Physics 145: LOUDSPEAKERS

What's the point? Bring all of your experience with data-analysis techniques, Fourier spectral measurements, automated experiment-control and data-acquisition methods, and AC impedances together to solve an interesting practical problem: design and build a two-way loudspeaker.

Equipment: Computer and interface box, dynamic microphone, woofer and tweeter drivers, variable-volume loudspeaker enclosure, preamplifier, power amplifier, frequency-response VI, crossover network components.

Introduction:

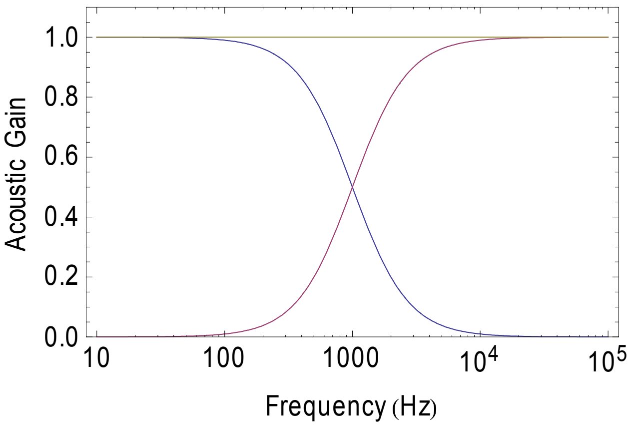
A loudspeaker is usually described as a multi-component system that includes an enclosure, two or more speaker drivers and an electrical crossover network. Have you ever wondered what physically differentiates a high-quality loudspeaker from a low-quality loudspeaker? Let's begin by exploring the properties of an individual speaker driver (e.g. a woofer, mid-range, or tweeter).

Recall that a "transducer" is a device that converts one type of physical stimulus (electrical, mechanical, acoustical, optical, etc.) into an electrical voltage, whereas a "generator" converts a voltage into another type of physical response. In this context, a microphone is a transducer and a speaker-driver is a generator, and both have physical responses that are time and frequency dependent. The time and frequency-response functions are related to one another via the Fourier transform. If the frequency-response function (FRF) were perfectly flat (i.e. frequency independent), then the output signal would always match the input signal -- a very desirable outcome. But inevitably, real transducers and generators perform better at some frequencies than at others. A woofer driver, for example, is designed to respond strongly to low-frequency audio signals, but does not respond well to high-frequency signals. A tweeter, in contrast, is designed to respond strongly to high-frequency audio signals, but not low-frequency signals.

While it would be nice to have a single driver with a uniform response throughout the audio range (0 to 20 kHz), conventional "dynamic" drivers are constrained by real physical properties and behaviors that are electrical (inductance, capacitance, resistance), mechanical (mass, spring stiffness, damping resistance to motion) and acoustical (enclosure volume and pressure, radiation, reflection, diffraction, etc.) in nature, all of which combine to produce a rather complicated effective impedance with a characteristic frequency response. While the FRF of a driver can be equivalently studied in any of the three domains: electrical (voltage → charge and current), mechanical (force → displacement and velocity) or acoustical (pressure → density and flux), we will focus our attention on the electrical domain. In the electrical domain, you should not expect that the mechanical and acoustical characteristics of the driver are ignored. Instead, they *transform* into the electrical domain as a network of effective resistors, capacitors and inductors. Thus, even though a conventional driver appears to be a simple inductive coil, its effective impedance will be far more complicated when the mechanical and acoustic impedances are accounted form. That's all we're going to say on this topic.

Because real drivers only perform well over a limited frequency range, a good loudspeaker will use two or more drivers in parallel, each of which does covers a different region of the audio frequency range. As long as the drivers have a little overlap (to avoid frequency gaps), this makes it possible to get a strong response all the way across the audio range.

