequency Analyzer. The Oscilloscope is used to measure peak-to-peak voltage from which PPL can be calculated.

When the system is being operated with high-velocity air flow, particularly at frequencies of high sample loss, the signal-to-flow noise ratio could be unfavorable. In anticipation of this problem two methods were proposed as solutions. The first idea was to use multiple microphone stations to average out the uncorrelated flow noise by adding the signals after delay and amplitude adjustments. The second was to use a Waveform Eductor before the Oscilloscope. This device works very well on repetitive pulses. A Waveform Eductor divides a signal into 100 time slots. A signal-in-noise is repeatedly run through the unit. Each time it passes it is accurately indexed so that identical portions of the signal are integrated in each time slot. Almost any amount of noise can be filtered out if one waits long enough. During each pass, and later from memory, the wanted signal wave shape can be displayed on an oscilloscope for measurement. A Waveform Eductor was obtained on loan and evaluated on tone-burst signals. The instrument showed the ability to greatly improve signal-to-noise ratio.

Integrated system: The completed physical system when assembled was as shown in Figure 7. The system consisted of a 291-16A driver; a 6-inch long, 1.4 to 2-inch exponential transition; a 10-inch long plenum with a 9-inch long, 2-inch diameter, 1/16-inch thick, 10 micron pore mixing tube; a 24-inch straight, 2-inch diameter, round tube section; a 6-inch long round-to-square transition; a 10-foot, round, 2-inch diameter tube; and a 500 Hz low-frequency cutoff exponential horn. One pressure transducer was flush-mounted 12 inches down the 10-foot pipe and another 8 inches upstream of the sample section inlet. A pneumatic compressor was connected to the plenum.

Sample measurements: The system was used to measure the insertion loss of a sample resonant absorber typical of those used on turbo-fan engines.

The test sample consisted of a 1.75-inch by 12-inch section of 10 Rayl felted metal 0.030 inches thick, bonded to a one inch thick transverse section of 1/4-inch cell aluminum honeycomb. This assembly was mounted in a box of 1/8-inch thick aluminum 2 inches by 12 inches covering all sides except the felted metal. Figure 8 shows the sample.

Measurements of insertion loss were made on the sample under the following conditions:

130 dB PPL Input	no flow
160 dB PPL Input	no flow
140 dB PPL Input	+130 ft/sec flow**
160 dB PPL Input	+130 ft/sec flow

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*25 dB is the filter loss one octave from center frequency. The bandwidth is approximately 1/3 octave.

**Flow designated + is the same direction as that of sound propagation.

Test Procedure: Test Procedure was as follows:

1. Mount sample with reflective side inside sample section

2. Air flow off

3. Set oscillator to 1000 Hz

4. Set tone-burst generator for 8 cycles on and 0.3 seconds off

5. Set spectrometer to 1000 Hz

6. Set audio frequency analyzer to 1000 Hz and 25 dB bandwidth

7. Observe output of pressure transducer before sample on oscilloscope and adjust power amplifier gain for the required PPL input

8. Read output of pressure transducer after sample on oscilloscope

9. Reverse the sample so that absorptive side is inside sample section

10. Read output of pressure transducer after sample on oscilloscope

11. Calculate the PPL at the transducer after the sample from steps 8 and 10

12. Determine insertion loss by subtracting the PPL of step 10 from that of step 8

13. Leaving the gain settings of the signal generating equipment as in step 7, change the oscillator, spectrometer, and analyzer frequency to the following in turn and repeat steps 1, 7, 8, 9, 10, 11, and 12: at 500, 630, 800, 1250, 1600, 2000, 2500, 3150, 4000, and 500 Hz.

14. Repeat the whole procedure for each set of conditions turning on and adjusting the air flow where appropriate.

Test results: The results of the measurement are shown in Figures 9, 10, 11, and 12. These measurements were preliminary. The signal generation gain was set at 1000 Hz at a point 8 inches upstream from the sample section inlet and not changed at other frequencies. Input PPL was not necessarily constant. Inspection of the curves does show that at an input of 160 dB PPL, the loss peak is shifted upward for the flow condition. Loss is similar in magnitude for the two conditions. At no flow, the loss peak is higher at 130 than at 160 dB PPL.

Final system. - The preliminary system demonstrated the validity of the approach. A program was pursued to improve the system without making any basic changes. All components received attention. High priority was given to increasing the PPL at the sample inlet and to defining the details of the insertion loss curve in the neighborhood of the peak loss.

Acoustic Source: The original plans called for developing a variable or adjustable resonant driver to increase significantly the peak pressure levels. The driver could take the form of a completely new transducer or the modification of an existing transducer.

The development of a new transducer is very expensive. If a resonant driver were developed it would be necessary not only to be able to vary the frequency of resonance but also the damping. The required output pulse is a "square" pulse passed through a one-third octave filter. If the "square" pulse were applied to a resonant driver with low damping (narrow bandwidth) the build-up and decay times would be too long. If the damping could be adjusted to give a one-third octave bandwidth,a good pulse might be obtained If the damping were high the advantage of the resonance would be lost. For these reasons a simpler approach was sought to increase driver output.

A simple and obvious approach to increasing sound level is to use multiple drivers. If that approach were coupled with an effort to reduce acoustic losses in the system prior to the sample section, sufficient sound pressure would be obtained. This course of action was pursued. Readily available drivers could be used. An additional advantage of this approach is that replacement diaphragm assemblies are available at low cost. Drivers can be run at higher levels and diaphragm assemblies replaced periodically. If a development model of a new transducer is used, a failure can cause long and expensive delays. It is necessary to supply twice the electrical power for two drivers in order to get maximum increase in sound pressure level, however.

The driver in the preliminary system was coupled to the mixing section by a 6-inch, 1.4 to 2-inch exponential transition. A "Y" coupler was obtained that consists of two short 1.4-inch diameter tubes that join in a short 2-inch diameter pipe. The length of each path from driver outlet to mixer inlet is 7.56 inches as compared with 6 inches for the single unit. The coupler not only permits two drivers to be used but acts as an area transition unit as well.

The output of one driver was compared with the output of two drivers. Two 291-16A drivers were used. First a single driver was coupled through the 6-inch transition to a terminated 2-inch tube. A pressure transducer was located 1.5 inches downstream from the transition outlet. The driver was pulsed in the usual way. Electrical input was adjusted to one watt peak at each frequency. PPL was measured at each frequency. A similar measurement was made on the other driver unit. Then the "Y" coupler was installed with the two drivers just tested. The units were connected in parallel and fed with a 2-watt input at the same frequencies. The results were as follows:

FREQ.	DRIVER #1 PPL_1 for 1 watt measured	DRIVER #2 PPL_2 for 1 watt measured	$PPL_1 + PPL_2$ calculated	DRIVER 1+2 on Y throat PPL for 2 watt measured	ΔdB
500	140.7	139.8	143.3	143.8	+0.5
630	141.3	140.7	144.1	144.5	+0.5
800	141.9	141.1	144.5	144.1	-0.4
1000	141.6	140.4	144.1	143.6	-0.5
1250	141.0	139.8	143.5	143.6	+0.1
1600	141.1	140.1	143.7	143.7	0.0
2000	140.1	139.5	142.9	142.8	-0.1
2500	137.9	138.0	141.0	141.4	+0.4
3150	137.0	137.6	140.2	140.3	+0.1
4000	135.0	134.9	138.0	137.2	-0.8
5000	132.4	132.1	135.3	135.2	-0.1
				AVERAGE	+0.04

The sum of the measured intensities for the 2 drivers at one watt each is shown in the table at $PPL_1 + PPL_2$. The fifth column is the measured PPL for the pair of drivers at 2 watts. The average difference is +0.04 dB which is negligible. It is evident that PPL can be increased by the factor 10 log N where N is the number of transducers if power per transducer is kept constant. It should be possible to use more than two drivers and obtain greater sound pressure. This arrangement has been used on some high-intensity progressive wave tubes with as many as 16 electromagnetic drivers. In this case the path length does get longer and some power is lost due to tube attenuation.

Air-Sound Mixing Section: The Air-Sound Mixing Section was not changed. Two characteristics of the section are important in operation of the system. One is the acoustic loss due to the impedance of the sintered cylinder. The other is its resistance to air flow. Measurements were taken in the mixing chamber of air flow velocity through the tone-burst tube versus the air pressure in the mixer plenum. For the maximum flow velocity desired of 310 ft/sec the pressure required was 10 psig for the 9-inch section of 1/16-inch thick, 10 micron pore, sintered tube. Any tube with lower pore size or greater thickness required more than 10 psig to produce 310 ft/sec. For use with an exhaust pump for reversed flow, 10 psig is about the maximum one can get. Previous measurements showed acoustic losses attributable to tube impedance of 0.5 to 3 dB depending upon frequency. To use a larger pore size or a thinner section would result in lower impedance and greater acoustic losses.

Flow Settling Section: The most profitable section in which to decrease losses is in the Flow Settling Section. The preliminary system had a 24-inch long section of 2-inch pipe between the outlet of the mixer and the inlet of the round-to-square transition before the sample section. This length was determined by the requirement for fully developed flow. Figure 13 shows the general loss characteristics of 2-inch tubes. Two approaches were taken to reducing the losses in the tube. The first was to increase the diameter of the tube from 2 to 5-5/8 inches.

To check the comparative losses of the two sizes of pipe a test was made. The setup consisted of a driver, an exponential transition, the test tube, and a 10-foot 2-inch tube and horn. First, a 10-foot section of 2-inch tube was connected between the transition and terminating tubes. Pulses of 166 dB PPL were applied at the transition. The PPL was measured immediately after the 10-foot section being tested. Then a tube section 10 feet long consisting of a 2 to 5 5/8-inch exponential transition, 6 inches long; a 108-inch long section of 5 5/8-inch tube; and a 5 5/8 to 2-inch exponential transition, 6 inches long. Figure 14 shows the test results. At 4000 Hz the 2-inch tube had 18.5 dB attenuation while the 5-5/8 inch tube had 11.0 dB. At lower frequencies the differences were less. The improvement by using a 2 foot tube of 5 5/8-inch diameter instead of 2 inches diameter would result in a reduction of only 1/5 of the 10 foot loss or 1.5 dB. The complication of using 2) 6-inch transitions and a 1 foot piece of 5 5/8-inch tubing to make up the 2 feet did not seem profitable. In addition there are bothersome cross modes in the larger pipe at high frequencies.

The second approach was to try to hasten the development of fully turbulent flow and to reduce the length of the 2-foot settling section. Experiments were made with "flow trippers" to determine whether the onset of fully-developed turbulence could be hastened.

The criterion for full turbulent flow used in this case was:

$$\frac{u}{U_R} = \frac{y}{\delta}^{\frac{1}{n}}$$

Where: U_R = maximum velocity of profile
u = velocity at y
y = radial distance from pipe circumference
δ = value of y where $u = 0.99\ U_R$

The value of n varies with the overall mass-flow rate. In the range of mass-flow rates used in the tone-burst system, n is in the order of 7-10.

Figure 15 shows flow profiles for n = 7, 8, and 10. The profile for laminar flow is also shown. In the transitional region where turbulent flow is developing, the profiles have intermediate shapes.

Several flow-tripping devices were selected for test. They were installed between the outlet flange of the mixing section and the inlet flange of the flow settling section. Flow profiles were measured at various distances down the flow tube. Flat-plate, vaned, and wire trippers were tested. Figure 16 shows the different types. The flat-plate trippers are 0.062-inches thick having center holes smaller than the inside pipe diameter. Hole sizes of 1.875, 1.750, and 1.625 inches in diameter were tried. Vaned trippers with various numbers and types of vanes were tested. Both types of wire trippers were used.

Of the three types tested, the flat-plate tripper with the 1.625-inch hole was the best, producing fully-developed flow at the shortest distance. Figure 17 shows the flow profile at 33-1/2 inches from the mixing section outlet and no flow tripper at an intermediate average flow velocity of 130 ft/sec. This profile is fairly close to the theoretical profile for n=7. For the preliminary system 40 inches was considered to have fully turbulent flow but 30 inches of settling section was used (24 inches of 2-inch pipe and a 6-inch round-to-square transition). Figure 18 shows the profile with no tripper at 9-1/2 inches and 130 ft/sec flow. Figure 18 also shows the profile at the same distance and flow with the 1.625-inch flat-plate tripper. There is an improvement but the profile indicates only partially developed flow. Figure 19 shows the profiles at 15-1/2 inches and flows of 42 and 320 ft/sec with the 1.625-inch flat-plate tripper. Both profiles have values of n greater than 10. Fully developed flow is indicated.

The Flow Settling Section could be reduced in length from 30 to 15 inches (9 inches of 2-inch pipe and the 6-inch round to square transition).

Acoustic insertion loss measurements on the 1.625-inch tripper at a PPL of 150 dB ranged from 0 at 500 Hz to 0.6 dB at 5000 Hz.

A Flow Settling Section of 12 inches was chosen for the final system. The reductions in tube losses should be as indicated by previous tests:

Frequency Hz	Decrease in Loss dB
500	0.1
1000	0.5
2000	1.5
5000	3.5

Reference-Sample Section: During tests on the preliminary system it was found that pressing the permanent 1/8-inch thick sides of the Sample Section affected the output pulses of the system. The section was redesigned and rebuilt with 1/4-inch sides having heavy ribs. Permanent clamps were built in for holding samples. Figure 20 shows a sketch of the section. No changes in basic dimensions were made.

Acoustic Termination: The effects of the type of horn on the 10-foot termination tube were measured. The system was set up with the two drivers, mixer, new settling section, and sample section with hard sample. The 10-foot termination tube was installed. A swept-sine wave at constant voltage was used to energize the drivers while the sample output transducer signal was recorded on a level recorder. Runs were made with: no horn; the 500 Hz sectoral horn; a 10-inch long, 14-inch mouth diameter, 370 Hz horn; a 45-inch long, 13-inch mouth diameter, 150 Hz horn; and a 64-inch long, 32-inch mouth diameter 150 Hz horn.

The transducer level-versus-frequency records are characterized by periodic fluctuations caused by horn throat impedance variations. To compare the various conditions, the transducer level fluctuation amplitude at various frequencies is shown in the following:

	Fluctuations in Transducer Level - dB				
Horn	125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz
NONE	30	20	18	13	6
500 Hz Sectoral	29	18	9	3	1
370 Hz 14 in Dia.	27	14	4	2	1
150 Hz 13 in Dia.	17	11	4	2	1.0
150 Hz 32 in Dia.	16	6	3	2	1.0

The data show the improvement made by using a lower frequency cutoff horn and a larger horn mouth. At 1000 and 2000 Hz all the horns perform equally well. At 500 Hz, the 500 Hz horn fluctuations are 9 dB as compared with 4 and 3 for the other horns.

The 500 Hz sectoral horn is as good as the larger low-frequency horns at 1000 Hz. At 500 Hz it shows larger fluctuations. These effects show up most strongly when an absorber is being measured with a swept-sine wave.

The 500 Hz horn is adequate for samples with resonances at higher frequencies when tone-bursts are used. For swept-sine wave measurements a better recorder trace is obtained below 1000 Hz with the 150 Hz, 32-inch mouth diameter horn.

Pressure transducer: During the preliminary system tests some difficulty was experienced with the mounting of the pressure transducer. The transducer body is threaded and can be screwed directly into the wall of a pipe. The thread-relief groove at the base of the mounting threads of the transducer leaves a wall approximately 10 mils thick. Even finger tightening altered the transducer sensitivity. The problem was eliminated by a different method of mounting. The transducer is mounted in a plate assembly which in turn is fastened to the tube. A sketch of the mounting is shown in Figure 21. Using the new method, the microphone assembly was mounted, dismounted, and remounted 10 times. For the same sound pressure the microphone readings showed a range of variations from -0.02 to +0.1 dB. In the final system the two pressure transducers were to be permanently mounted, one in each of the round-to-square transitions. It would not be necessary to move them thereafter. For that reason the transducers were screwed directly into threaded holes.

Instrumentation: Some additions were made to the instrumentation for the final system.

In tone-burst tests a special one-third octave filter that could be varied in center frequency was at times substituted for the Audio Spectrometer. The variable filter was used to make measurements at frequencies other than the standard fixed values in the spectrometer.

For making swept-sine wave measurements on absorbent samples, a different instrument set up was used. Figure 22 shows a block diagram of the system.

The Beat Frequency Oscillator is set to sweep in synchronism with the Level Recorder. The oscillator output feeds the Power Amplifier, Transformer, and Drivers. The sample-input pressure transducer is connected through the Audio Spectrometer set on Linear (all-pass) to the Compressor Input of the Beat Frequency Oscillator. When the oscillator drive is started, the output of the Input Microphone operates the compressor to maintain the sound pressure level constant at that point. The Level Recorder can then be used to record the input or output sound pressure levels versus frequency at the sample.

The output curve is the transmission loss or attenuation of the sample versus frequency once the proper dB scale is assigned to the record. The scale is determined from the calibrations of the microphones and the input level.

Integrated system: The complete physical system when assembled is shown diagramatically in Figure 23. A photograph of the system is shown in Figure 24. The system consists of 2 type 291-16A driver units; the "Y" coupler; the 10-inch long plenum with 9-inch long, 2-inch diameter, 1/16-inch thick, 10 micron pore mixing tube; the 1.625-inch flat-plate flow tripper; the 6-inch long, 2-inch diameter settling tube; the 6-inch long round-to-square transition; the new 14-inch long, 1.75-inch square sample duct; the 6-inch long square-to-round transition. The pressure transducers are 2.25 inches from the start and end of the sample. Figure 25 shows the drivers, mixing section, and sample section. Either a pneumatic compressor or exhaust pump can be connected to the plenum through a flow meter.

The instrumentation for tone-burst tests is shown in Figure 26.

Test Procedure Development

General test procedure. - The objective of tone-burst flow tube technique development is to make available a simple means of evaluating the acoustic losses of absorptive duct linings under conditions of very high acoustic intensity and air flow. The physical system has been described. The development of test procedures occurred concurrently with the development of the system and affected that development. The procedures were developed to obtain certain specific data on duct linings.

The data to be obtained were the insertion losses of a duct-lining sample at medium and very high sound pressure levels over the frequency range of 500 to 5000 Hz in the presence of appropriate air flow with and against the direction of sound propagation. The data were to be compared with similar data taken by other laboratories or in the field.

The general test procedure was to use tone-bursts in a point-to-point comparison against frequency between the transmission loss of a reflecting reference duct and a substituted absorptive sample duct. The input bursts to the reference duct were to be adjusted to the specified PPL at each frequency. The input electrical power to the drivers was to be the same for the sample. Insertion loss could then be determined by subtracting the output PPL for the sample from the output PPL for the reference. Air flow was to be maintained constant throughout a test run.

Samples. - The conformation and size of sample orignially chosen was a 1.75 x 11.75 inch rectangle. This sample was mounted in a rectangular 1/8-inch aluminum holder with outside dimensions of 2 by 12 inches. A sample in its holder is shown in Figure 8. The width of the sample had been made to match one side of the square sample section whose dimensions were derived from the 2-inch pipe area. For a wider sample, the cross sectional area would be larger for a square section, reducing the intensity of the acoustic signa. The length was arbitrarily chosen to be one foot. The sample section will accommodate shorter samples. Shorter samples can be made by facing part of the sample holder length with solid metal. For samples longer than 11.75 inches a longer sample section could be made, using the same basic design. Time did not permit an extensive study to determine optimum sample size. Results of tests on several samples indicate that the choice of sample size was satisfactory.

The sample holder is designed so that samples can be mounted on two facing sides of the square duct. A sample holder could easily be designed to hold samples on three or four sides.

The procedure for making and mounting samples is discussed in Appendix A and Reference 5. Samples with holders are installed by loosening the nuts on the bar clamps shown in Figure 20, swinging the bars around, pulling the sample or reference away from the sample section, removing the mold seal, placing the sample into the section, swinging back the bars and tightening the nuts, and applying a bead of mold seal around the sample-to-holder joint.

Pulse characteristics. - The investigation of pulse behavior in the system was a major effort. Most of the work concerned the transmission of pulses through narrow-band networks such as resonant absorbers. The effect of amplitude distortion on pulses was also considered.

Pulse shape: The shape of pulse will not be changed if it is transmitted through a network that has either no phase shift or phase shift proportional to frequency. Networks characterized by phase shift which is neither zero nor proportional to frequency will change the shape of pulses. In general, networks with that type of phase shift also have non-flat frequency responses. A straight piece of lossless pipe with a diameter much smaller than the shortest wavelength is a network with phase shift proportional to frequency. The only effect on a signal is a time delay. A tone-burst transmitted through such a pipe experiences only time delay. The oscillogram of the received pulse is identical to that of the transmitted pulse. On the other hand, the "square" pulse turns into a gently rising and falling pulse in being transmitted through a one-third octave filter.

Comparison of PPL's to determine transmission or insertion loss was originally predicated upon ducts with phase shift proportional to frequency. A resonant absorber fulfills that condition only at both low and high frequencies far from the frequency of resonance. It was noted during the preliminary system tests that near resonance the pulses at the absorber output had changed shape. When one of the standard one-third octave frequencies was near resonance, the output pulse looked like two half-length input pulses joined together. Figure 27 shows the type of pulse observed. The problem was where in the pulse to measure the PPL. The test operator chose to measure amplitude at the center of the pulse which was actually the minimum pulse amplitude. This was not a good choice. An investigation was made to determine the best method of measuring such "distorted" pulses.

It was decided to study electrical pulse behavior in resonant electrical circuits analogous to the preconceived nature of resonant absorbers. It was assumed that a resonant absorber made of felted metal over honeycomb could be represented as a single Helmholtz resonator. Figure 28 shows results of some measurements on a simple electrical analogy to a single Helmholtz resonator. Insertion loss measurements were made with sinusoidal signals and tone bursts formed from 8 cycle "square" pulses modified by one-third octave filters. The input and output pulses of the analog network were identical to those of Figure 27. The amplitude of the output pulse was measured at its minimum point. The pulse measurement indicates an attenuation 3 dB less than the sinusoidal measurement.

Further measurements of a similar nature were made on a network with variable bandwidth and variable damping. Here four types of measurement were compared: sine wave; peak-to-peak tone burst at center of pulse; integral of tone burst; and one-third octave random noise. At the frequency of resonance the measured insertion losses by the different methods were

sine wave	32.04
peak-to-peak	29.96
integral	30.62
1/3 octave	26.84

The sine wave measurement was not matched by any of the methods.

Using a narrower band version of the same network, a comparison was made at resonance between the behavior of a normal input tone burst and an elongated "square" pulse modified by a one-third octave filter. The pulses are shown in Figure 29. Here the peak-to-peak measurement made with the long pulse at the uniform center section gave a loss equal to a sine wave measurement. Computer calculations confirmed the pulse behavior.

The conclusion was that when the sample output pulse shape, for a normal tone-burst input is distorted, a longer "square" pulse should be used. The length should be determined by gradually increasing the pulse length until a meaningful measurement can be made. Later measurements made on actual resonant samples gave more complicated pulses.

Very careful tone-burst measurements at small frequency intervals and the swept-sine tests on resonant samples revealed a series of dips and peaks where only one peak had been expected. As a confirmation, analog measurements were made on an electrical network with two dips close together. Figures 30, 31, and 32 show the loss characteristics of the network and typical output pulses.

Amplitude distortion. - If a high-intensity sinusoidal pressure wave is propagated through a pipe or duct it degenerates into a stable sawtooth wave after a certain distance.

In the process, harmonics of the sine wave are produced. A sawtooth wave with a maximum amplitude of "a" has harmonic components with amplitudes according to the following relationship:

$$Cn = \frac{-2a}{n\pi} \cos (n\pi)$$

where Cn is the amplitude of nth harmonic. The amplitudes of the first four harmonics are:

n	Cn
1	0.637a
2	0.319a
3	0.212a
4	0.157a

and a rms = 0.576a

Assume that a sinusoidal wave of maximum amplitude "b" is converted to a sawtooth wave of amplitude "a" without loss of energy.

then a rms = b rms

and 0.576a = 0.707b

and a = 1.225b.

If peak amplitudes only are measured the conversion appears to produce a gain of 1.8 dB.

If the sawtooth wave is passed through a narrow-band filter to separate out the fundamental "c"

c = 0.637a = 0.78b

Then for peak amplitudes there appears to be a loss of 2.2 dB. This simple analysis indicates that for non-linear systems care must be taken in interpreting measurements made on a peak amplitude basis.

Some experiments were made in a 2-inch tube to determine the magnitude of the effect of non-linear distortion. An electromagnetic driver was mounted on a 2-inch terminated tube. Microphones were mounted in the tube wall at a reference point close to the source and at distances of 1 and 4 feet downstream from that point. The driver input was an 8-cycle long, one-third octave-filtered, pulse. The outputs of the microphone were displayed on an oscilloscope.

Patterns were observed for microphone outputs directly and passed through a 1/3-octave filter. Tests were made at 1000, 2000, and 4000 Hz.

The test results are as follows:

Frequency Hz	Filter	Reference PPLdB	1 foot PPLdB	1 foot Loss	4 foot PPLdB	4 foot Loss
1000	in	166.2	164.0	2.2	163.2	3.0
	out	166.3	164.3	2.0	164.2	2.1
2000	in	166.2	163.2	3.0	160.6	5.6
	out	166.7	164.2	2.5	163.0	3.7
4000	in	164.6	160.6	4.0	156.3	8.3
	out	165.1	162.1	3.0	158.5	6.6

At 4 feet, for example, and 4000 Hz the measured loss is 8.3 dB for a filtered signal and only 6.6 dB for an unfiltered signal. The difference is 1.7 dB. Oscillograms of some of these pulses are shown in Figure 33. The effect will be somewhat different in the use of the system for actual absorbers.

The effects of amplitude distortion on tone-burst measurements made on actual samples cannot readily be determined. Whether it is better to use filtered or unfiltered output pulses is also an open question. It was decided that filtered pulses probably gave better results. For the final system filtered pulses will always be used.

<u>Pressure transducers</u>. - The Piezoelectric pressure transducers originally selected were satisfactory and were retained for the final system. The locations of the transducers were changed. One transducer is now located in the round-to-square transition, 2.25 inches from the sample outlet. The first location allows the input level to the reference and sample to be accurately measured because of proximity. The second location is close to the sample output. The noise due to flow turbulence is the same throughout the flow tube. By measuring the signal as its point of maximum level in the system the best signal-to-noise ratio is obtained.

<u>New test procedure</u>. - The new test procedure was worked out over an extended period of time while system components were being developed. The major sub-procedures are:

Sample screening by sine-wave sweep
Tone burst evaluation.

Sample screening: It is valuable to have a rough idea of the transmission loss characteristics of an absorber sample before a tone-burst test is made. It permits the operator to pre-select the tone-burst test frequencies. Samples with fabrication faults may also be uncovered in this procedure. The test also has an intrinsic value as a valid measurement of transmission loss giving fine detail in terms of a true sinusoidal response.

The instrumentation for this procedure is shown in Figure 22. The sample is mounted. The level recorder is switched to the input pressure transducer circuit. The oscillator is set at a convenient frequency above 500 Hz. The input to the drivers is adjusted to produce the desired SPL at the sample inlet. The compressor circuit of the oscillator is adjusted to hold the SPL at the sample inlet. The compressor circuit of the oscillator is adjusted to hold the SPL constant over the measurement frequency range. The oscillator is set at 500 Hz and a swept-sine run is made on the level recorder. The record should be essentially flat from 500 to above 4000 Hz. This chart represents the input SPL to the sample. The level recorder is switched to the output pressure transducer circuit. Without changing any settings another swept-sine run is made. The record is the output SPL of the sample. Since the input SPL was constant, the output chart represents the relative attenuation of the sample.

Figure 34 shows typical results of swept-sine measurements on a sample of resonant absorber. The characteristic pattern for a felted metal over honeycomb sample is shown. The curve provides a guide to selection of frequencies for a tone-burst test.

Tone burst evaluation: A typical tone-burst evaluation will include the following:

No-flow test at low and high PPL.
Positive* flow test at medium and high flow rates at low and high PPL.
Negative* flow test at medium and high flow rates at low and high PPL.

The procedure for all tests is the same except that appropriate flow conditions are set up and maintained.

Instrumentation for tone-burst evaluation is shown in Figures 5 and 6. In Figure 5 a one-third octave filter with adjustable center frequency is substituted for the Audio Spectrometer.

The general procedure for a tone-burst evaluation of a sample is first to establish a baseline for the particular test conditions on the reference duct. This is essentially an input-output transmission measurement on the reference duct at a constant input. With the same driver power, an input-output transmission measurement is made on the sample duct. The difference between the output level of the reference and the output level of the sample is the insertion loss of the sample. The difference between the input level to the reference and the input level to the sample indicates differences in input impedance between the two duct configurations.

A completely detailed test procedure is given in Appendix A. A condensation of the procedure will suffice for an understanding of the procedure. The condensation is as follows:

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*Positive flow is in the direction of sound propagation. Negative flow is in the opposite direction.

1. Mount the sample holders with the reflective sides inside the sample section.

2. Set the air flow as required.

3. Set the oscillator at the lowest frequency selected from the swept-sine wave test.

4. Set the pulse generator for 8 cycles on and 0.3 seconds off.

5. Set the spectrometer** or adjustable input filter to the frequency of step 3.

6. Set the frequency analyzer to the frequency of step 3.

7. Observe the peak-to-peak output of the pressure transducer before the sample section on the oscilloscope and adjust the oscillator output and power amplifier gain for the required PPL.

8. Read the peak-to-peak driver voltage with the oscilloscope.

9. Read the peak-to-peak output of the pressure transducer after the sample section on the oscilloscope.

10. Repeat steps 3-9 for all test frequencies.

11. Remove and reverse the sample holder(s) to place the absorbent side(s) inside the sample section.

12. Set the oscillator to the lowest frequency required.

13. Set input and output filters to the frequency of step 12.

14. Read the peak-to-peak driver voltage and adjust the oscillator output and power amplifier gain for the voltage from step 8 for the test frequency being used.

15. Read the peak-to-peak output of the pressure transducer before the sample and of the pressure transducer after the sample. (See note)

16. Repeat steps 12-15 for all test frequencies.

NOTE: When the output of the pressure transducer after the sample is a typical tone-burst, peak-to-peak voltage is measured at maximum pulse amplitude. When the transducer output pulse is double or odd shaped, increase the "on"

**Spectrometer may be used for standard one-third octave center frequencies.

setting of the pulse generator from 8 to 16, 32, or 64 until a pulse is obtained with a constant amplitude center section. Read the peak-to-peak voltage of that section. Figures 35 and 36 show pulses illustrating the procedure.

The raw data consist of peak-to-peak pressure transducer output voltages. All the peak-to-peak voltages are reduced to peak voltages by dividing by two. The peak voltages are then converted to PPL in dB re: $2 \times 10^{-5} N/m^2$. This is done by taking account of the pressure transducer calibration sensitivities and the transducer circuit gains. Then insertion loss is computed by subtracting the sample output PPL's from the reference output PPL's.

For comparison with data in Reference 2 for resonant absorbers an insertion loss versus frequency curve is plotted on log frequency, linear dB paper. An insertion loss is found where the bandwidth of the curve is one octave. This value is the effective insertion loss of the absorbing duct. Figure 37 shows the insertion loss curve of the sample of Figure 34 as measured at no flow and 155 dB PPL input.

Assembly and operating instructions. - For complete and detailed assembly, operating, and testing procedures see Appendix A.

Specifications for instrumentation. - For complete, detailed specifications for all system components see Appendix B.

Maximum system performance. - The maximum PPL obtainable at the sample inlet was significantly increased during the development of the final tone-burst system. A direct comparison between the preliminary and final systems was made. In each system the PPL input to the sample was measured at 2.25 inches upstream of the sample. The systems were run without flow. In any system, maximum PPL is limited either by the driver(s) or by the power amplifier.

To ensure maximum power from the amplifier, for each measurement the transformer impedance ratio is adjusted to optimum. Limitation by the amplifier is evidenced by operation of the overload circuit and cutting off of the amplifier. The amplifier overload circuit will cut the amplifier off if the output voltage exceeds 48 volts peak or if the output current exceeds 42.4 amperes peak. The amplifier has a one ohm output impedance. By observing the input voltage of the driver at the transformer output with the oscilloscope voltage cutoff can be anticipated. Cutoff voltages at the driver will be as follows for the most commonly used transformer impedance taps:

Transformer Impedance ohms	Peak Voltage
No transformer	48
4.	96
5.8	115
7.8	134

10.2	153
13.	172
16.	192
19.	209

The nominal volt-ampere rating of the amplifier is 1150 RMS and 2300 peak. Driver PPL limitation is indicated in any of a number of ways none of which is as clear cut as amplifier cutoff. The oscilloscope trace of the driver voltage bouncing up and down at a peak voltage lower than amplifier cutoff usually indicates impending driver failure. Sharp spikes in the driver voltage indicate driver limitation. Clicks and erratic pulses heard from the system are indicators. Complete failure of the driver may also occur.

Three system configurations were measured:

Preliminary system - one driver

Hybrid system - two drivers

Final system - two drivers

The preliminary comparison was to be between the preliminary and final systems. The hybrid system was the same as the preliminary except that two drivers and the Y coupler were used. Another condition measured was on the preliminary system at driver voltages equal to maximum final driver system voltages.

In the final system the two drivers are connected in parallel, lowering the transformer load impedance to one-half of the single driver impedance. If one 16-ohm driver is amplifier limited in voltage it is absorbing the total amplifier power. If two drivers are connected (8 ohms) and the transformer tap is reduced to half impedance the total power into the drivers is still the same. There will be no increase in PPL. If the single driver is using only half the available amplifier power then when two drivers are connected, the amplifier gain can be increased and they will receive the full power. In this case the PPL will be increased by 3 dB.

As will be evident from the test results, at some frequencies the systems were transducer limited and at others amplifier limited.

Test results and discussion: Table 1 shows the primary result of the test, the direct comparison of the preliminary and final systems operating at maximum power. Maximum PPL at the sample inlet is essentially constant at about 165 dB from 800 to 1600 Hz in the preliminary system. Above 1600 Hz, PPL falls off at about -2dB per one-third octave. The increase in sample inlet PPL of the final system over the preliminary system is an average of 2.6 dB. The increases are somewhat greater at the lower frequencies.

Table 2 shows the maximum PPL in the hybrid system, which is the preliminary system with two drivers. The portion of the total increase of the final system due to shortening the flow tube is the difference between the PPL in the hybrid and final systems. The average increase due to shortening the flow tube is 1.8 dB being a minimum at 800 Hz and a maximum of 3150 Hz.

Table 3 lists the difference between the maximum PPL in the preliminary and hybrid systems. The difference represents the portion of the final system increase due to using two instead of one driver. The increase is 2.6 dB at 800 Hz and 2.2 dB at 1000 Hz. From 1250 Hz upward in frequency the increases are random and less than 1 dB. The reason for the limited increase will be explained in detail later. It is a deficiency of amplifier output power.

Table 4 shows the increase in PPL caused by adding a second driver when the same drive voltage is applied to each driver. The increases average 3.5 dB which is a little better than the 3 dB one might expect. The two drivers must be better matched to the flow system. These data indicate the doubling of acoustic power at the sample by adding a driver. The difference between these results and those of Table 3 is that in the one case equal power was applied to each driver and in the other total available amplifier power was applied to one driver and to the pair of drivers.

Table 5 is the basis for the explanation of why the addition of a driver did not increase the final system output by 3 dB at all frequencies. The table lists for both the preliminary and final systems the transformer tap, the maximum transformer output voltage available, the limiting driver voltage, and whether the system is driver or amplifier limited. In the preliminary system from 800 to 1250 Hz the limit was set by the driver. The driver voltages are all lower than the amplifier limit. At frequencies of 1600 and higher the limit appears to be set by the amplifier. It is obvious at this point that two drivers are going to be more severely amplifier limited than one. In the final system at 800, 1000, and 1250 Hz the limits are still set by the drivers. At higher frequencies, final system driver voltages are much lower than preliminary system voltages. In order to get the full benefit of two drivers the voltage on two must be the same as on one. The last column in the table gives the amount by which the amplifier output must be raised to fully drive two drivers. The values were calculated from the relationship

$$\text{Deficiency} = 20 \log_{10} \frac{\text{Max. preliminary driver volts}}{\text{Max. final driver volts}}$$

The average deficiency is 3 dB. To get full advantage of the final system each driver should have its own amplifier. The best system would be one in which the driver is the limiting component of all frequencies. There is a possibility with a larger amplifier that higher output could be obtained from each driver at high frequencies. The possibility could not be confirmed since a higher-powered amplifier was not available.

Table 6 shows the estimated maximum PPL that might be produced in the final system if the power deficiencies were made up. PPL of over 166 dB would be available from 800 to 2500 Hz. Even at 5000 Hz the PPL would be almost 162 dB.

Table 7 lists the total peak acoustic power available at the sample for preliminary, final, and estimated final. These data are plotted in Figure 38. Also plotted are two curves taken from the proposal[6] for the project. The preliminary system power is higher than the original capability curve. The final system power is somewhat lower than the original projected curve. The power of the estimated final system with two amplifiers would exceed the projected power above 1250 Hz.

Evaluation of Materials

Background. - The purpose of this portion of the program was to demonstrate the utility of the tone-burst technique, using the final system. The method of demonstration was to evaluate typical lining materials and to compare the results with available data on similar duct treatments. It was decided to make one of the sample materials fiber-metal over honeycomb. This material was extensively studied by Atvars and Mangiarotty in "Parametric Studies of the Acoustic Behavior of Duct-Lining Materials" (Reference 2). The influence of significant parameters was determined. Empirical equations in terms of the parameters were developed for single-layer lining behavior. The characteristics of a single-layer lining evaluated in the tone-burst flow tube system could then be conveniently compared with results given in the paper. "A Study of Turbofan-Engine Compressor-Noise-Suppression Techniques" by Marsh, et al (Reference 3) gave results of tests using the same test apparatus as Atvars and Mangiarotty. Among the absorbers tested were various grades of fiber metal over free-air spaces. The acoustic characteristics here were measured in specific configurations representing full-size engine ducts. The comparisons between those results and tone-burst tube results would not be expected to be as straightforward.

Samples. - Two resonant and two broad-band samples were assembled for the tone-burst tests. The pairs of samples were made as alike as possible.

Resonant absorbers: The resonant samples were designated #1 and #2. The elements of a sample are a sample holder, honeycomb filler, and fiber metal surface plate. The sample holder is a box made of 1/8-inch aluminum with inside dimensions of 1-3/4 inches wide, 1 inch deep, and 11-3/4 inches long. One of the 1-3/4 by 11-3/4 inch sides is open. A flat piece of 10-Rayl fiber metal 1-3/4 by 11-3/4 is cut out. Then a piece of aluminum honeycomb with 1/4-inch cells 1 inch thick is cut into a 1-3/4 by 11-3/4-inch rectangle. The axes of the cells will be at right angles to the fiber metal. One face of the honeycomb is dipped in adhesive. The felted metal is placed on top of the honeycomb with the adhesive and weighted down with a uniformly distributed weight. After the adhesive has cured the assembly is placed in the holder so that the open side of the honeycomb is against the bottom of the holder and cemented to it. The fiber metal is cemented to the holder around its edges. Reference 5 shows a number of ways to fabricate this type of resonant absorber.

Broadband absorber: The broadband absorber samples were designated #3 and #4. These samples were made by cementing a 1-3/4 by 11-3/4 inch flat piece of 50-Rayl fiber metal into the open side of a sample holder 1/2-inch deep.

Test Conditions. - Five sample-duct configurations were used:

a. 1-3/4 inch square duct, 11-3/4 inches long, sample #1 on one wall, three walls hard

b. 1-3/4 inch square duct, 11-3/4 inches long, sample #2 on one wall (opposite to side of configuration a), three walls hard

c. 1-3/4 inch square duct with sample #1 on one wall, sample #2 on opposite wall, and two walls hard

d. 1-3/4 inch square duct with sample #3 on one wall, three walls hard

e. 1-3/4 inch square duct with samples #3 and #4 on opposite walls.

The general test procedure on a particular sample was to first make a swept-sine transmission measurement at an input SPL of 130 or 140 dB. Then tone-burst insertion loss measurements were made for a number of PPL's and flow conditions according to the procedure of Appendix A. A full series of measurements was made on sample #1 for condition a. For the other samples only some of the measurements were made. The series was:

1.	No flow	155 dB PPL input
2.	No flow	160 dB PPL input
3.	No flow	165 dB PPL input
4.	+130 ft/sec (0.115 M)	155 dB PPL input
5.	+350 ft/sec (0.32M)	155 dB PPL input
6.	+350 ft/sec (0.32M)	160 dB PPL input
7.	+350 ft/sec (0.32M)	165 dB PPL input
8.	-130 ft/sec (-0.115M)	155 dB PPL input
9.	-130 ft/sec (-0.115M)	160 dB PPL input
10.	-130 ft/sec (-0.115M)	165 dB PPL input

The data were converted to plots of insertion loss versus frequency. The octave insertion loss was graphically determined from the plots for resonant absorbers. Octave insertion loss is the insertion loss where the curve is exactly one octave wide. The center frequency of the octave bandwidth was also determined.

Test results. - Test results for sample #1 are given in Table 8 and Figures 39-44.

Test results for sample #2 and samples #1 and #2 together are given in Table 9 and Figures 45-47.

Test results for samples #3 and#4 are given in Table 10 and Figure 48.

Discussion. - Reference 2 gives empirical equations for determining the resonance frequency and insertion loss at octave bandwidth for fiber-metal honeycomb absorbers.

Resonant absorbers: The expression for the peak attenuation (insertion loss) frequency is:

$$f_p = 4.86\ (h + 5.3d - 19.3)^2 + 1530\ (d-1.2)^2 + 800\ M + 980\ \text{(in Hz)}$$

where f_p = frequency of peak attenuation

h = height of duct

d = depth of absorber (thickness of honeycomb)

M = Mach number

for $0.25 \leq d \leq 1.0$, $4 \leq h \leq 12$, $-0.4 \leq M \leq 0.4$

For one octave attenuation bandwidth

$$f_{bc} \simeq f_p$$

where f_{bc} = center frequency of octave attenuation band

A chart is given for correcting f_p for metallic low-reactance linings since the expression was derived for high reactance non-metallic lining.

The expression for octave-bandwidth attenuation is:

A $$A = [1.26\ (10^{-0.09h})\ \ell + 4.5] \times [1.06 - 0.96(d-0.75)^2]\ \text{(in decibels)}$$

where A = attenuation (insertion loss) for 1-octave bandwidth

for $\ell \geq 10$ in., $0.25 \leq d \leq 1.0$ in., $4 \leq h \leq 12$ in
$0 \leq M \leq 0.4$

Two families of curves are given to correct A* for air flow.

$$A^* = 1.06 - 0.96\ (d-0.75)^2$$

One more restriction is placed on the expressions. The width of the lining should be approximately twice the duct height or more. It is also stated that results of tests with one wall lined in a duct of height h/2 is equivalent to two walls lined in a duct of height h.

