Acoustics Lab
Measurement of the Propagation of Sound

Objective

This experiment employs a loudspeaker, microphone, function generator, amplifier, and a dual channel oscilloscope to investigate the generation and propagation of sound. The speed of sound in air will be measured by determining the transit time of a burst of sound as a function of propagation distance. This speed can be compared with an elementary result from fluid dynamics. The transition from near field to far field propagation will be investigated, and the Rayleigh distance, which indicates the general location of this transition, will be identified. The directional properties of a representative loudspeaker at a distance relatively far from the speaker will be measured at two frequencies, one moderately low and another moderately high. This directional performance can be compared with a model from acoustics theory.

Background

Sound is a pressure fluctuation that propagates in a compressible medium. A sound wave is a traveling longitudinal wave that represents a propagating pattern of compression and rarefaction. (A "longitudinal" is a wave in which the fluid particle displacement associated with the pressure and density disturbance is essentially parallel to the direction of wave propagation.)

Sound is considered to be infrasonic, audio or ultrasonic depending on whether its frequency content is below, within or above the range of frequencies which can be heard by humans (40 – 20,000 Hz). Audible sound pressure ranges over many orders of magnitude, and the sensitivity of the ear is basically logarithmic. Consequently, it is convenient to represent the magnitude of sound in terms of the Sound Pressure Level (SPL), L_P , on a logarithmic scale. The SPL is expressed in decibels (dB) as

$$L_P = 20\log\left(\frac{P}{P_{\text{ref}}}\right) \tag{1}$$

The reference pressure is essentially arbitrary. For absolute loudness measurements, the threshold of audibility at 1 kHz, 20 µPa, is commonly used. Approximate SPL (re 20 µPa) for representative sounds is shown in Table I.

SPL (dB re 20µPa)	Comment
140	Gunfire at the gunner's ear; jet engine at 30 m
130	Threshold of pain
120	Pneumatic hammer at I m; Concorde take off at 500 m.
110	Rock Concert; disco, car horn at 1m
100	Discomfort; hearing damage long term; heavy industry
90	Inside an underground train
80	Inside a bus; shouting at I m
70	Heavy city traffic at the curbside
60	Busy restaurant or department store; conversation
50	Quiet car Interior; average business office.
40	Quiet residential neighborhood, living room
30	Library; empty lecture theatre
20	Recording studio; bedroom at night, whisper
0	Threshold of hearing (average for a young person @1 kHz)

Table I. Sounds representative of various Sound Pressure Levels (SPL) in dB re 1µP

As example of how the decibel scale is used, consider the value of 50 dB given as the SPL for an average business office. What is the value of this sound pressure in Pascals? [Note that for noise the pressure values are r.m.s.].

A one dimensional acoustical plane wave propagating in the x direction at speed c is represented mathematically by

$$P(x,t) = P_0 f\left(t - \frac{x}{c}\right) \tag{2}$$

where f is an arbitrary function. According to equation (2) the pressure is $P_0 f(t)$ at $x = 0$. At any other range x *the* pressure time history is identical to what it was $x = 0$ except that its delayed *by x/c,* the time required for the wave to propagate from *0* to *x*.

A one-dimensional pure tone or single frequency wave is a simple harmonic wave that may be represented as

$$P(x,t) = P_S \sin\left(\frac{2\pi}{T}\left(t - \frac{x}{c}\right)\right) \tag{3}$$

Where the period is T, and c is the speed of sound. Alternatively the wave may be represented in terms of the frequency, f, and wavelength, λ, or in terms of angular frequency, ω, and wave number, k, as

$$P(x,t) = P_S \sin\left(2\pi f t - \frac{2\pi}{\lambda}x\right) = P_S \sin(\omega t - k x) \tag{4}$$

For acoustic sources which are "small", *i.e.* sources which are smaller than the smallest wavelength of interest, the sound radiated from the source will be an omnidirectional spherical wave. For a source located at the origin, such a wave is given by

$$p(r,t) = \frac{\rho_0}{4\pi r}\frac{dV(t - r/c)}{dt} \tag{5}$$

Where V is the volume velocity of the source and *r is* the range. Like the plane wave this wave propagates without change in shape at velocity c but its amplitude falls off like *1/r* (spherical spreading) in order to conserve energy.