But we aren't done yet because *strong* isn't necessarily *flat*. Ideally, we are looking for a flat FRF that leaves the input signal undistorted. First, the drivers themselves need to be flat in their respective frequency ranges. A more expensive driver will usually be flatter over a wider range. Second, the FRFs of the various drivers used need to be balanced, because we don't want the tweeter to have a much larger response at 10 kHz than the woofer has at 500 Hz. And finally, we want to selectively send each to driver only the frequencies in its designed range. The purpose of the electrical crossover network is to split the input signal into several different outputs, each of which filters out all but the right frequency range for its driver. This last issue begs another question. What could it hurt to just send the total input signal to each driver, since they won't respond well to frequencies outside their respective ranges anyway? There are at least two important reasons to use a crossover network. (1) In an frequency-overlap region, where two adjacent drivers both produce significant output, the two drivers become acoustically coupled via the pressure waves that they generate. In this way, one driver can modify (i.e. degrade) the FRF of another driver. (2) Interference between the waves produced by the two drivers can result in a grossly non-dipolar directional sound-intensity distribution.



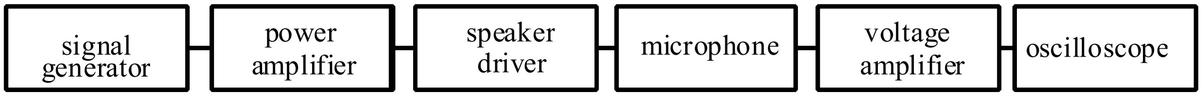
In this lab, we will build a simple first-order two-way (i.e. two driver) loudspeaker system with one woofer and one tweeter. After splitting the signal into two branches, we filter the woofer signal with a low-pass RL filter and the tweeter signal with a high-pass RC filter. By setting the cutoff frequencies of the two filters equal to one another, we optimize the flatness frequency response. In the figure below, you can see the separate responses of a low-pass RL filter and a high-pass RC filter that have been tuned to the same cutoff frequency, together with their sum, which is flat. More complicated filter circuits with sharper (i.e. higher-order) cutoffs can further reduce the woofer/tweeter overlap. Most nice commercial loudspeakers employ higher-order crossover networks.

The loudspeaker enclosure is also an important loudspeaker component, and is designed to extend the low-frequency end of the response curve of the woofer. No woofer has a low-frequency response that stays flat all the way to zero frequency -- they typically drop off somewhere between 100 and 200 Hz. The enclosure adds a mechanical potential-energy (i.e. analogous to a spring or a capacitance) to the system since the enclosed volume of air is compressed by a pressure wave. Combined with the mass of the driver cone, you get a mass-spring oscillator that can be tuned to resonate broadly just below the woofer's low-frequency cutoff, effectively extending the low-frequency response of the system.

There are several approaches to measuring the FRF of a loudspeaker (or driver). We will mention three here. (1) The most basic technique is analogous to frequency-response measurements that you have already made on electrical, mechanical and acoustical systems in the Impedance labs, where you manually swept the frequency of a sinusoidal input signal through the range of interest and measured the gain at several points along the way. This is called the *continuous wave* (CW) approach. (2) In the Fourier Spectra lab, you found that a pulsed signal (e.g. a hand clap or a balloon popping) produced a very broad input spectrum. An infinitely-sharp pulse, if it could be produced, would produce a perfectly flat input spectrum. Thus, sending a very sharp input pulse into the driver, and measuring the acoustic frequency response with a microphone, will yield the FRF of the driver. This is the *pulsed* approach. (3) Random noise (sometimes called static) actually has a remarkably uniform FRF. By analogy to visible light, we often refer to noise as *white* because is contains a uniform mixture of frequencies. It is appropriate to think of white noise as a random train of infinitely-sharp pulses. The *white noise* approach involves sending an artificially-generated white-noise electrical signal into the driver, and measuring the acoustic FRF with a microphone. You will use this approach to characterize your drivers.

One important caveat: in each of the three FRF measurements described above, we have ignored the fact that the other components in the experiment (the power amplifier, the microphone, the preamplifier and the room itself) also have their own FRFs. The combined FRF is actually the multiplicative product of the individual FRFs of these components. This situation is illustrated below, where the output of each component in the train becomes the input for the next component. The combined FRF is the multiplicative product of the individual FRFs of each of these components. The gain of the whole train is also the multiplicative product of the component gains. Because any component with a flat response can be harmlessly ignored (multiplying by a constant doesn't change the shape of the FRF), one might hope to assume that the auxiliary components all have flat frequency responses. But in reality, you have to check auxiliary components very carefully to make sure that they don't have severe frequency-response limitations. A multimeter, for example, loses sensitivity at frequencies above a few hundred Hz. A good oscilloscope, on the other hand, has a flat response from 0 to well beyond 2 MHz, and a rise time of less than 0.1 μs. Thus the oscilloscope can be used to directly determine the FRFs of the signal generator and amplifiers. Our signal generators tend to exhibit flat FRFs when the impedance of the load being driven is light. But for heavy loads (where the impedance is comparable to or smaller than the 600 Ω output impedance of