The tone burst technique is different in several respects from the tests described in Reference 2. In the latter technique sample ducts were generally six inches in height. For the frequencies of interest, 2000 Hz ± one octave, wavelength is comparable to or smaller than the duct cross secitonal dimensions. The sound enters the duct from a reverberant chamber in random directions. Both these conditions tend to encourage cross modes and non-parallel sound incidence on duct walls. In the tone-burst system progressive waves are used and the 2-inch diameter is generally smaller than the wavelengths used. The tone-burst system gives a sine-wave response whereas the other system uses random noise (measurement bandwidth not stated). The tone-burst samples are only half as wide as the effective duct height for single samples and equal to it for double samples. Duct heights in the tone burst system are 3.50 and 1.75 inches. The empirical expressions are limited to 4-inch duct height. However, the differences are not very great and a meaningful comparison of results can be made for the 3.50-inch height.

Using the expression for f_p and correcting for the low-reactance linings and for f_{bc}, the following values are calculated for one resonant sample in the tone-burst system:

$M = 0$, $f_{bc} = 2107$ Hz
$M = 0.115$, $f_{bc} = 2199$ Hz
$M = 0.32$, $f_{bc} = 2363$ Hz
$M = -0.115$, $f_{bc} = 2015$ Hz

For two samples in the tone-burst system:

$M = 0$, $f_{bc} = 2561$ Hz
$M = 0.115$, $f_{bc} = 2653$ Hz

Using the expression for A and correcting for Mach number for a single sample:

$M = 0$ or $+$ $A = 11.6$ dB
$M = -0.115$ $A = 12.8$ dB

No correction procedure was given for flow resistance (10 Rayl versus 30 Rayl for Reference 2). There is indication that A would be somewhat smaller for 10 Rayl material.

For two samples:

$M = 0$ or $+$ $A = 14.8$ dB

These estimated values can now be compared with the tone burst data.

Comparison of f_{bc} values gives the following from Tables 8 and 9.

Sample	Flow M	Sine	f_{bc} Tone Burst Average all PPL's	Ref. 2	Error
#1	0	2000		2107	-4%
#1	0		2066	2107	-2%
#1	0.115		2260	2199	+3%
#1	0.320		2420	2363	-2%
#1	-0.115		1917	2015	-5%
#2	0		2120	2107	+0.5%
#2	0.115		2280	2199	+4%
#2 and #3	0	2400	2630	2561	+3%
#2 and #3	0.115		2600	2653	-2%

The tone-burst frequencies match those calculated from Reference 2 within better than ±5%.

Comparison of A values from Tables 8 and 9 and those calculated from Reference 2 gives the following for sample #1:

Sample	Data Averaged over Flow M	Data Averaged over PPL dB	A - dB Sine	A - dB Tone Burst	A - dB Ref. 2	Error dB
#1	0	140 SPL	11.5		11.6	-0.1
#1	0	155-165		13.3	11.6	+1.7
#1	0.115 to 0.32M	155-165		10.8	11.6	-0.8
#1	0 to 0.32M	155-165		11.2	11.6	-0.4
#1	-0.115M	155-165		14.5	12.8	+1.7

The sine-swept measurement made at an SPL of 140 dB agrees well with the value calculated from Reference 2.

Tone-burst measurements of A at no flow and averaged over the PPL range of 155-165 dB are 1.7 dB high.

Measurements of A averaged over positive flow and PPL ranges are 0.8 dB low.

If all measurements for no flow and positive flow at all levels are averaged the value of A is only 0.4 dB low

At negative flow and average level the A value is 1.7 dB high.

For Sample #2 the comparisons are:

Sample	Data Averaged over Flow M	PPL dB	Sine	A - dB Tone Burst	Ref. 2	Error dB
#2	0	155		11.0	11.6	-0.6
#2	0.115	155		9.8	11.6	-1.8
#2	0 to 0.115	155		10.4	11.6	-1.2

All the measured values of A are lower than those calculated from Reference 2. The maximum difference is 1.8 dB.

At no flow and 155 dB PPL the A values are 13.0 (avg) for #1 and 11.0 for #2. At 0.115M and 155 dB PPL the A values are 11.5 for #1 and 9.8 for #2. The samples were supposed to be identical.

Another interesting comparison can be made by rearranging the data as follows:

Sample	Flow M	PPL dB	A-dB Tone Burst
#1	0	155	13.0
	0	160	13.8
	0	165	13.5
Avg	0		13.3
#1	0.115	155	11.5
#1	0.32	155	10.5
	0.32	160	10.5
	0.32	165	10.5
Avg	0.32		10.5
#1	-0.115	155	14.0
	-0.115	160	14.5
	-0.115	165	15.0
Avg.	-0.115		14.5

For each flow condition A appears independent of PPL. Comparing the A values over the PPL range shows a descending value of A from -0.115M to 0.32M of 14.5 to 10.5 dB.

For samples #1 and #2 together, the measured values of A compare with those calculated from Reference 2 as follows:

Sample	Flow M	PPL dB	A Sine	A Tone Burst	Ref. 2	Error
#1 and #2	0	140 SPL	24.0		14.8	+9.2
	0	155		24.5	14.8	+9.7
	0.115	155		21.5	14.8	+7

For the double sample the errors are very large. The only explanation possible is that the height of the tone-burst duct in this case is just too far outside the limits of Reference 2.

Shape of resonant absorber curves. - The measurements of the broadband absorbers showed smooth curves which more or less matched measurements of other investigations. The measurements of resonant absorbers, however, showed multiple small, closely spaced, peaks and dips in the neighborhood of the "frequency of resonance" of the equivalent Helmholtz resonator of the sample. These irregularities appeared in the swept-sine measurements at sinusoidal sound pressure levels as low as 130 dB as well as in the tone burst measurements. The effect does not appear to be related to non-linearity. The phenomenon was observed on the first resonant sample made as well as two subsequent samples. The fiber metal and honeycomb design has a multitude of individual sections which form a series of coupled resonators. As far as could be determined the irregularities are actually in the sample and not artifacts of the method.

Broadband samples: Results of attenuation measurements on broadband absorbers are reported in Reference 3. Tone-burst tests were run on samples of 50-Rayl fiber metal backed by a 1/2-inch air space. Tests on 60 Rayl fiber metal backed by 1/2-inch air space were included in Reference 3. It was originally thought that it might be possible to correlate the results of the two tests. The duct used in Reference 3 simulated a section of center fan-discharge duct which naturally is an odd shape. The average cross section was 13.5 x 16.8 inches and the length 26.3 inches. The two wide walls were covered with absorbers, about 70% of wall area. There was no way to obtain empirical relationships that might be applied to the tone-burst samples. The results of Reference 3 relate to specific designs rather than duct parameters. No attempt was made to compare the results except in a very general way.

Figure 48a compares the tone-burst measurements of insertion loss for 25% and 50% of the 1.75 x 1.75 inch by 11.75 inch long sample tube area covered with broadband absorber. The doubling of absorber area only increased the loss by 1 dB. Maximum insertion loss for two samples was 4.5 dB.

Figure 48b shows the loss measured by the tone-burst system with 0.115M air velocity. Loss was reduced slightly by flow at 1000 and 1250 Hz. At higher frequencies, the flow seemed to induce higher losses. Also in Figure 48b is shown the attenuation from Reference 3 for similar flow conditions for duct configuration A19-2 described above. The two curves are smiliar in shape although displaced by 3 dB. It is possible that there is some correlation between the tests because of the similarity of shape.

CONCLUSIONS

The measurements made with the tone-burst flow system on single resonant absorber samples show good agreement with values predicted by Reference 2. Octave center frequencies agree within ± 5%. Octave insertion loss values agree within ± 1.8 dB. For two resonant absorber samples, the tone-burst measurements of insertion loss do not agree with the predictions of Reference 2. In this case the sample duct height of the tone burst system was less than half the limiting value for the empirical equation in Reference 2. Measured curves of insertion loss versus frequency by the tone-burst method for broad-band samples show the same shape as measurements in Reference 3 on a similar absorber in a much larger duct.

The tone-burst flow tube system is a valid method of measuring resonant absorber materials when duct dimensions are within the limits prescribed in Reference 2. Peak sound pressure levels of over 166 dB can be produced at the sample absorber up to frequencies of 3150 Hz. This frequency range can be extended by the addition of a second power amplifier. Positive and negative air flow can be produced in the system at least to 0.32M. Higher velocities can be obtained with proper pumps.

The tone-burst flow system is compact, simple, and direct. It is made entirely of simple machined parts, standard tubing, and off-the-shelf instrumentation. Samples are quickly and easily fabricated. Sample sizes other than the 1-3/4 x 11-3/4 inch size used in this program can be accommodated by fabricating larger or smaller reference-sample sections with appropriate transition sections from and to 2-inch pipe. For a sample section 3.5 x 3.5 inches maximum peak sound pressure of over 160 dB could be attained. For a sample section of 3 square inches peak sound pressure levels of over 169 dB could be attained.

It is most probable that the range of validity of the tone-burst system could be extended by a concentrated program of absorber evaluation. The purpose of this program was to develop a method. The results of the limited number of measurements that were made are sufficiently encouraging to make extended tests of its use valuable.

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APPENDIX A

TONE BURST FLOW DUCT SYSTEM

ASSEMBLY AND OPERATING INSTRUCTIONS

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TONE BURST FLOW DUCT SYSTEM

ASSEMBLY AND OPERATING INSTRUCTIONS

Assembly of Flow Duct System

Refer to Figures 49 and 50. Thoroughly clean the interior of housing SK689A. Lightly grease two of the O-rings and install one into the recess in the end of the housing that has the smaller diameter opening and install the other in the recess in the insert-seal SK692A. (Use vacuum grease for this and all subsequent operations requiring grease.) Lightly grease the larger O-ring and install it into the recess in the other end of the housing.

Lightly grease the entire surface of the insert-seal SK692A and fit it onto one end of the porous tube SK693A. Insert the other end of the porous tube into the housing and carefully push it as far in as possible. The insert-seal will normally protrude about 1/32" at this time.

Lightly grease both sides of the flow trip-ring SK712A, see Figure 51, and install over the studs on the housing. Install the 6" long flow section SK690A-3, see Figure 52,and tighten the nuts to about 10 ft.-lbs. torque.

Look into the sound inlet end of the housing to make sure the porous tube has fully seated. Also look downstream through the porous tube to be sure no grease has squeezed out around the flow trip-ring.

Install the pressure/vacuum monitoring tap in the threaded hole in the housing. Use pipe thread tape at all threaded joints to prevent leaks.

Refer to Figures 53 and 54. Lightly grease the flange faces of both round-to-square transition sections D11460. Align the A and B on the rim of each transition flange with the appropriate letter on the rim of test section D11526 flange. Install nuts and bolts and tighten to about 10 ft.-lbs. torque.

Install a pitot-static tube into the 10 ft. long tube SK690A-1, Figure 52. Align the pitot tube with the axis of tube SK690A-1 with the stagnation port facing into the air flow stream.

Connect this end of tube SK690A-1 to the test section. Align the reinforcing ribs on the test section parallel with the air inlet pipes on the housing SK689A and assemble on flow tube SK690A-3. Install nuts and bolts and tighten to about 10 ft.-lbs. torque.

Assemble the acoustical termination on the other end of the 10 ft. long tube.

Connect the air source to the 1 1/2" male pipe nipples on the housing. The lines between the main air source and these inlets must be of equal length and area (i.e., equal loss) to provide a balance of air velocities entering the air-sound mixing chamber. Use pipe thread tape on all joints to prevent leaks.

Install driver(s) on appropriate transition section and assemble on housing.

Test Section Assembly

Refer to Figure 20. Very lightly grease the threads and O-ring of two previously calibrated pressure transducers.[1] (Do not get grease on the sensitive surface of the transducer. If this occurs carefully clean the surface with a cotton swab.) Install the transducers in the tapped holes provided in each round-to-square transition section. The transducer nearest the housing shall be designated the input transducer (microphone) and the other the output transducer (microphone). Their serial numbers and locations must be recorded for later use. Tighten each transducer to 4 1/2-5 1/2 in.-lbs. torque.

[1-The pressure transducers selected in accordance with the contract schedule, Part II, A, Item 7 are Atlantic Research Corp. LC-71. This transducer is supplied by the maker with a pressure sensitivity calibration at one point only. For this application it is necessary to obtain a pressure sensitivity vs. frequency calibration for the range of 100 Hz to 10,000 Hz. This calibration should be made with a sound level of about 150 dB re 2.0×10^{-5} N/m^2 at standard pressure-temperature while tightened to 4 1/2 - 5 1/2 in.-lbs. torque.]

Test specimens are assembled into a standard size specimen holder. (Covered more fully in next section.) The specimen holder fits into the rectangular recess on each side of the test section. (See Figures 54 and 20) Specimen holders are held in place with bar clamps SK711A. (See Figure 20) The nuts are tightened to about 5 in.-lbs torque. Seal all around the specimen holder with mold seal[2] or modeling clay. The sealant should be applied sparingly, using just enough to insure an air-tight seal while under pressure or vacuum from air flow.

[2-"Mold Seal" TC527 was selected for use in this application because it is easy to apply and will cure in a very short time. When cured "Mold Seal" has a very high tensile strength and may be removed completely without leaving any residue. TC527 Mold Seal is a product of EPD Laboratories, 2055 E. 223rd. Street, Long Beach, California 90810.]

Sample Preparation

Refer to Figures 8 and 55. Sample absorber systems are assembled into a standard size specimen holder which permits a maximum of 20 in.^{2} of absorber area. The test section (Figure 20) of this system can accept two of these standard holders. Other size specimens can be fabricated up to this maximum area provided appropriate non-absorbant fillers are used to take up the remaining area.

The number and variety of absorber specimens are seemingly endless, therefore, no attempt will be made to describe an assembly procedure. Assembly of any particular specimen should be in accordance with the procedure which would be used in the ultimate application of the absorber into an engine or duct.

Sample Evaluation

Sample characteristic screening, sine wave sweep. - Equipment for sample screening is as follows:

Beat Frequency Oscillator*
Audio Frequency Spectrometer*
Audio Frequency Analyzer*
Graphic Level Recorder*
Power Amplifier*
Driver Unit (2)*
Pressure Transducer (2)*
Transducer Amplifier (2)*
D.C. Power Supply
V.T.V.M.
Ocilloscope*
Impedance Matching Transformer

Equipment set-up and check out: Interconnect equipment as shown in Figures 56 and 57 in accordance with manufacturer's recommendations.

Install sample(s) with hard wall side facing sound stream and seal.

Set the Beat Frequency Oscillator (BFO) frequency to 1000 Hz, "Attenuator"** to 4000, and adjust "Output Voltage" until the meter indicates 4.0 volts. Switch on the power amplifier and adjust "Gain" until the VTVM indicates 10.0 volts. Set the "Compressor Voltage" control on the BFO fully clock-wise. Set "Compressor Speed" to 1000. On the Audio Frequency

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*(Appendix B, Equipment Specifications)

**Items in quotation marks are control designations on instrument panel.

Spectrometer (AFS) set "Function Selector" to 1/3 oct, and "Filter Switch" to Lin. Slowly rotate the "Compressor Voltage" control counter-clockwise until the AFS meter indicates the proper level for a sound pressure level (SPL) of 140 dB. This level may be determined from the following relationship,

$$dBV = SPL+SENS-74 \qquad (1)$$

where dBV = level in dB re. 1.0 volt
SPL = sound pressure level re: $2.0x10^{-5}N/m^2$
SENS = sensitivity of transducer in $dBV/N/m^2$

For example, determine the level for a transducer with a sensitivity of -113.0dBV and a sound pressure level of 140 dB.

$$dBV = 140+(-113.0)-74 = -47dBV$$

This is the level which the meter on the AFS should indicate when the levels are properly set.

Manually sweep through the frequency range of 500 Hz to 5000 Hz and see that the level remains relatively unchanged. Observe the output on the ociloscope and see that there is no clipping or severe distortion.

On the Audio Frequency Analyzer (AFA) set the "Function Selection" to selective section off. Set the gain as necessary to produce an output level as nearly equal the output of the AFS as possible. While observing this output level on the ociloscope manually sweep through the frequency range of 500 Hz to 5000 Hz. There must not be any clipping or severe distortion of the waveform.

Place the BFO frequency dial off scale. Observe meter indications on both the AFS and AFA. They should indicate -90 dBV or less.

Specimen measurements: Place the transducer selector switch in the input position and run a curve of the voltage response on the Graphic Level Recorder (GLR) from 500 Hz to 20,000 Hz. Place the switch in the output position and run this response over the previous one using a different color ink.

The latter response curve is the baseline from which the specimen insertion loss will be directly obtainable.

Place the absorbent side of the specimen into the sound stream and seal with the mold seal. Record the input and output response curves as before.

By comparing the response curve without an absorber specimen to the curve with the specimen, the absorber insertion loss is obtained directly in dB vs. frequency.

Tone burst evaluation of absorber specimens. - Equipment for tone burst evaluation of specimens is as follows:

Beat Frequency Oscillator*
Audio Frequency Spectrometer*3***
Audio Frequency Analyzer*3
Tone Burst Generator*
Frequency Counter
Power Amplifier*
Driver Unit (2)*
Pressure Transducer (2)*
Transducer Amplifier (2)*
D.C. Power Supply
Ociloscope*
Waveform Eductor*
Impedance Matching Transformer

Equipment set-up and check out: Interconnect equipment as shown in Figures 58 and 26 in accordance with manufacturer's recommendations.

Set the BFO frequency to 1000 Hz, "Attenuator" to 1200, and adjust "Output Voltage" until the BFO meter indicates 1.0 volt.

Set the Tone Burst Generator (TBG) "Trigger level" to 0, "Trigger Slope" to +, "Output On" to 8 cycles, "Output Off" to SEC, and "Output Off Vernier" to about 0.3.

On the AFS set "Input Switch" to input potentiometer, "Input Potentiometer" to 0, "Function Selector" to 1/3 oct, and "Filer Switch" to 1000 Hz. Place "Meter Switch" in off position.

On the AFA set "Input Switch" to direct, " Weighting Network" to linear, 20-40 000c/s, "Frequency Range" to 630-2000, "Meter Switch" to off, "Range Multiplier" to 0 dB, "Meter Range" to -60 dB, "Frequency Analysis Octave Sensitivity" to 25 dB, "Function Selector" to freq. analysis, and "Frequency Tuning" to 1000 Hz.

Set transducer selector switch to input.

Set the scope vertical sensitivity to 1.0 volt/cm. Set horizontal sweep speed to 1.0 cm/sec and triggering as required to produce a trace in sync with the TBG.

Strap the impedance matching transformer taps as indicated in Table II for 5.8 ohm load.

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*(Appendix B, Equipment Specifications)

*** See Note 3 Page

Turn on the power amplifier and set the "Gain" control to about the 10 o'clock position.

Slowly rotate the "Input Potentiometer" on the AFS until a pulse is visible on the scope. Adjust the "Meter Range" and/or "Range Multiplier" controls on the AFS as necessary to drive the power amplifier, Always rotate the "Input Potentiometer" back to 0 before making any change in the "Meter Range" or "Range Multiplier" (Running the "Range Multiplier" as near 0 dB as practical results in the highest signal to noise ratio.) Observe that the overload light on either the AFS or AFA does not flicker or light at any time during each pulse. If this should occur, reduce the gain by rotating the "Meter Range" control clockwise. If more gain is then required it must be made up in the "Range Multiplier" control.

Continue to increase the drive level until the scope indicates a pulse envelope of about 6.0 volts P-P. (With a transudcer sensitivity of -113.0 dBV and gain settings on the AFA as described above 6.0 volts P-P equals a peak sound pressure level of about 137 dB at the input transducer).

Switch the TBG "Output Off" control to single pulse and the "Function Selector" on the AFA to selective section off. Switch the "Meter Switch" on the AFA to RMS SLOW and read the wide-band noise level for each position of the transducer selector switch. It should be lower than -90 dBV. Observe that there are no strong line-frequency components present in the noise wave form. Correct any ground loops or other errors in the set up before proceeding with any tests on specimens.

Return the "Meter Switch" to off and the "Function Selector" to freq. analysis.

Determination of gains, levels, and data recording: For the purpose of data recording the chart shown in Figure 59 has been designed. Its use is recommended. There are rows and columns for all quantities to be measured as well as spaces for other pertinent information.

From the sine sweep sample screening a frequency range of interest for a particular sample can be determined.[3] Enter the 1/3 octave band center frequencies for this range on the chart in the "Band Center Frequency" column.

[3-In some cases, the 1/3 octave band center frequencies in the range of interest do not occur at the standard 1/3 octave band center frequencies. It will then be necessary to replace the Audio Frequency Spectrometer with an Audio Frequency Analyzer. This gives a continuously variable center frequency with a 1/3 octave bandwidth capability.]

From the input transducer (microphone) calibration curve obtain a sensitivity for each of the bands chosen above. With this band sensitivity and the PPL at which the specimen is to be evaluated determine an input transducer output level in dBV from the following:

$$dBV = PPL+SENS-74 \qquad (2)$$

where dBV = level in dB re 1.0 volt
PPL = peak sound pressure level re: $2.0\times10^{-5} N/m^2$
SENS = sensitivity of transducer in $dBV/N/m^2$

For example, determine the output in dBV for a PPL of 160 dB and transducer sensitivity of -113.0 dBV. Substitution into formula (2) yields

$$dBV = 160+(-113.0)-74 = -27$$

From the output level determined from (2) determine a gain setting, to the nearest 10 dB, that will produce a maximum voltage out of the AFA that does not exceed 40 volts peak.

For example, 40 volts ≃ +32 dBV, the maximum permissible level. The difference between this level and that obtained from (2) will be the gain required. (+32)-(-27) = +59. Since 60 dB is the nearest 10 dB increment one would normally use this figure. However, 60 dB gain would result in an output of +33 dBV which is 44.6 volts and exceeds our 40 volt limit. Therefore, the gain would be set at 50 dB with the resulting output level of +23 dBV.

The output level in dBV determined must now be converted to voltage using the following:

$$dBV = 20 \log \frac{V_o}{V_r} \qquad (3)$$

where V_r = 1.0 volt
V_o = output level in volts

Rearranged $$V_o = \log^{-1} \frac{dBV}{20} \qquad (4)$$

From the example above substitution into (4) yields

$$V_o = \log^{-1} \frac{23}{20} = \log^{-1} 1.150 = 14.1 \text{ volts peak}$$

Having determined a voltage and gain for each of the bands listed on the chart enter these values in the column headed "Baseline", input mic, beside the appropriate frequency band.

Determination of a baseline: Place the specimen holders in the test section with the hard wall side facing the sound stream. Seal all around with mold seal and cure.

Set the BFO, Spectrometer, and Analyzer to the center frequency of the first band listed on the chart. Place the transducer selector switch in the input position and set the AFA gain to the value recorded on the chart for this band and PPL.

Switch the TBG " Output Off" control to sec. Slowly increase the drive level until the input transducer output voltage reaches the value on the chart for this frequency band and PPL. Readings may be made in either peak volts or peak-peak volts. See Figure 60. The example cited is in peak volts. If P-P measurements are desired merely double the values recorded. (In some high sound levels it is more desirable to read P-P because distortion can cause a non-symetrical pulse envelope). It does not matter what units are used so long as all readings are taken in the same units.

Place the transducer selector switch in the output position and read and record the volts/gain for the output transducer. Record on the chart under the column headed "Baseline" output mic.

Read the driver signal coil voltage and transformer impedance tap and record in the column provided.

Repeat the above three operations for each band-center frequency listed on the chart. All readings are made from the scope trace.

If the PPL specified on the chart for this particular specimen is to be the maximum obtainable for this system (Ref. PPL. MAX) the following additional operations will be required.

Determine as nearly as possible the driver signal coil impedance for each band center frequency.[4] Enter this value beside the appropriate frequency in the " Signal Coil", volts/z column on the chart.

[4-Determination of Driver signal coil impedance. Connect a 0.1 ohm non-inductive resistor in one side of the driving signal line. With the system operating at a fairly high level (160 dB) measure the peak voltage (Ep) across the signal coil and the peak voltage drop across the 0.1 ohm resistor (Ip) on the scope. The impedance,

$$Z = \frac{\frac{Ep}{Ip}}{10} .$$

For example, suppose Ep = 120 volts and Ip = 2.07, then

$$Z = \frac{120/2.07}{10} = \frac{58}{10} = 5.8\ \Omega]$$

Now instead of predetermining an input transducer output voltage as described above, measure the maximum voltage obtainable as follows.

Set the transformer tap (Table II) to the proper impedance for the frequency band being evaluated. Set the filters and BFO to the proper frequency and start the pulses.

Slowly increase the drive level while carefully watching the input transducer output voltage on the scope. Continue to increase the drive level until the power amplifies circuit breaker trips. Make a note of the input transducer output voltage just prior to this "trip". Reset the circuit breaker and slowly increase the drive level until the power amplifies again "trips". This level should be very near the previous one. Decrease this input transducer output voltage maximum about 0.3 dB and record this value on the chart for the proper band. Reset the power amplifies circuit breaker and set the input transducer output voltage to the reduced value determined above. Record the driver signal coil voltage and output transducer voltage.

If when the input transducer output voltage is set to the reduced value as determined above the power amplifies trips, further reduce the level until the amplifier holds. If this "holding" level is more than 1.5 dB below the first maximum level obtained above, it will probably be necessary to replace the driver diaphragm assemblies and rerun any points on the baseline previously determined.

It may also be possible to achieve a higher PPL by readjusting the transformer tap (Table II) to the next impedance higher or lower than that which was determined for this band. Usually a higher impedance is optimum (provided it is not too much higher) because it compensates for the heat rise in the driver signal coil. If the transformer tap is changed, record the information on the chart for reference so the proper tap can be selected whenever the specimen evaluation is underway.

An additional gain in PPL of 2-5 dB may be achieved, if no air flow is necessary, by removing the air-sound mixing section, flow tripper, and flow tube, and connecting the driver transition section directly onto the round-to-square transition section.

All of the information obtained in the preceding section is the baseline calibration for a given set of conditions. These data may be re-used at a later date under the same set of conditions. Should any part of the system fail, such as the driver diaphragms, transducer, etc., a new calibration must be made.

Enter the maximum PPL obtained in the column headed "Baseline" under PPL. The value of PPL may be obtained from equation (3) in conjunction with the gain setting and equation (2), rearranged here for convenience.

$$\mathrm{PPL} = 74 + \mathrm{dBV} - \mathrm{SENS}. \tag{5}$$

where dBV = level in dB re 1.0 volt
PPL = peak sound pressure level re: $2.0\times10^{-5}\,\mathrm{N/m^2}$
SENS = sensitivity of transducer in $\mathrm{dBV/N/m^2}$

For example, find the PPL if the maximum voltage is 28.0 volts, the transducer sensitivity is -113.0 dBV, and the gain is 50 dB.

Substitute into eq (3)

$$\mathrm{dBV} = 20 \log \frac{28.0}{1.0} = +28.9$$

Subtract the gain,

$$+28.9 - 50 = -21.1 \text{ dBV}$$

generator described in Appendix B is continuously variable from 1Hz pulse to a continuous wave output.)

If the specimen test conditions require air flow it is desirable to measure and monitor the flow velocity during the test sequence. This may be done in the following manner.

Make sure the pitot-static tube is facing upstream of the anticipated air flow. From the following equation determine a value for the differential pressure Δp for the desired flow velocity.

$$\Delta p = \left(\frac{V}{43.6}\right)^2 \qquad (6)$$

where Δp = the differential pressure of the pitot-static tube in mm Hg
V = air flow velocity - feet/second

The specimen evaluation is conducted in the same manner with or without air flow.

One precaution with air flow is that the mold seal around the specimen holder must be cured before the air is turned on. There will be a differential pressure between the atmosphere and the inside of the test section that will blow out the mold seal while in its liquid state.

Another precaution with air flow is that under some flow velocity/sound level combinations it will be very difficult to read the level of the sound pulse envelope and at others virtually impossible to separate the sound pulse from the air flow noise. If the sound pulse is modulated by the noise ± 10% of its P-P value an accuracy of measurement within 1.0 dB is easily obtained. However if the noise interference is greater than this some form of signal processing is necessary to reduce this noise level.

The most practical signal processor for this application is a Waveform Eductor TM (Appendix B)

The Waveform Eductor is connected between the AFA (transducer bandpass filters) output and the scope input with its gain set to unity. The maximum output voltage capability is ± 10 volts (20V P-P), therefore the gain settings of the preceding equipment must be reduced accordingly. A sync pulse for the Waveform Eductor is taken from the sync output of the tone burst generator. The Eductor "Trigger Delay" time is set to about the transmission time of the electrical/acoustic system. The "Sweep Duration" is set to equal the pulse duration. Set the scope "Trigger Delay" to the same value as the Eductor "Trigger Delay" and the scope horizontal sweep so one sweep is completed in about the "Sweep Duration" time set on the Eductor. Set the Eductor "Time Constant" to its lowest value and "Smoothing" (at the rear of the instrument) to on. With the "Mode" switch in analyze and readout, a pulse should begin to form on the scope. Once the pulse has been located, the delay times and sweep durations may be adjusted to display any portion of the waveform. The best compromise appears to be to display the middle two cycles of a normal

8 Hz pulse. After the sampling window is determined and set, sample the pulse for about 200 times the time constant selected (i.e., if TC is 0.5 sec, sample for about 2 minutes) before measuring the voltage. The voltage measurement is then made in the same manner as before, with the oscilloscope. (A slightly higher degree of accuracy may be obtained in the reading by switching off the "Smoothing" and reading the peak points which occur. Also on the lower time constants a slight shifting of the pulse envelope may still occur where the signal is equal to or less than the air noise. This can be eliminated by shifting to a longer time constant after the prescribed sample period is completed at the lower time constant without affecting the accuracy of the read-out).

Interpretation of Test Results

Reduction of data: The data now contained in the chart must be reduced to a useful form. The main item of interest is the insertion loss of the specimen. The specimen transmission loss is also available.

Specimen insertion loss: The specimen insertion loss in dB is calculated from

$$dB_{IL} = 20 \log \frac{O_B}{O_S} \tag{7}$$

where dB_{IL} = insertion loss

O_B = output voltage of output transducer from " Baseline" column of chart.

O_S = output voltage of output transducer from " Sample" column chart

Any difference in gain settings between O_B and O_S must be corrected for in these calculations. For example, suppose O_B = 14.1/50 (14.1 volts and 50 dB gain) and O_S = 0.10/70. Then

$$dB_{IL} = 20 \log \frac{14.1}{0.10} - (50-70) = 43 \text{ dB}$$

Calculate dB_{IL} for each frequency band and enter it on the chart under "Sample Insertion Loss". The insertion loss is then plotted on semi-log graph paper in dB vs. Frequency.

Specimen transmission loss: To get the specimen transmission loss the voltage/gain values under the column headed "Sample" must be converted to their equivalent sound level by using equations (3) and (5). The transmission loss is then the input sound level minus the output sound level $\pm$ any difference between the input and output transducer calibration sensitivities for each band.

APPENDIX B

SPECIFICATIONS OF EQUIPMENT

BEAT FREQUENCY OSCILLATOR

Frequency Range: 20-20000 Hz

Frequency Scales:

Main Scale: Logarithmic from 20-20000 Hz.
Tolerance ± 0.7 degrees of theoretical logarithmic curve. Vernier driven.
Increment Scale: Range -50 to +50Hz of main scale reading. Both scales illuminated.

Frequency Accuracy:

Main Scale: 1% ± 1 Hz
Increment Scale: ± 0.5 Hz

Outputs:

Matching: Switchable matching impedance for 6,60, 600 or 6000 ohms load.
Max. power output 2.5 watts approx.
Attenuator: Variable in steps of 10 dB (within ± 0.2 dB) from 125 μV to 12.5 V. Continuously variable by potentiometer within each step.

Output Voltage Accuracy: In frequency range 20 to 20000 Hz.

Better than ± 0.3 dB on "Attenuator Output".
Better than ± 1 dB on "Load" 1 watt loaded.

Voltmeter: Vacuum-tube voltmeter. Moving-coil. Illuminated mirrored scale.

Highly accurate, better than 1.5% of full-scale deflection. Perfectly safeguarded against overload.

Distortion:

Frequency in Hz	20	200	2000	20000
"Attenuator" terminal. No load with 10V output approx	1.0%	0.1%	0.1%	0.7%
"Load" terminal. (Loaded 1 watt)	2.0%	0.3%	0.3%	1.2%

Automatic Output Regulator: Output voltage automatically regulated when required. Built-in compressor amplifier maintains regulation up to 45 dB and a constant voltage, current or sound pressure to within ± 2 dB in frequency range 50-20000 Hz. Input impedance 100 kohms. Regulation speed variable in steps: 30-100-300 and 1000 dB/sec.

Frequency Modulation: Continuously variable modulation swing 0 $\pm$ 200 Hz. Selectable modulated frequency by built-in saw-tooth oscillator of 1-2-4-8-16 and 32 Hz.

Oscillator Stop: Push-button Oscillator Stop for noiseless switching in reverberation measurements. Remote control available.

Frequency Scan: Worm gear in oscillator permits variable capacitor to be driven from motor of Level Recorder. Connection achieved by flexible shaft. Magnetic clutch for set and release of drive. Clutch can be remotely controlled. Accurate synchronization with Level Recorder. Frequency Calibrated Paper.

GRAPHIC LEVEL RECORDER

Electrical

Frequency Range:

AC:	10 Hz - 200 KHz ± 0.5 dB (re. 1000 Hz) for "Input Potentiometer" in position around "0" and "10" ± 1 dB for other positions
DC:	Chopped at twice the mains frequency.
Maximum Sensitivity:	Minimum voltage to give zero deflection of stylus: AC: 5 mV r.m.s. approx. DC: 10 mV approx.
Dynamic Range:	Determined by the interchangeable Range Potentiometers which are available as:

Range	Accuracy
10-35 mV. Linear	2% of full scale
10-110 mV. Linear	2% of full scale
10 dB. Logarithmic (100 mV-316 mV)	± 0.1 dB
25 dB. Logarithmic (10 mV-180 mV)	± 0.2 dB
50 dB. Logarithmic (10 mV-3.16V)	± 0.3 dB
75 dB. Logarithmic (10 mV-56 V)	± 0.5 dB

Resolving Power:	Better than 0.25 mm on scale when adjusted for 50 mm paper 0.5 mm on scale when adjusted for 100 mm paper.

Input Circuit:

Input Impedance:	16-18 KΩ, dependent of position of "Input Potentiometer", parallel with 100-120 pF approx.
Maximum Input Voltage:	100 volts.
Input Potentiometer:	Non-calibrated Covers 0-12 dB, continuously variable.
Input Attenuator:	Six steps of 10 dB Within ± 0.25 dB, relative position "0".

Rectifier Response:	Selectable by a control knob: R.M.S. ± 0.5 dB for crest factors up to 5. Arithmetic Average. Peak (half peak-to-peak).
Writing Speeds:	Selectable by a control knob. 50 mm paper: 2-4-8-16-25-40-63-100-160-250-400-500-630-800-1000 mm/sec. 100 mm paper: 4-8-16-31.5-50-80-125-200-315-500-800-1000-1250-1600-2000 mm/sec.
Recording System:	Electrodynamic. Pulling force 1 kg approx. External arm connection possible.
Overall Stability:	Better than ± 0.2 dB in deflection for ± 10% deviation in power supply voltage.
Calibration Voltage:	Built-in square-wave signal at power supply frequency. 100 mV r.m.s. Stability: Within ± 1% for ± 10% deviation of power voltage.
Lower Limiting Frequency:	Selectable to 10, 20, 50, and 200 Hz.

Mechanical

Paper Speeds:	Selectable by a control knob. 0.0003-0.001-0.003-0.01-0.03-0.1-0.3-1-3-10-30-100 mm/sec. derived from a reversible self-starting synchronous motor.
Types of Recording:	Rectilinear. Polar: Synchronous drive Turntable.
Types of Transcription:	Pens for ink writing, easily interchangeable for writing in different colors. Sapphire Stylus for writing on wax-coated paper.
Paper:	Rolls of 50 and 100 mm paper width for ink writing and 50 mm for sapphire writing (wax-coat). Various types of preprint. Polar, preprinted in degrees, radius 100 mm, for ink writing.
Event Marker:	Manually and remotely operated. Suitable for ink and sapphire stylus writing.

Drives for External Instruments:	Drive Shaft I and Drive Shaft II can be given speeds of rotation (independent of each other) of 0.0036-0.012-0.036-0.12-0.36-1.2-3.6-12-36 and 120 r.p.m. The two shafts can, by means of a Flexible Shaft, be connected to the scanning mechanism of other instruments, e.g. Beat Frequency Oscillators and Frequency Analyzers. Allowable max. mechanical load on the Drive Shaft outputs: 2 kg cm. Built-in switch supplied for control of Spectrometer filter selector.
Remote Control:	Various, such as: Start-stop, single chart, lifting of pens and event marking.
Two-Channel Selector:	Can be used for successively recording of two signal levels, time marking, etc.

AUDIO FREQUENCY SPECTROMETER

Frequency Characteristics:

"Linear":	2 Hz -45000 Hz within ± 0.5 dB relative to 1000 Hz. 5 Hz -20000 Hz witin ± 0.3 dB relative to 1000 Hz.
Band Pass Filters:	33 filters of 1/3 octave bandwidth, 3 dB, center frequencies 25 Hz to 40000 Hz. 11 filters of 1/1 octave bandwidth, 3 dB, center frequencies 31.5 Hz to 31500 Hz.
Selectivity	1/3 octave filters, approximately 50 dB at 1 octave from center frequency. 1/1 octave filters, approximately 35 dB at 1 octave from center frequency.
Deviation in Pass Band	Within ± 0.5 dB for 1/3 octave, and ± 1 dB for 1/1 octave filters, ± 2 dB for the 31.5 and 63 Hz filters.
Weighting Networks:	"A", "B" and "C", frequency response of Spectrometer in accordance with the prposed IEC standards for Precision Sound Level Meters. (Helsinki 1961). "Lin" gives linear range 20-45000 Hz. Cuts off by maximum 18 dB per octave below 20 Hz.

Attenuators:

Meter Range:	20 dB between steps. Within ± 0.15 dB relative to position "10mV", at 1000 Hz.
Range Multiplier:	10 dB between steps. Within ± 0.1 dB relative to position "X0.01", at 1000 Hz.

Input Impedances:

"Direct":	2.2 Megohm paralled with 30 pF.
"Input Potentiometer":	0.7-1 Megohm approximately,the parallel capacity being dependent on the setting.
"Condenser Microphone":	2.2 Megohm parallel with 30 pF.

Meter:

Meter Rectifier Circuits, Accuracy:	"RMS", ± 0.5 dB for crest factors up to 5. "Average", ± 0.1 dB of the read value. "Peak" (half peak-to-peak) ± 0.2 dB of the read value.
Meter Scale:	± 1% of full deflection for "RMS", "Average", and "Peak".
Meter Damping:	"Slow" and "Fast" both in accordance with the proposed IEC standards for Precision Sound Level Meters (Helsinki 1961). The meter is perfectly protected against overload.

Calibration Voltage: Built in, mains frequency square wave. Within ± 0.03 dB for ± 10% deviation of the power voltage.

Sensitivity: Maximum 100 µV and minimum 1000 V for full deflection on the indicating meter.

Distortion, Input Amplifier: Within 0.1% in the frequency range 5-45000 Hz, when voltage on FILTER INPUT EXTENSION FILTER does not exceed 1 V r.m.s., sine-wave, and load is formed by the built-in filters or an impedance higher than 500 ohm. With load impedance higher than 5 kΩ the voltage can be 10 V rms sine-wave.

Overload Indicator: Indicates by red light when output voltage of input amplifier section exceeds 4.5 V peak approx.

Signal/Noise Ratio, (2-45000 Hz) (as Linear Amplifier): Better than 60 dB with RANGE MULTIPLIER switch in position "X1, 0 dB", full deflection on the indicating meter and short-circuited AMPLIFIER INPUT. The ratio is made the same number of dB smaller as the gain is increased in dB by the RANGE MULTIPLIER relative to the said position.

Noise, (2-45000 Hz)

Input Amplifier:	Referred to the input at maximum gain, 40 dB. Approximately 5 µV with short circuited input. Approximately 15 µV with open input.
Output Amplifier:	On RECORDER output. Approximately 10 mV with RANGE MULTIPLIER in position "X 1". Approximately 20 mV with RANGE MULTIPLIER in position "X 0.01".

Output at the Terminal "Recorder":

Voltages:	8-10 V for full deflection on the meter scale. Maximum available peak voltage approximately 45 Volts.
Output Impedance:	Smaller than 50 Ω in series with 24 μF.
Load Impedance:	Highest possible. At 3kΩ, 15 V peak available with METER SWITCH on "off". At 8kΩ, 45 V peak available with METER SWITCH ON "Off".

Terminal "Filer Input-Extension Filter":

Voltage:	1 V Max: for full deflection on indicating meter. (Spectrometer set as linear amplifier.) 75 V DC approx. independent of indicating meter deflection.
Output Impedance:	10 Ω approximately.

Terminal " Filter Output":

Input Impedance:	1.5 Megohm paralleled by 40 pF approximately.

Stability:	Better than ± 0.3 dB for a deviation of ± 10% of power supply voltage.
Remote Control:	The filter switching can be remotely controlled by a single poled switch in series with a 24 V DC supply (180 mA).
Polarization Voltage:	The polarization voltage for the Condenser Microphone Cartridges can be adjusted between 150 and 250 Volts.

AUDIO FREQUENCY ANALYZER

Frequency Responses: Linear: 2-40000 Hz to within ± 0.5 dB (± 0.3 dB in the range 5-20000 Hz).

Weighted: According to Curve A, B, and C as proposed in the IEC standard for Precision Sound Level Meters (Helsinki 1961).

Selective: Can be used both as frequency analyzer and as distortion factor meter. Frequency scale continously tunable from 20 to 20000 Hz, through six ranges. Automatic tuning from external motor possible.

Band pass characteristics of the constant percentage bandwidth type with adjustable selectivity: 20-25-30-35-40 dB and max. (approx. 46 dB) attenuation 1 octave from the tuned-in frequency. Accuracy of filter top at 40 dB octave selectivity is ± 0.5 dB with respect to "Linear" ranges. The frequency to which the instrument is tuned is marked on a large, illuminated frequency dial. Frequency accuracy better than ± 1%. When used as distortion factor meter, the attenuation of the fundamental frequency is more than 60 dB. Attenuation 1 octave away from fundamental < 0.5 dB.

Sensitivity: Full scale deflection from 100 μV to 1000 V in 10 dB steps on two attenuators. Maximum amplification 100 dB.

Harmonic Distortion: Input Amplifier, less than 0.1%.