At higher frequencies (where the source is no longer small with respect to a wavelength), sufficiently far from the source the sound spreads spherically but is no longer omnidirectional. The directivity pattern of the source will be frequency dependent. The acoustic pressure for a wave at angular frequency ω at a point r, θ, ϕ in the far field is expressed in terms of the directivity function $D_\omega(\theta,\phi)$ by

$$p_\omega(r,\theta,\phi) = p_0 \frac{r_0}{r} D_\omega(\theta,\phi) \tag{6}$$

The quantity p_o is called the *source level* of the source and r_o is a reference range usually chosen to be one meter.

The formula for the speed of sound is derived in most standard texts on fluid mechanics or acoustics such as Munson, *et al*. (1990, pg. 706). In general the speed of sound is the square root of the partial derivative of pressure with respect to density at constant entropy, or

$$c^2 = \left(\frac{\partial P}{\partial \rho}\right)_s \tag{7}$$

For ideal gases such as air, the preceding relationship reduces to

$$c = \sqrt{\gamma R T} \tag{8}$$

Where γ is the specific heat ratio, R is the gas constant on the unit mass basis, and T is the absolute temperature. For air c is around 340 m/s.

Investigation of the generation and propagation of sound from an ideal vibrating diaphragm or an actual extended source such as a loudspeaker reveals two regimes. In the near field region, the sound is significantly collimated and exhibits significant complexity. . Here the maximum sound pressure fluctuates from point to point but has little systematic decrease with distance from the source. In the far field, as discussed above, the sound pressure drops off inversely with the distance *r* from the source. The transition from near to far field is not abrupt, but it is in the general vicinity of the Rayleigh distance given by,

$$R_0 = \frac{\pi a^2}{\lambda} \tag{9}$$

Here *a* is the radius of the source, which has been taken to be circular.

An appropriate theoretical model for the experimental situation was developed by McLachan (1929). This model considers a cylindrically symmetric, vibrating circular diaphragm in an infinite rigid baffle and gives the following result for the far field dependence of the sound pressure on distance and angle, θ, from the normal:

$$P(r,\theta) = \frac{\rho_{\text{air}} a^2 \omega u_0}{2} \left| \frac{2\mathrm{J}_1(k a \sin\theta)}{k a \sin\theta} \right| \frac{1}{r} \tag{10}$$

Here $\mathrm{J}_1(x)$ is the Bessel function of the first kind of order 1, ρ_{air} is the density of air, and u_0 is the amplitude of the velocity of the vibrating piston. In the ME4053 laboratory the baffle is a piece of plywood with a circular hole in it. The parameter *a* in (10) is the radius of the hole. The piston is effectively the air moving back and forth in the hole, driven by the vibrating speaker cone. Note that this result is a particular case of Eq. (6) with the directivity *D* of Eq. 6 being the term in | | in equation (10)

At a fixed distance from the source, the relative sound pressure is determined only by the angle off normal, and a convenient sound pressure ratio can be defined as

$$SPR_{\text{th}} = \left(\frac{P(r_{\text{fixed}},\theta)}{P(r_{\text{fixed}},0)} \right)_r = \left| \frac{2\mathrm{J}_1(k a \sin\theta)}{k a \sin\theta} \right| \tag{11}$$

Since the response of the microphone and amplifier can be assumed to be linear over a reasonable range, the peak output voltage, *V*, as displayed on the oscilloscope should be proportional to the peak sound pressure, *P*, so

$$SPR_{\text{exp}} = \left(\frac{P}{P_{\text{ref}}} \right) = \left(\frac{V}{V_{\text{ref}}} \right) \tag{12}$$