$$\left(\frac{\text{output}}{\text{input}}\right)_{\text{train}} = \left(\frac{\text{output}}{\text{input}}\right)_1 \left(\frac{\text{output}}{\text{input}}\right)_2 \left(\frac{\text{output}}{\text{input}}\right)_3 \cdots \left(\frac{\text{output}}{\text{input}}\right)_n$$



The train of components required to measure the frequency response of a speaker driver. The output of each component becomes the input of the next component.

the signal generator), the response weakens dramatically at high frequencies. Because we really hate a frequency-dependent input signal, we often use our blue power amplifiers, which have a much smaller output impedance, to drive the load. They have an impressively flat frequency response from 0 to 1 MHz.

For simplicity, we will assume that all components except our speaker driver have flat responses that can be neglected. In reality, our amplifier responses are flat enough to ignore, but the responses of the microphone and the room are not. Thus, even your flattest FRF won't look very ideal. A really expensive mic would help, as would doing the experiment downstairs in the anechoic (i.e. no echo) chamber.

EQUIPMENT WARNINGS: Your speaker drivers are quite expensive and easily damaged.
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1) The diaphragms can be easily ripped or punctured. Touch them only lightly.
2) The speaker enclosure is heavy and should be lifted and moved with care. Do not drop it.
3) The quickest way to destroy your woofer driver is to change the enclosure volume without first breaking the seal on the adjustable back plate, thereby creating a large positive or negative pressure inside. First set the plate position with the plate laying face down, and then rotate it up to make and lock the seal.
4) The quickest way to destroy your tweeter driver is to feed it too strong a signal. A signal level that sounds great from the woofer will fry your tweeter. After changing the cables on any driver, always start at zero gain on the power amplifier and turn the gain up slowly until the sound level just begins to annoy your neighbors.

PROCEDURE: Form teams of no more than three students per station.

A: Isolate the acoustical frequency responses of a pair of speaker drivers.

1) Equipment configuration: The computer sends a white-noise signal out to your blue power amplifier from analog output A of your interface box (ao0), and then on to your speaker driver. Your microphone picks up the signal and feeds it through your preamplifier and into analog input 2 (ai1) of the interface box, where it is Fourier transformed into a frequency response. A third BNC cable tees off from output A and runs into input 1 (ai0) of the interface box as a reference signal. On the microphone preamplifier, set the coarse gain to 20 and the fine gain all the way up. On the power amplifier, use the line input and set the gain knob **ALL THE WAY DOWN**. The location of the speaker and microphone are important due to room-echo effects (you want them to be sitting right where they were when the instructor collected calibration data earlier in the year). The speaker enclosure should placed on the side of the table closest to the whiteboards with its back facing the wall, and as close to the wall as possible. The microphone should be placed 2 or 3 inches from the driver and along the driver axis.
2) Labview configuration: Copy *FrequencyResponse.vi* from the usual location into your own workspace in a new directory associated with this experiment. Explore the front panel and block diagram of the VI. The *Getting Started* panel shows you how to hook up the external cables. The *Time Record* panel shows the measured audio time signal. The FRF panel shows both the magnitude and phase of the computed frequency response function (i.e. the Fourier transform). The *DAQ settings* tab should be preset with correct default values so that you don't need to modify them. In the *FRF settings* tab, set the number of samples to 10,000 (i.e. 0.5 seconds) with a *Hanning*-type apodization window. Set the averaging controls to 30 measurements, *RMS* averaging, *exponential* weighing, and *one-shot* repetition. The *view controls* can be modified as you see fit. Set the output path to *frf.txt* so that you can export your FRF if you need to. Try a few test measurements -- start the VI and turn the power amplifier gain up slowly until the speaker volume just begins to annoy your neighbors.
3) Isolate the acoustical frequency response of your woofer driver over the range from 0 to 10 kHz