Noise and Hum Level: When switched as wide band amplifier, the noise level is approx. 15 μV with open input and 5 μV with short circuited input. When switched as narrow band analyzer: The hum level is approx. 3 μV. All figures are referred to the input and maximum gain.

Input:

1. "Direct". Input impdance 2.22 MΩ paralleled by 30 pF.

2. "Input Potentiometer". Input resistance approx. 1 MΩ, the parallel capacity depends on setting. To be used for relative measurements.

3. "Condenser Microphone". 7-poled socket.

Output : "Recorder" output impedance smaller than 50Ω in series with 24 μF. Output voltage corresponding to full scale meter deflection approx. 10 V. Maximum available output voltage approx. 45 V peak.

External Filters: "Ext. Filter Input": Impedance approx 12 Ω. "Ext. Filter Output": Impedance approx. 1.5 MΩ.

Meter: Conveniently illuminated and mirrored instrument scale. Perfectly safeguarded against overload. Accuracy: Approx. 2% of full scale deflection. Scale 0-10 and 0-31 V (linear), 0-20 dB (logarithmic) as well as 3 scales calibrated in % absorption.

Switch selection of two meter damping characteristics "Fast" and "Slow" which are in accordance with the proposed IEC standard for Precision Sound Level Meters (Helsinki 1961).

Meter Rectifier: R.M.S., peak (half the peak to peak), or arithmetic average type rectification can be selected by means of a switch. The R.M.S. meter indication is accurate to within 0.5 dB for any signal wave form with crest factor smaller than 5.

Polarization Voltage: The polarization voltage for Condenser Microphone Cartridges can be adjusted between 150 and 250 volts.

OCILLOSCOPE

Vertical Deflection System

Plug-In Unit	Calibrated Deflection Factor	Bandpass at -3 dB	Fastest Rise Time	Approx. Input Capacitance
Type W	1 mv/cm to 50 v/cm	dc to 8 MHz dc to 23 MHz	44 nsec 16 nsec	20 pf

Sweep Generation

Sweep Rates (at 1X magnification)	0.1 μsec/cm to 5 sec/cm in 24 calibrated steps. Displayed Sweep-rate accuracy is ± 2% for both sweeps. An uncalibrated variable sweep-rate control permits either sweep to be slowed to at least 0.4 of the indicated rate.
Sweep Magnification	Any sweep rate can be increased by expanding the center portion of the display horizontally in fixed steps of 2X, 5X, and 10X. Sweep-rate accuracy is within ± 5% in the magnified positions.
Trigger Source Selection	Internal normal, internal plug in, external, and line
Trigger Coupling Selection	Dc, ac, and ac low-frequency rejection.
Trigger Signal Requirements	Internal (ac): Minimum deflection is 2 mm, rising to 1 cm at about 50 MHz. Internal (dc): Minimum deflection is 5 mm at dc. Internal (ac low-frequency rejection): Minimum deflection is 2 mm with signals at about 2 KHz, rising to 1 cm at about 50 MHz.

	External: Frequency ranges are the same as internal. Minimum amplitude is 200 mvolts peak-to-peak (ac), 200 mvolts change in dc level (dc) and 200 mvolts peak-to-peak (ac low-frequency reject). A MAXIMUM INPUT of ± 30 VOLTS must not be exceeded in the EXTERNAL trigger position. Minimum trigger level range is greater than ± 2 volts with the TRIGGER LEVEL control pushed in and ± 20 volts with the control pulled out.
Sweep Delay	The time-base A sweep can be delayed by the main time base (B) sweep. Delay is continuously variable over the range of 0.1 μsec to 50 sec with the DELAY TIME and DELAY-TIME MULTIPLIER controls. Delay time is accurate to ± 1% of indicated delay ± 2 minor divisions of the DELAY-TIME MULTIPLIER at sweep rates from 50 μsec to 50 sec. At delay times shorter than 50 sec, indicated delay accuracy is the same as above plus approximately 75-100 nsec. The 75-100 nsec represents the fixed inherent delay of the internal trigger circuitry. Incremental delay accuracy is ± 4 minor divisions of the DELAY-time multiplier dial at sweep rates from 1 sec to 50 sec. Incremental accuracy at the three fastest sweep rates (0.1, 0.2, and 0.5 μsec) is ± 10 minor divisions. Stated accuracies apply only when the VARIABLE controls are set to CALIB. Delay jitter is not greater than 1 part in 20,000..

Horizontal Deflection System

The following characteristics apply when the HORIZONTAL DISPLAY switch is set to the EXT positions.

Deflection Factor	Continuously variable from approximately 0.1 volt/cm to 10 volts/cm.
Frequency Response	Dc to 400 KHz (3-dB down).
Input Characteristics	1 megohm paralleled by approximately 55 pf.

Amplitude Calibrator

Output Voltages	0.2 mvolts to 100 volts peak-to-peak in 18 steps. In addition, a 100 volt dc output is available.
Frequency	Approximately 1 KHz square wave.
Output Current	5 ma square wave available at the front-panel current loop.

Output Impedance	50Ω in .2 to 200 mVOLTS positions. Progressively higher output impedances in the .5 to 50 VOLT positions up to about 4 k in the 50 VOLT position. Output impedance of the 100 VOLT Position (ac and dc) is about 420Ω.
Amplitude Accuracy	Peak-to-Peak amplitude accuracy is ± 3% of indicated value when working into an impedance of 1 megohm. The .2 to 200 mVolts positions will be within ± 3% of one-half of the indicated voltage when working into an impedance of 50 ohms. The 5 ma current accuracy is ± 3%.

<u>Front-Panel Output Signals</u>

+ GATE B	Approximately 20-volt peak-to-peak square-wave pulse having the same duration as the B sweep. Minimum dc load resistance is 5 kΩ.
DLY'D TRIG	Approximately a 10 volt peak-to-peak pulse occurring at the end of the delay period.
SWEEP A	Approximately a 90 volt, peak-to-peak sawtooth voltage having the same duration as the A sweep. Minimum load impedance is 10 kΩ.
+ GATE A	Approximately 20 volt peak-to-peak square-wave pulse having the same duration as the A sweep. Minimum dc load resistance is 5 kΩ.
VERT SIG OUT	Vertical signal output connector. Output amplitude is approximately 0.3 volt per centimeter of deflection on the crt. Rise time is 20 nsec or faster. Output i-s ac coupled.
External Single-Sweep Reset Input-Signal Requirements	Requires a positive-going step or pulse of at least +20 volts with a rise time of 0.5 μsec or faster.

<u>Cathode-Ray Tube</u>

Type	T5470-31-2
Unblanking	Dc coupled.
Accelerating Potential	10 kv.
Usable Viewing Area	6-cm high by 10-cm wide.

Focus	Vertical: 2 horizontal lines/mm distinguishable over the center 4 cm. 1.5 horizontal lines/mm distinguishable in the top and bottom 1 cm.
	Horizontal: 2 time markers/mm distinguishable over the middle 8 cm. 1.5 time markers/mm distinguishable in the first and tenth cm.
Graticule	Internal, adjustable edge lighting. 6X10 cm with vertical and horizontal 1-cm divisions with 2-mm markings on the centerlines. Provision made for rise time measurement.

TONE BURST GENERATOR

Signal Input (Signal to be switched)

Amplitude: Proper operation results from input signals of not greater than 10 V pk(7 V rms) and not less than 1 V pk-pk.

Frequency Range: Dc to 2 MHz.

Input Impedance: 50 kΩ, approx.

Timing Input (signal that controls switch timing). Same specifications as Signal Input except:

Input Impedance: 20 k , approx.

Signal Output

Output On: Replica of Signal Input at approx. same voltage level; dc coupled; down 3 dB at >1 MHz. Output current limits at>25 mA pk, decreasing to >15 ma at 2 MHz. Output source impedance typically 25Ω increasing above 0.2 MHz. Total distortion contribution <0.3% at 1 kHz and 10 kHz.

Output Off: Input-to-output transfer (feedthrough), -60 dB, dc to 1 MHz, increasing above 1 MHz.

Spurious Outputs: Dc component and change in dc component due to on-off switching (pedestal) can be nulled with front-panel control. Output switching transients are typically 0.2 V pk-pk and 0.2 μs in duration (120-pF load).

On-Off Timing Timing is phase-coherent with, and controlled by, either the signal at the Signal Input connector or a different signal applied to the Ext Timing connector. The on interval (duration of burst) and the off interval (between bursts) can be determined by cycle counting, timing, or direct external control.

Cycle-Count Mode: On or off intervals can be set independently, to be of 1, 2, 4, 8, 16, 32, 64, or 128 cycles (i.e periods) duration or to be 2, 3, 5, 9, 17, 33, 65, or 129 cycles with +1 switch operated.

Timed Mode: On and off intervals can be set, independently, for durations of 10 μs to 10 s. On and off times occur at first proper phase point of controlling signal occurring after time interval set on controls; one interval can be timed while other is counted.

Switching Phase: In above modes, input controls determine phase of timing signal at which on and off switching occurs. Slope control selects either positive or negative slope of timing signal; Trigger Level control sets voltage level at which both on and off switching occur.

Direct External Control: A 10-V pulse applied to rear-panel connection will directly control switching.

Synchronizing Pulse: A dc-coupled pulse that alternates between +8 V for output on, and -8 V when off. Source resistance approx. 0.8 kΩ for positive output and 2 kΩ for negative.

TONE BURST GENERATOR

Signal Input (Signal to be switched)

Amplitude: Proper operation results from input signals of not greater than 10 V pk(7 V rms) and not less than 1 V pk-pk.

Frequency Range: Dc to 2 MHz.

Input Impedance: 50 kΩ, approx.

Timing Input (signal that controls switch timing). Same specifications as Signal Input except:

Input Impedance: 20 k , approx.

Signal Output

Output On: Replica of Signal Input at approx. same voltage level; dc coupled; down 3 dB at >1 MHz. Output current limits at>25 mA pk, decreasing to >15 ma at 2 MHz. Output source impedance typically 25Ω increasing above 0.2 MHz. Total distortion contribution <0.3% at 1 kHz and 10 kHz.

Output Off: Input-to-output transfer (feedthrough), -60 dB, dc to 1 MHz, increasing above 1 MHz.

Spurious Outputs: Dc component and change in dc component due to on-off switching (pedestal) can be nulled with front-panel control. Output switching transients are typically 0.2 V pk-pk and 0.2 μs in duration (120-pF load).

On-Off Timing Timing is phase-coherent with, and controlled by, either the signal at the Signal Input connector or a different signal applied to the Ext Timing connector. The on interval (duration of burst) and the off interval (between bursts) can be determined by cycle counting, timing, or direct external control.

Cycle-Count Mode: On or off intervals can be set independently, to be of 1, 2, 4, 8, 16, 32, 64, or 128 cycles (i.e periods) duration or to be 2, 3, 5, 9, 17, 33, 65, or 129 cycles with +1 switch operated.

Timed Mode: On and off intervals can be set, independently, for durations of 10 μs to 10 s. On and off times occur at first proper phase point of controlling signal occurring after time interval set on controls; one interval can be timed while other is counted.

Switching Phase: In above modes, input controls determine phase of timing signal at which on and off switching occurs. Slope control selects either positive or negative slope of timing signal; Trigger Level control sets voltage level at which both on and off switching occur.

Direct External Control: A 10-V pulse applied to rear-panel connection will directly control switching.

<u>Synchronizing Pulse:</u> A dc-coupled pulse that alternates between +8 V for output on, and -8 V when off. Source resistance approx. 0.8 kΩ for positive output and 2 kΩ for negative.

WAVEFORM EDUCTOR

Triggering

(1) External: An external waveform of approximately 0.5 volts peak-to-peak minimum is required. The trigger threshold is front panel adjustable over a ± 5 volt range on either the positive or negative slope.

(2) Internal: A 5 volt gate is provided at the start of each scan which can be used to initiate the external measurement.

(3) Manual: A front panel push-button is provided to initiate the signal scan.

Delay: A time delay, continuously variable from 10 microseconds to 11 seconds can be inserted between the triggering signal and the initiation of the signal scan when in the external trigger mode. In the internal trigger mode the delay setting adjusts the time between the end of one scan and the start of the next.

Signal Scan

(1) Sweep Duration: 100 microseconds minimum. Continuously variable from 100 microseconds to 11 seconds in five ranges plus a fixed 100 second slow readout.

(2) Minimum Resolving Time*: Approximately 1 microsecond.

Characteristic Time Constant: (Time constant with which the stored waveform asymptotically approaches its final value): Five to 500 seconds in 1-2-5 increments. Shorter time constants available on special order.

Input Impedance: 100 KΩ shunted by approximately 30 pF.

Gain: Adjustable front panel settings of 1, 2 5 and 10 with continuously adjustable vernier.

Noise: Internally generated noise with shorted input is a function of both Sweep duration and Characteristic Time Constant. With shorted input, output noise is typically 0.2% of full scale.

Output: ± 10 volts. Will drive oscilloscope, stripchart or x-y recorders.

* The minimum time separation between two input pulses that permits resolution of the two pulses at the output.

ALTEC 291-16A DRIVER LOUDSPEAKER

Specifications

Power:	40 watts (continuous white noise over the range 500 -16000 Hz)
Frequency:	500-16000 Hz
Sound Pressure Level:	115 dB re 2.0 x 10^{-5} N/m^2 with 1 watt input measured at 4 feet from the mouth of a 30" trumpet
Impedance:	16 ohms
Voice Coil Diameter:	2.8"
Protection:	12 dB/octave attenuation network
	Low Frequency, 500 Hz Cutoff
Usage:	For sheltered indoor or outdoor applications
Dimensions:	Diameter 6 1/2" Height 4 7/8"
Weight:	20 lbs.

Field replaceable diaphragm/voice coil assembly.

ALTEC 290E DRIVER LOUDSPEAKER

Specifications

Power:	100 watts (above 300 Hz)
Frequency:	300-8000 Hz
Sound Pressure Level:	114 dB re 2.0 x $10^{-5} N/m^2$ with 1 watt input with warble frequency 500-2500 Hz, 4 feet from mouth of 30" trumpet; or 135 dB at 100 watts.
Impedance:	4 ohms
Voice Coil Diameter:	2.8"
Protection:	Low frequencies should be attenuated by network providing 6 dB slope per octave below 300 Hz when used at or near maximum power rating.
Usage:	Weather proofed construction for outdoor use.
Dimensions:	Diameter 6 1/2" Height 7 1/2"
Weight:	21 lbs.

Field replaceable diaphragm/voice coil assembly.

ALTEC 802D DRIVER LOUDSPEAKER

Specifications

Power:	30 watts (above 500 Hz)
Frequency:	500-22000 Hz.
Sound Pressure Level:	111.7 dB re $2.0 \times 10^{-5} N/m^2$ with 1 watt input with warble frequency 500-2500 Hz, 4 feet from mouth of 30" trumpet;or 126.4 dB at 30 watts.
Impedance:	16 ohms
Voice Coil Diameter:	1.75"
Protection:	Low frequencies should be attenuated by network providing 6 dB slope per octave below 500 Hz.
Usage:	HF Driver unit for wide range, two-way studio "Playback" systems.
Dimensions:	Diameter 4 1/2" Height 3 11/16"
Weight:	7 lbs.

Field replaceable diaphragm/voice coil assembly.

ATLANTIC RESEARCH LC-71 PRESSURE TRANSDUCER

Nominal Characteristics

Sensitivity:	0.15 V/Psi, -113 dB, 120 pC/Psi
Capacitance:	800 pf
Maximum Static Pressure:	300 Psi
Rise Time Face On (max):	4μs
Resonant Frequency:	150 KHz
Polarity with Positve Pressure:	Positive
D.C. Resistance (min):	2500 MΩ
Operating Temperature Range:	-40 to +225°F
Thermal Sensitivity:	0.14 %/°F
Acceleration Sensitivity:	Longitudinal 0.013 Psi/g Transverse 0.013 Psi/g
Acoustic Area:	0.034 in^2
Total Weight (max):	0.2 oz
Sensing Element:	Lead Zirconate Titanate

ATLANTIC RESEARCH LG-1310 TRANSDUCER PREAMPLIFERS

Nominal Characteristics

Gain:	0dB
Input Impedance:	1000 MΩ (shunted by 15 pF)
Output Impedance:	300 Ω (in series with 2.2 mF)
Rated Load:	10,000 Ω
Maximum Output Voltage:	With rated load - 3 Vrms With open circuit - 6 Vrms
Nominal Current Drain:	0.5 MA
Equivalent Input Noise with 1000 pF Input:	Broadband -108 dBV
Length:	3 inch
Diameter:	0.75 Inch

LING ELECTRONICS TP-850 POWER AMPLIFIER

Characteristics

Output:	1000 Va at any load power factor from 0.5 leading or lagging. DC-10 KHz (direct coupled) 15 Hz - 10 KHz (input transformer coupled)
Dissipation Capability:	2000 watts, continuous (500 watts DC-25 Hz)
Harmonic Distortion:	Less than 0.5% total at any output level.
Power Response: (at rated output)	± 0.5 dB DC-10 KHz (direct coupled) ± 0.5 dB 15 Hz -10 KHz (input transformer coupled)
Output Voltage:	34 Vrms maximum with 115/230 V line voltage.
Output Current:	30 A rms maximum
Noise and Hum:	At least 80 dB below full output.
Effective Output Impedance:	Less than 0.1 Ω
Input Voltage:	Less than 1 V rms required for full output (direct coupled) Less than 2 V rms required for full output (input transformer coupled-normal connection).
Input Impedance:	1500 Ω (direct coupled) 4000 Ω (input transformer coupled-normal connection)
DC Stability:	Less than 70 mv offset for combination of the following conditions: Line Voltage: 105-125, 210-250 v Ambient Temperature: 10°-45°C Load Impedance: 0.1 Ω to open circuit
Output Current Meter:	0-30 amperes, (thermocouple type meter)
Protective Circuits:	Fast acting (20 μs) solid-state switch in both positive and negative supplies. Protects against over-current, short circuit, and over-temperature.
Power Requirements:	115/230v, 50/60 Hz, single phase, 2900 watts, maximum

Size:	10 1/2" H x 19" W x 25" D
Weight:	125 lbs.

TABLE 1

MAXIMUM SAMPLE INPUT PPL AND TOTAL INCREASE
PRELIMINARY VERSUS FINAL SYSTEM

Frequency Hz	PPL-dB Maximum Preliminary	Maximum Final	Increase*
800	164.8	168.0	3.2
1000	164.8	168.3	3.5
1250	165.8	168.3	2.5
1600	164.4	167.6	3.2
2000	163.2	165.7	2.5
2500	161.1	163.2	2.1
3150	158.9	161.7	1.9
4000	157.5	159.8	2.3
5000	155.7	158.2	2.5
Average			2.6

*Total increase including driver doubling and settling section shortening.

TABLE 2

MAXIMUM SAMPLE INPUT PPL AND TUBE SHORTENING INCREASE
HYBRID VERSUS FINAL SYSTEM

Frequency Hz	PPL-dB Maximum Hybrid	PPL-dB Maximum Final	Increase*
800	167.4	168.0	0.6
1000	167.0	168.3	1.3
1250	166.4	167.8**	1.4
1600	164.4	166.7**	2.3
2000	163.5	165.4**	1.9
2500	161.0	163.2	2.2
3150	159.3	161.7	2.4
4000	157.5	159.8	2.3
5000	156.4	158.2	1.8
Average			1.8

*Increase due to shorter tube

**Levels lowered slightly for this test.

TABLE 3

MAXIMUM SAMPLE INPUT PPL AND DRIVER INCREASE

Frequency Hz	PPL-dB Maximum Preliminary	Maximum Hybrid	Increase*
800	164.8	167.4	2.6
1000	164.8	167.0	2.2
1250	165.8	166.4	0.6
1600	164.4	164.4	0.0
2000	163.2	163.5	0.3
2500	161.1	161.0	-0.1
3150	158.9	159.3	0.4
4000	157.5	157.5	0.0
5000	155.7	156.4	0.7

*Increase due to two drivers instead of one.

TABLE 4

ONE VERSUS TWO DRIVERS ON PRELIMINARY SYSTEM
EQUAL INPUT VOLTAGE

Frequency Hz	PPL-dB One Driver	Two Drivers	Increase*
800	164.8	168.0	3.2
1000	164.8	168.3	3.5
1250	164.2	167.8	3.6
1600	162.8	166.7	3.9
2000	162.1	165.4	3.3
2500	159.9	163.2	3.3
3150	157.8	161.7	3.9
4000	156.3	159.8	3.5
5000	154.6	158.2	3.6
Average			3.5

*Increase due to adding another driver at same voltage as first driver.

TABLE 5

PRELIMINARY VERSUS FINAL VOLTAGE AND POWER DEFICIENCY

	Preliminary				Final				
Frequency Hz	Trans Ohms	Max Volts	Driver Volts	Limit	Trans Ohms	Max Volts	Driver Volts	Limit	Two Driver* Input Power Deficiency dB
800	19	209	120	Driver	10.2	153	120	Driver	0
1000	16	192	122	Driver	7.8	134	122	Driver	0
1250	13	172	127	Driver	7.8	134	127	Driver	0
1600	13	172	178	AMPL	5.8	115	121	AMPL	3.3
2000	13	172	160	AMPL	5.8	115	116	AMPL	2.7
2500	10.2	153	155	AMPL	5.8	115	104	AMPL	3.6
3150	10.2	153	144	AMPL	5.8	115	101	AMPL	2.8
4000	10.2	153	144	AMPL	5.8	115	104	AMPL	2.8
5000	10.2	153	143	AMPL	5.8	115	105	AMPL	2.8
Average 1600-5000									3.0

*Additional amplifier power necessary to drive two drivers at same voltage as one driver.

TABLE 6

ESTIMATED FINAL PPL FOR TWO AMPLIFIERS

Frequency Hz	Measured Maximum Final	Deficiency	Estimated Maximum Final
800	168.0	0	168.0
1000	168.3	0	168.3
1250	168.3	0	168.3
1600	167.6	3.3	170.9
2000	165.7	2.7	168.4
2500	163.2	3.6	166.8
3150	161.7	2.8	164.5
4000	159.8	2.8	162.6
5000	158.2	2.8	161.8

TABLE 7

PRELIMINARY, FINAL, AND ESTIMATED FINAL ACOUSTIC POWER

Frequency Hz	Acoustic Power Watts Preliminary	Final	Estimated Final
800	61.5	129.0	129.0
1000	61.5	138.0	138.0
1250	77.5	138.0	138.0
1600	56.3	117.5	253.0
2000	43.7	76.0	141.0
2500	26.0	42.7	98.0
3150	15.9	30.2	58.0
4000	11.5	19.5	37.0
5000	7.6	13.5	31.0

TABLE 8

FINAL SYSTEM EVALUATION OF RESONANT ABSORBER

Sample	Test	Flow Condition M	Input PPL dB	Figure No.	Octave Insertion Loss-dB	Octave Center Frequency No.
1	Sine Sweep	0	140 rms	39 a)	11.5	2000
	Tone Burst	0	155	39 b)	13.5	2100
	Tone Burst	0	155		13.5	2030
	Tone Burst	0	155		12.0	2100
	Tone Burst	-0.115	155	40 a)	14.0	1930
	Tone Burst	+0.115	155	40 b)	11.5	2260
	Tone Burst	+0.320	155	41 a)	10.5	2460
	Tone Burst	0	160	41 b)	13.8	2040
	Tone Burst	-0.115	160	42 a)	14.5	1930
	Tone Burst	+0.320	160	42 b)	10.5	2400
	Tone Burst	0	165	43 a)	13.5	2060
	Tone Burst	-0.115	165	43 b)	15.0	1890
	Tone Burst	+0.320	165	44 a)	10.5	2400

TABLE 9

FINAL SYSTEM EVALUATION OF RESONANT ABSORBERS

Sample	Test	Flow Condition M	Input PPL dB	Figure No.	Octave Insertion Loss-dB	Octave Center Frequency No.
2	Tone Burst	0	155	45 a)	11.0	2120
"	Tone Burst	+0.115	155	45 b)	9.8	2280
1 + 2	Sine Sweep	0	140 rms	46 a)	24.0	2400
"	Tone Burst	0	155	46 b)	24.5	2630
"	Tone Burst	+0.115	155	47	21.8	2600

TABLE 10

FINAL SYSTEM EVALUATION OF BROADBAND ABSORBERS

Sample	Flow Condition M	Input PPL dB	Figure No.	2500 Hz Insertion Loss dB
3	0	155	48 a)	1.3
3 + 4	0	155	48 a)	2.2
3 + 4	+0.115	155	48 b)	3.0

TABLE II

IMPEDANCE MATCHING TRANSFORMER TAPS

LOAD IMPEDANCE	TERMINALS STRAPPING
4Ω	CONNECT 3, 7, 11, & 15 TOGETHER (ONE LINE)
	CONNECT 5, 9, 13, & 17 TOGETHER (ONE LINE)
5.8Ω	CONNECT 4, 8, 12 & 16 TOGETHER (ONE LINE)
	CONNECT 6, 10, 14 & 18 TOGETHER (ONE LINE)
7.8Ω	CONNECT 3, 7, 11 & 15 TOGETHER (ONE LINE)
	CONNECT 6, 10, 14 & 18 TOGETHER (ONE LINE)
10.2Ω	CONNECT 5 TO 8 & 13 TO 16
	CONNECT 4 TO 12 (ONE LINE)
	CONNECT 9 TO 17 (ONE LINE)
13Ω	CONNECT 3 TO 11 (ONE LINE)
	CONNECT 9 TO 17 (ONE LINE)
	CONNECT 5 TO 8 & 13 TO 16
16Ω	CONNECT 5 TO 7 & 13 TO 15
	CONNECT 3 TO 11 (ONE LINE)
	CONNECT 9 TO 17 (ONE LINE)
19Ω	CONNECT 3 TO 11 (ONE LINE)
	CONNECT 9 TO 17 (ONE LINE)
	CONNECT 6 TO 8 & 14 TO 16
23Ω	CONNECT 6 TO 8 & 14 TO 16
	CONNECT 4 TO 12 (ONE LINE)
	CONNECT 10 TO 18 (ONE LINE)
31Ω	CONNECT 6 TO 7 & 14 TO 15
	CONNECT 3 TO 11 (ONE LINE)
	CONNECT 10 TO 18 (ONE LINE)
41Ω	SERIES CONNECT 5-8, 9-12, 13-16
	4 (ONE LINE)
	12 (ONE LINE)

TABLE II (cont'd)

IMPEDANCE MATCHING TRANSFORMER TAPS

LOAD IMPEDANCE	TERMINALS STRAPPING
64Ω	SERIES CONNECT 5-7, 9-11, 13-15
	3 (ONE LINE)
	17 (ONE LINE)
88Ω	SERIES CONNECT 6-8, 10-12, 14-16
	4 (ONE LINE)
	18 (ONE LINE)
126Ω	SERIES CONNECT 6-7, 10-11, 14-15
	3 (ONE LINE)
	18 (ONE LINE)

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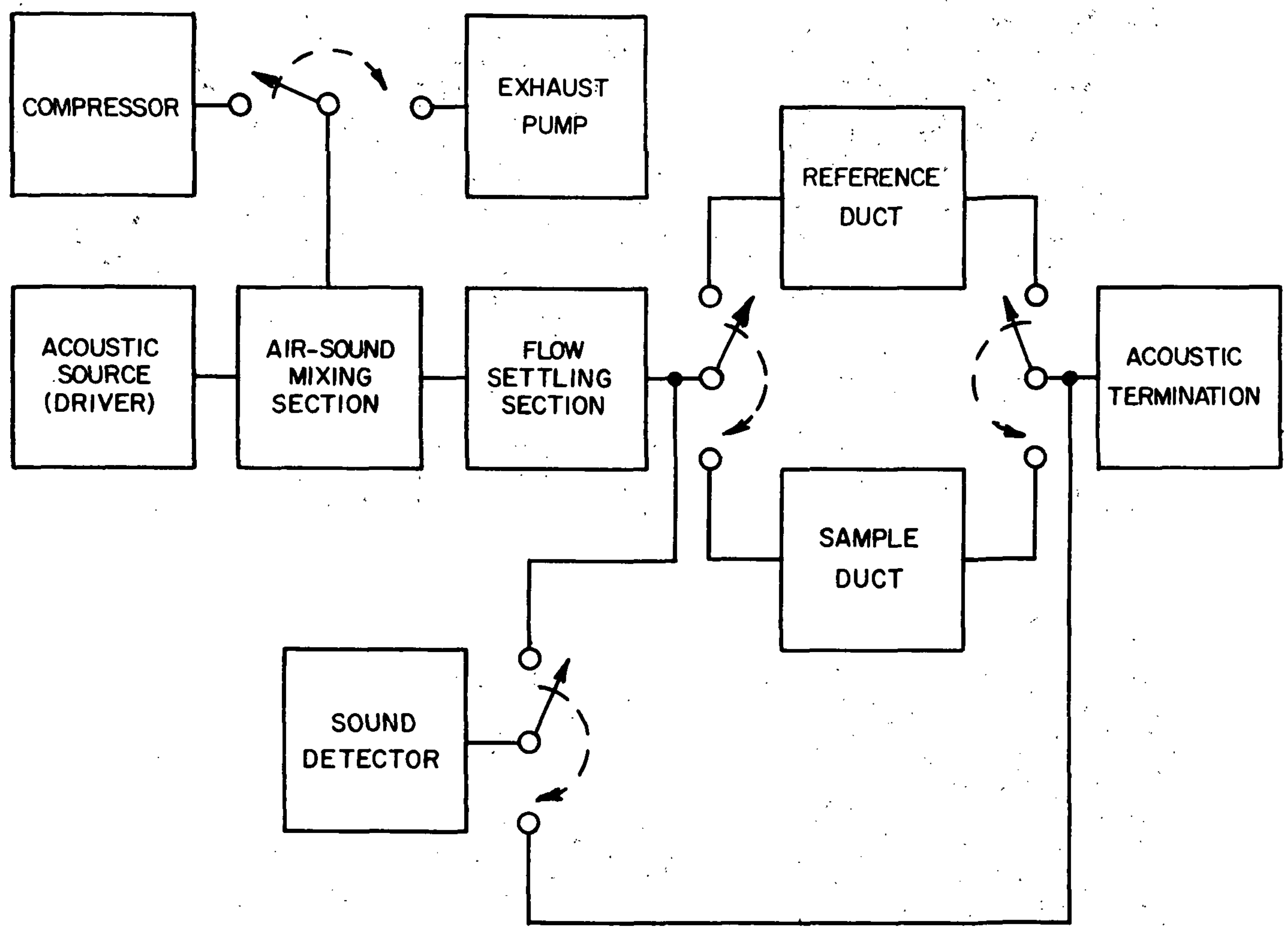
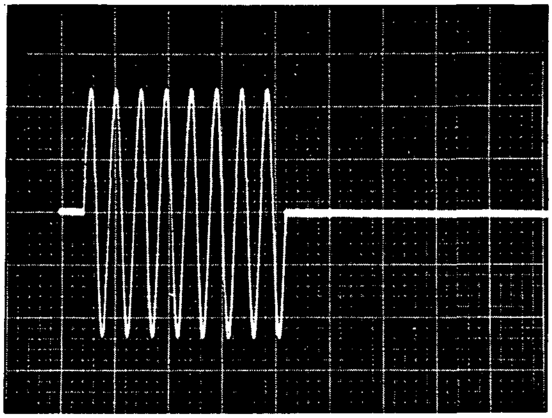
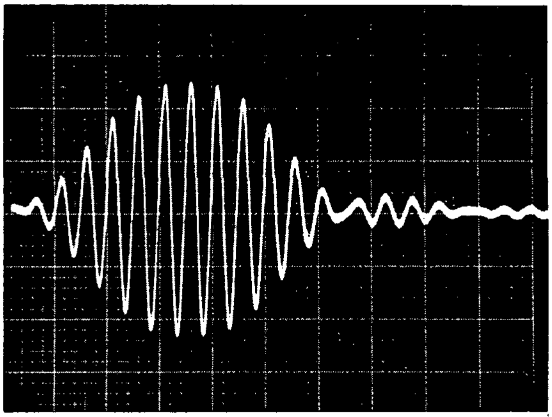


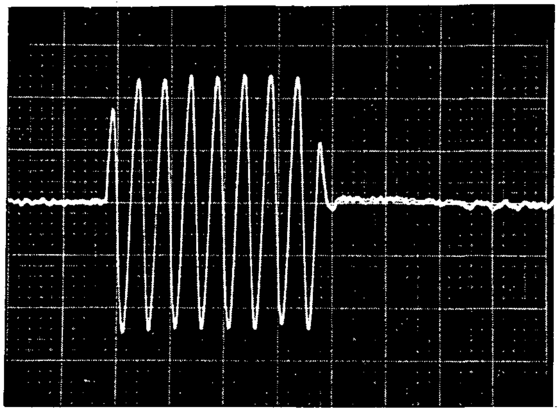
FIGURE I. TONE - BURST FLOW SYSTEM DIAGRAM



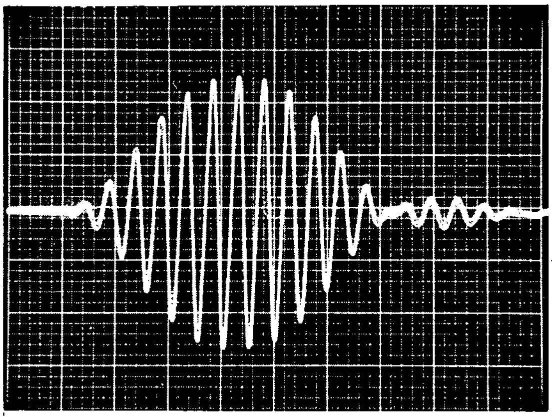
a. TONE BURST GENERATOR OUTPUT



b. FILTERED TONE BURST (ONE-THIRD OCTAVE FILTER)



c. DRIVER OUTPUT WITH UNFILTERED TONE BURST INPUT



d. DRIVER OUTPUT WITH FILTERED TONE BURST INPUT. (ONE-THIRD OCTAVE FILTER)

FIG. 2. TONE BURST INPUT SIGNAL AFTER VARIOUS STAGES OF PROCESSING. 1K Hz CENTER FREQUENCY, 8 CYCLE DURATION.