For relative measurements, any convenient reference can be used such as the pressure at minimum distance and normal to the source, as in

$$L_P = 20\log\left(\frac{P(r,\theta)}{P(r_{\min},0)} \right) = 20\log\left(\frac{V(r,\theta)}{V(r_{\min},0)} \right) \tag{13}$$

In the far field of a loudspeaker or any compact source, the wavefront is spherical and the acoustic pressure drops off in magnitude inversely with the radial distance from the source. Algebraically such an inverse relationship is

$$\frac{P(r)}{P_{\text{ref}}} = \frac{K}{r} \tag{14}$$

For this inverse dependence on radial distance the SPL has a slope of –20 dB/decade when plotted against the common logarithm of the radius ratio. The voltage from a microphone with a linear voltage response to sound pressure will exhibit the same trend, so

$$L_P = 20\log\left(\frac{P(r)}{P_{\text{ref}}} \right) = 20\log\left(\frac{V(r)}{V_{\text{ref}}} \right) = K' - 20\log\left(\frac{r}{r_{\text{ref}}} \right) \tag{15}$$

This systematic variation is a simple and unambiguous indicator of far field behavior.

Apparatus and General Proceduures

The function generators used in this lab are combination frequency generators and modulators. The frequency generator produces a nearly pure sinusoidal signal. This signal is modulated by a boxcar wave with an adjustable repetition rate and duty cycle. The number of cycles (and hence the length of the burst) and the interval between bursts with vary from class to class and experiment to experiment. You should record in your notebook the, frequency, number of cycles and pulse interval. For example the signal might consist of a four cycle 5 kHz sine burst (.8 msec) intervals of around 10 msec. Bursts are used to minimize the effects of room reflections as well as interference between adjacent experimental stations in the directivity experiment, and they are also convenient signals for determining the speed of sound by this time of flight method. The use of bursts (with a digital data acquisition system) also permits signal averaging to increase the signal to noise ratio. It is the ability to signal average that makes it possible to perform valid acoustical measurements without an anechoic chamber.

A laboratory thermometer with traceable calibration should be used to measure the room temperature before and after the speed of sound measurements.

The function generator drives the loudspeaker directly, and its modulator signal is also used to trigger the oscilloscope. The function generator output is displayed on one oscilloscope channel. A miniature microphone is used to detect the sound. The microphone generates a signal proportional to the sound pressure, and its input-output response is very nearly linear over a wide range. The microphone signal is boosted by a linear amplifier and displayed on the other oscilloscope channel. Signal parameters such as rms or peak-to-peakvoltage can be read directly off the oscilloscope. Waveforms obtained on the scope can also be recorded on a floppy disc and transferred to a pc for further analysis and display.

The loudspeaker is a moving coil dynamic loudspeaker (typically a Radio Shack 6 Inch Full-Range Speaker, Catalog No. 40-1242E). The microphone is an electret capacitance microphone (typically a Realistic Model 33-10520). The microphone output is amplified by an instrumentation amplifier based on an integrated circuit operational amplifier (typically a Motorola MC 3408 High Performance JFET Input Operational Amplifier).

A linear relationship between the input sound pressure and the output voltage of the microphone over a range of sound level is critical in this experiment. This response is called the sensitivity. Unfortunately the limits on the variation in sensitivity are not typically specified, probably because even inexpensive commercial devices have adequate performance. Consequently, these limits are probably not critical in most applications and certainly are not as important as having a uniform response over a range of frequencies. Some representative data is available, however. For example, Gibbins (1981) reports that a representative high-quality capacitance microphone has a variation in sensitivity of no more than .0001 dB over the range of 0 to 110 dB. This variation corresponds to an error in the pressure ratio of no more than .01 % for this huge amplitude range. The amplitude range in the ME 4053 experiment is on the order of only 10 dB, and it can be reasonably expected that the sensitivity of the moderate quality device used in our lab varies much less than 0.1 % over this range. Variation in the sensitivity of this order of magnitude is insignificant, and for the relative measurements in this lab it should not be necessary to calibrate the response of the inherently highly linear electret capacitance microphone.

