

Spherical Waves and Reverberation: Sound pressure measurements a reverberant room

(Report due 7-October-2004)

Laboratory 1 consists of acoustic measurements.

Each student should prepare a report on the lab. Since you work in pairs, the data presented in each report may be shared among partners, however the words: introducing the report, describing the data and describing the conclusions must be the work of each individual.

A. Objectives

The primary objectives of this laboratory session are:

1. to demonstrate the properties of spherical acoustic waves.
2. to demonstrate the effects of room reverberation on sound transmission.

B. Equipment

- A speaker (on a 1.2 m pole) driven by a power amplifier,
- A microphone (on a pole),
- A microphone amplifier,
- A two-channel oscilloscope,
- A PC compatible micro-computer with a DSP-16+ and SysId software, and
- A highly reverberant room
- A sound level meter is optional

C. Theory

1. Spherical Acoustic Waves (Section 5.11 of Kinsler et al., pg. 112-115).

Spherical acoustic waves can be treated as one dimensional waves in which the magnitude and angle of sound pressure \underline{P} vary only with the radial distance from the source r , such that

$$\underline{P}(r) = \frac{\underline{A}^+}{r} e^{-jkr} \quad , \quad (\text{Eq. 1})$$

where: $k = \omega/c$ and \underline{A}^+ is the complex amplitude describing the outward going wave, (Eq. 1 assumes that there are no reflected waves). Reflections and the production of backward traveling waves complicate the relationship between $\underline{P}(r)$ and r .

2. Room Acoustics (Chapter 13 of Kinsler et al.)

a. *The influence of reverberations on sound pressure and volume velocity*

The importance of reflections in an acoustic environment is primarily dependent on the sound absorptive properties of the environment's boundaries and furnishings. Anechoic environments are bounded by surfaces that are matched to the characteristic impedance of air. Such surfaces absorb all incident sound waves, such that there are no reflections. Rigid boundaries absorb little sound energy and cause large reflections. If the walls and all of the furnishings within a room were completely rigid and air were a truly lossless medium, sound emitted within such an reverberant environment would theoretically continue forever, bouncing

from surface to surface without loss.

The Sabine equation

$$T \propto \text{Volume} / \text{Absorption} \quad (\text{Eq. 13.1 of Kinsler et al.})$$

relates the reverberation time (a measure of the decay time of an acoustic transient) to the losses (absorption) within an environment. The reverberation time T is usually defined as the time needed for an acoustic transient to be reduced to 60 dB below its peak value. The 60 dB dynamic range required of the measurement equipment is difficult to achieve without the use of large sound pressures or long averaging times. The techniques in this laboratory exercise will allow you to estimate reverberation times from measurements of smaller dynamic range.

b. The use of narrow-band noise

The presence of large reflections increases the spatial variation within a sound field. The sound-pressure magnitude in response to a tonal stimulus in a reverberant environment would show large spatial derivatives corresponding to alterations between peaks and troughs of the “standing waves”. This rapid spatial dependence complicates measurements of sound propagation within the room. In order to smooth out the spatial dependence, it is common practice to use narrow-band noise in measurements of the acoustic properties of a room. Part of the measurement protocol makes use of a third-octave bandwidth noise centered at 1 kHz to help determine how sound pressure and velocity vary in a room.

c. Direct and Diffuse field.

In general, the sound pressure at any point in space can be characterized as the sum of waves propagating in all directions. In the one-dimensional case, we need only consider waves propagating in the positive and negative directions. In a three-dimensional reverberant environment, the number of possible wave directions is infinite. However, there are a few rules, which simplify the sound field. If the dimensions of the environment are large compared to sound wavelengths, there will be a region near the sound source where the sound field is dominated by the one-dimensional “direct field”. This region occurs because losses in air and the spread of sound power through the room reduce the magnitude of the reflected waves relative to the output wave near the source. As the stimulus wave propagates from the sound source its pressure amplitude is reduced by a factor of $1/r$ (Eqn 1) until the reflected waves dominate. If the sound stimulus continues for a long enough time, the reflected waves come from all directions and are all of similar average power density such that the sound pressure is relatively independent of position. This condition is called the diffuse field. The location where the direct and diffuse field meet is called the *critical distance*.

3. Sources of Variation in Acoustic Measurements

In general, acoustic measurements are inherently variable, e.g. small variations in microphone positions in the sound field or uncontrolled noise sources can cause significant variations in measured sound pressure. Your presence and the presence of others in the rooms where you make measurements, as well as the ringing of the elevator bell, doors opening and closing, etc., will increase the uncertainty associated with any one measurement. In order to reduce these uncertainties you should plan to repeat measurements. A well thought out plan of measurement repetitions will not only determine the variability of your estimates, but will allow you to partition the variability between uncontrolled room-noise and variations and errors in microphone

positioning.

D. Protocol

1. Set up

- a. Place the speaker in the center of the room and connect it to the power amplifier output.
- b. Place the microphone assembly 0.25 meter from the speaker and connect the pressure microphone to the amplifier.
- c. Confirm that the output lead of the DSP-16+ board is connected to both oscilloscope channel 1 and the input to the power amplifier.
- d. Confirm that the output of the microphone amplifier is connected to both oscilloscope channel 2 and the input lead of the SysId board.
- e. Confirm that the microphone amplifier is on, working and the gain is set to 20 dB.
- f. Confirm that the power amplifier is on and the attenuator is set to 20 dB. You may adjust the attenuator to change the intensity of the sound produced. Smaller attenuations yield more intense sounds. Assume the attenuator and the microphone amplifier are linear.