(loudspeaker enclosure should be completely open in back). When other teams are taking measurements (you will know because the light will be turned off), everyone else will need to be quiet in order to avoid contaminating their data. The frequency response that you measure is not representative of your woofer driver alone, but rather contains the combined responses of your microphone, preamp, power amp, woofer driver, and the room that you are working in. The total response is actually the product (multiplication) of the responses of these individual components. The power amp and preamp responses are flat enough to be ignored. The mic response can't be ignored, though we will ignore it anyway. The room response can be minimized by moving the mic up close to the driver (2 or 3 inches). After having your TA check to make sure that your frequency response looks reasonable, plot it (cut and paste from VI front panel into MS Word) and describe it in your lab notebook.

4) After re-reading the warning above, install and seal the back plate of your loudspeaker enclosure. Now repeat the FRF measurement and plot the new FRF. Describe how the enclosure modifies the low-frequency end of the FRF? Try several different enclosure volumes by adjusting the location of the back plate, and describe the effect of enclosure volume on the FRF.
5) After re-reading the warnings above, measure the acoustical frequency response of your tweeter driver over the range from 0 to 10 kHz. Because the tweeter is self-enclosed, it doesn't matter whether or not the large enclosure is sealed. Plot and describe the tweeter FRF. How does it differ from the woofer FRF?
6) Consider that your challenge is to use these two drivers together to obtain the flattest possible combined frequency response over the 0 to 10 kHz frequency range. Take some time to explain the importance of a flat frequency response when judging the sound quality of a loudspeaker (ask your TA for pointers).

B: Design and build a loudspeaker that includes your drivers, an enclosure, and a 2-way crossover network.

1) Take a moment to read the hyperphysics entry on crossover networks. This page is located at http://hyperphysics.phy-astr.gsu.edu/hbase/audio/cross.html, and contains some very useful information. Notice that the electrical resistance of the driver itself can be considered to be a filter compoment. Use a multimeter to measure the resistances of your drivers. The hyperphysics site also shows you how to use a parallel resistance to reduce the output from the tweeter. This "balance" resistor allows you to reduce the sound intensity of the tweeter to match that of the woofer. In this lab, we will compensate for any imbalance by simply moving the mic closer to one driver than the other. But now you know how to apply a balance resistance.
2) Design a first-order two-way Butterworth crossover network that will optimally combine your woofer and tweeter drivers to obtain the flattest possible frequency response function. The http://www.lalena.com/Audio/Calculator/XOver/ site may be helpful, but don't skip doing the calculation by hand. Review your woofer and tweeter FRFs, and do your best to pick a crossover frequency near the middle of the region where their FRFs overlap. Then set the 3dB rollover points of the high and low-frequency channels of your crossover equal to the crossover frequency. You should consider which audio resistors, capacitors, and inductors are available in

the lab as you develop your design. RLC meters are available.

3) Build the crossover network that you designed (use a breadboard for any resistors or capacitors that you select). Use *FrequencyResponse.vi* to visually check the frequency-responses of the woofer and tweeter output channels to make sure that they work as expected (not for the notebook). Now drive both output channels simultaneously. Adjust the microphone position to balance the sound intensity from the two drivers. Try to obtain the flattest possible frequency response over the widest possible range. Use the optimal enclosure volume that you identified earlier. You may want to try a smaller capacitance if you get a lot of distortion in the frequency-overlap region. Plot your best FRF for you lab notebook. Consider the imperfections in your FRF between 10 Hz and 10 kHz, and describe further improvements (a better mic, a more sophisticated 3rd-order crossover network, an echo-free room, etc.) that would improve your FRF.
4) Bring a laptop or an MP3 player and let the whole class judge your sound quality and your taste in music. The stereo-BNC adaptor cable will allow you to pipe your music into the power-amplifer input. Have fun, but try to accomodate those who are still collecting data. DO NOT put any music on the lab computers -- thanks in advance!
5) If you find that your tweeter is a little too dominant (due to the fact that we didn't require you to properly balance your drivers), suppress the tweeter intensity by placing an appropriate resistor in parallel with the tweeter driver and modifying the associated capacitance accordingly.
6) Optional: cooperate with another team at an adjacent station by sending the stereo-right signal to one loudspeaker and the stereo-left signal to the other.