Students should assess the uncertainty of all the instruments used to make direct measurements in this and every other lab. Uncertainties can be assessed by referring to calibration reports, by referring to manufacturer's manuals or data sheets, by referring to the general technical literature, or by making direct observations and inspections.

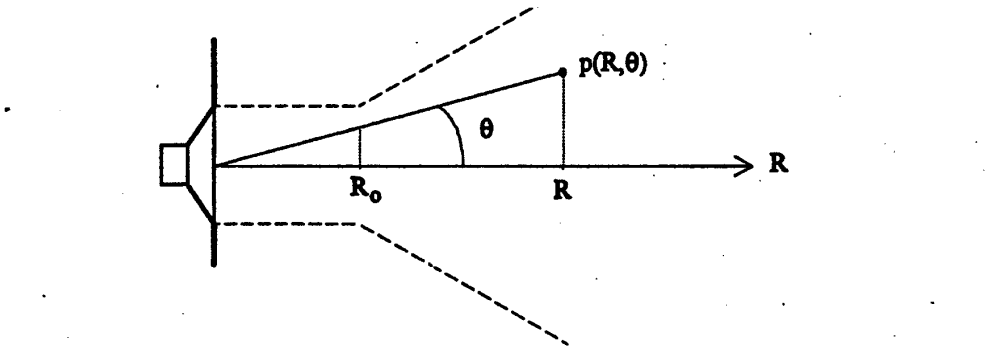
Students should sketch the actual system schematic and identify the actual instrumentation and equipment used at their station in their notebooks. Include a very neat but only schematic diagram in your report. Be sure to record all relevant experimental parameters such the nature and parameters of the input signal (e.g. frequency, pulse length, amplitude), scope settings and amplifier gains.

Experimental Setup

The specific procedures to be followed for this experiment will be given to you by the TA prior to the experiment. Some general remarks concerning the experiments you will do follow:

Theoretical model:

The pressure field radiated from the speaker is divided into two regions: the near-field and the far- field. The approximate theoretical boundary between these two regions is called the Rayleigh distance,R_0. R_0:is given by the area of the transducer divided by the wavelength.



The near-field is a more or less cylindrical region that extends from the front of the speaker to a distance of order to R_o. In this region, the axial pressure amplitude goes through a series of peaks and nulls.

In the far-field, R >> R_0, the pressure falls off like 1/R. The theoretical expression for the sound pressure for a circular piston transducer (which is a reasonable approximation to a circular loudspeaker) is given by:

$$p(R,\theta)=\frac{1}{2}\frac{\rho_0 a^2 \omega u_0}{R}\left|\frac{2J_1(ka\sin\theta)}{ka\sin\theta}\right| \qquad (1)$$

The different quantities appearing in the above equation are: the density of air ρ_0, the piston radius a (the radius of the hole in the plywood baffle), the angular frequency at which the speaker is driven ($\omega = 2\pi f$) and the velocity amplitude of the cone of the speaker (u_o). [Note that the farfield pressure is proportional to ωu_0 which is the acceleration of the cone]. The most important features of this equation are:

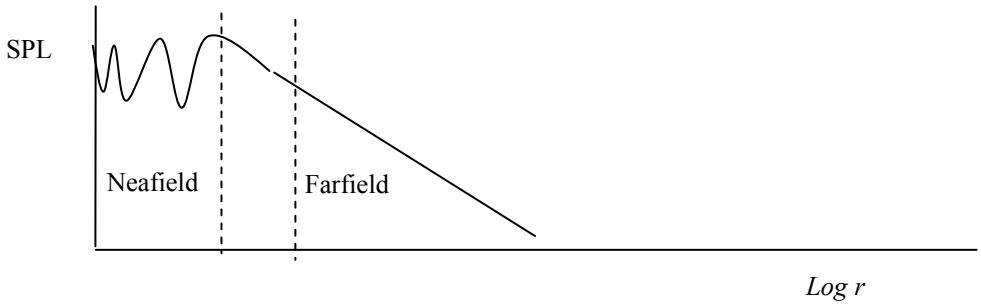
- The 1/R term, that describes how the pressure decreases with distance;
- The term between the absolute value bars, that describes how the pressure varies with the angle θ and frequency.