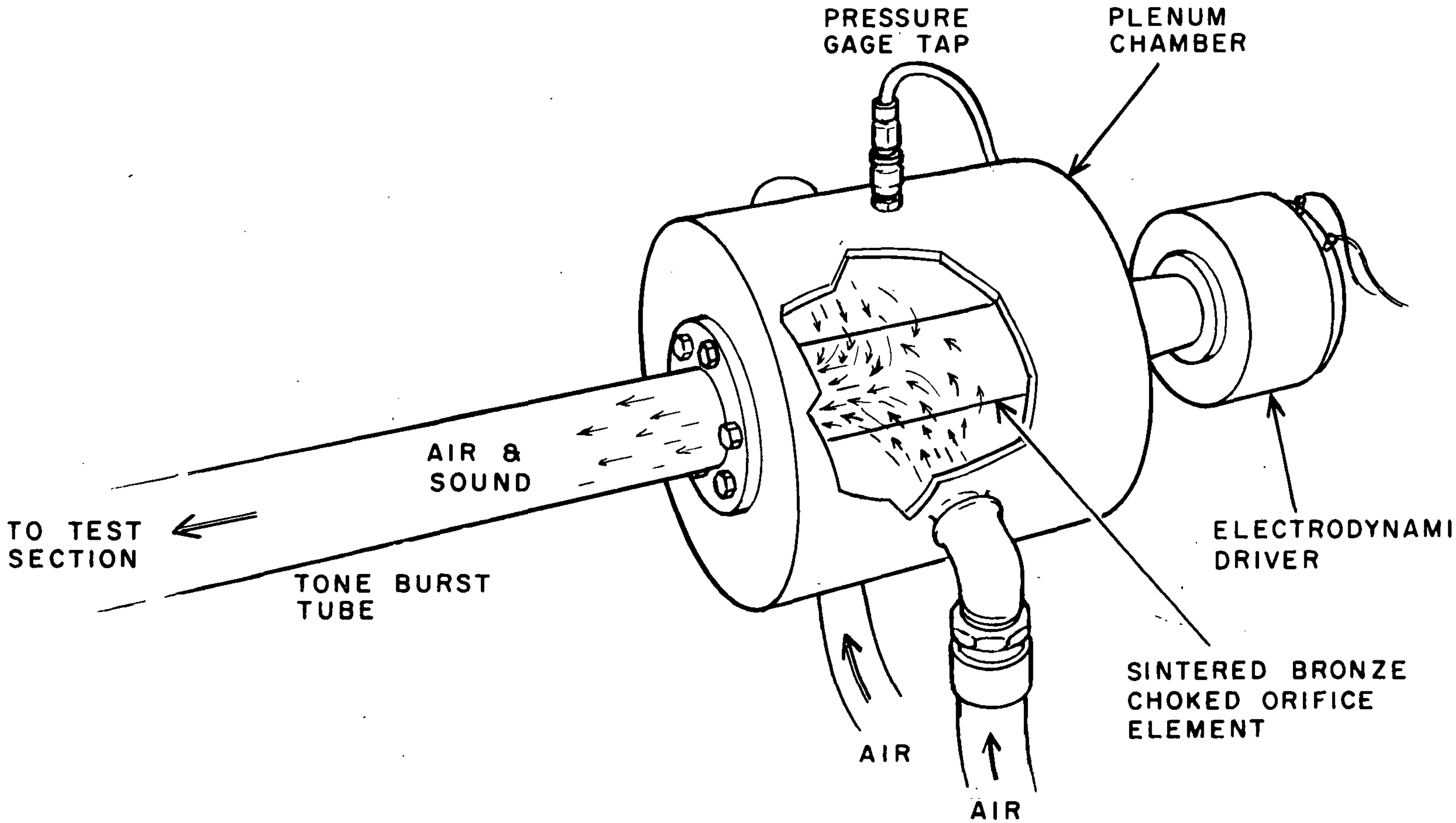


FIGURE 3. AIR-SOUND INTERMIXING GEOMETRY

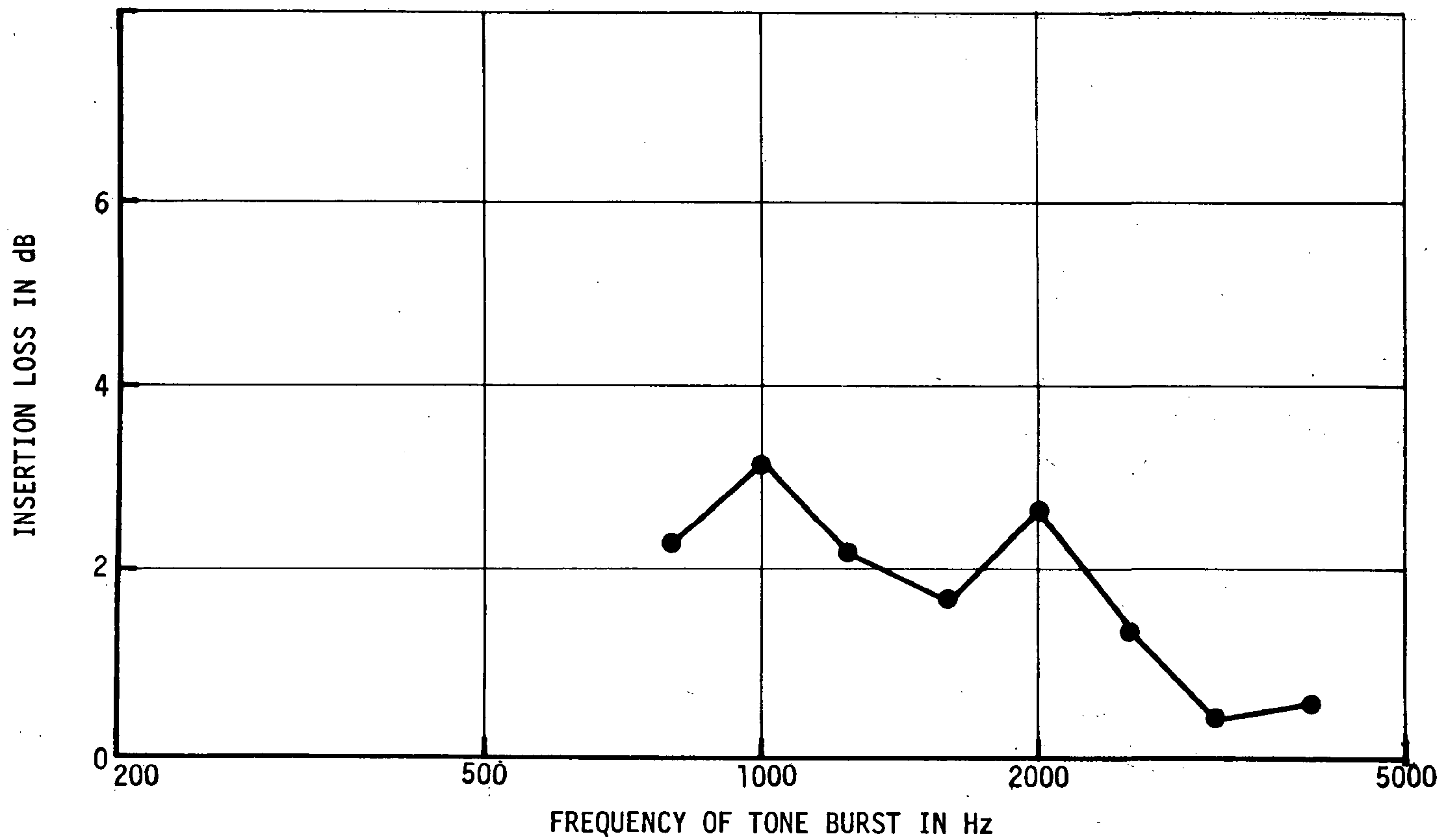


FIGURE 4. MIXING SECTION INSERTION LOSS

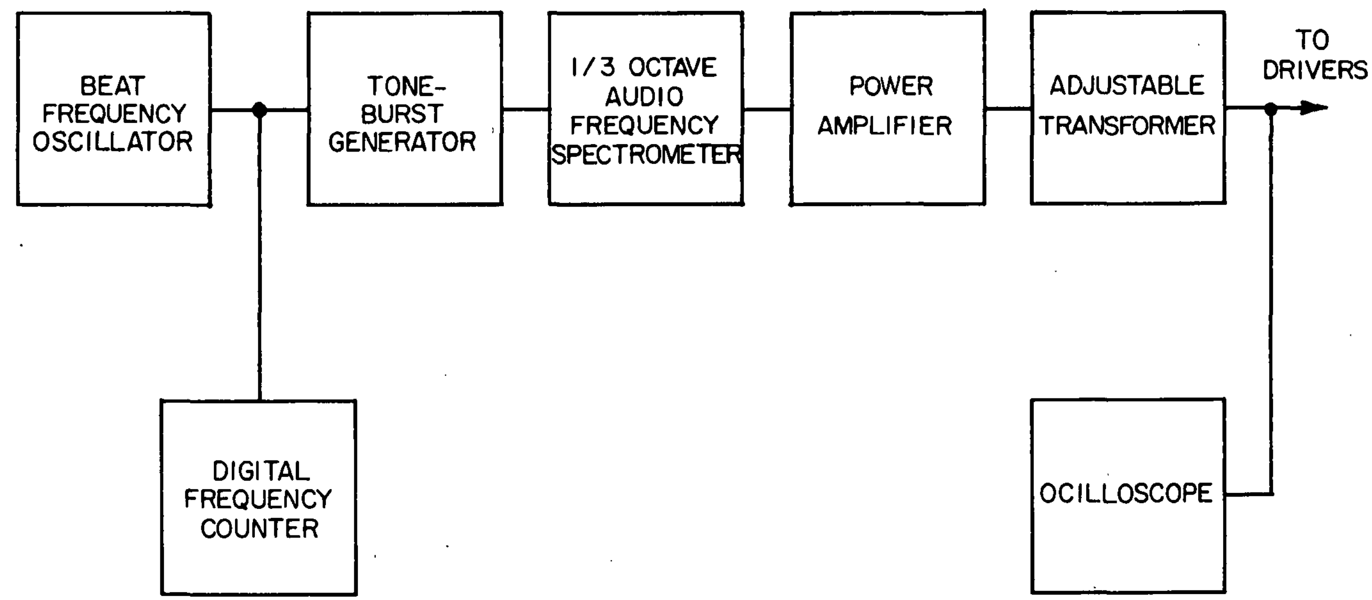


FIGURE 5. SIGNAL GENERATING INSTRUMENTATION

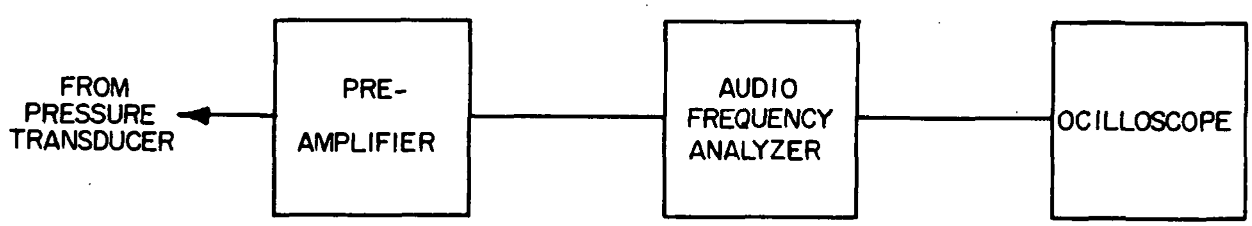


FIGURE 6. SIGNAL PROCESSING INSTRUMENTATION

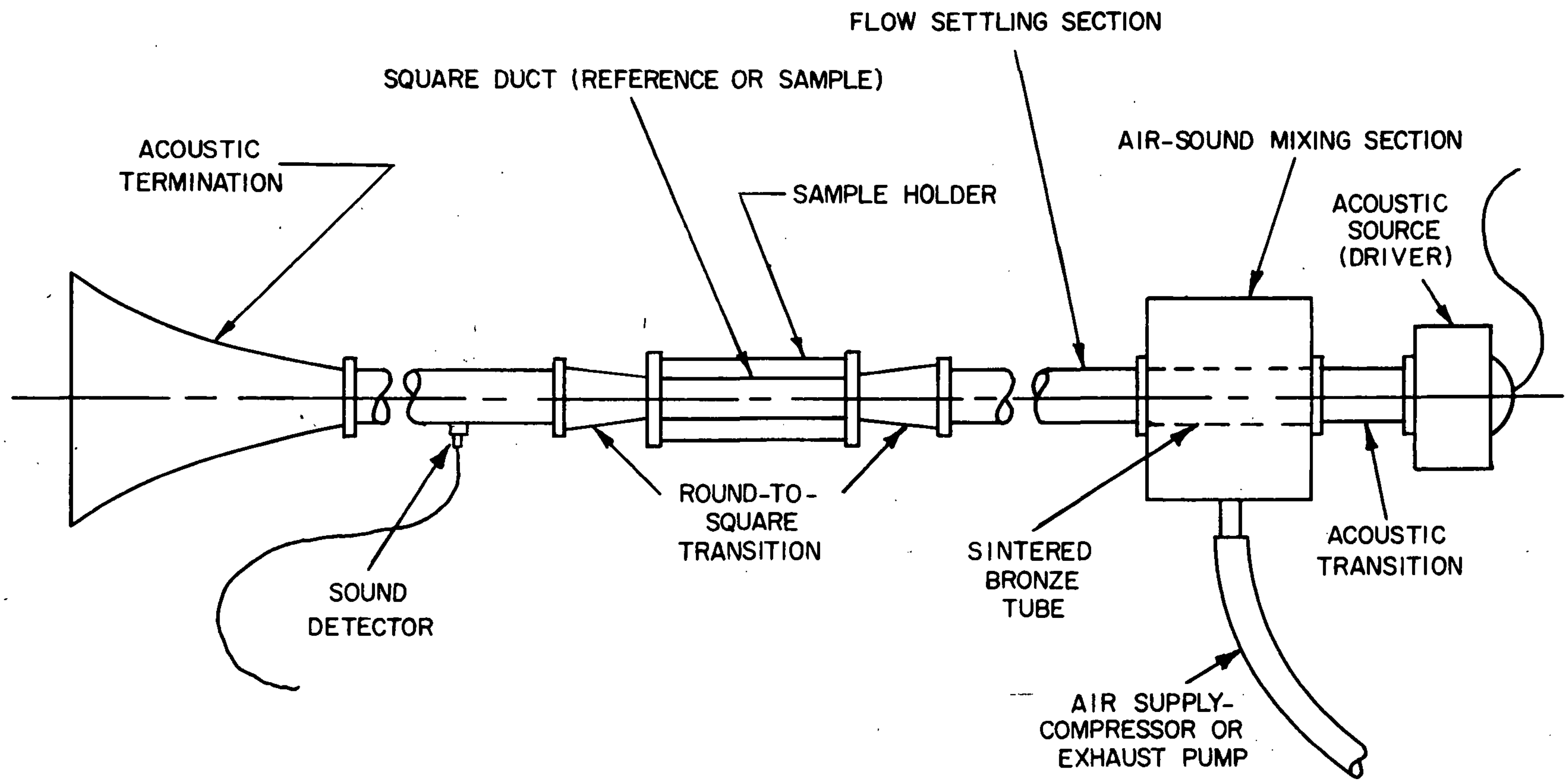


FIGURE 7. PRELIMINARY TONE-BURST FLOW TUBE SYSTEM

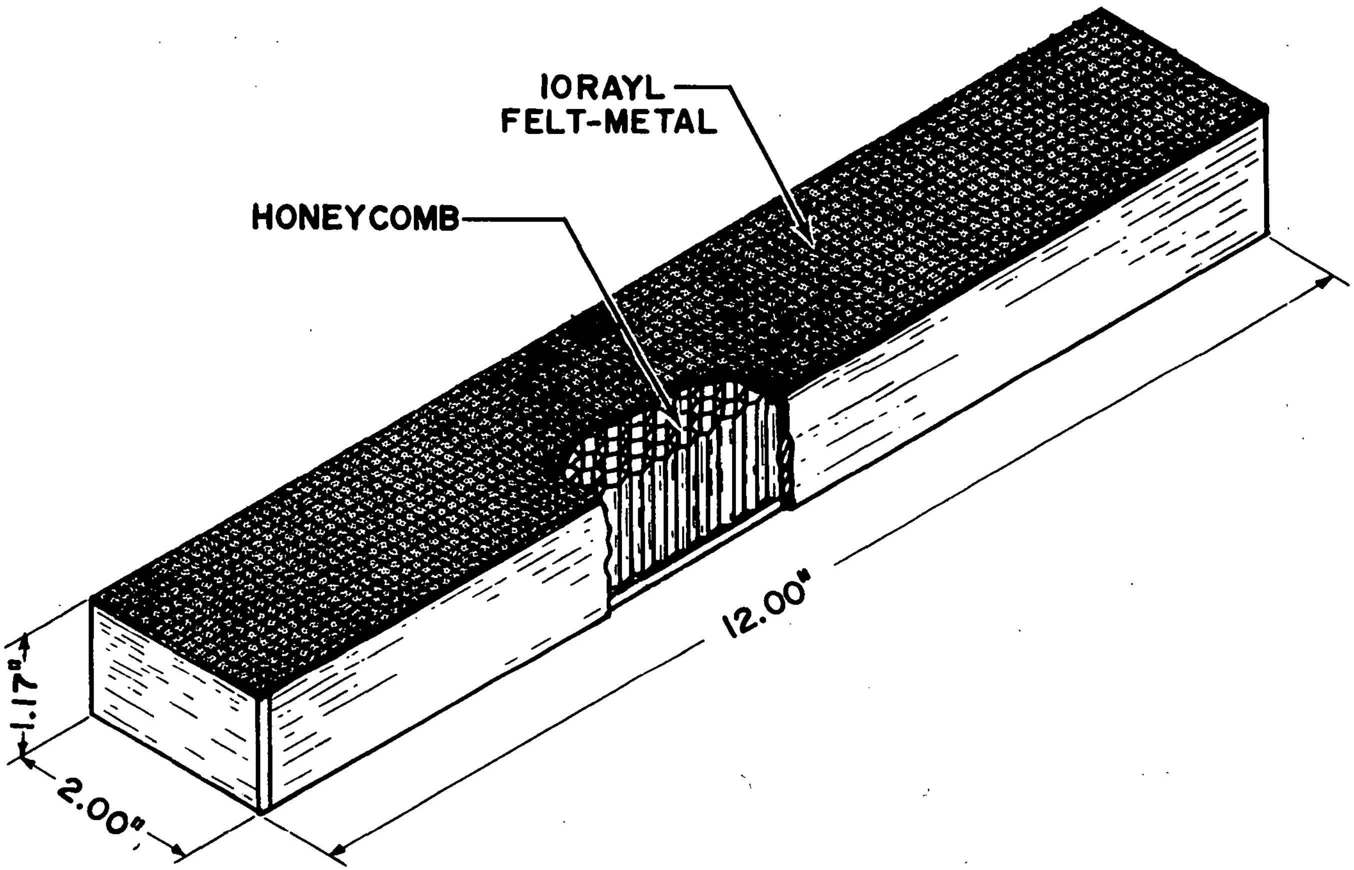


FIGURE 8 TEST SAMPLE

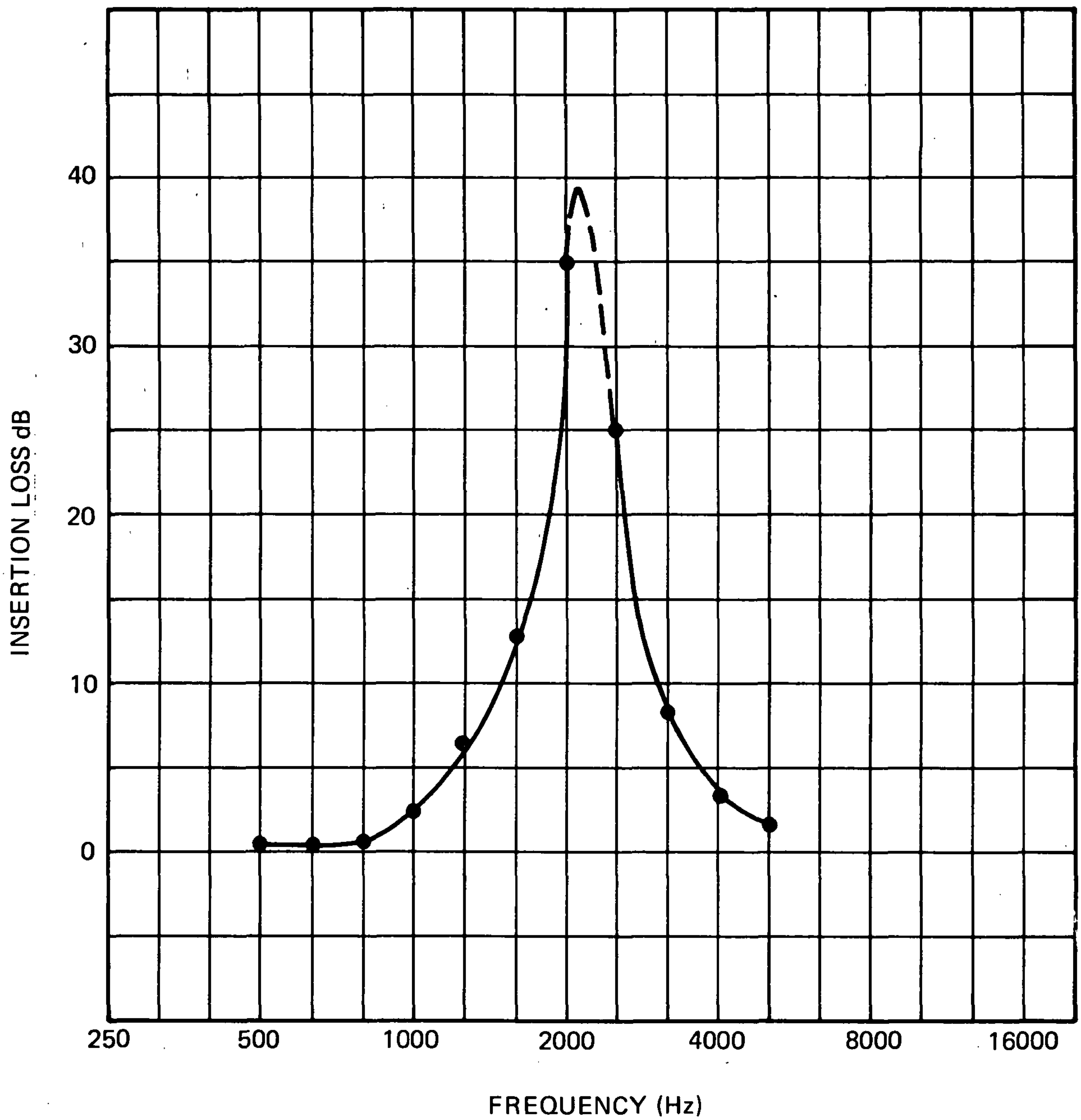


FIGURE 9. INSERTION LOSS RESONANT ABSORBER AT 130dB PPL NO FLOW

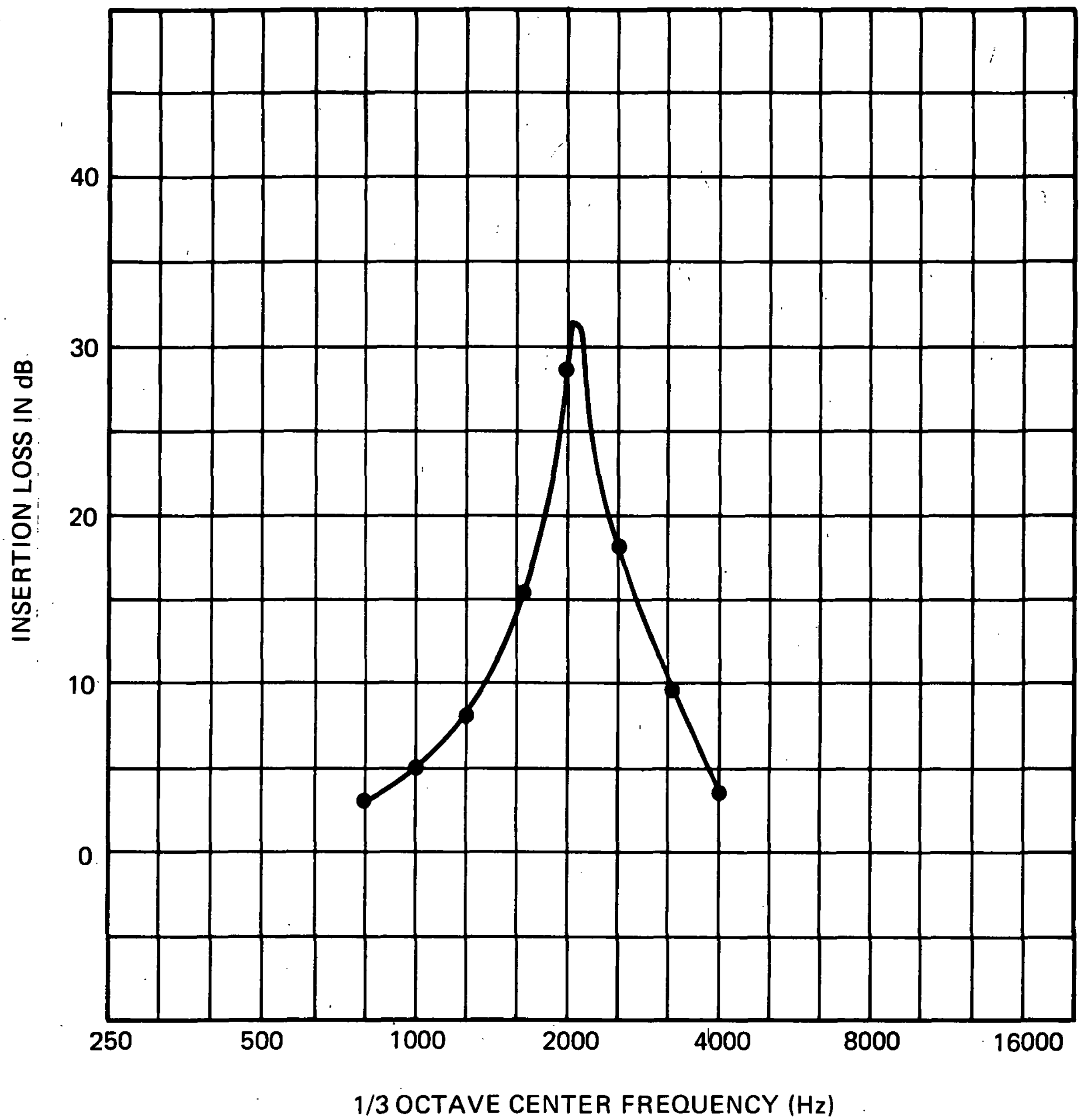


FIGURE 10. INSERTION LOSS RESONANT ABSORBER AT 160 dB PPL NO FLOW

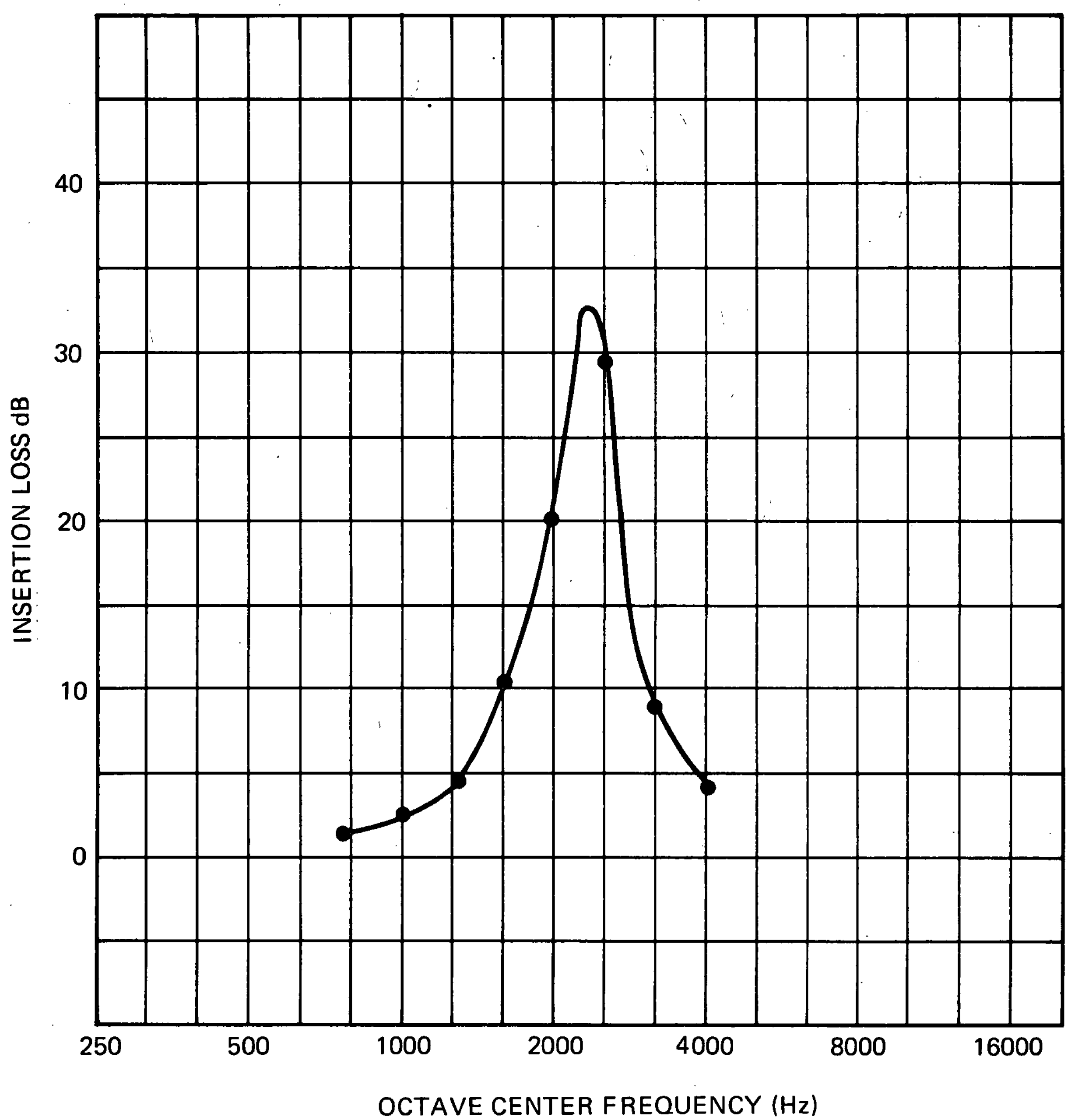


FIGURE 11. INSERTION LOSS RESONANT ABSORBER AT 140 dB PPL 130 FT/SEC FLOW

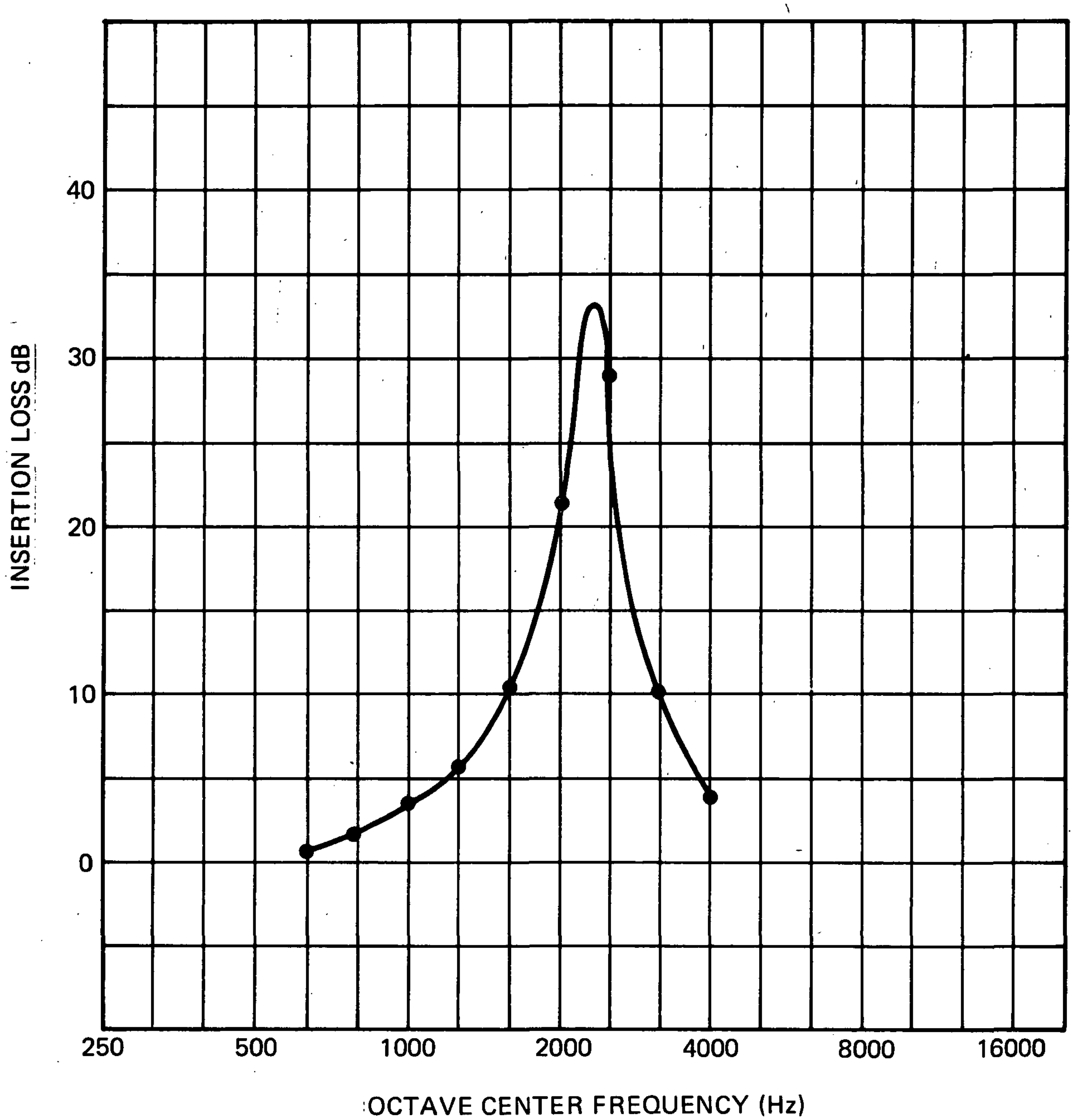


FIGURE 12. INSERTION LOSS RESONANT ABSORBER AT 160 dB PPL 130 FT/SEC FLOW

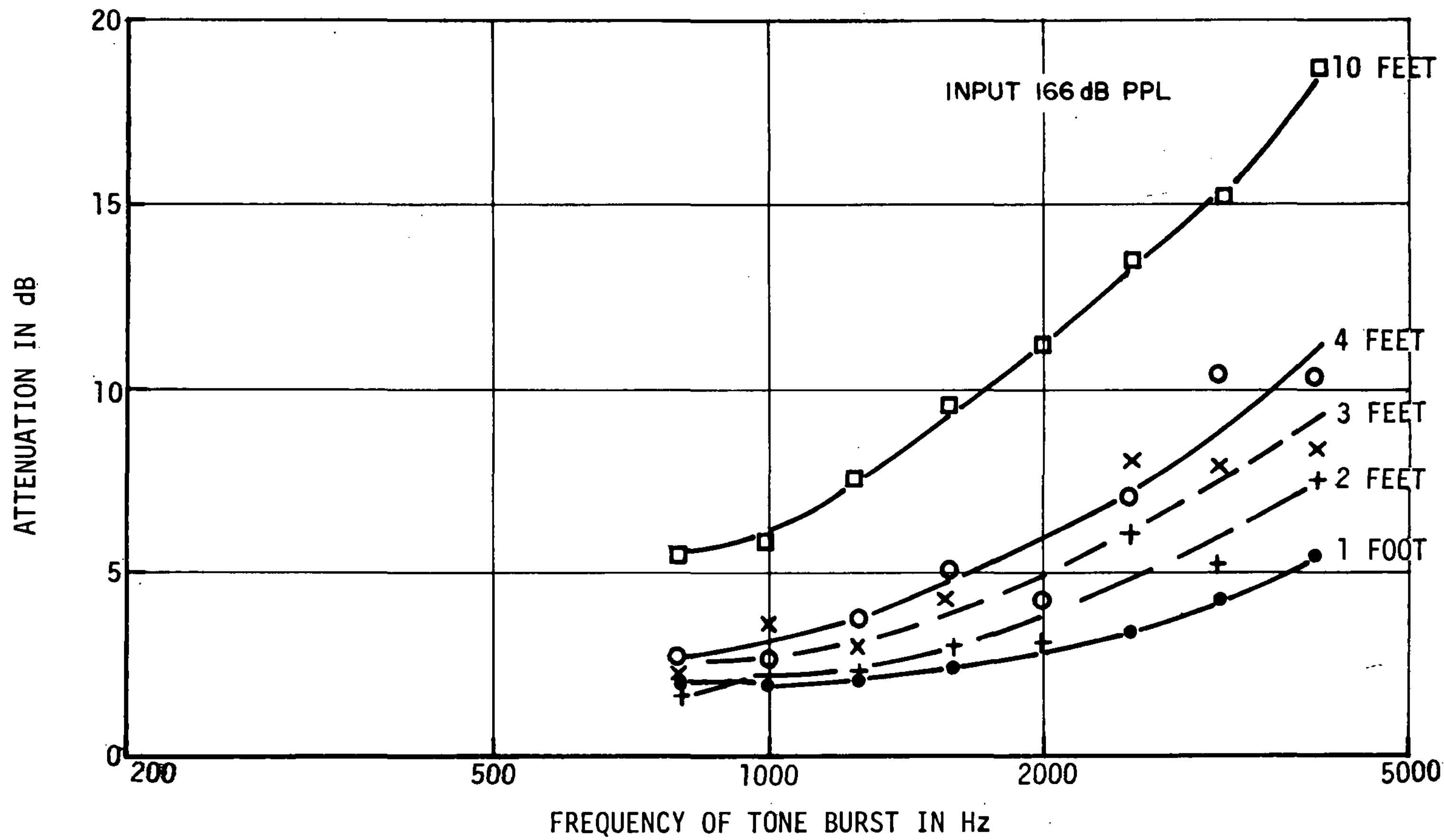


FIGURE 13. 2-INCH TUBE ATTENUATION VS LENGTH

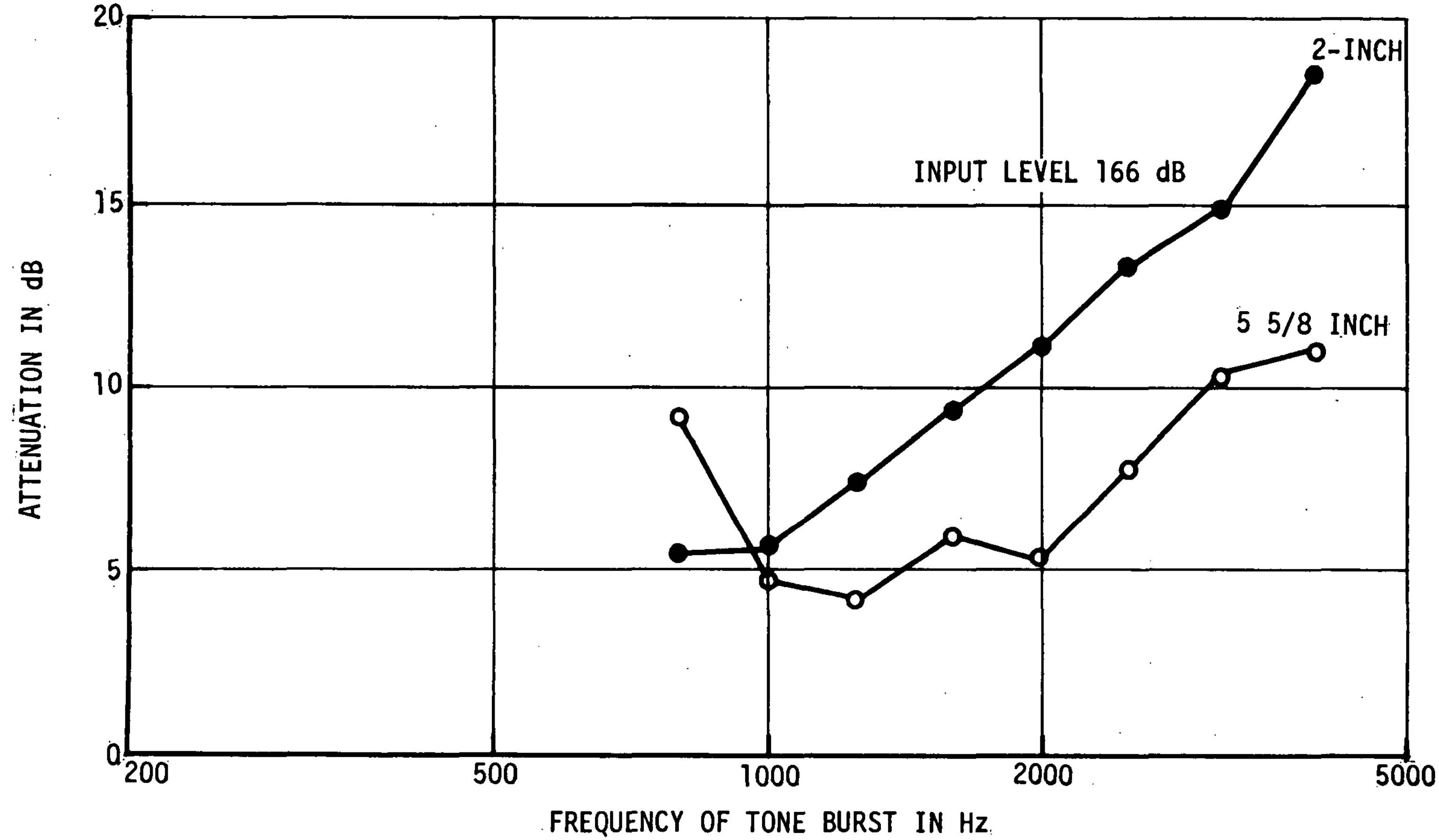


FIGURE 14. 2-INCH AND 5 5/8 INCH TUBE ATTENUATION FOR 10 FOOT LENGTH

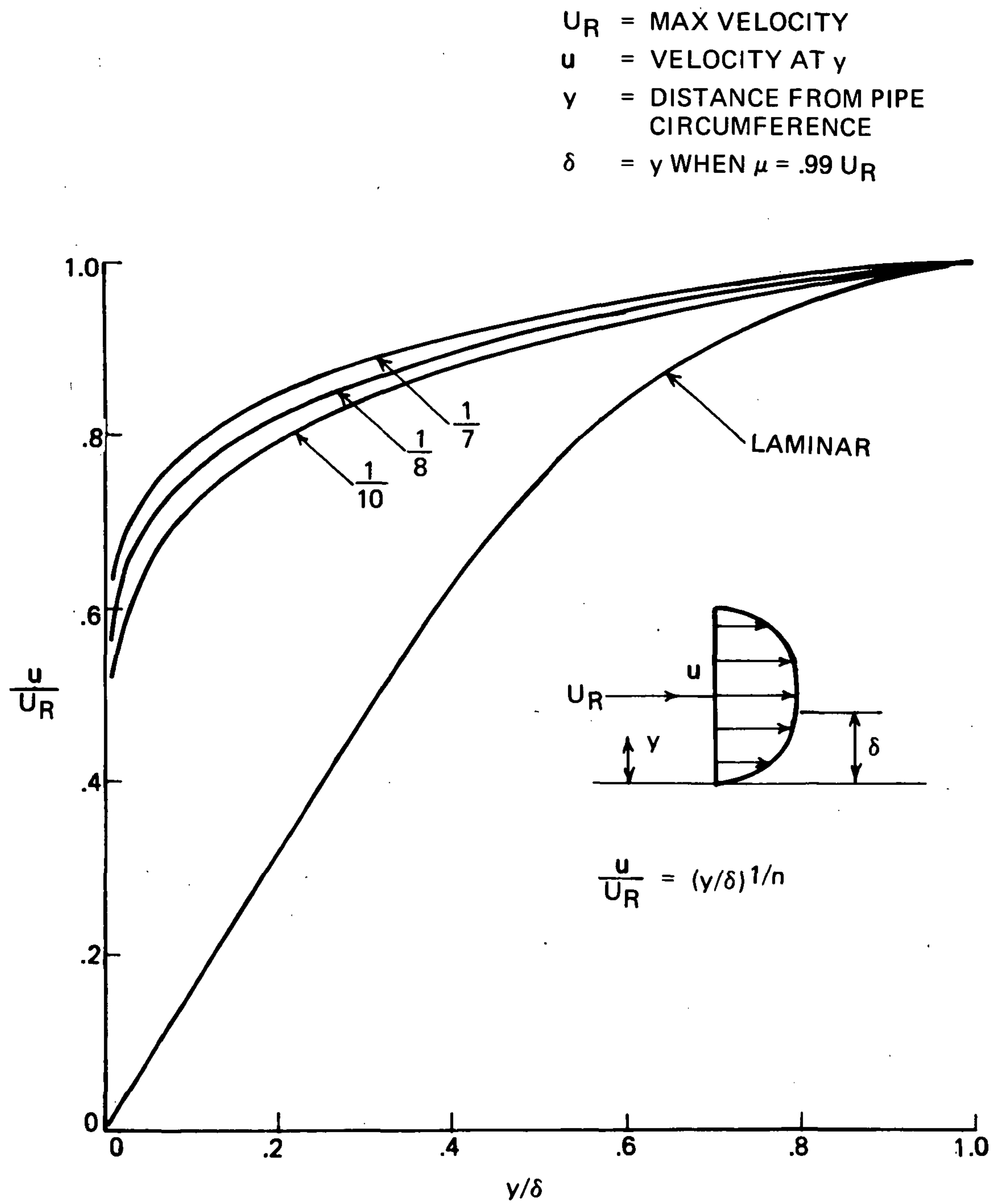


FIGURE 15. FLOW PROFILES IN A PIPE