Use the attenuator to adjust the sound to a comfortable level. Keep track of the attenuator level and microphone gain in your protocol. Undocumented variations in gain or attenuation can cause large misinterpretations. Also be conscious of noise floor issues. If your measured pressures vary randomly about some level, is it because the stimulus is indistinguishable from measurement noise?

- g. Turn on the computer, and move to the appropriate disk directory.

–Type the bold in response to the prompt

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C:\>cd 6551LABS.04\LAB1.DIR
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- h. Start SYSID by typing

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C:\6551LABS.04\LAB1.DIR>sysid
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2. Impulse Response

Measure the sound pressure produced by an acoustic click and estimate the reverberation time of the system.

- a. Load the impulse measurement parameters with a macro command
 - hit the “F1” key
 - at prompt hit “ENTER” key.
- b. Initiate the measurement (Type **F**).
- c. Describe the response spectrum. Is it flat? upward sloping? downward sloping? What is the frequency of the maximum and what are the critical frequencies for describing roll-offs? (Make sure you choose an attenuation and microphone gain that give a good signal to noise level. To differentiate between signal and noise, make repeated measurements at varied attenuator levels. The signal should change regularly with attenuation, the noise will not.)
- d. Describe the time wave form.
 - Type **T**

- What is the duration and overall shape of the click response?
 - How long does it take for the click response to decay from its peak to the noise level?
 - Adjust the display limits (use the **L** command) to more closely observe the first 200 ms of the response.
 - What is the latency between the electric pulse to the speaker and the first maximum of the click response?
- e. Redisplay the full time span of the response and use a log-representation of the envelope of the wave form to investigate reverberation time. (The waveform envelope traces measurements of the average amplitude of the waveform made over a small integration time.)
- Type **V** and then **G**.
 - What is the magnitude range (in decades) between the peak response and noise?
 - How much time is required for the stimulus to go from the reverberation maximum, just after the click peak, to the noise floor?
 - Use these plots to estimate the reverberation time of the room. In a dB vs. linear time plot, the reverberation should decrease in a near linear fashion. Use the slope of the decrease to estimate how much time it would take for the reverberation to fall a full 60 dB. Remember that you won't see a full 60 dB fall because of noise.

3. Sound Pressure as a Function of Distance from the Source.

Spherical wave theory predicts that the pressure varies in regular patterns with distance from the source. This exercise asks you to use tones and a third-octave noise band to determine these spatial variations in both environments.

- a. Place the microphones 25 cm from the speaker and connect the pressure microphone to the amplifier.
- b. Initiate a noise band centered at 1 kHz.
 - Load the measurement parameters and start the measurement with macro “F3”
 - at prompt hit “ENTER” key.
- c. Measure the response magnitude, it may be helpful to use third-octave filtering from the Process menu: type **P**, **O**, and return. Also clicking the mouse activates a cursor.
- d. Repeat the measurement a few times in order to estimate the variability.
 - From the Measure menu type **F**.
- e. Repeat the process with a few pure tones of 1000 Hz,
 - from the Measure menu type **O** and then 1000 and then return.
- f. Vary the distance of the microphone from the speaker and re-measure with the noise and tone. Plot your results. Is there a regular variation of $|P|$ with r ? What would you predict for a spherical source? Is that pattern observable in your data? Can you define a critical distance where the reverberation begins to dominate? Use a constant spatial increment, say 25 cm for your measurements. Are the measurements with the noise, more or less regular than the measurement with the tones?

Construct all plots while you are taking the data. Inspection of the results will help you think about the measurements and may point out errors.

4. Other things you can try.

- a. A spherical sound source is omni-directional. Does this speaker approximate an omni-directional source?. Does the directionality depend on sound wavelength?

- b. Using impulses or tone bursts (hit the F5 or F6 keys), change the position of the microphone in the room to try to exaggerate and identify specific echoes.
 - c. Walk around the room with the sound level meter while either tones or the noise band is being played. Can you find evidence of nulls.
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E. Report

Describe your results in a brief but complete report. Your report should include:

- A brief review of your methods: Where was the sound source placed? Over what distance range did you make acoustical measurements, etc.
- A description of the impulse response and the estimates of reverberation time. How reproducible are these estimates. What determines the first few msec of the impulse response? What determines the later times? How good are these estimates?.
- Describe in detail how sound pressure varies with distance from the source in the elevator lobby. Does any part of these measurements fit the theory for spherical acoustic waves? Does any part of these measurements fit the theory for the diffuse field? What is the critical distance in the two environments? Are there differences between the tone and noise-band measurements? Why or why not?
- If you investigated the directionality of the speaker or the location dependence of the reverberation, or any thing else, summarize those results.
- Remember each lab report is 10% of your grade. The best reports will be concise and clear, and will discuss the measured data in terms of some aspect of acoustic theory, e.g. you may use the data to test the “inverse square law” hypothesis and then discuss to what degree it holds in the two environments.