In the acoustic lab, you will experimentally verify the theoretical predictions provided by these two terms.

Theoretical distance dependence

A quantity very often used in acoustics is the sound pressure level (SPL) expressed in decibels (dB). It is a logarithmic quantity that is essentially proportional to the log of the pressure. If you were to plot the SPL as a

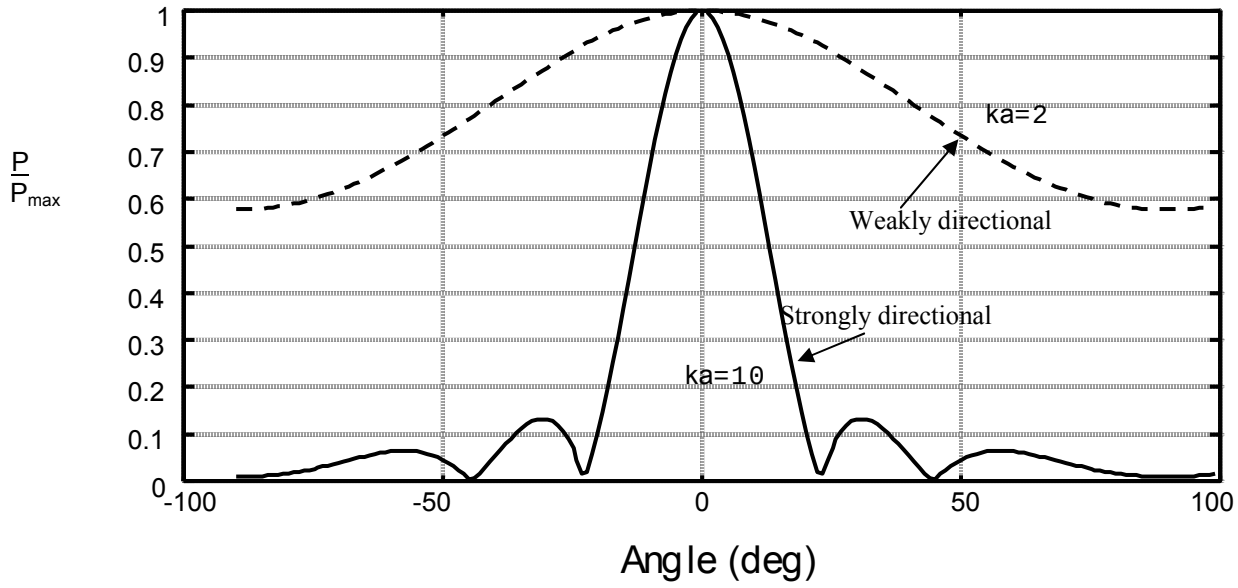
function of log(R) (keeping θ constant), the above equation reveals that, in the far-field, this plot would look like a straight line with a negative slope of 20dB/decade:



See Fig. 1 , on page 27, for an example of an actual pressure vs range data.

Theoretical angle dependence

The theoretical angular dependence given by (1) is shown below for two different values of $ka = 2\pi a/\lambda$



The above plots show pressure vs angle for fixed angle. A similar plot could be obtained from (1) which would show pressure versus frequency at a a fixed angle. See Fig. 2, on page 27, for an example a measured directivity pattern

Sound Speed Measurement

To measure the speed of sound we transmit a short pulse using the function generator. We measure the time delay between the trigger and the arrival of the beginning of the acoustic pulse as a function of the distance between the loudspeaker and the microphone as shown in Fig. 3 on page 27. The data should fall on a straight line. The slope of this line is the propagation time per meter which is called the slowness. [To get this slope, we use a least square technique to fit the data to the best straight line.] The inverse of the slowness is the sound speed. A typical example of an input pulse and a received pulse are shown in Fig. 4 on page 17. Note the delay due to propagation.