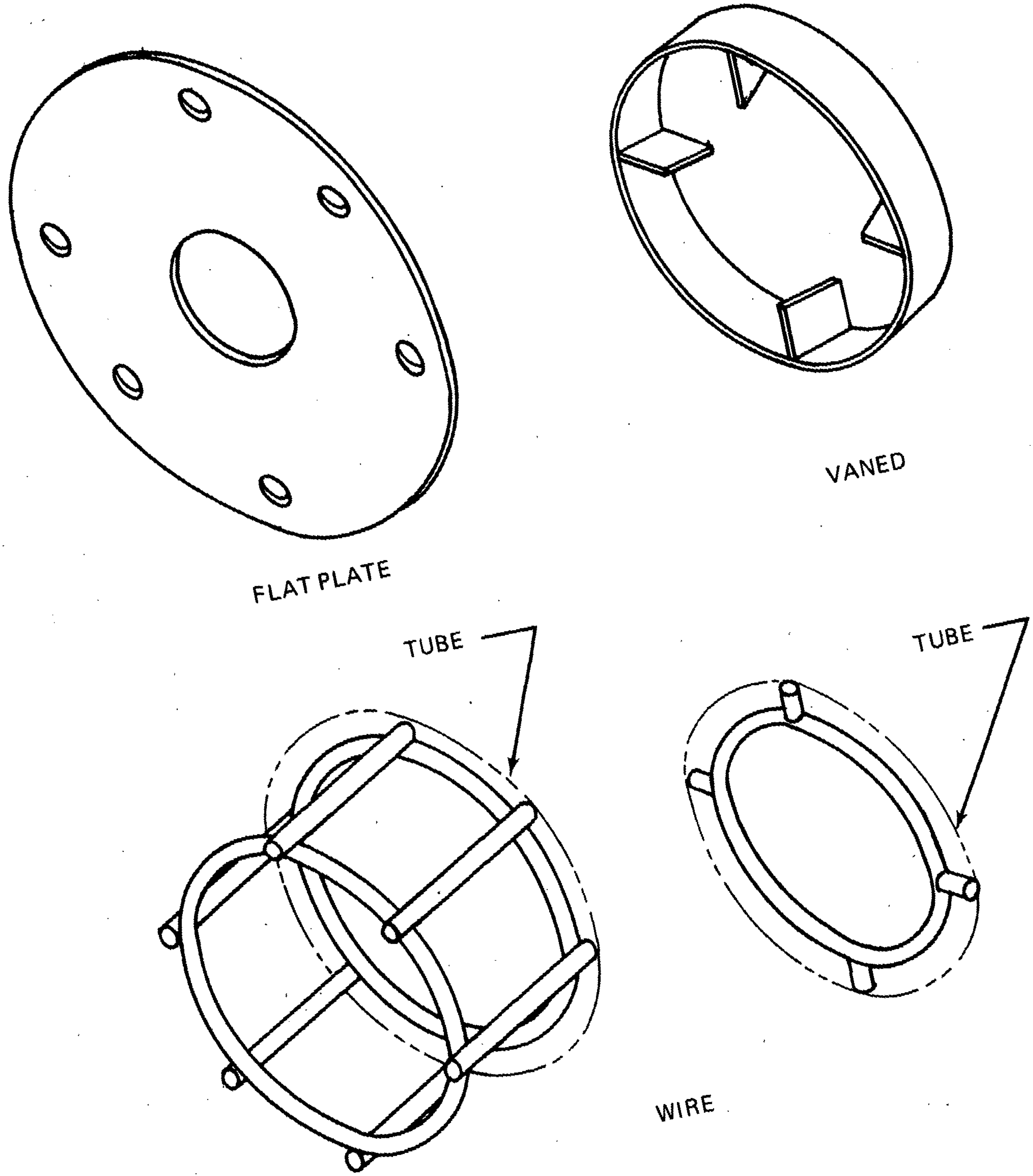


FIGURE 16 FLOW TRIPPERS

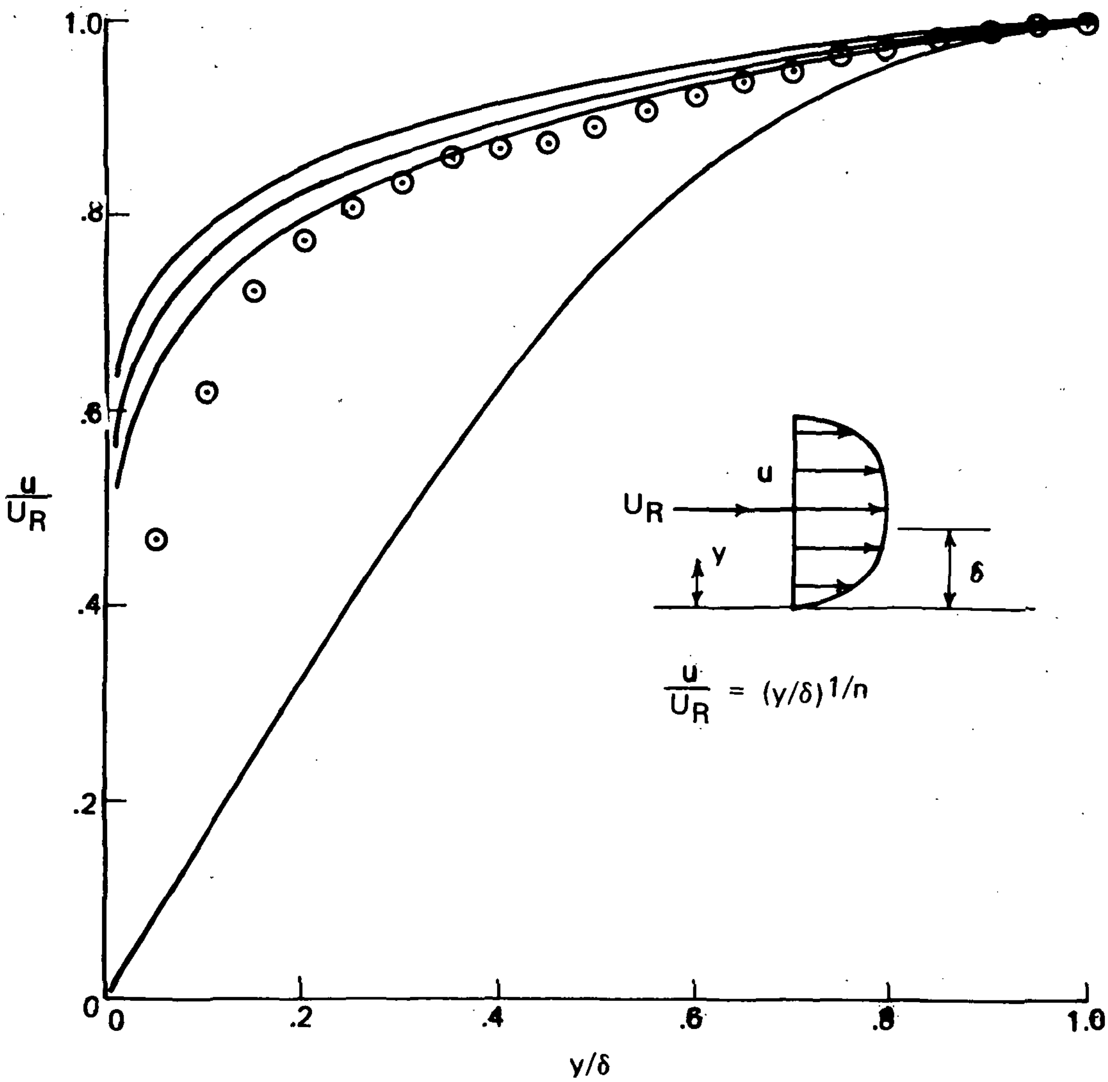


FIGURE 17. FLOW PROFILE NO TRIPPER AT 33½ INCHES AND 130 FT/SEC

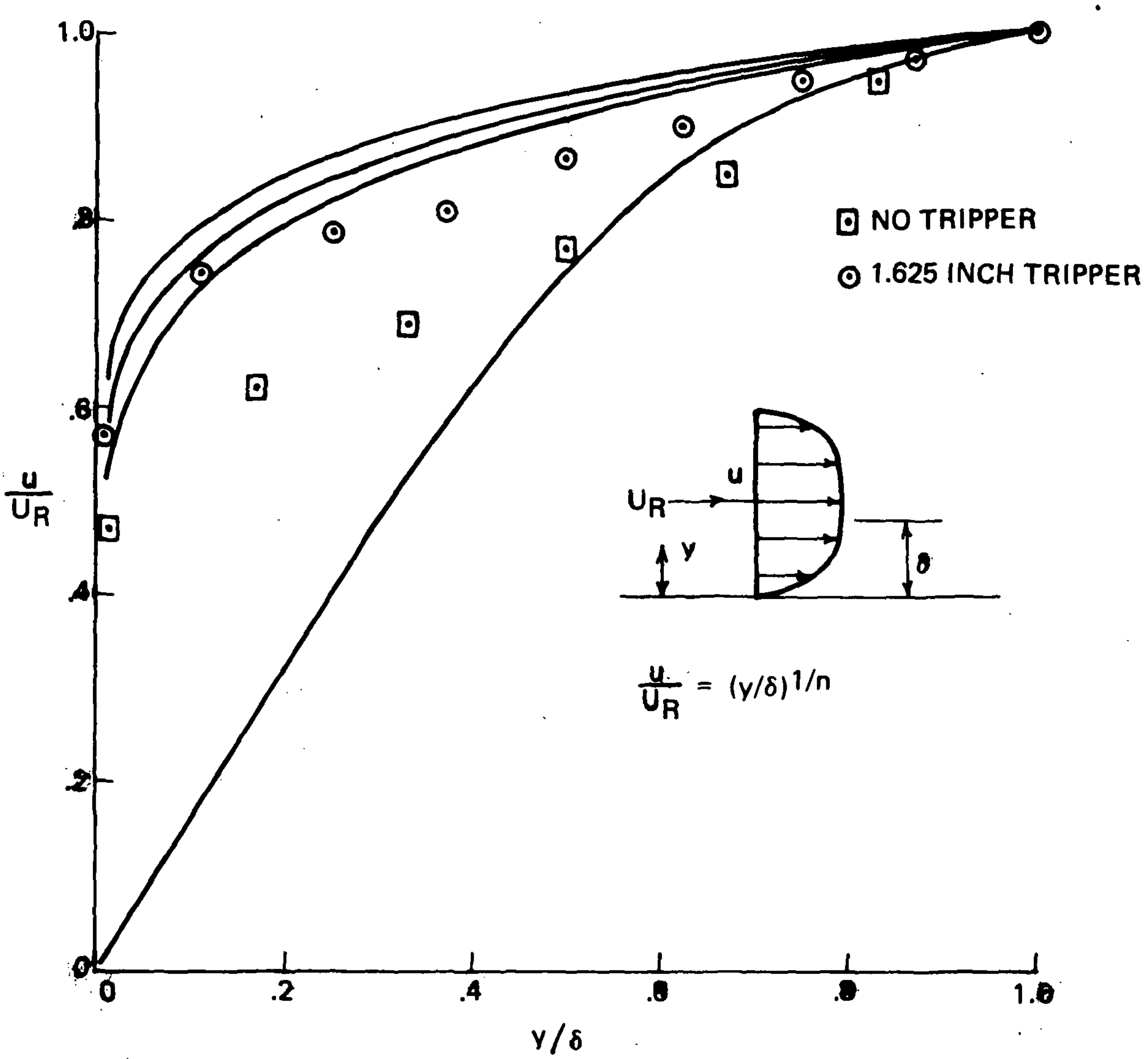


FIGURE 18. FLOW PROFILES AT 9½ INCHES AND 130 FT/SEC

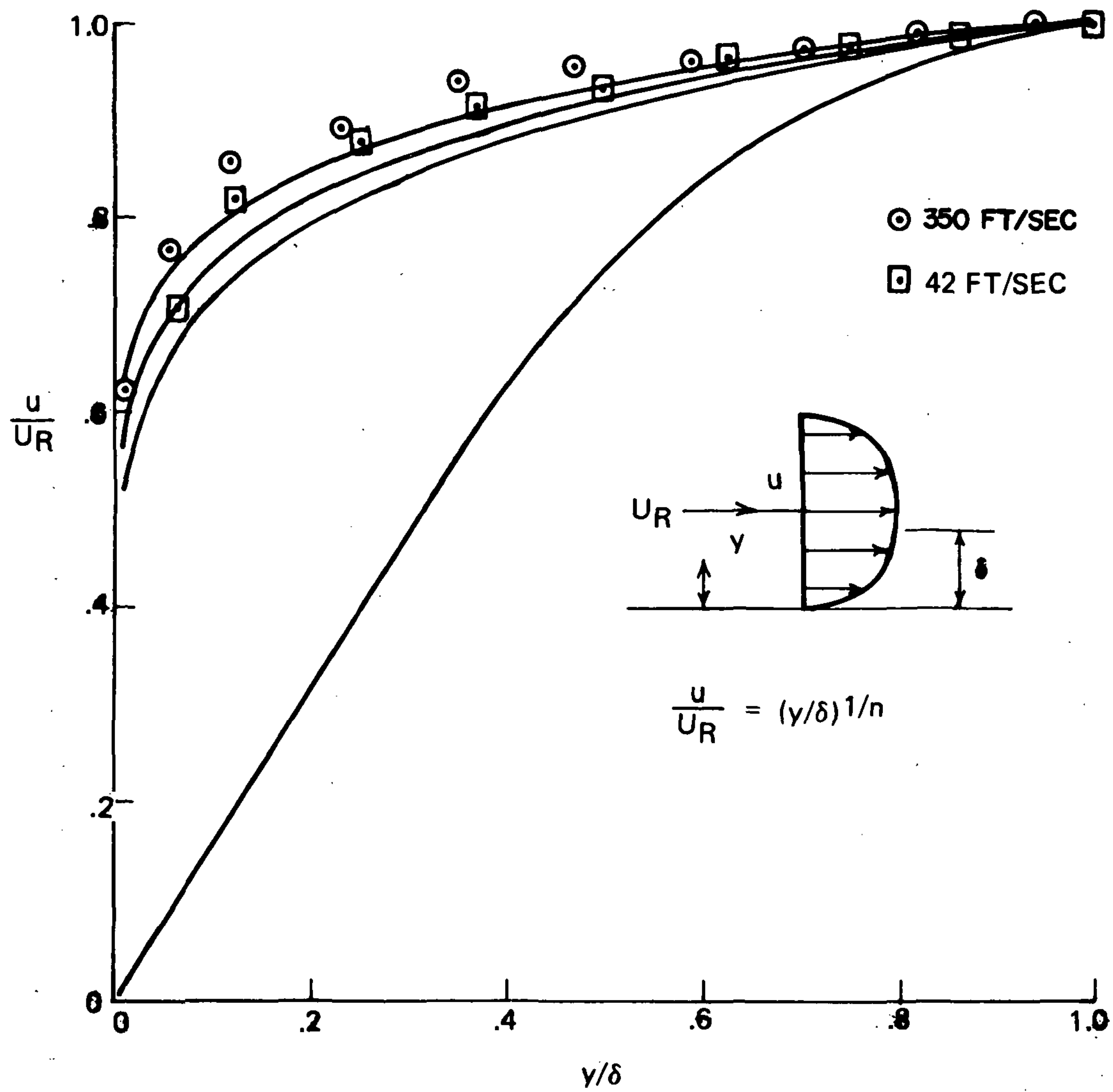


FIGURE 19. FLOW PROFILES WITH 1.625 INCH TRIPPER AT 15½ INCHES

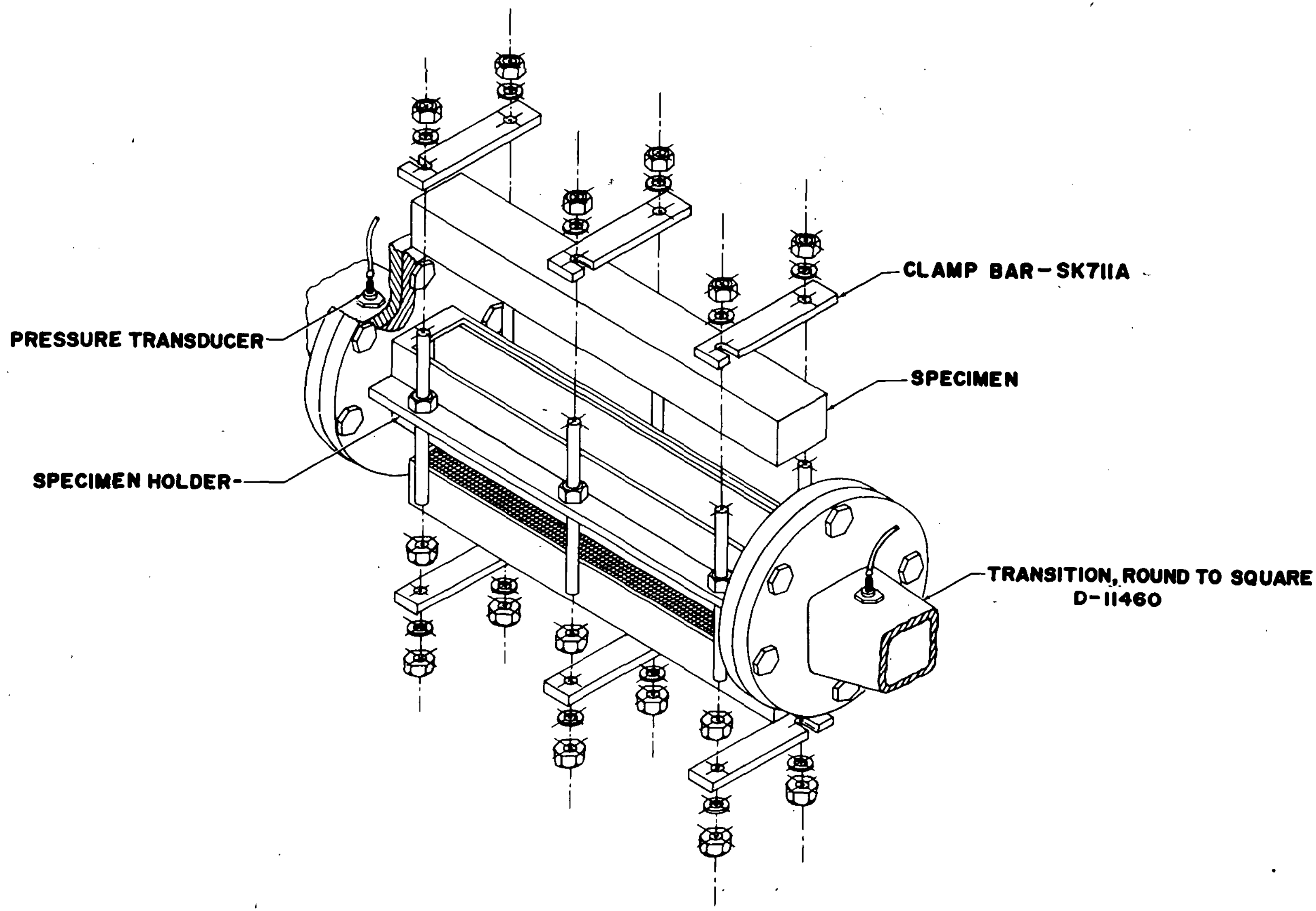


FIGURE 20 IMPROVED REFERENCE – SAMPLE SECTION

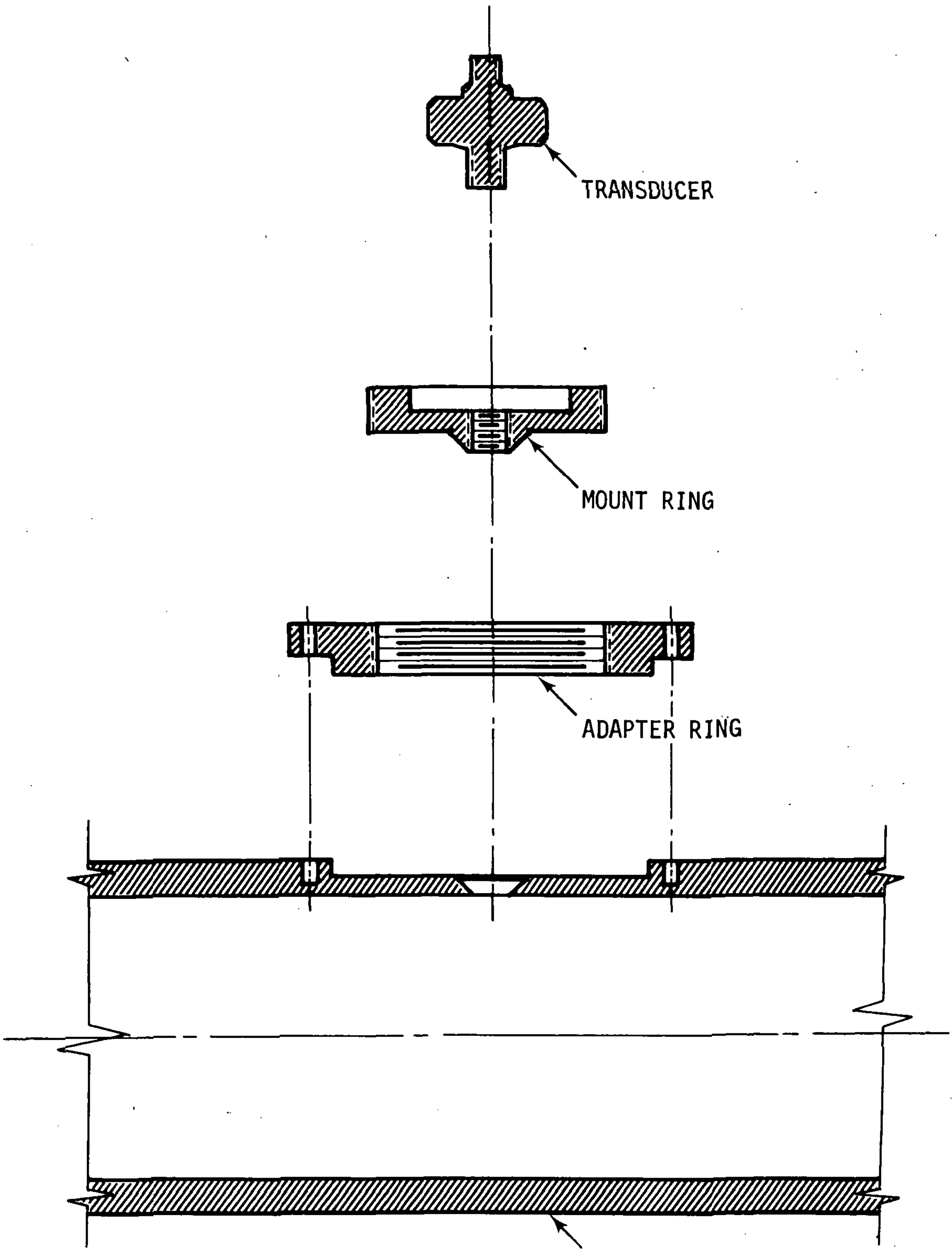


FIGURE 21. MICROPHONE MOUNTING

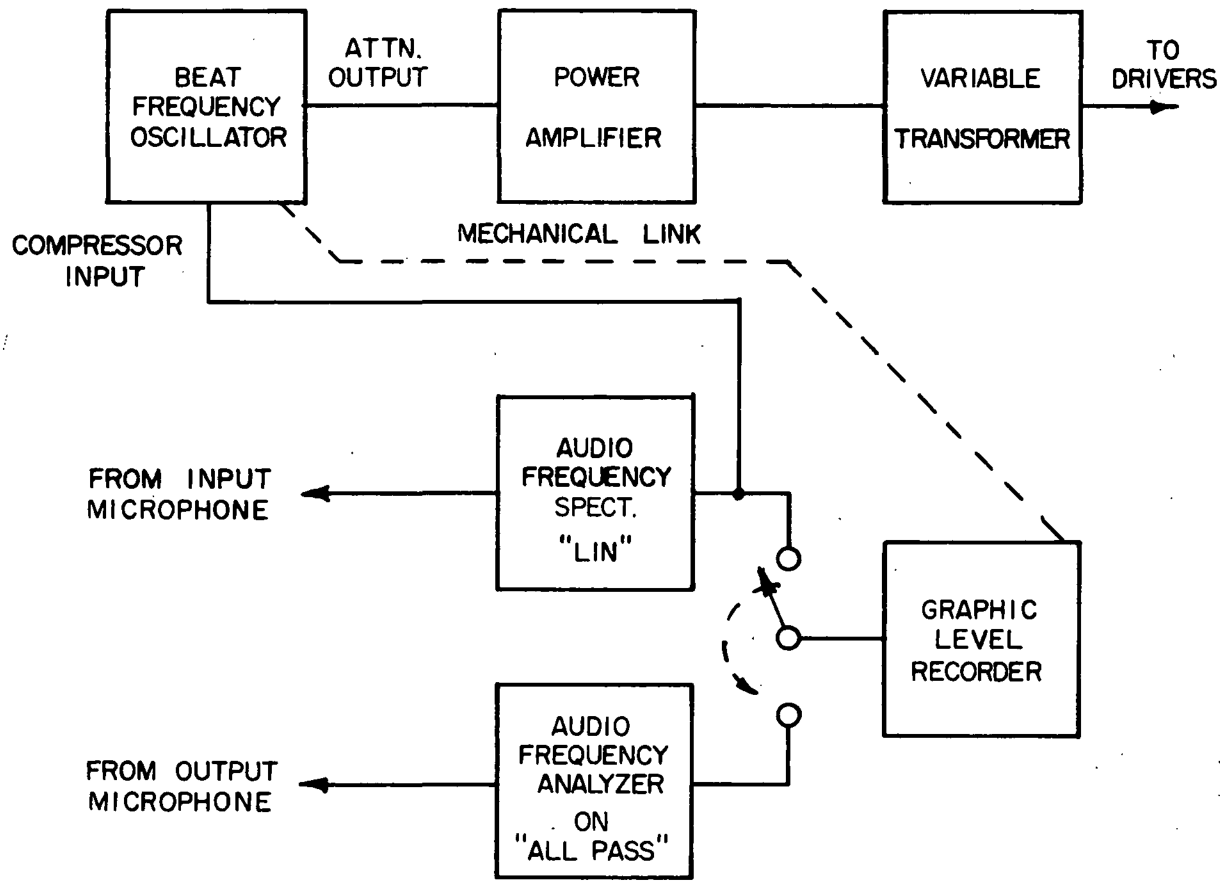


FIGURE 22. INSTRUMENTATION FOR SWEPT-SINE WAVE MEASUREMENTS

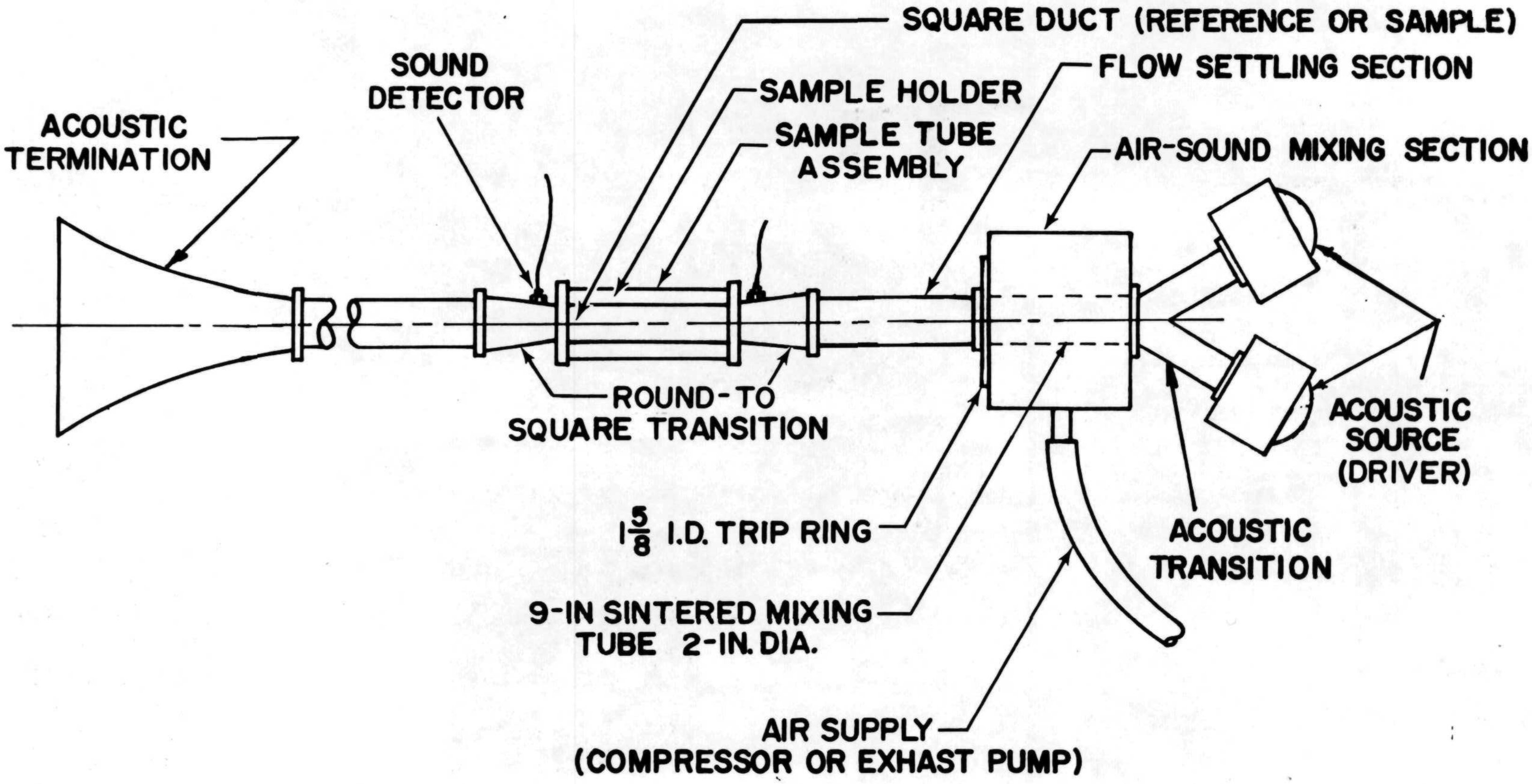


FIGURE 23. FINAL TONE-BURST FLOW TUBE SYSTEM

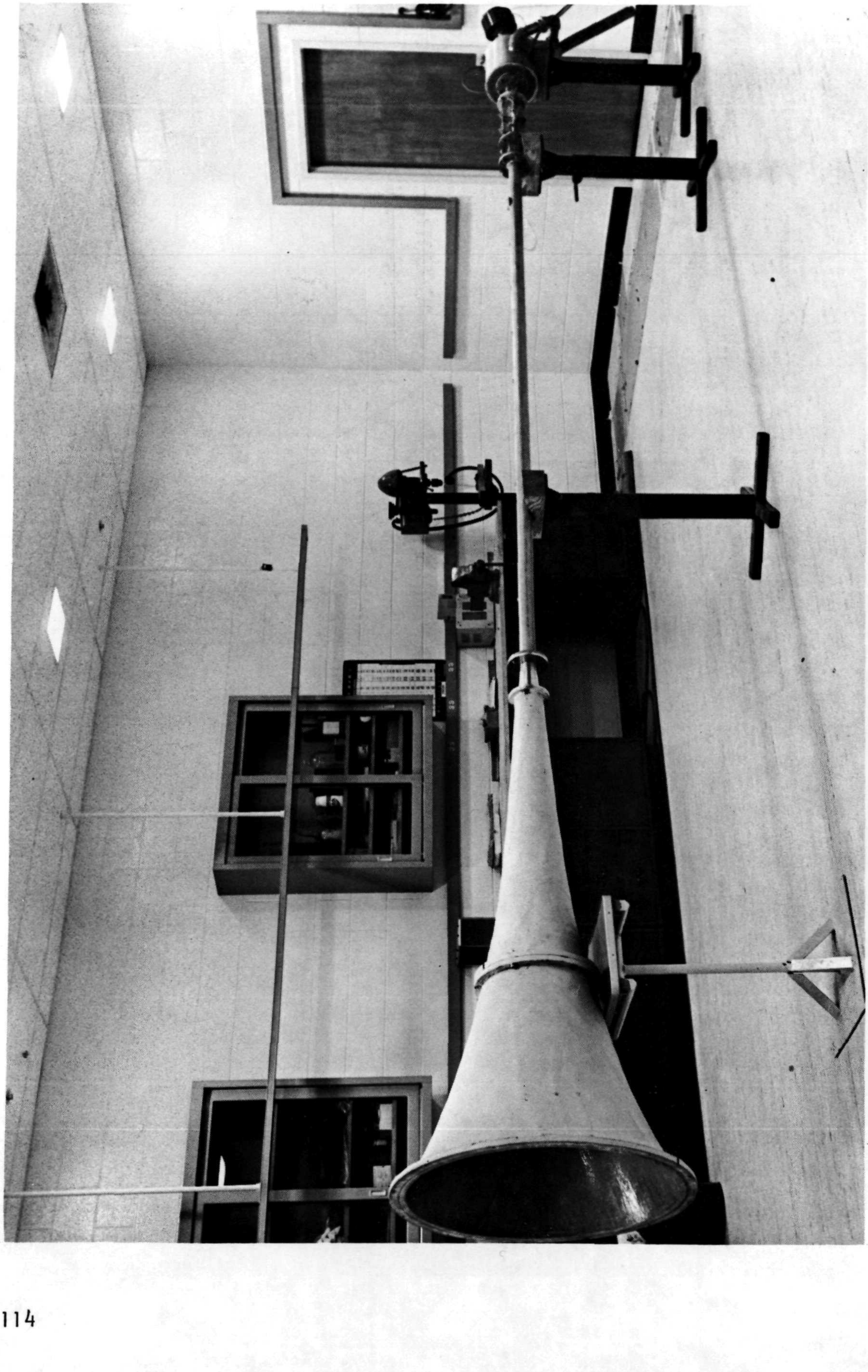


FIGURE 24. FINAL TONE - BURST FLOW TUBE SYSTEM

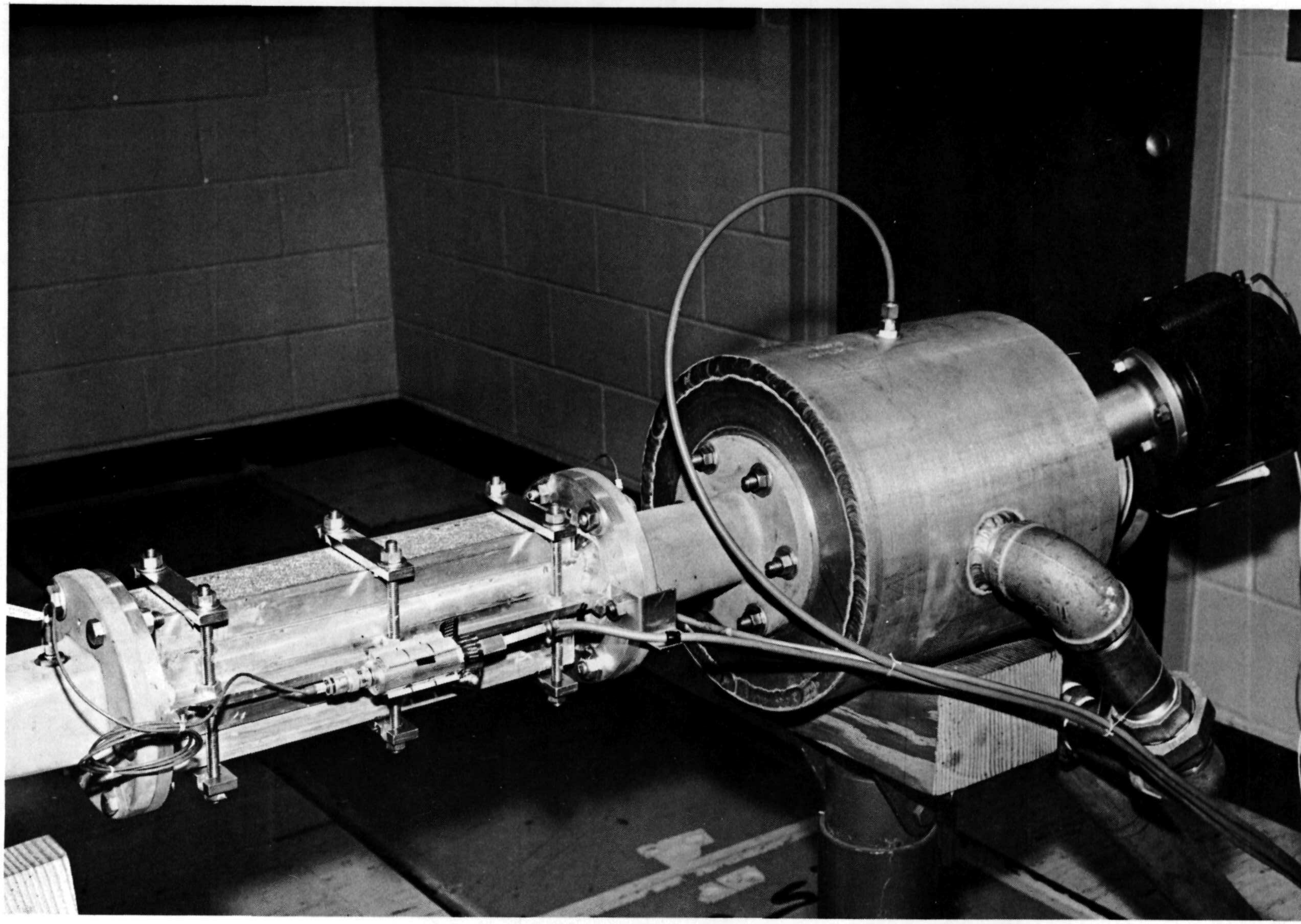


FIGURE 25. FINAL SYSTEM—DRIVERS, MIXER, AND SAMPLE

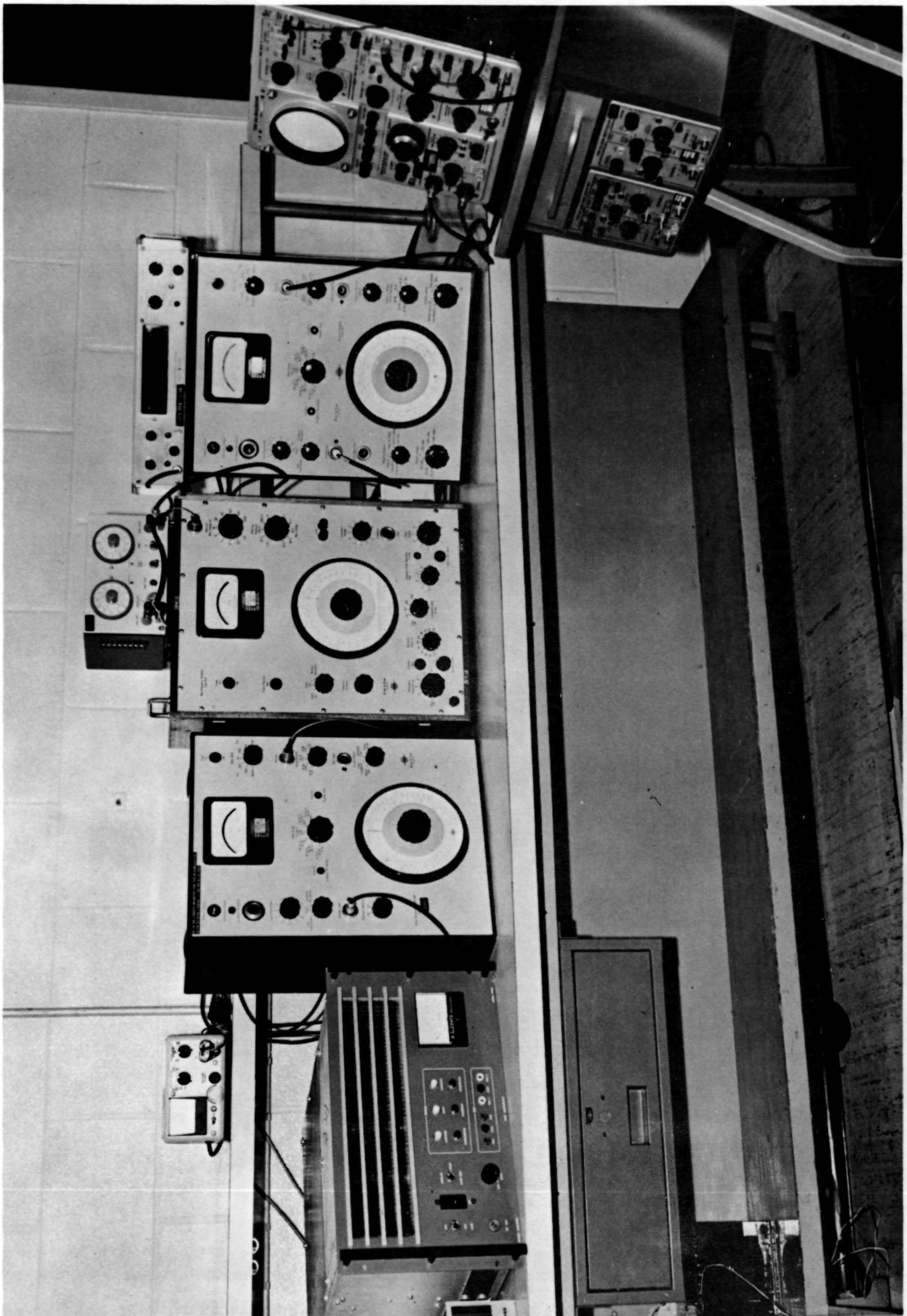


FIGURE 26. TONE - BURST INSTRUMENTATION

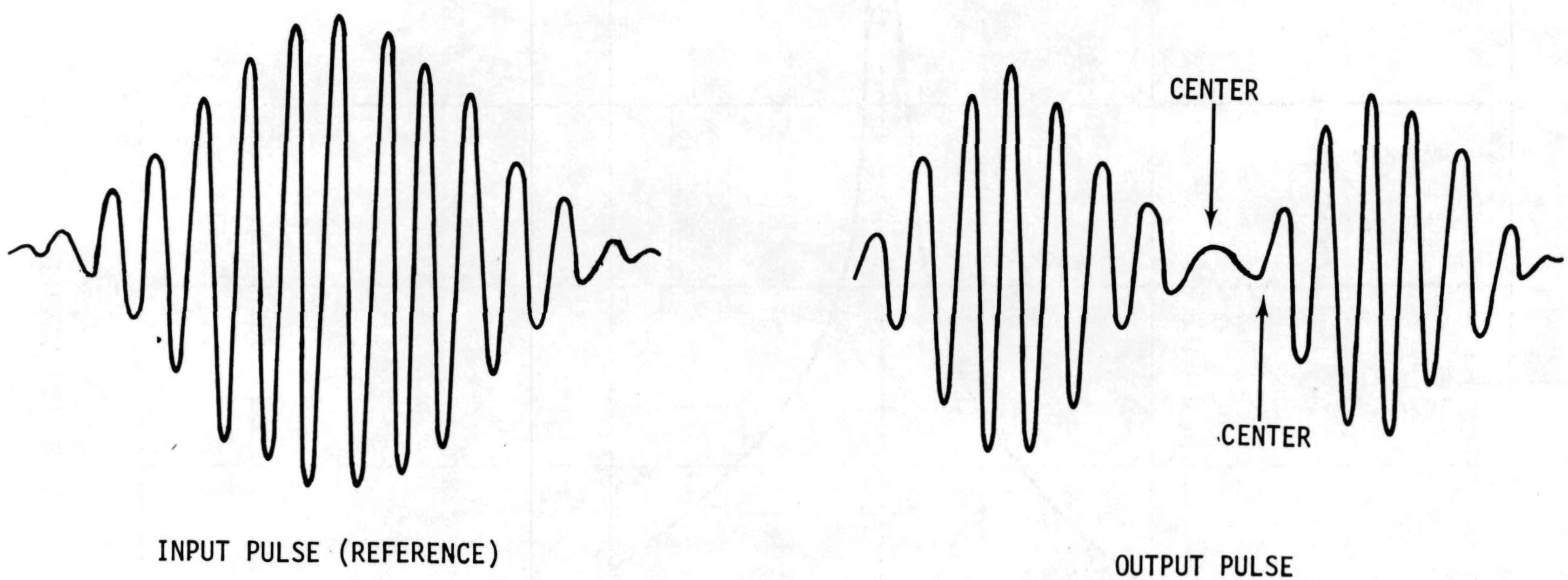


FIGURE 27. OBSERVED PULSE SHAPES

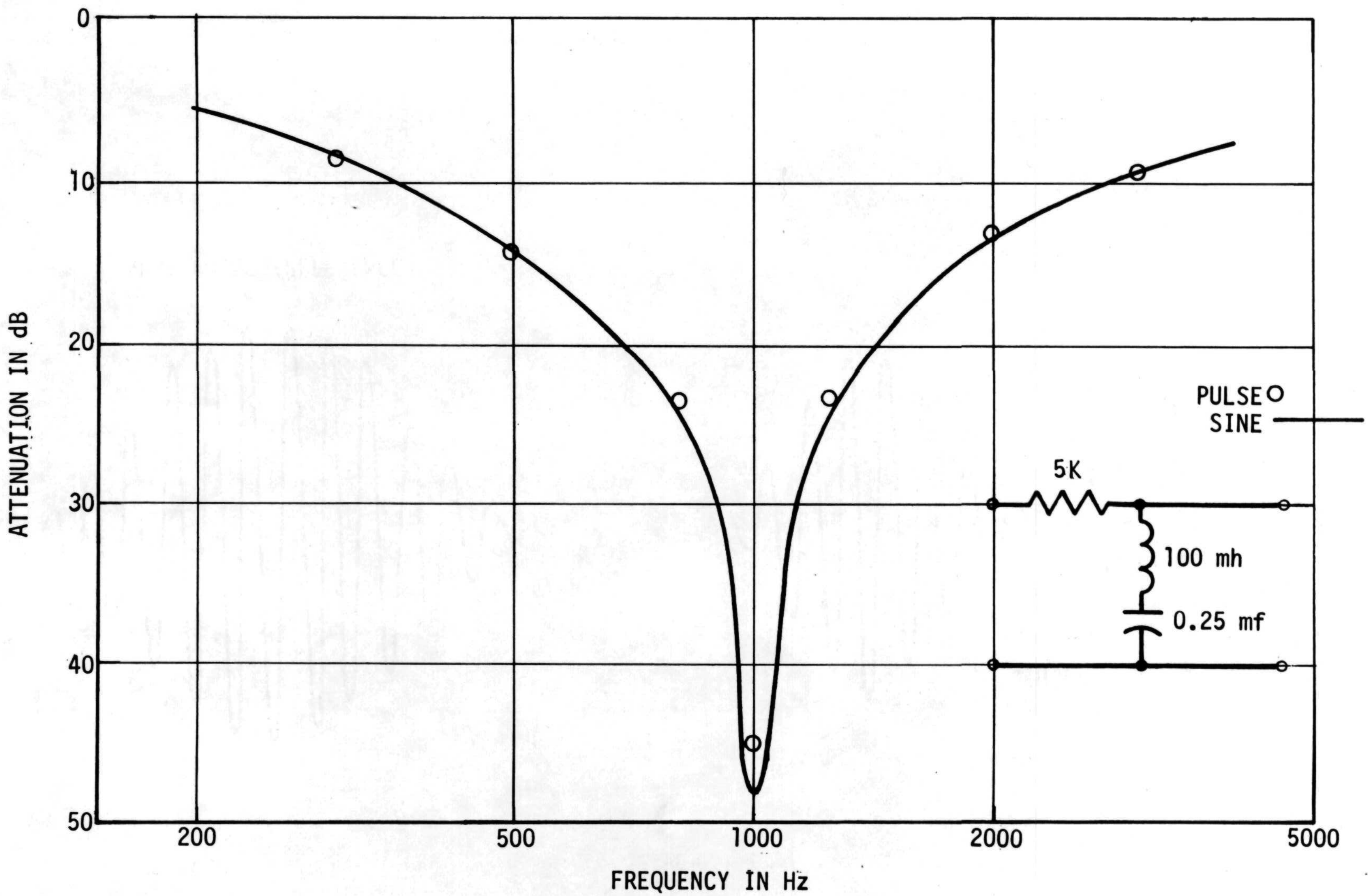
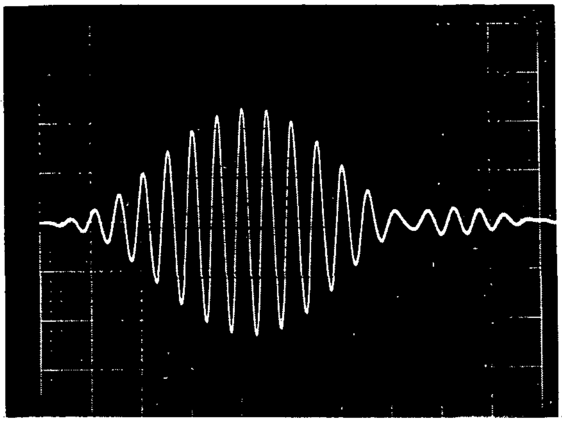
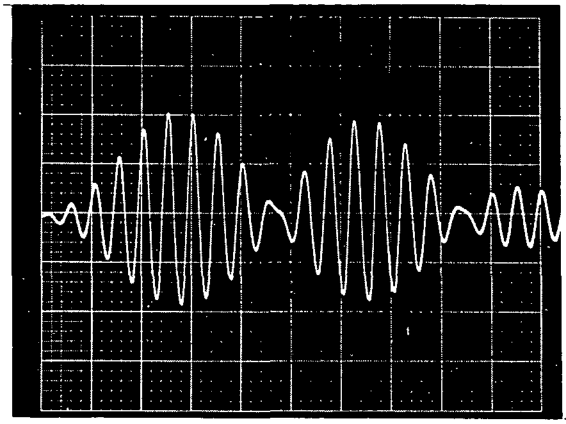