The theoretical expression for the sound speed in air is

$$c = \sqrt{\gamma RT}$$

where γ is the ratio of specific heats, R is the gas constant and T is the absolute temperature.

For air

$$\gamma = 1.4$$

and

$$R = \frac{R_0}{M_{mol}} = \frac{8314(J/mol\,^\circ K)}{29(kgm/mol)}$$

so that at 0°

$$c = \sqrt{\frac{1.4 \cdot 8314 \cdot 273}{29}} = 331 m/\sec$$

and at 20°

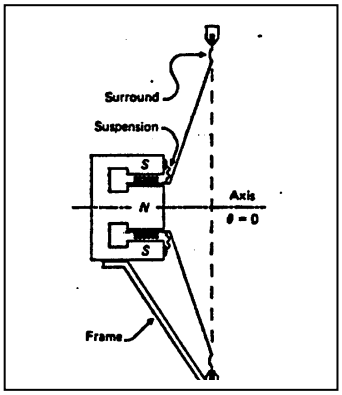
$$c = 331\sqrt{\frac{293}{273}} = 343 m/\sec$$

Be sure to measure the temperature of the room when you measure the sound speed so you can correctly calculate the theoretical value for the sound speed

Acoustic Instrumentation

The Loudspeaker

The loudspeaker used in the acoustics experiment is a standard moving cone dynamic loudspeaker, as shown in the figure. A light cardboard cone is weakly mounted to a rigid metal frame. A coil consisting of many turns of thin copper wire is attached to the end of the cone. This coil is located between the poles of a magnet. An ac voltage applied to the coil generates a current which interacts with the magnetic field to produce the driving force which causes the cone to vibrate and radiate sound. The farfield radiated pressure is proportional to the acceleration of the cone. The speaker is broadband, capable of generating --10 watts of acoustic power in the frequency range 40 -16000 Hz.



The microphone

The microphone is a back electret type from Radio Shack. A diagram showing the main components of the microphone is given in Figure 2. A thin, metalized polyester diaphragm is stretched tightly across the front face of the microphone. There is a small air gap between the polyester film and a layer of electret material mounted on a

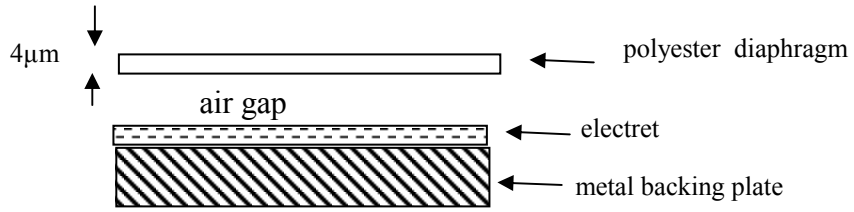


Figure 2 Back Electret Microphone

metal back plate. The electret material is a plastic layer with electric charge permanently embedded in it. Incident sound causes vibration of the polyester diaphragm which produces dynamic changes in the air gap between the polyester film and the electret. The fluctuations in the air gap in turn lead to an ac voltage, V_o, between the film and the electret. This voltage is proportional to the incident sound pressure, P_i. The ac voltage is amplified with an FET amplifier.

For the purpose of the acoustics lab it is sufficient to know that the output voltage from the microphone is proportional to the incident sound pressure.

$$V_o = MP_i$$

The receiving sensitivity M, is a function of frequency but a good microphone will be relatively flat across the audible range of frequencies. The microphone used in the laboratory is flat within +/- 4dB over the range of frequencies from 50 to 15,000 Hz. Its electrical impedance is 800 Ohms and its sensitivity is -72dB re 1V/µbar at 1 kHz. (1µbar is 0.1Pa)

Data Examples

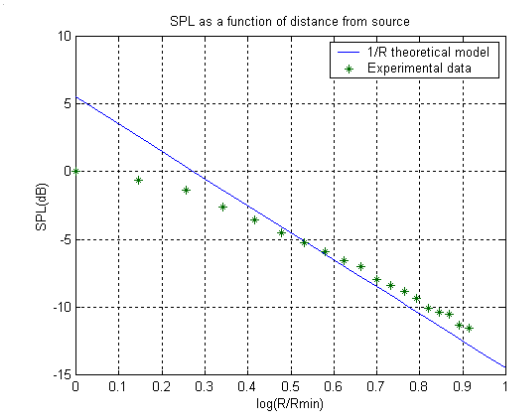


Figure 1. Typical Data Illustrating Transition from near field to Far Field Behavior.

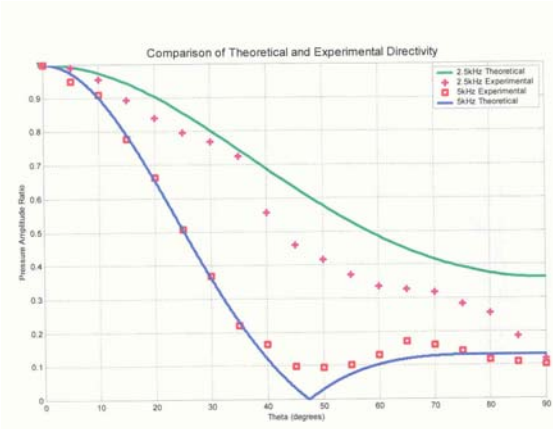


Figure 2. Results of Typical Directivity Tests.

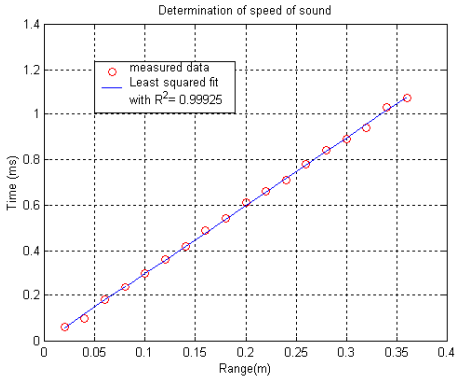


Figure 3. Typical data and models for speed of sound measurement

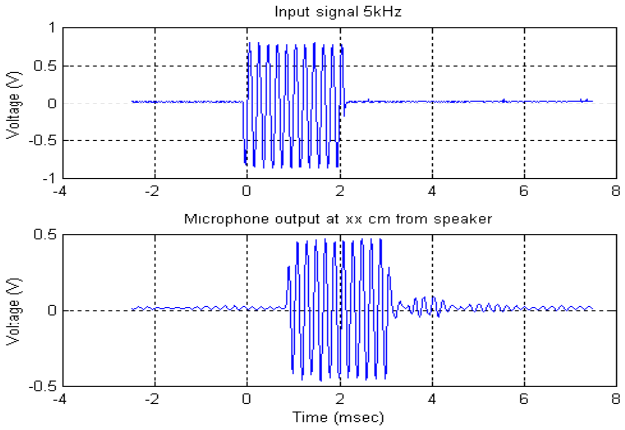


Figure 4. Example of voltage input to loudspeaker and measured by the microphone

Reference

McLachan, N. W., 1929, "Pressure Distribution in a Fluid due to the Axial Vibration of a Rigid Disc", *Proceedings of the Royal Society of London, Series A, Containing Papers of a Mathematical and Physical Character*, Vol. 122, Issue 790, 1 February 1929, pp. 604-609, full text available on-line at www.Jstor.org.