FIGURE 28. RESPONSE OF PRELIMINARY ANALOG ABSORBER

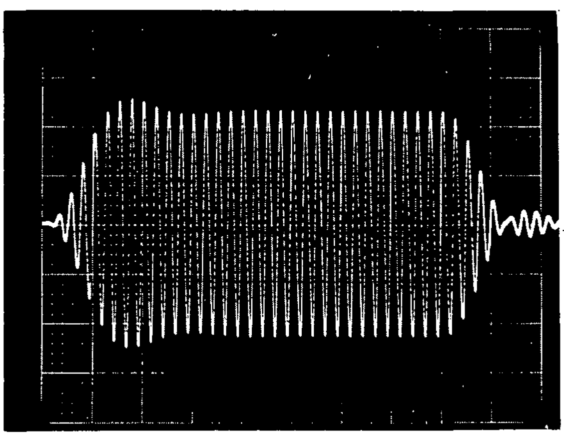


(1) 2000Hz REFERENCE 9.2 V P-P
FILTERED

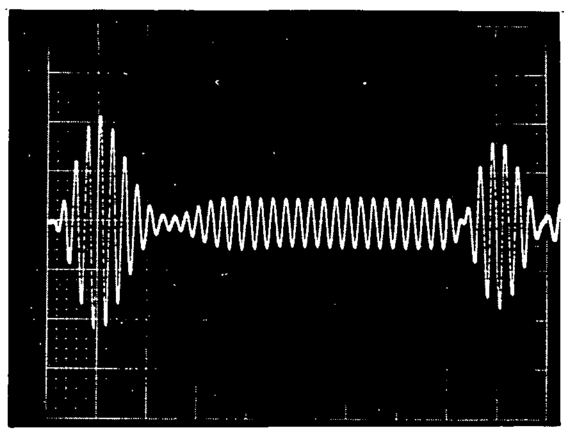


(2) 2000Hz OUTPUT 0.425 V P-P
FILTERED

LOSS = 26.7 dB



(3) 2000Hz REFERENCE 9.2 V P-P
UNFILTERED



(4) 2000Hz OUTPUT 0.51 V P-P
UNFILTERED

LOSS = 25.0 dB

FIGURE 29. PULSE RESPONSE OF NARROW BAND ANALOG

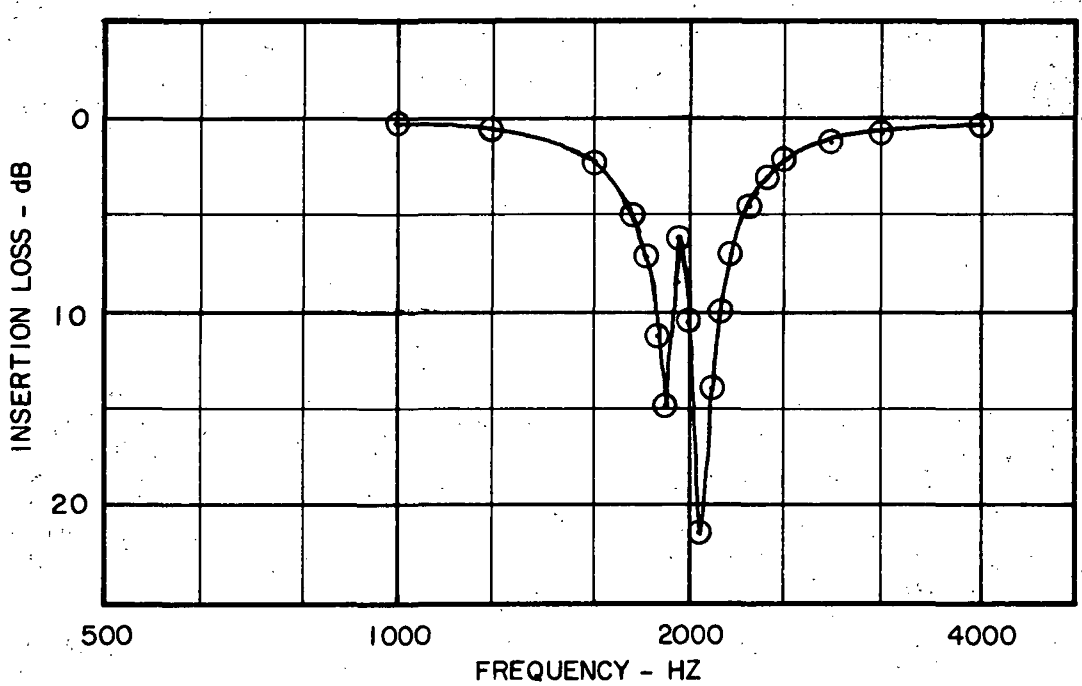
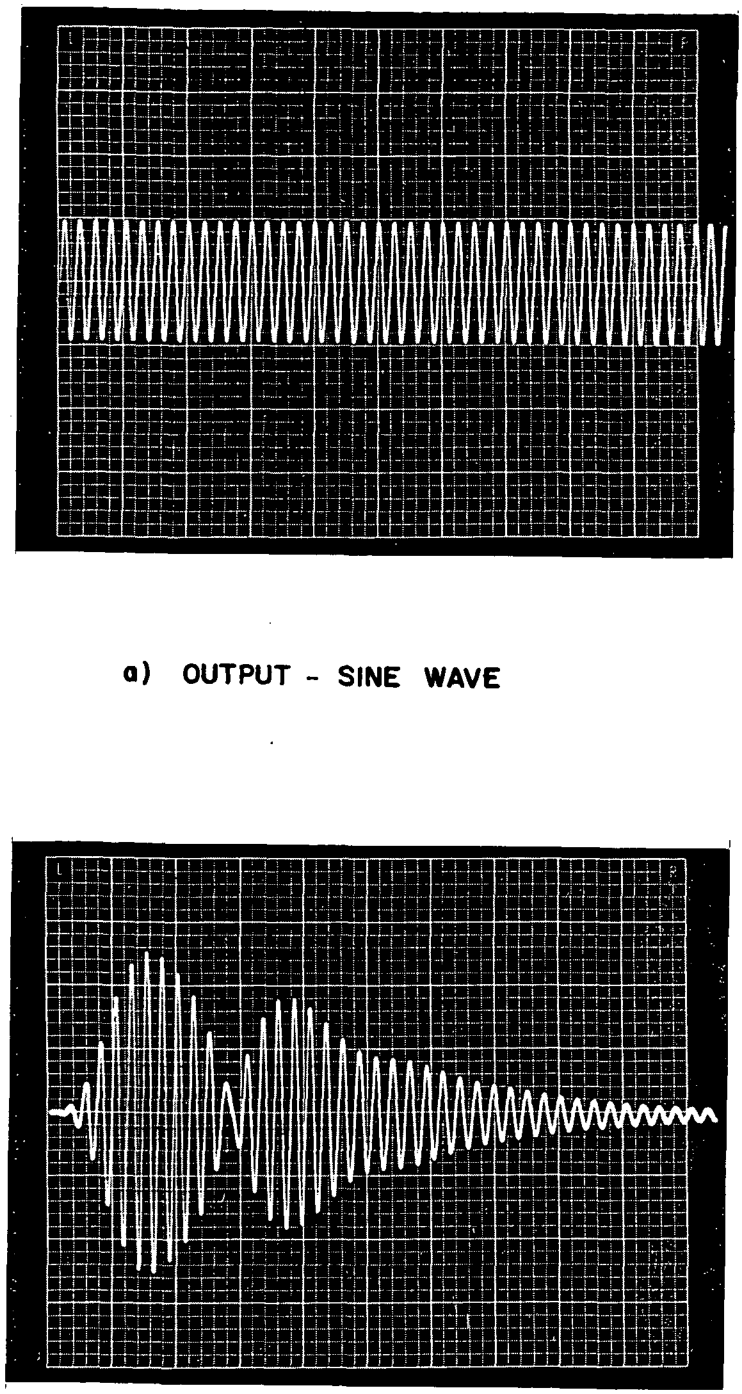


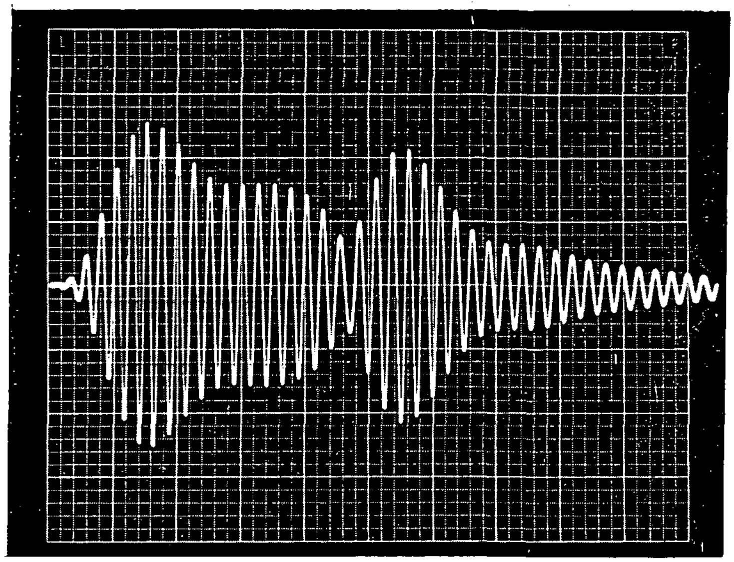
FIGURE 30. MULTIPLE-RESONANCE ANALOG NETWORK RESPONSE CHARACTERISTIC



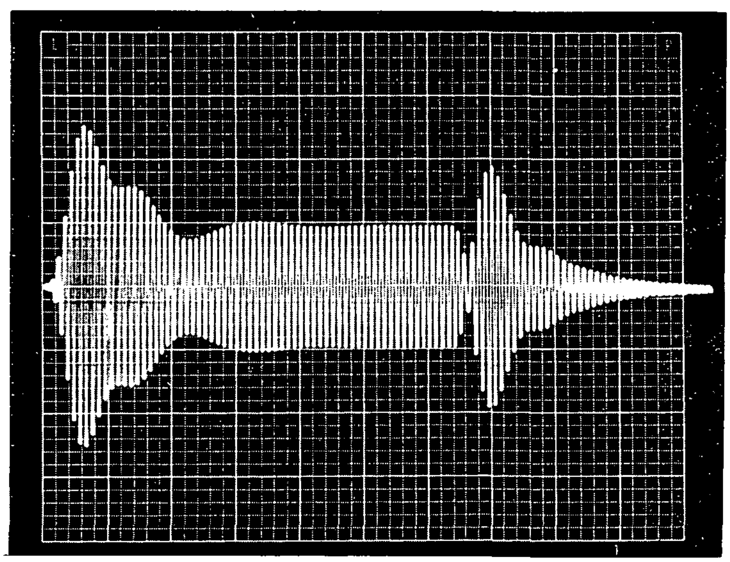
a) OUTPUT - SINE WAVE

b) OUTPUT - 8 HZ PULSE

FIGURE 31. MULTIPLE-RESONANCE ANALOG NETWORK PULSE RESPONSE

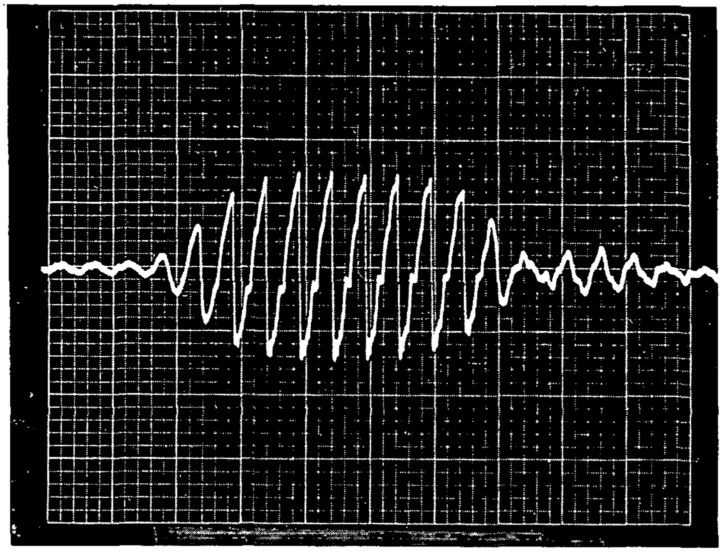


a) OUTPUT - 16 HZ PULSE

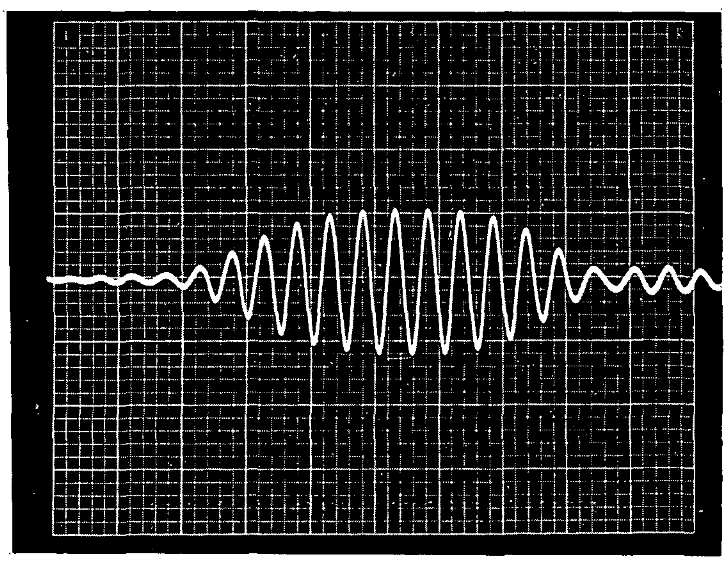


b) OUTPUT - 64 HZ PULSE

FIGURE 32. MULTIPLE-RESONANCE ANALOG NETWORK PULSE RESPONSE



a) 4000 HZ 4 FEET UNFILTERED



b) 4000 HZ 4 FEET FILTERED

FIGURE 33. DISTORTED TONE - BURSTS

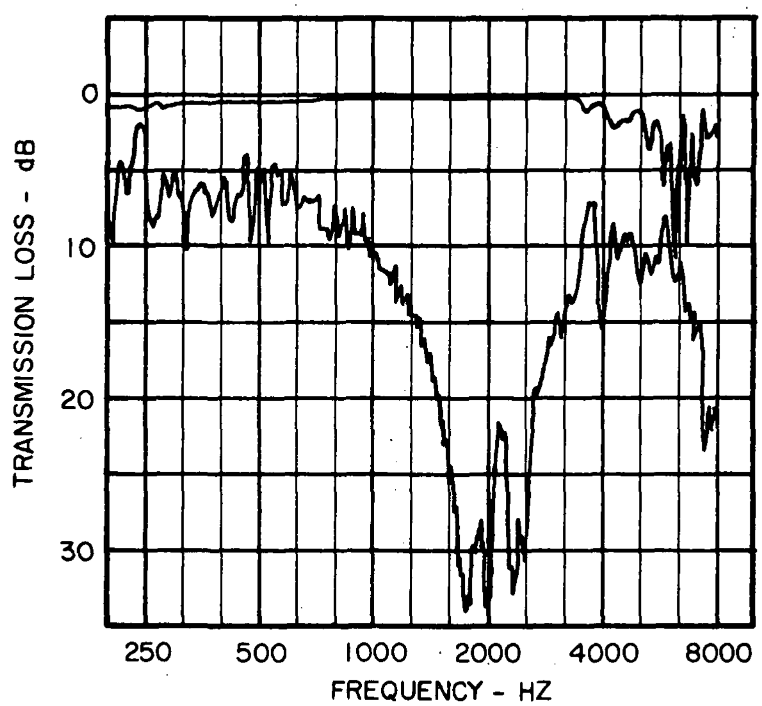
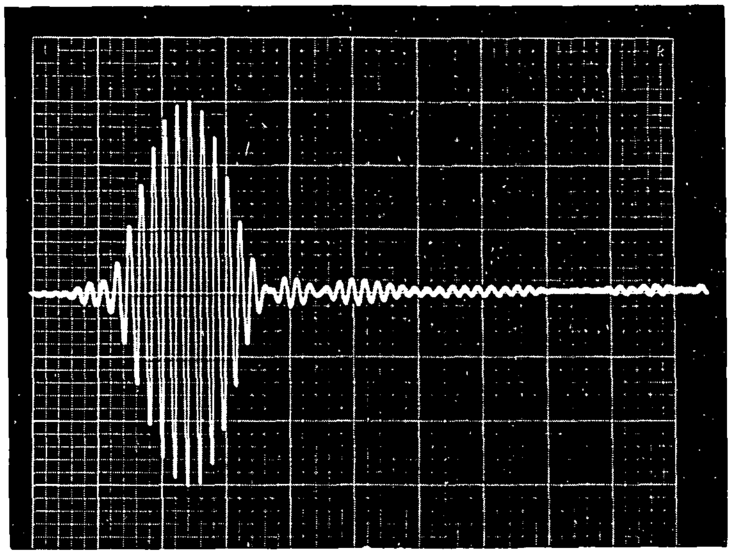
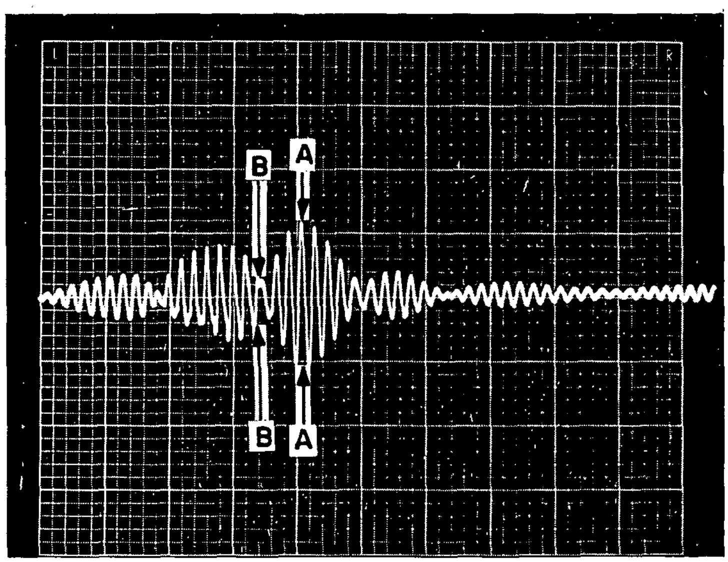


FIGURE 34. SWEPT - SINE TRANSMISSION LOSS OF RESONANT ABSORBER

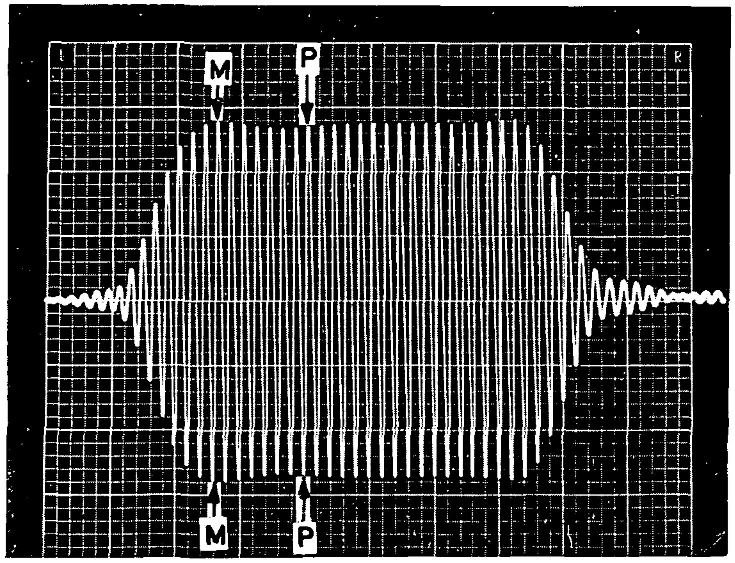


a) INPUT PULSE - 8 HZ

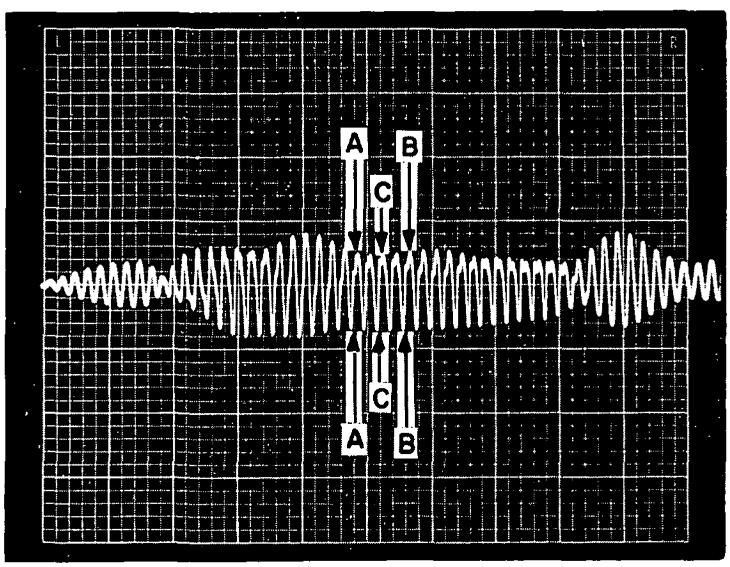


b) OUTPUT PULSE - 8 HZ

FIGURE 35 PULSE RESPONSE OF RESONANT ABSORBER



a) INPUT PULSE - 32 HZ



b) OUTPUT PULSE - 32 HZ

FIGURE 36 PULSE RESPONSE OF RESONANT ABSORBER

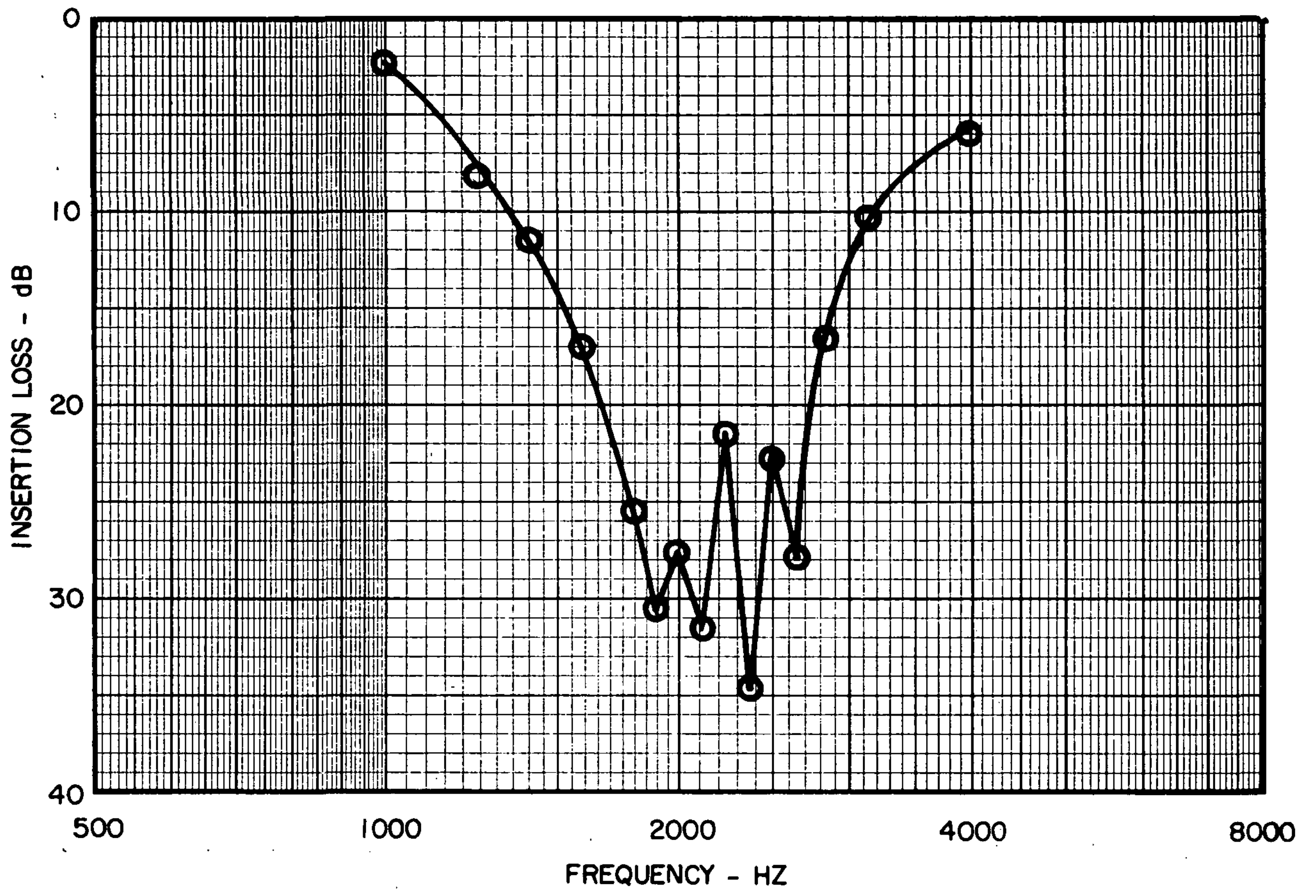


FIGURE 37. TONE-BURST INSERTION LOSS OF RESONANT ABSORBER

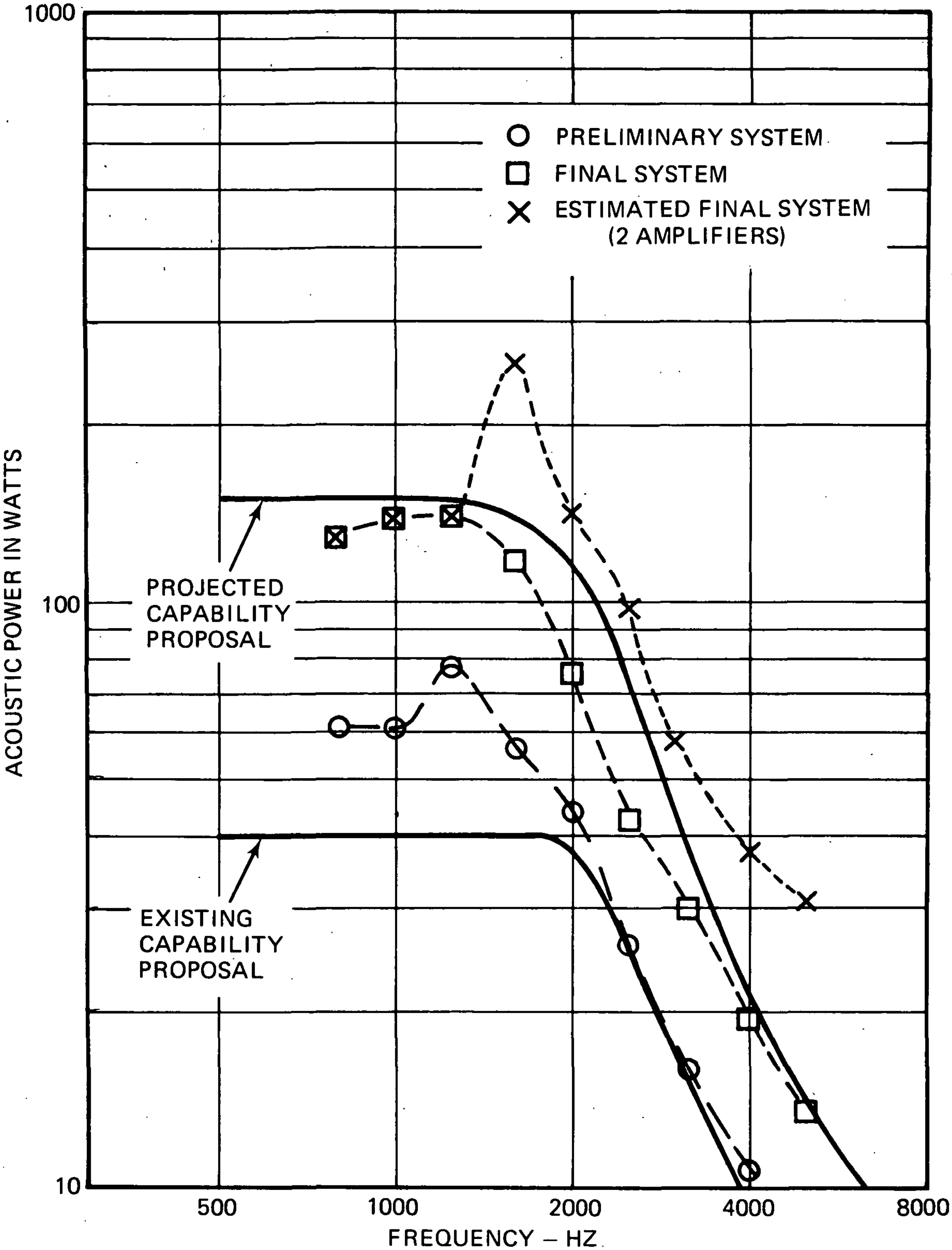
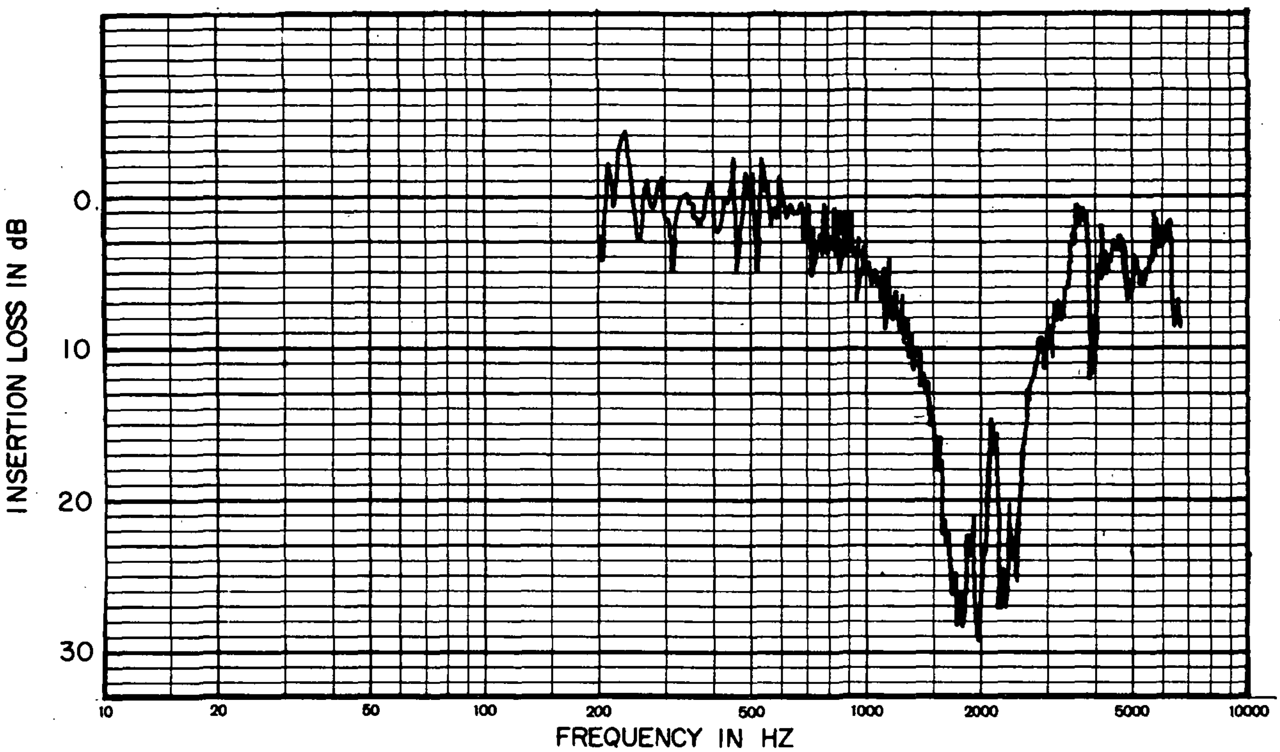
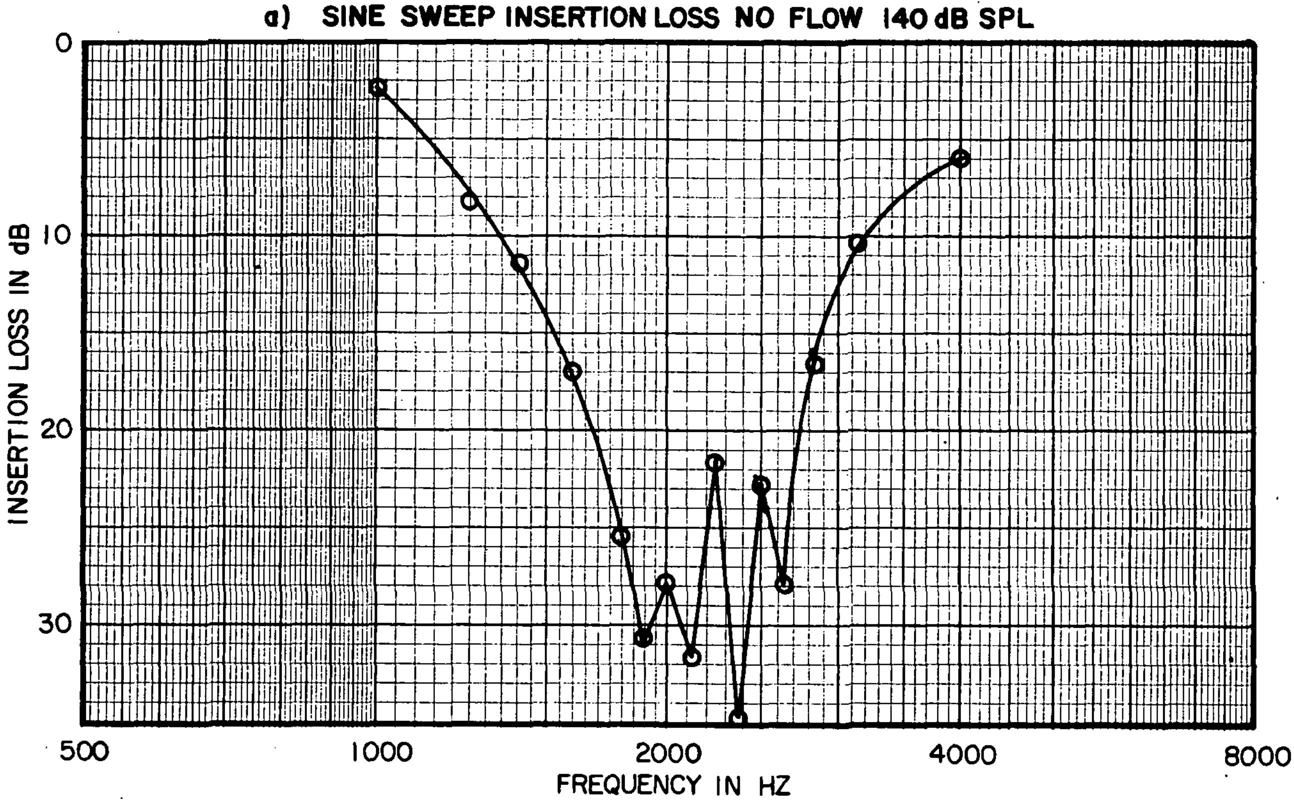


FIGURE 38 MAXIMUM ACOUSTIC POWER TONE-BURST SYSTEM

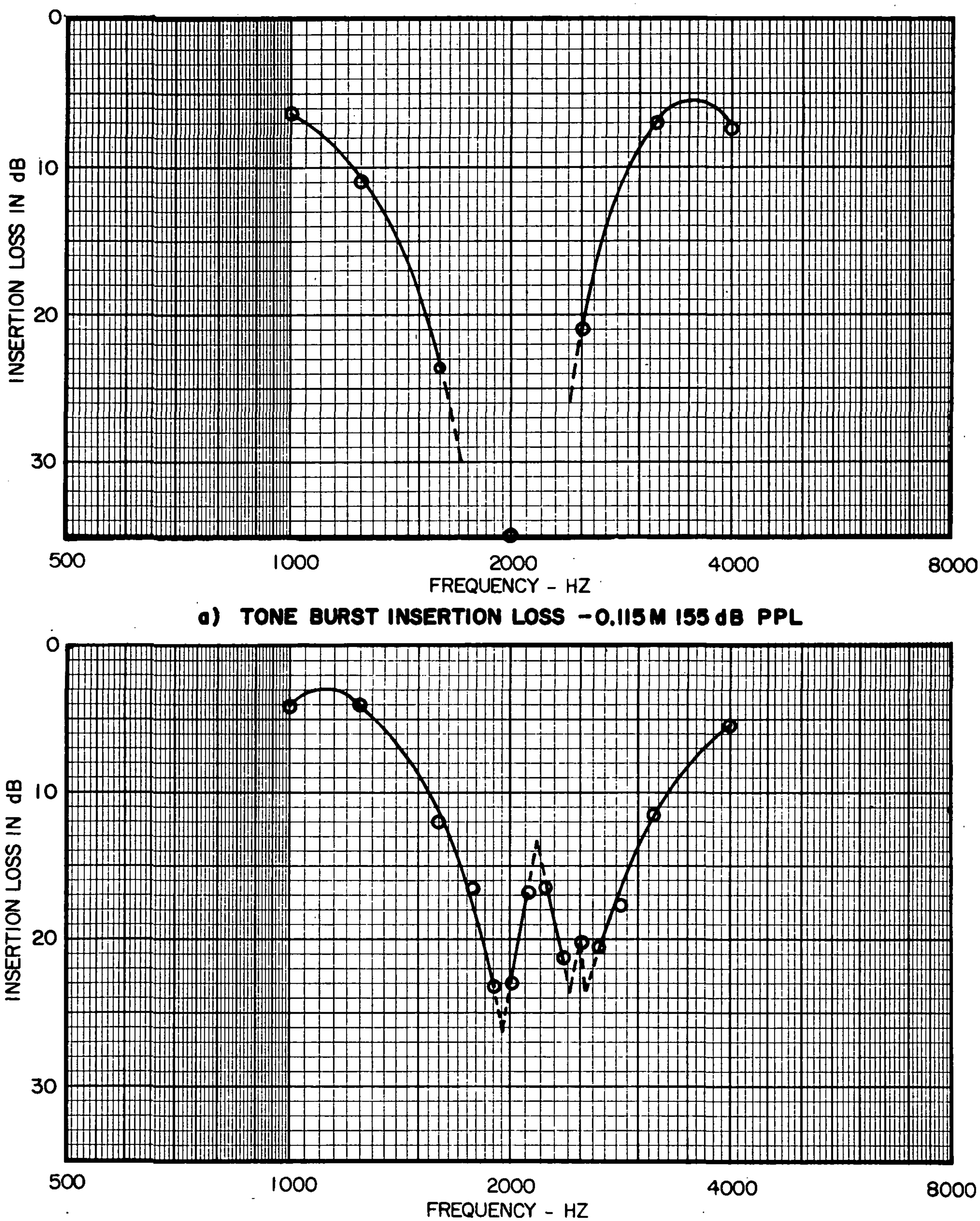


a) SINE SWEEP INSERTION LOSS NO FLOW 140 dB SPL



b) TONE BURST INSERTION LOSS NO FLOW 155 dB PPL

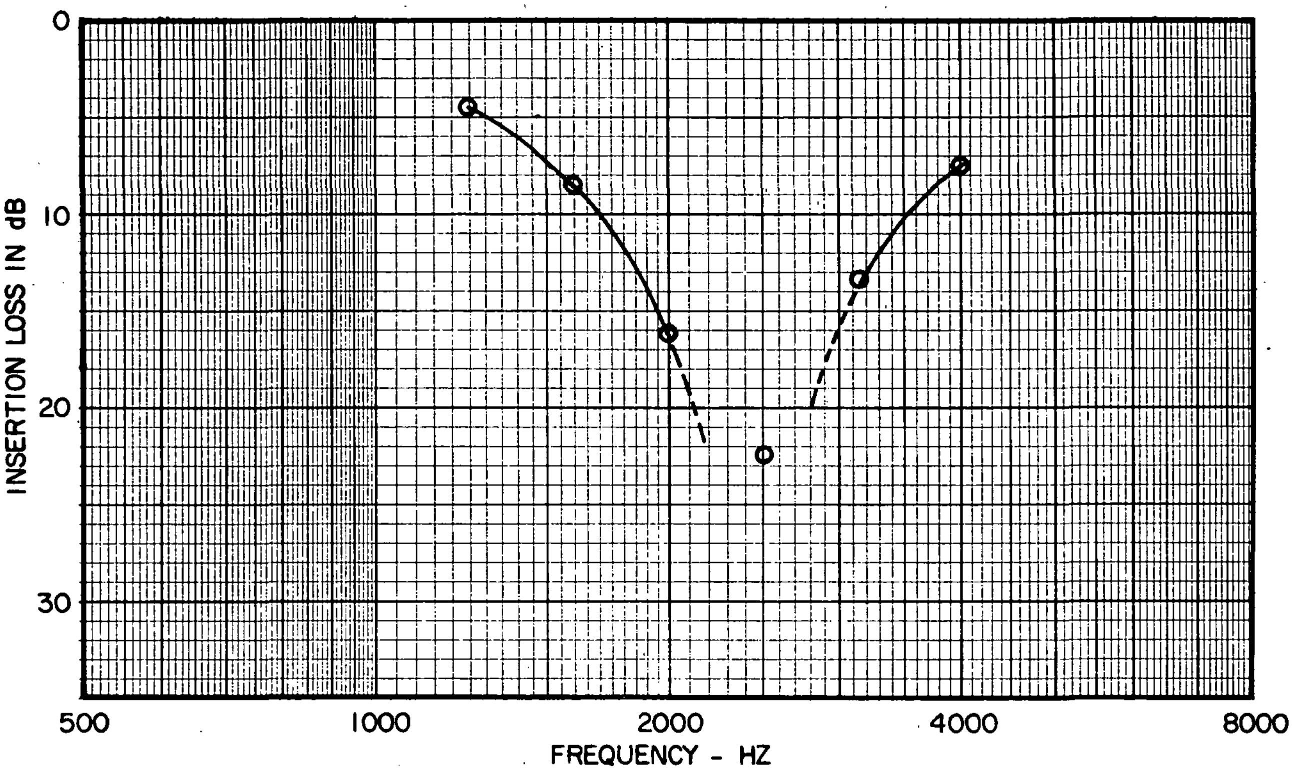
FIGURE 39. SAMPLE NO. I RESONANT ABSORBER



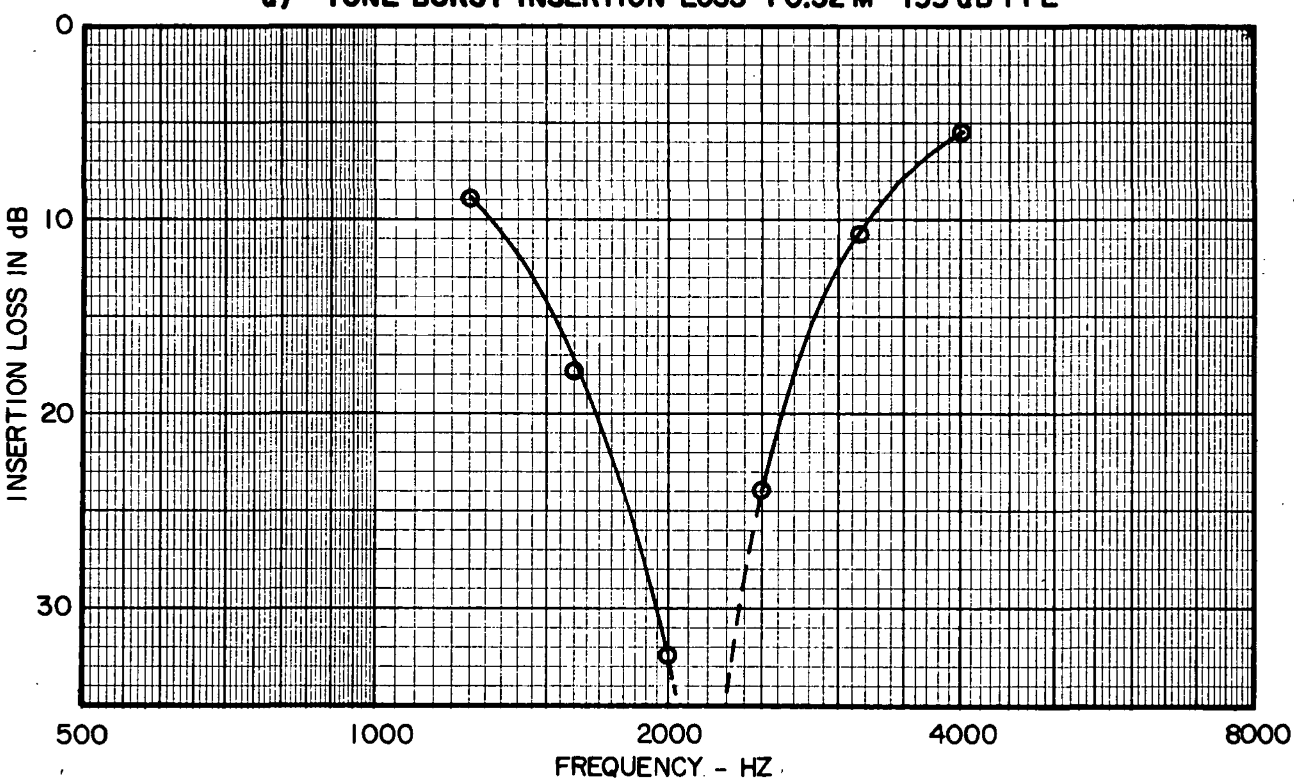
a) TONE BURST INSERTION LOSS -0.115M 155 dB PPL

b) TONE BURST INSERTION LOSS +0.115M 155 dB PPL

FIGURE 40. SAMPLE NO. I RESONANT ABSORBER

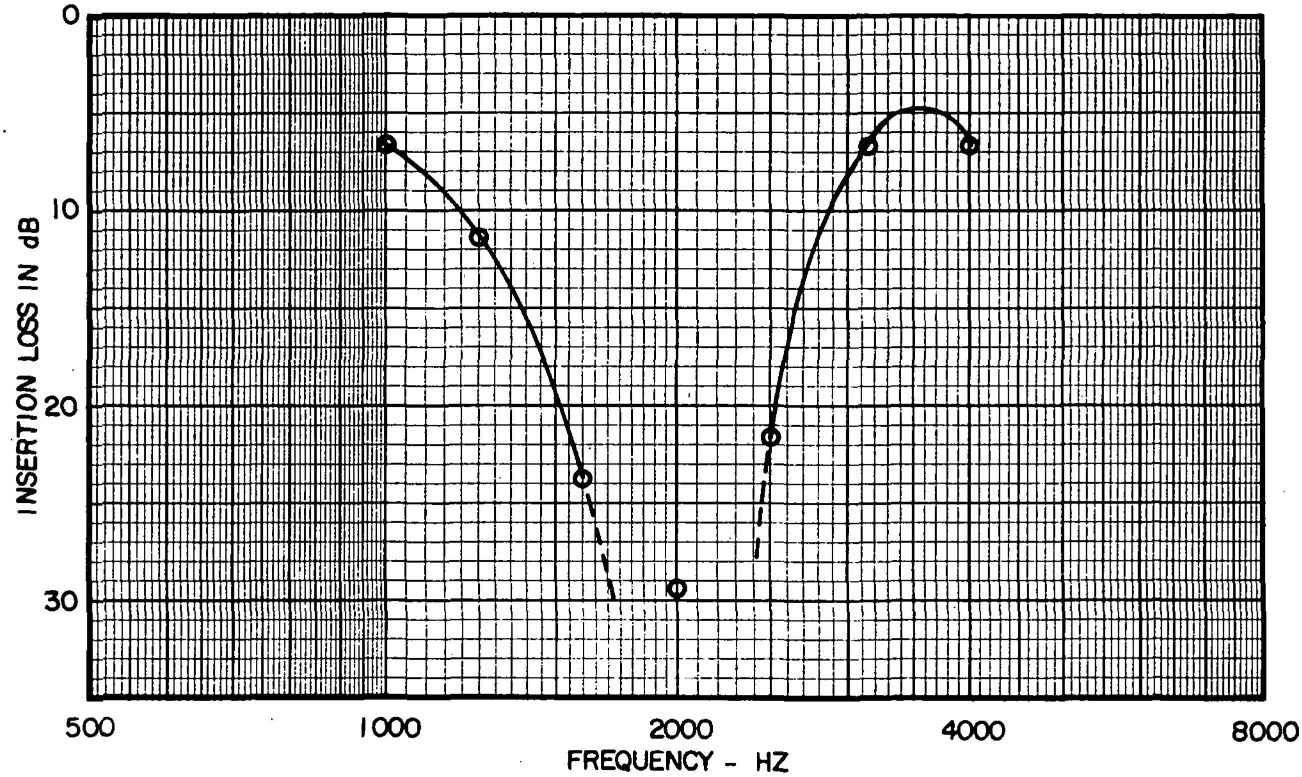


a) TONE BURST INSERTION LOSS +0.32 M 155 dB PPL

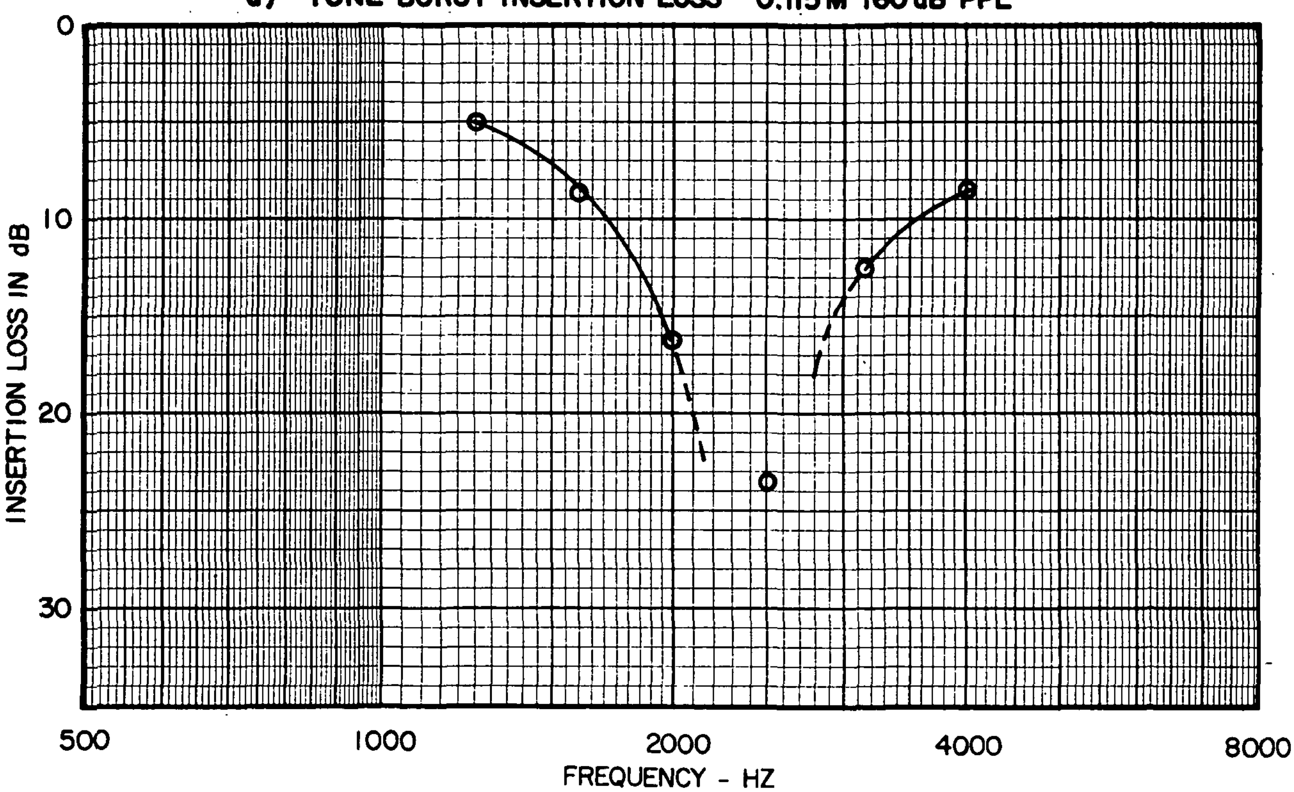


b) TONE BURST INSERTION LOSS NO FLOW 160 dB PPL

FIGURE 41. SAMPLE NO. 1 RESONANT ABSORBER

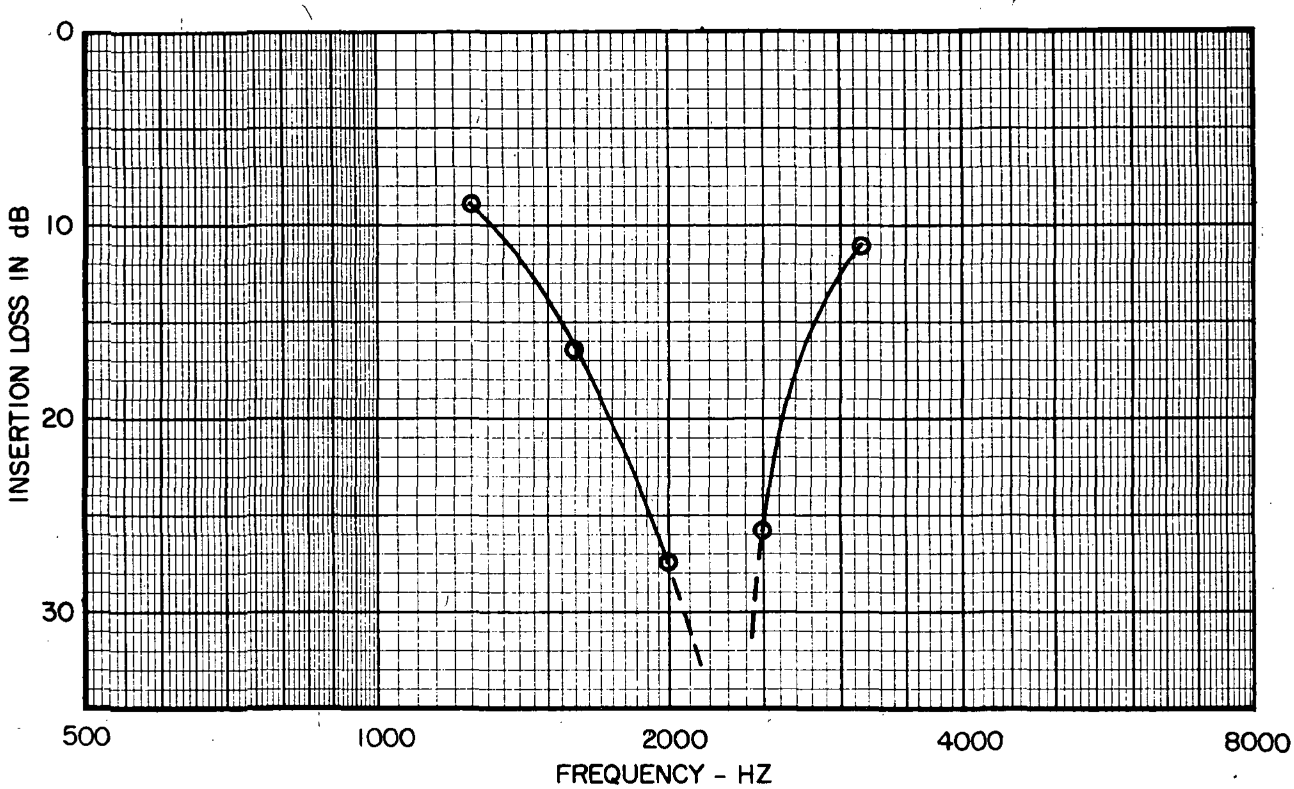


a) TONE BURST INSERTION LOSS -0.115 M 160 dB PPL

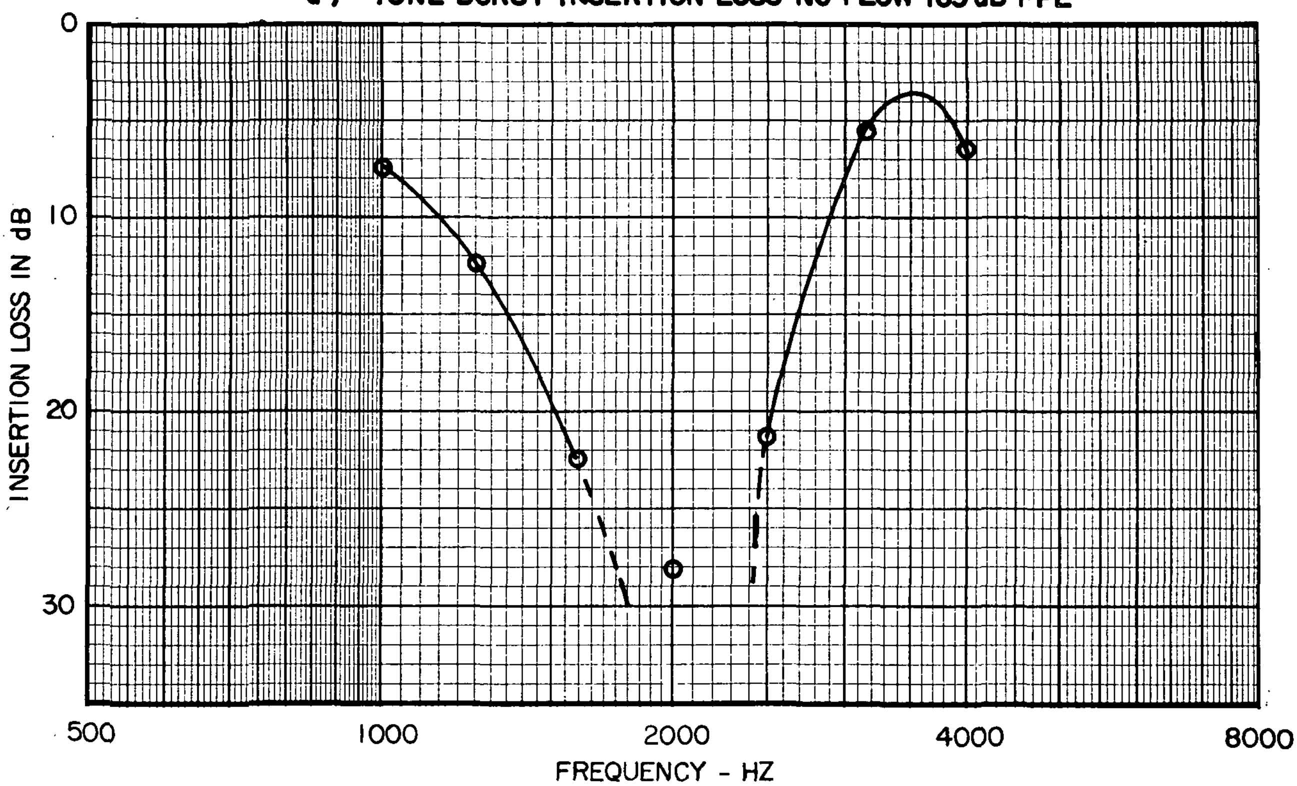


b) TONE BURST INSERTION LOSS +0.32 M 160 dB PPL

FIGURE 42. SAMPLE NO. 1 RESONANT ABSORBER

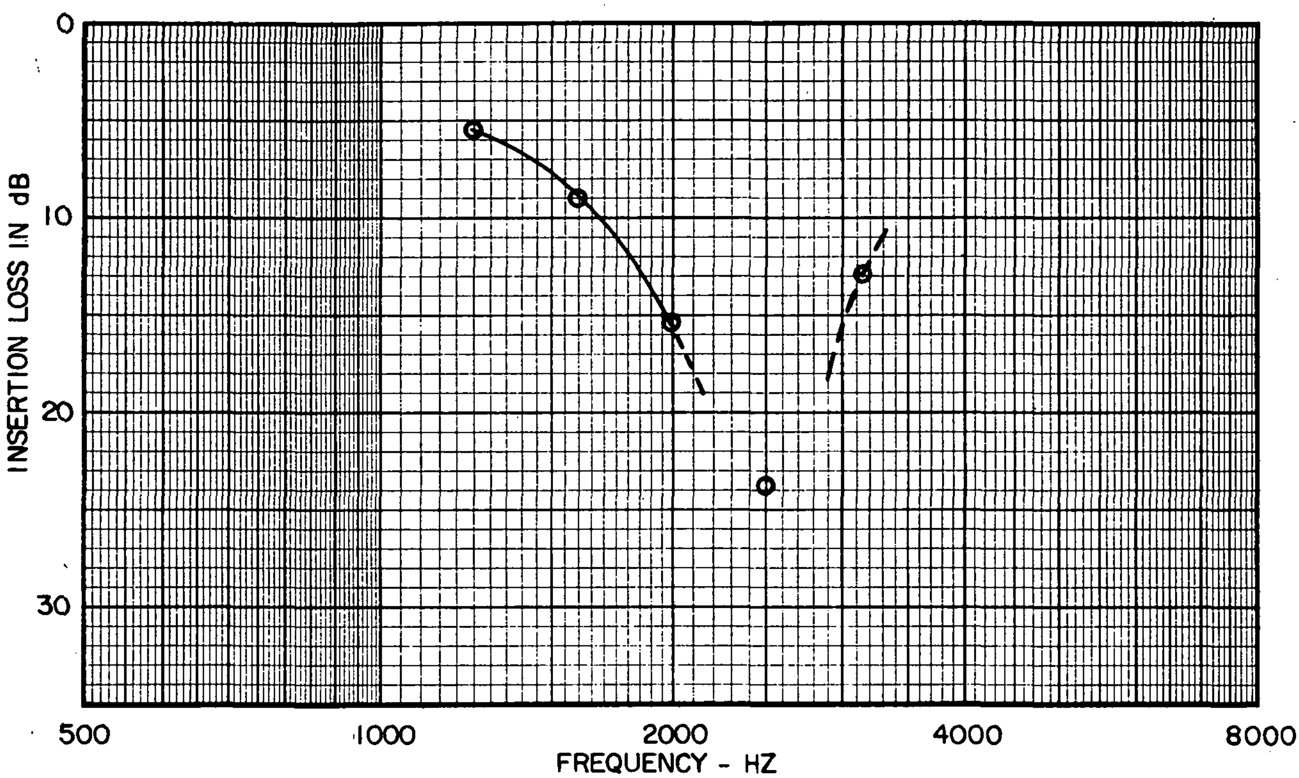


a) TONE BURST INSERTION LOSS NO FLOW 165 dB PPL



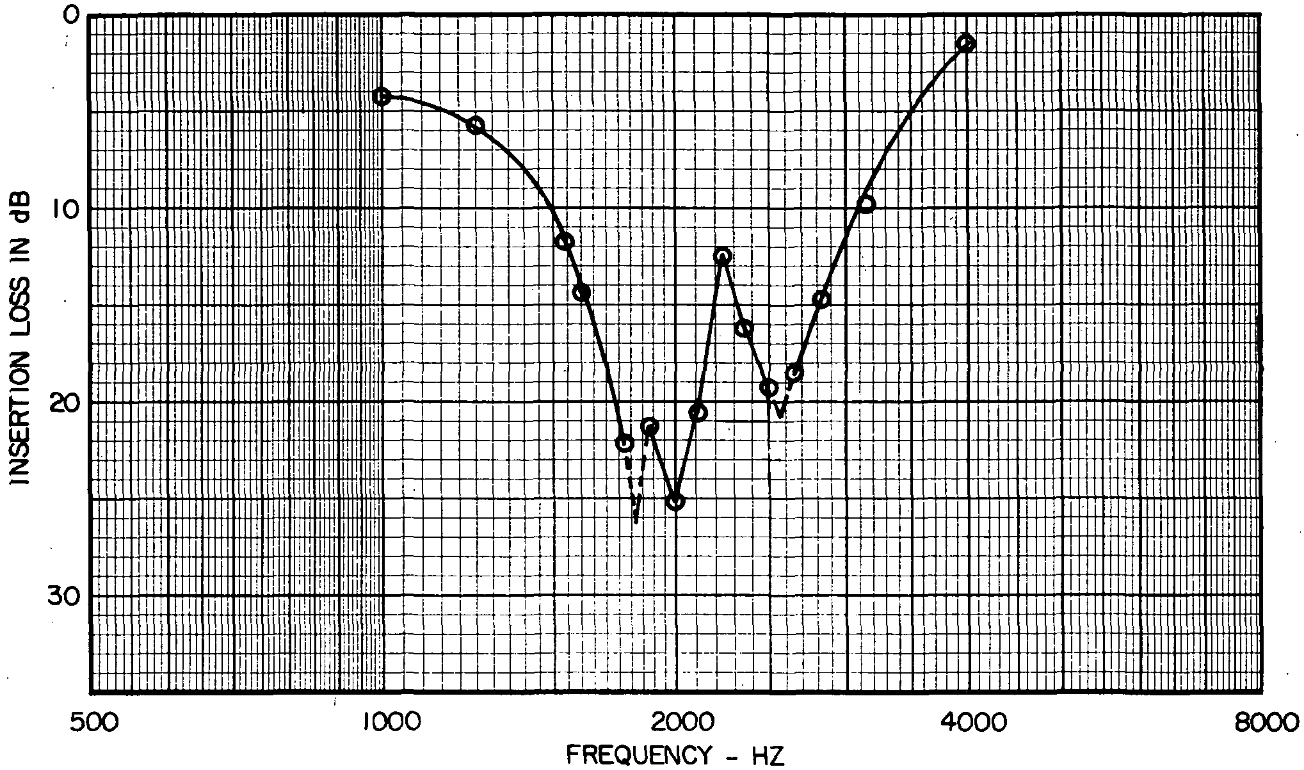
b) TONE BURST INSERTION LOSS −0.115 M 165 dB PPL

FIGURE 43. SAMPLE NO. 1 RESONANT ABSORBER

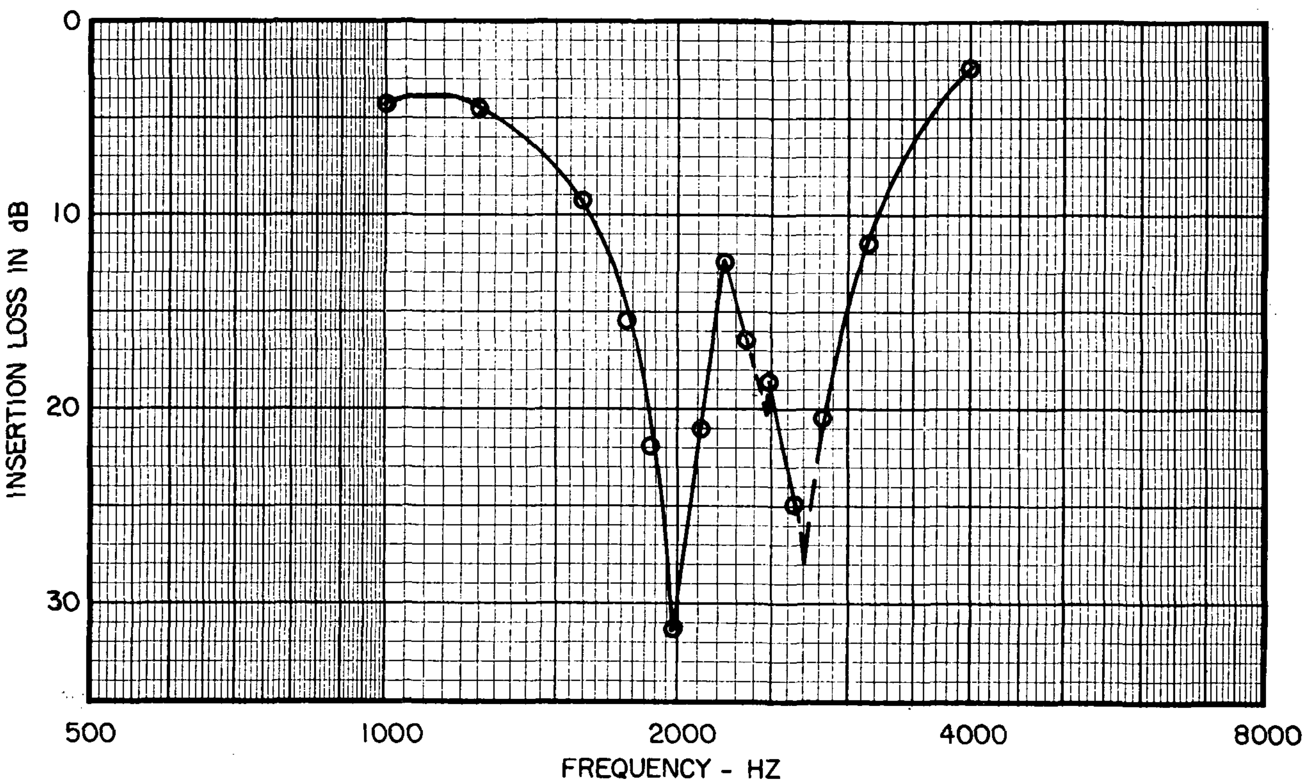


TONE BURST INSERTION LOSS + 0.32 M 165 dB PPL

FIGURE 44. SAMPLE NO. 1 RESONANT ABSORBER

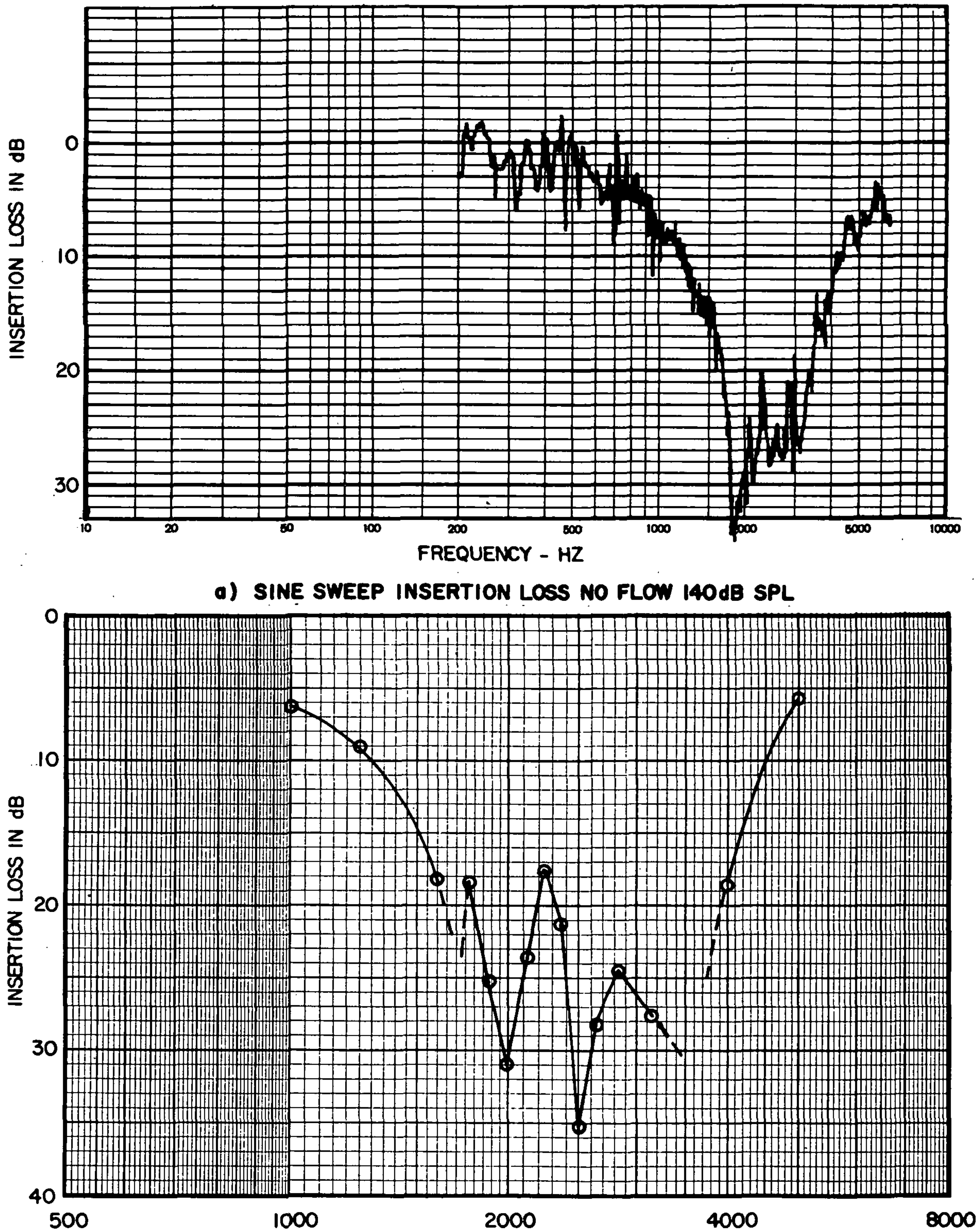


a) TONE BURST INSERTION LOSS NO FLOW 155 dB PPL



b) TONE BURST INSERTION LOSS +0.115 M 155 dB PPL

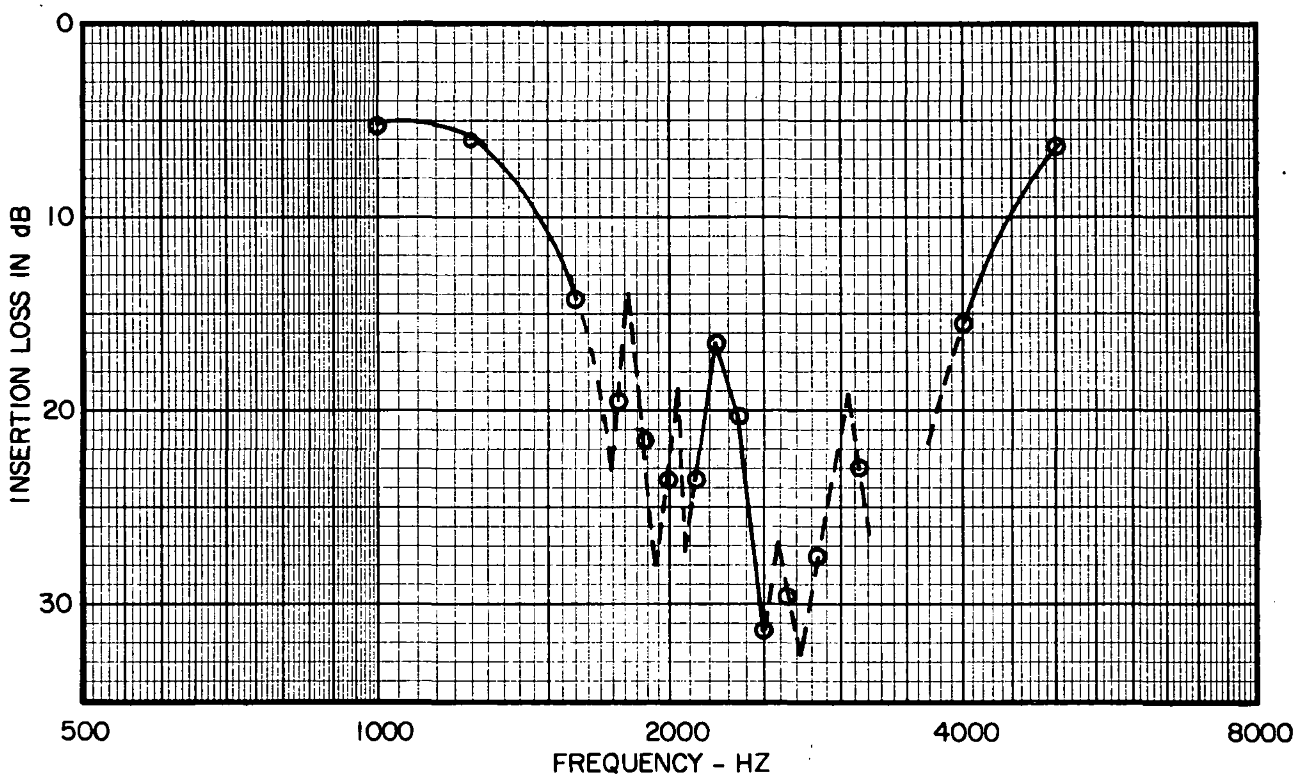
FIGURE 45. SAMPLE NO. 2 RESONANT ABSORBER



a) SINE SWEEP INSERTION LOSS NO FLOW 140dB SPL

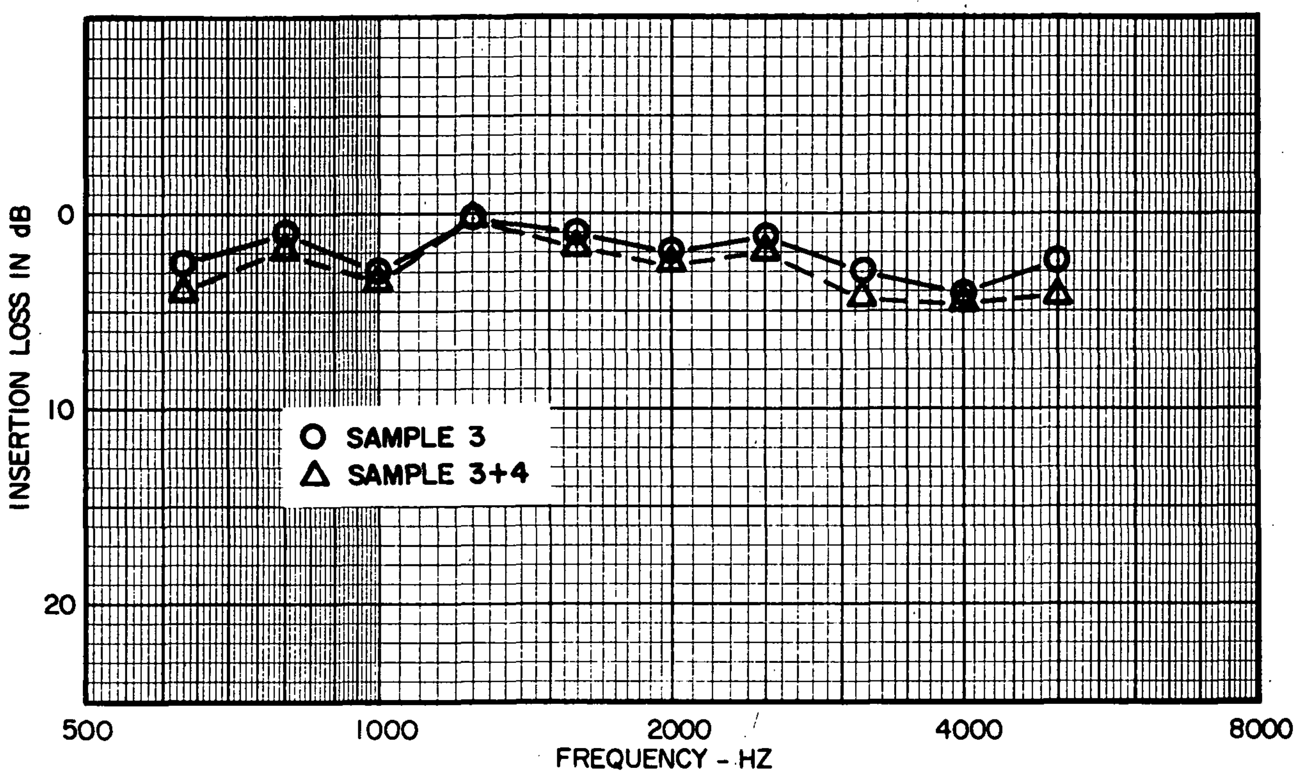
b) TONE BURST INSERTION LOSS NO FLOW 155dB PPL

FIGURE 46. SAMPLE NO. 1 AND NO. 2 RESONANT ABSORBER

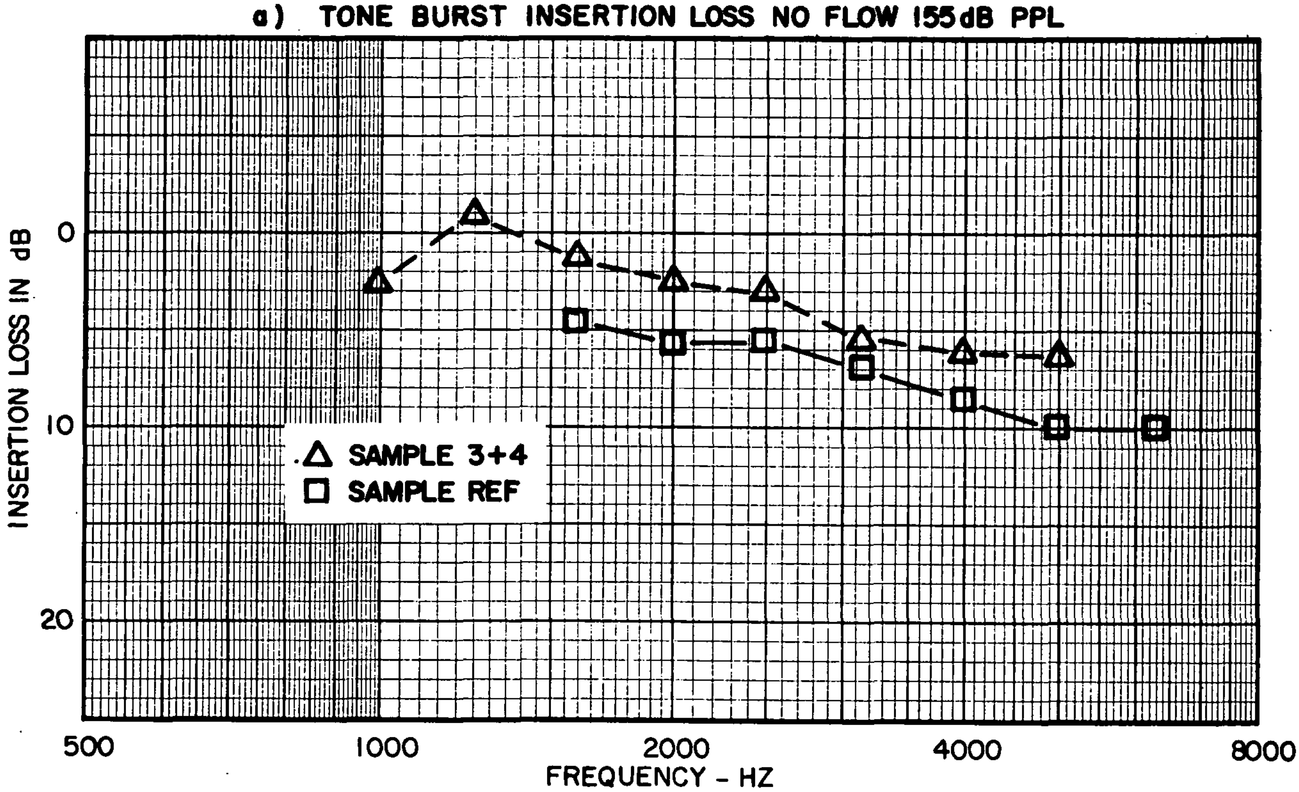


TONE BURST INSERTION LOSS +0.115M 155dB PPL

FIGURE 47. SAMPLE NO. 1 AND NO. 2 RESONANT ABSORBER



a) TONE BURST INSERTION LOSS NO FLOW 155dB PPL



b) TONE BURST INSERTION LOSS +0.115M 155dB PPL

FIGURE 48. SAMPLE NO. 3 AND NO. 4 BROADBAND ABSORBER

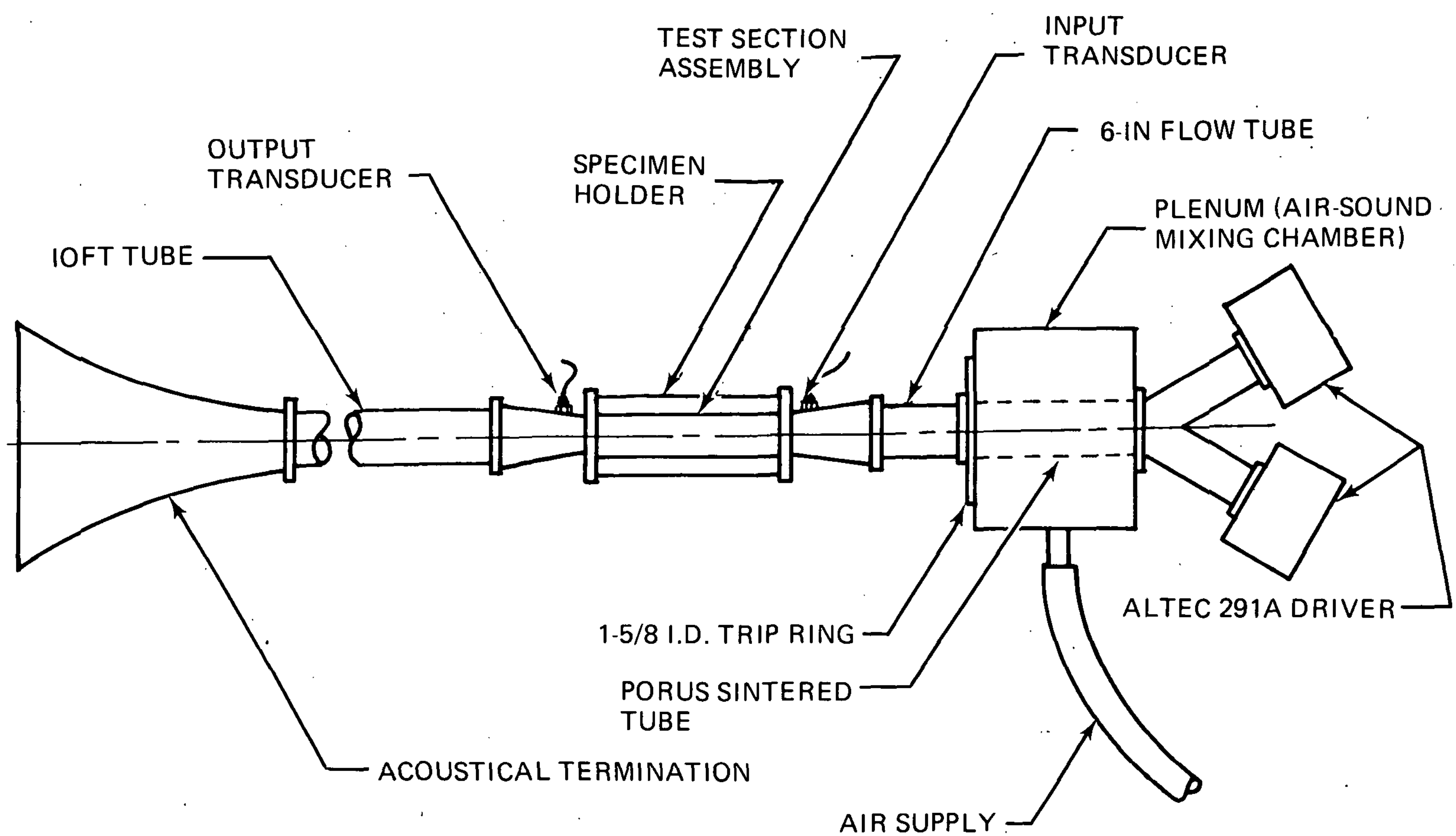


FIGURE 49 TONE BURST FLOW TUBE SYSTEM

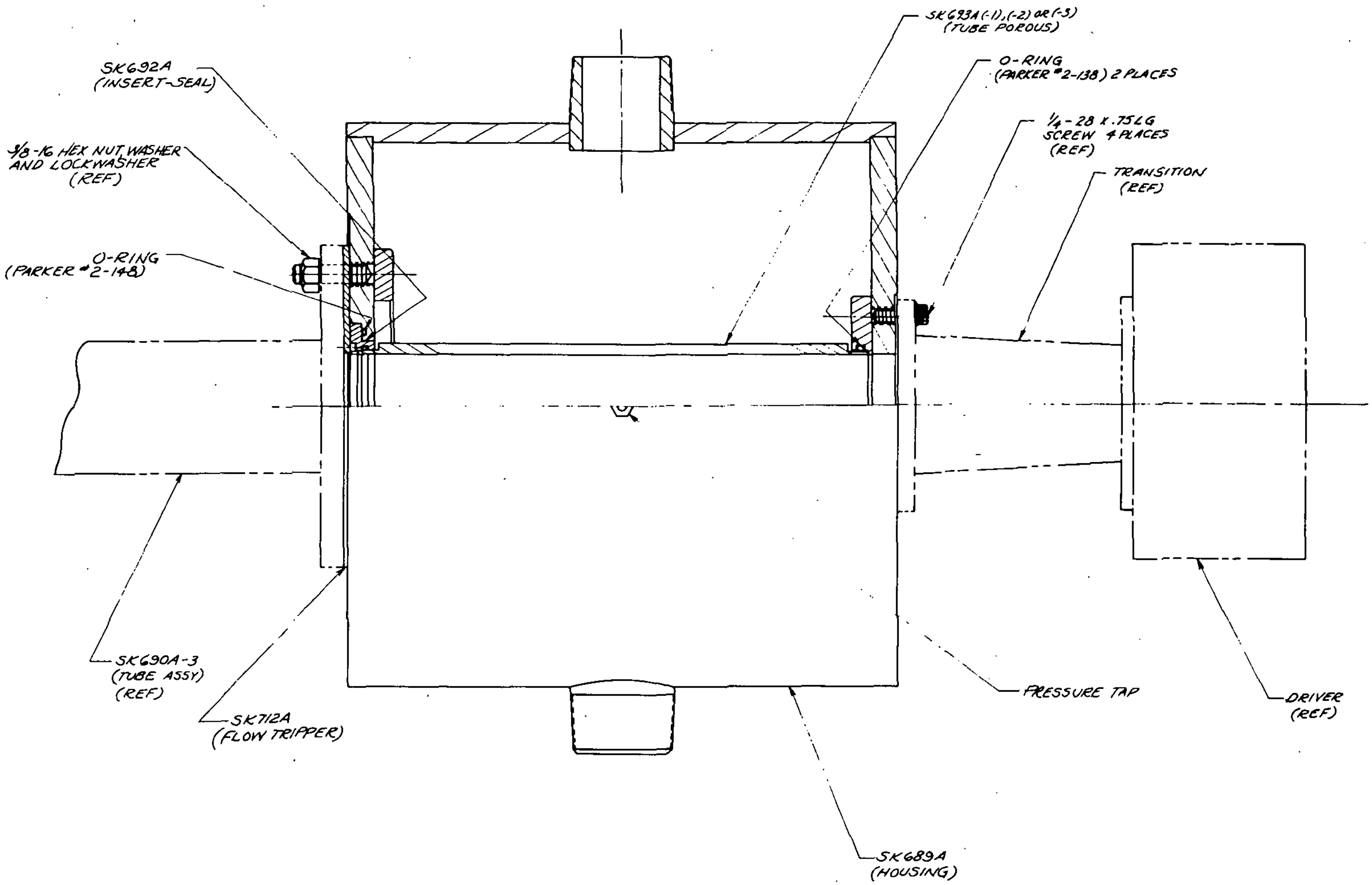
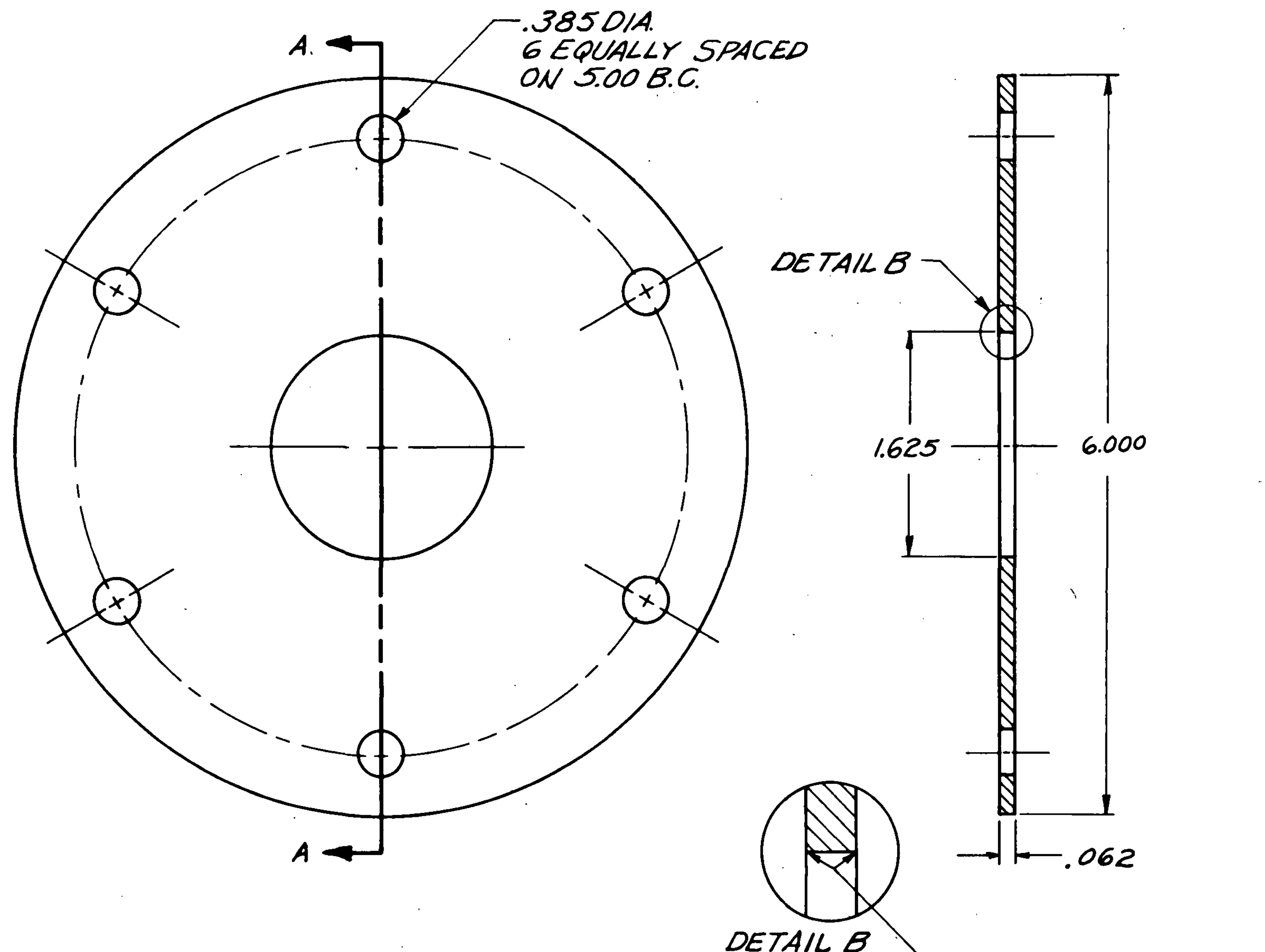
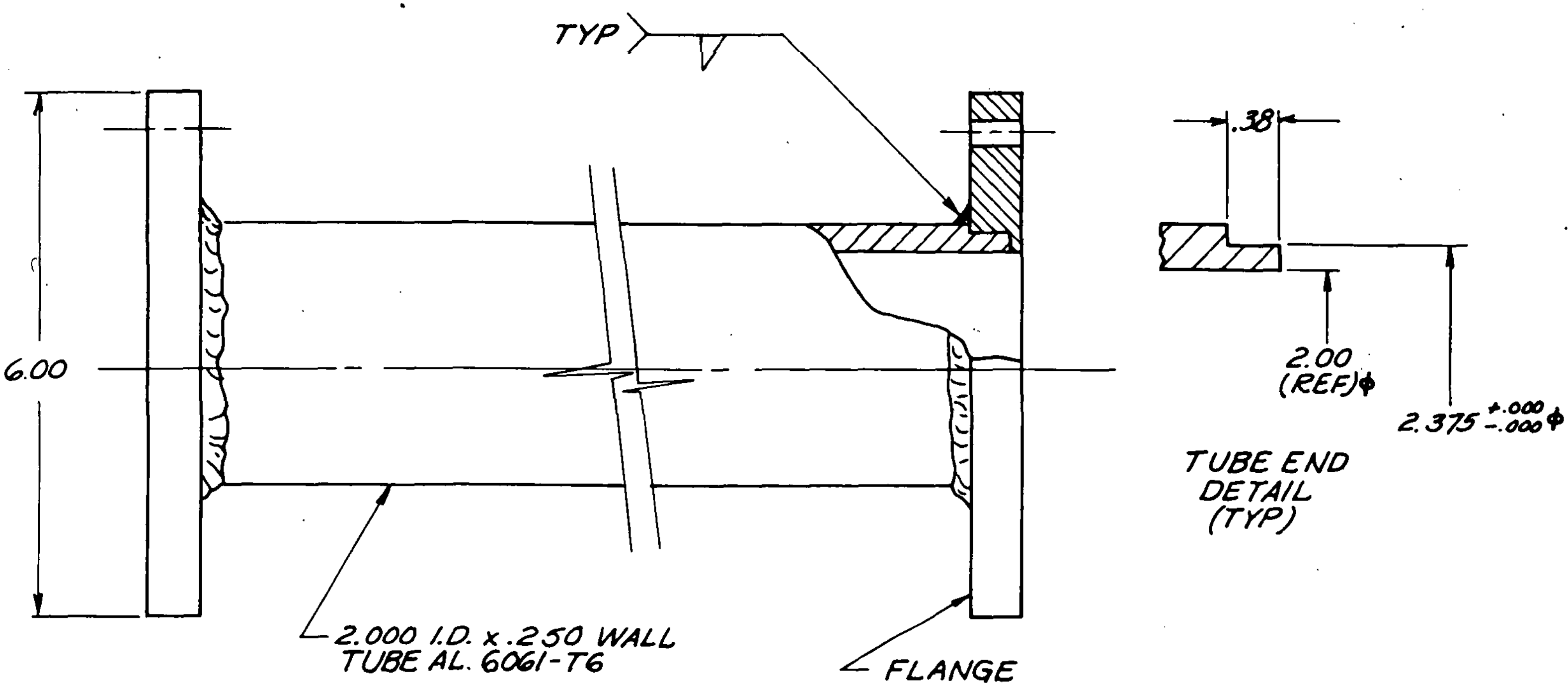


FIGURE 50 DWG. NO. D-11517 AIR SOUND INTERMIXING SECTION ASSY



MATERIAL: ALUMINUM 6061-T6 OR EQUAL

FIGURE 51 DWG. NO. SK712A AIR FLOW TRIP RING

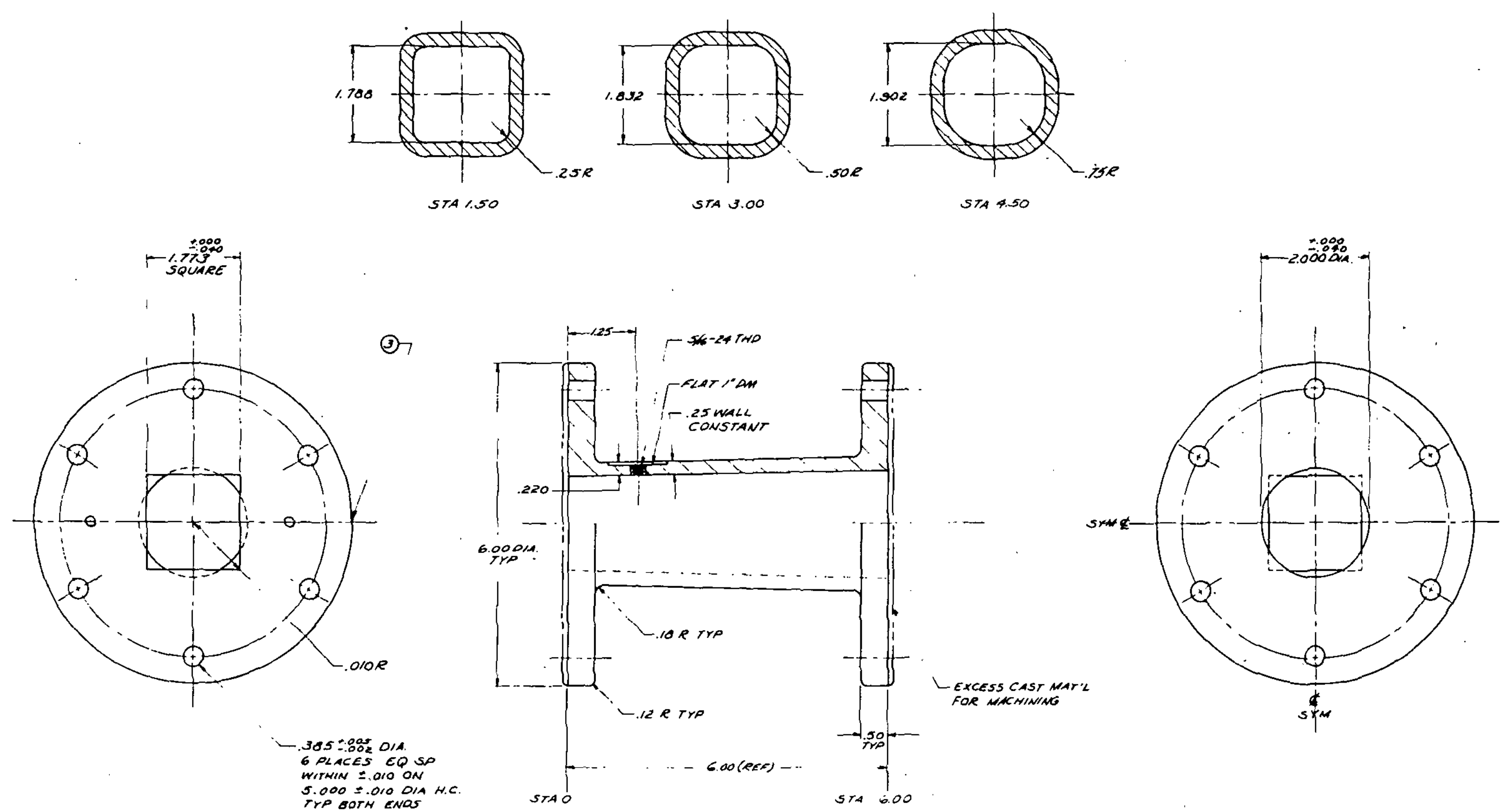


DASH NO.	L
-01	120"
-03	6"

NOTES: 1. WELD AS SHOWN

P.N. 08700

FIGURE 52 DWG. NO. SK690A TUBE ASSY



TRANSITION FROM SQUARE END OF TRANSITION INTO TEST SECTION MUST BE SMOOTH AND FREE OF STEPS. MATE & MACHINE EACH END.

(3) AFTER EACH TRANSITION IS MATCHED TO THE TEST SECTION STAMP THE LETTER A, OR B, ON EACH MATING FLANGE, AND INSERT DOWEL PINS FOR CORRECT POSITIONING WHEN REASSEMBLED

2. TRANSITION BLEND TO BE SMOOTH & CONTINUOUS WITH NO ABRUPT CHANGE IN DIRECTION OF MORE THAN .010

1. CAST SURFACE 125/

FIGURE 53 DWG. NO. D11460 ROUND TO SQUARE TRANSITION

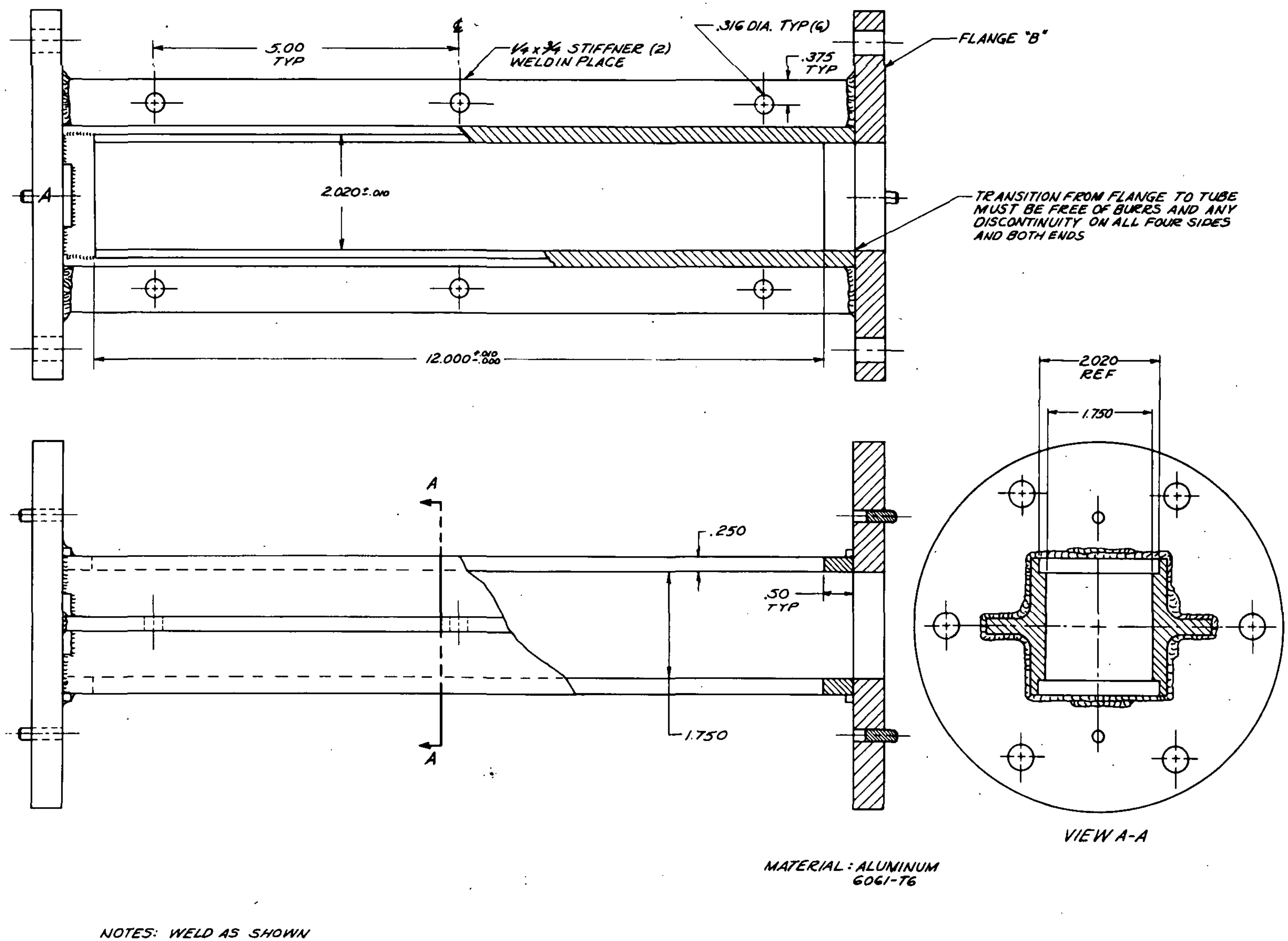
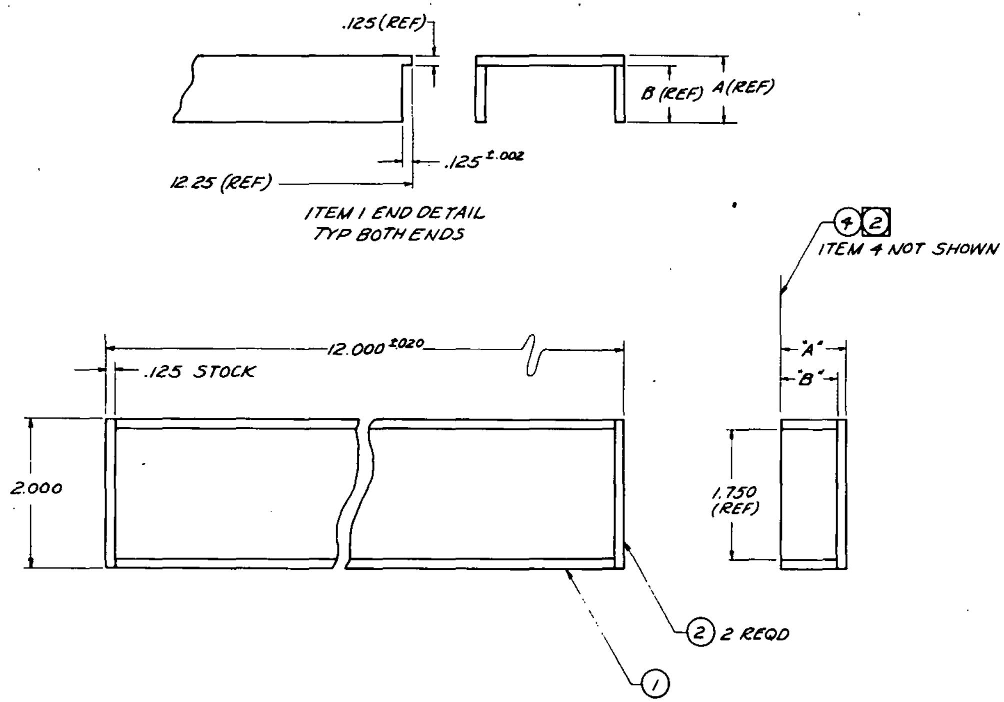


FIGURE 54 DWG. NO. C-11526 TEST SECTION



PART NO	"A"	"B" ±.002
-1	.275	.150
-2	.775	.650
-3	1.275	1.150
-4		
-5		

QTY REQD	PART NO.	DESCRIPTION	SPECIFICATION	ITEM NO.
(2) A/R	#1910-.062 THK	RUBBER SHEET 15-20S.H. (RUBBERCRAFT)		4
				3
2	-102	.125 THK SHT ALUM 6061-T6		2
1	-101	1.750 I.D. x .125 WALL SQ TUBE ALUM 6063-T5		1

LIST OF MATERIALS

(2) BOND ITEM 4 ON SURFACES SHOWN (AFTER ANODIZE) USING RUBBER ADHESIVE #1300, "3M CORP." 100% SURFACE CONTACT REQD FOR SEALING AT NEXT ASSY.
1. DIP BRAZE ITEMS 1 & 2 AS SHOWN.
NOTES:

FIGURE 55 DWG. NO. C-11523

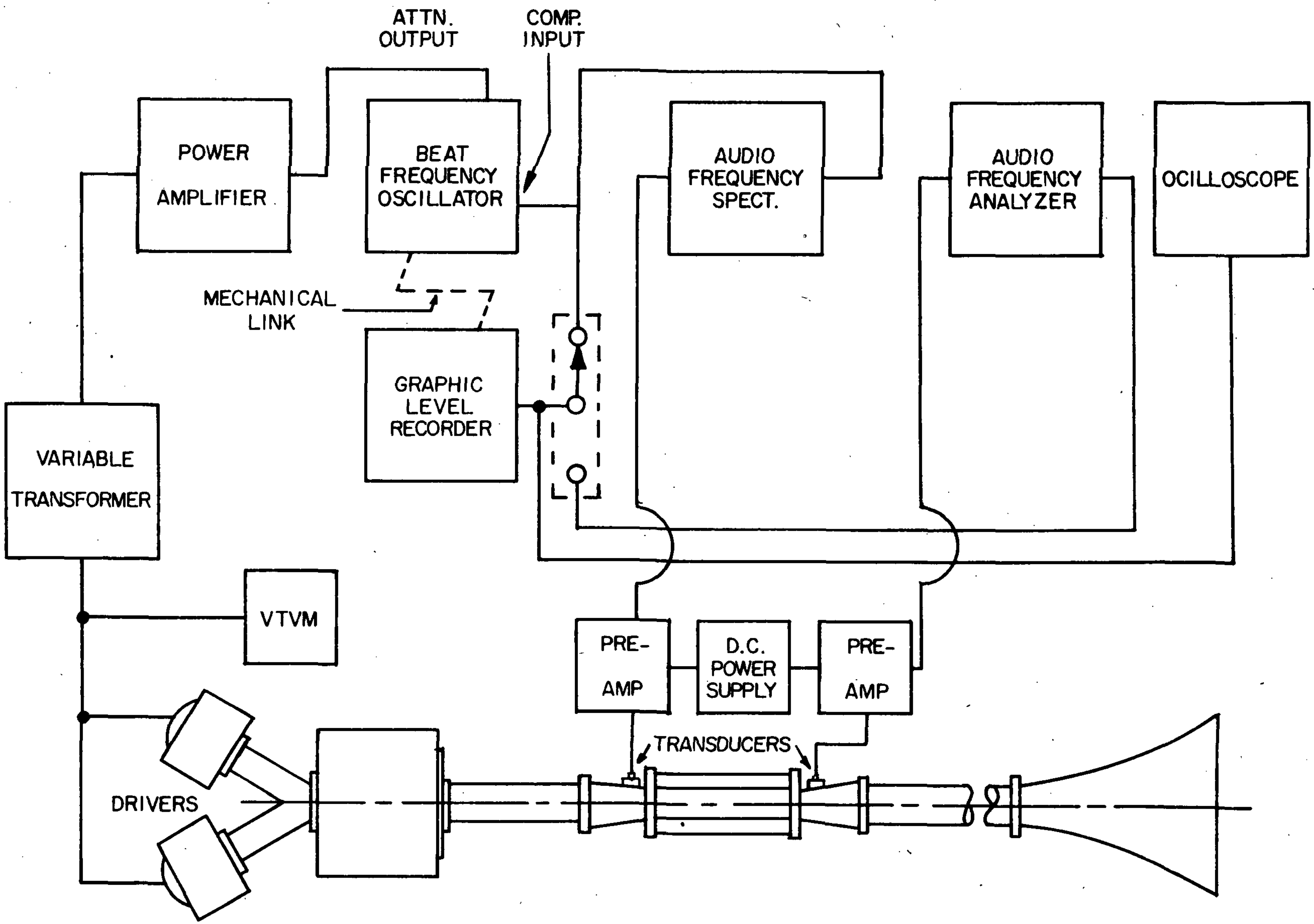


FIGURE 56. BLOCK DIAGRAM - SINE SWEEP SAMPLE SCREENING

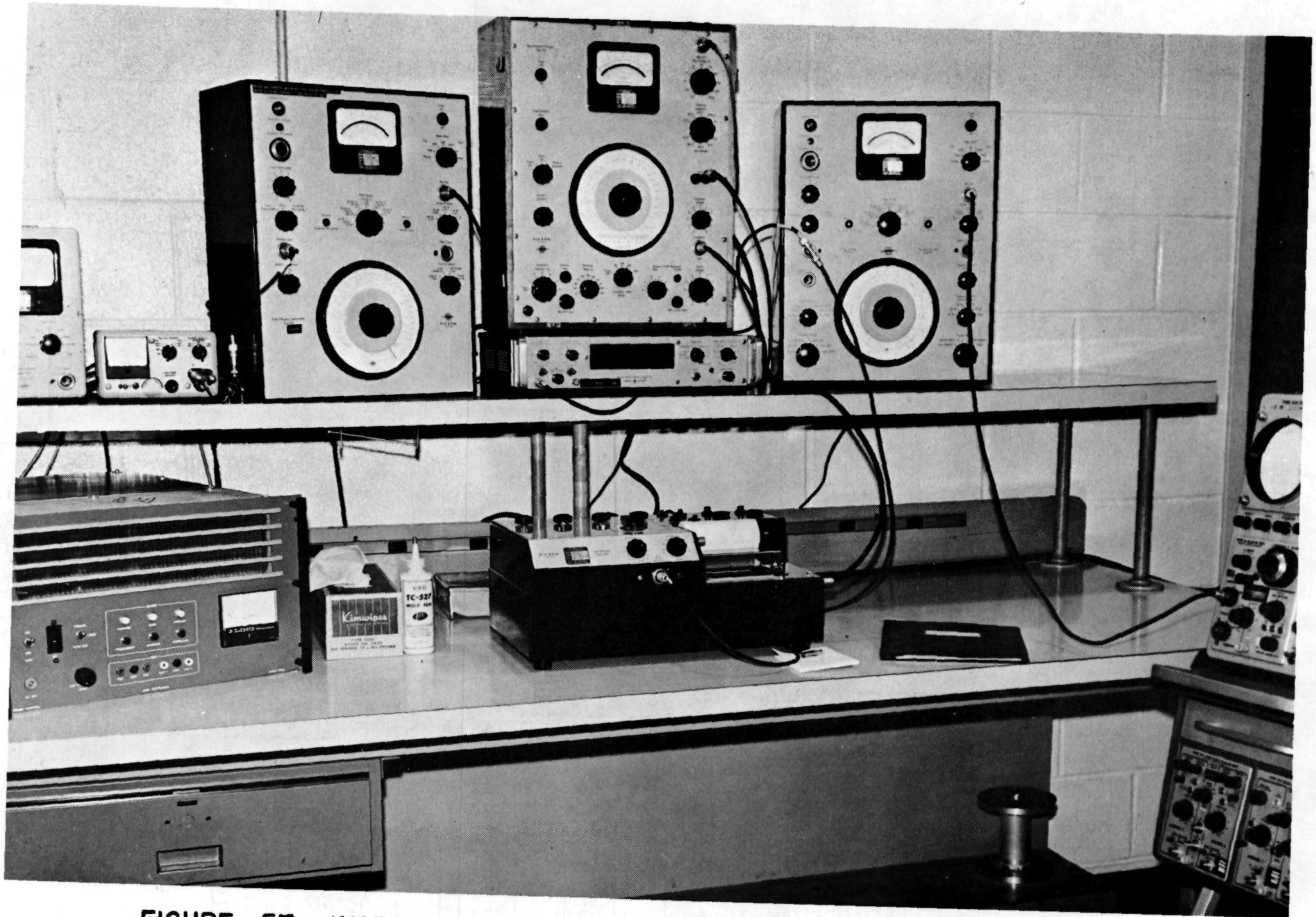


FIGURE 57. INSTRUMENTATION — SINE-SWEEP SAMPLE SCREENING

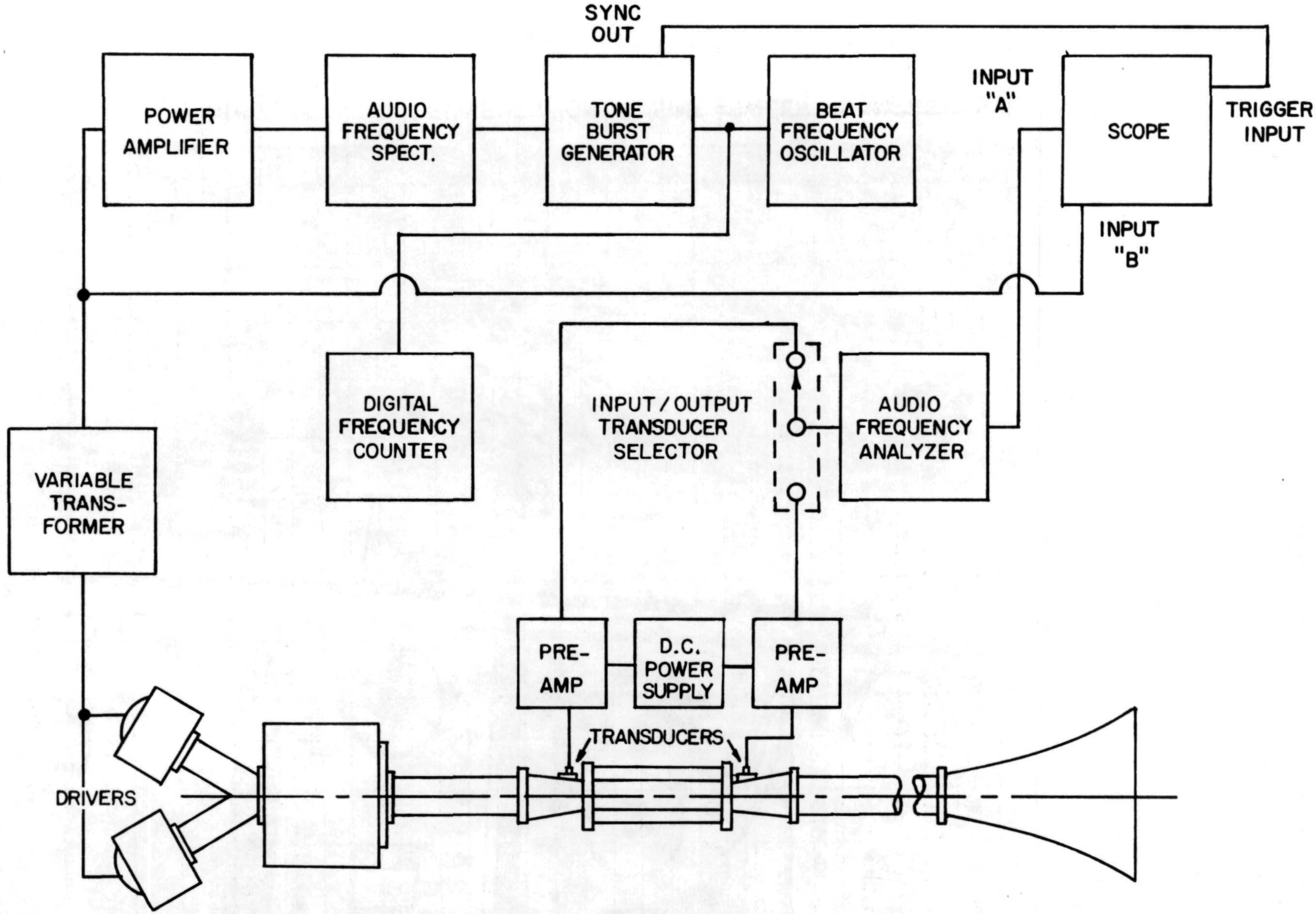


FIGURE 58. BLOCK DIAGRAM — TONE-BURST EVALUATION OF SAMPLE

DATA RECORD
SOUND ABSORBER SYSTEMS
TONE-BURST EVALUATION

Date ________ PPL Ref. ________ Air Velocity ________ Air Direction ____

Microphone Sensitivities (1) Input S/N____ Sens.______dBV (2) Output S/N____ Sens.______dBV

Sample Description ____

____

____

Band Center Freq.	BASELINE					Driver Signal Coil Volts/Z	SAMPLE						
	Input Mic			Output Mic			Input Mic		Output Mic			Inser-tion loss	Trans-mission loss
	volts	gain	ppl	volts	gain		volts	gain	volts	gain	pulse length		

Remarks ____

____

Observer ____________

FIGURE 59 DATA RECORDING CHART

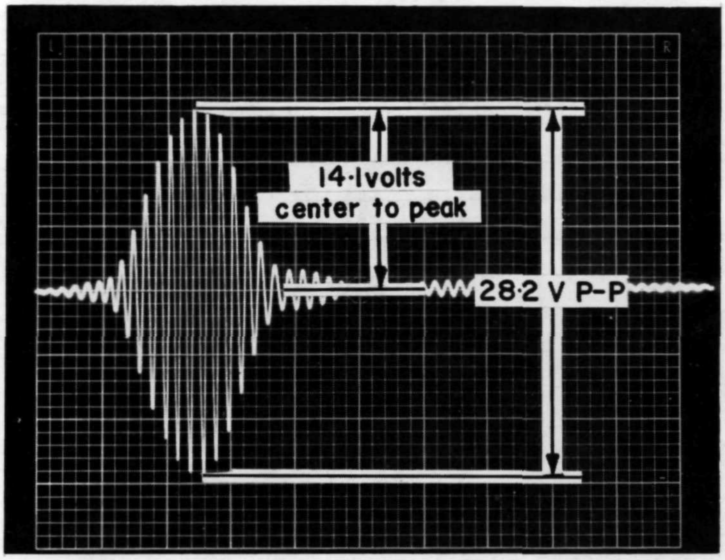
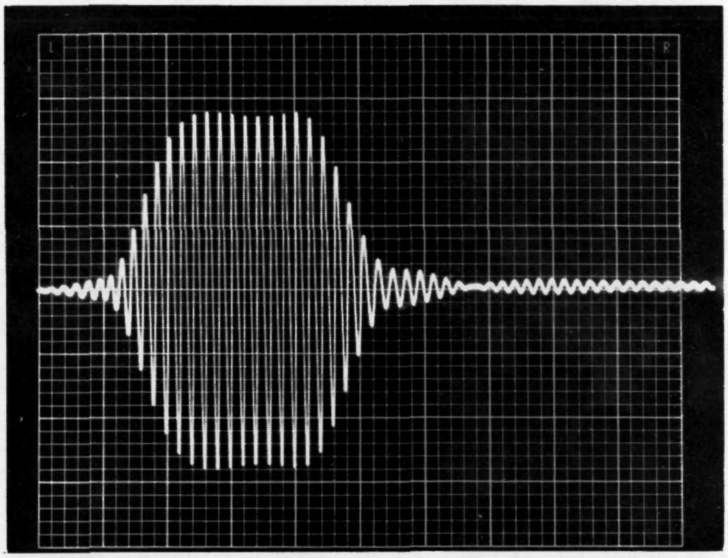
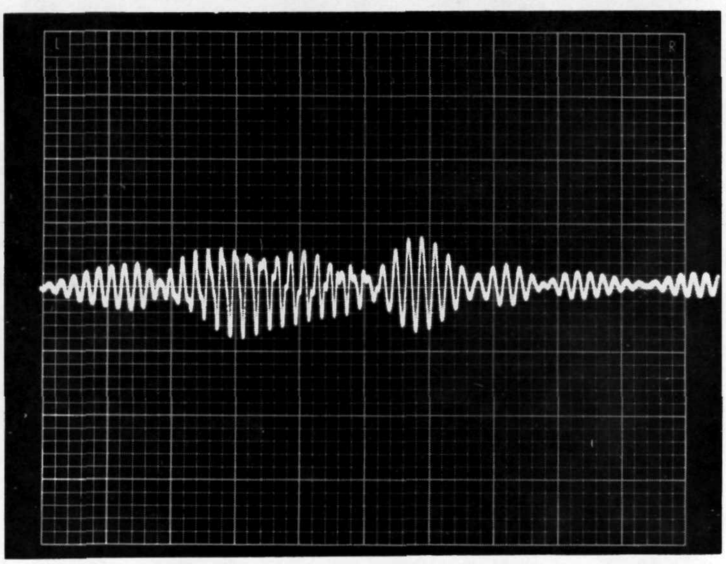


FIGURE 60. NORMAL 8 HZ PULSE ENVELOPE



a) INPUT PULSE - 16 HZ



b) OUTPUT PULSE - 16 HZ

FIGURE 61. PULSE RESPONSE OF RESONANT ABSORBER