C.2 Properties of Dynamic and Electrostatic Loudspeakers

Introduction: The basics, very briefly

Since essentially all signal sampling these days is done digitally it is very important to understand digital sampling techniques for representing signals accurately in the time and frequency domains. See elsewhere in the course material for background.

Loudspeaker types

The dynamic loudspeaker is an electro-inductance transducer that contains a permanent magnet, and has low impedance. The diaphragm, through which the "voice coil" runs, is usually circular and made of light material. The so-called "ribbon tweeter" has a very low-mass, rectangular diaphragm and covers a broad frequency range extending well into the ultrasonics. The electrostatic loudspeaker is made of a specially milled or porous metal back plate that is covered by thin metalized plastic film, and has a high impedance. The loudspeaker is polarized with about 150 v dc applied across the film and back plate. The ac voltage to be transduced is applied to the polarizing voltage using specially constructed amplifiers. Electrostatic loudspeakers can cover a very broad frequency range depending of how they are constructed.

Distortion

Overloading a loudspeaker will first cause distortion harmonics to increase, especially the 3rd harmonic. Further overloading with burn out the loudspeaker or the power amplifier. or both, or short-circuit the electrostatic loudspeaker. The loudspeaker (and amplifier) should be protected against burnout.

Frequency characteristics

The frequency characteristics are primarily determined by the dimensions and mechanics of the loudspeaker. A good example is the old ionophone that created a plasma as a mass-less "diaphragm" thus giving the ability to produce very high frequencies at high intensities. This is normally not possible since large loudspeakers with relatively heavy diaphragms cannot produce high frequencies. An exception to the size rule is the electrostatic loudspeaker. Here the membrane breaks up into an array of small areas, each of which can radiate high frequency sound.

At low frequencies, the loudspeaker's response can be further improved by placing it in a baffle. This reduces "acoustic short circuiting", or where sound from the backside of the loudspeaker interferes with that from the front. The point at which the baffle will begin to improve the low frequency response is given as: $\mathbf{f\, a\, (2\pi/c) = 2}$, where $\mathbf{f}$ is frequency in Hz, $\mathbf{a}$ is the radius of the loudspeaker in m, and $\mathbf{c}$ is the velocity of sound in m/s.

Directional characteristics

The directional characteristics are determined primarily by the dimensions of the loudspeaker and the wavelength of sound. The sound field becomes more directional as the frequency increases and less directional as the size of the transducer decreases. The loudspeaker's directionality is usually measured at specific frequencies and the primary beam width is expressed in degrees at -3 dB points, or some other defined dB value. A so called Directivity Index can also be calculated.

Near and far acoustic fields

The sound field close to the source is acoustically very complicated. It is called the "near field" and is appropriately introduced by the following quote from Au, 1993: "The concept of a near field is often difficult to grasp since many explanations are embroiled in mathematics that seem to confuse rather than clarify the issue". However, the near field can be important for acoustic communication in some animals (bees and fruit flies for example). The near field is dominated by the movements of air (water) particles, which can best be detected by movement or pressure gradient receivers. The sound pressure can vary drastically with distance since it is affected by the wave length (λ) of sound and the dimensions of the source. The border between the near and far field, where the sound waves become planar, is often defined as the range where the **1/r** law for spherical spreading loss takes over, or where the sound pressure decreases by 6 dB for each doubling of distance. According to most physical acoustics textbooks this occurs at a distance (**r**) where: $\mathbf{r = 2\pi a^2 / \lambda}$ and **a** is the radius of a circular radiator.

The sound level output of a loudspeaker (sound source)

The sound level output of a loudspeaker should be expressed as the sound pressure level (SPL), which is an RMS (root mean square) measure, at a reference distance (1 m). The best method is to calibrate the microphone system using a known source (a calibrator, for example at 1 Pa=94 dB SPL re. 20 μPa for sound in air). Using an oscilloscope, the peak-to-peak value of the source output (a loudspeaker or an animal sound) is compared to that of the RMS calibrating signal. After making the proper level compensations (for example changes in amplification relative to the calibration) the source is expressed as dB peak equivalent (pe) SPL. This technique is especially useful for transient signals. Warning! If one measures the peak-to-peak source output on an oscilloscope and forgets to compensate for the RMS voltage sensitivity of the microphone an error of +9 dB can occur. (See pg. 44 of the B&K pocket hand book and the Hydrophone Practical). If the sound source is an animal (animal sound) then it might be more appropriate to express the sound level in energy (μPa·s), (see elsewhere in the course material).

The exercises:

Please note: **ALL PARTICIPANTS MUST USE EAR PLUGS IN THESE EXERCISES**

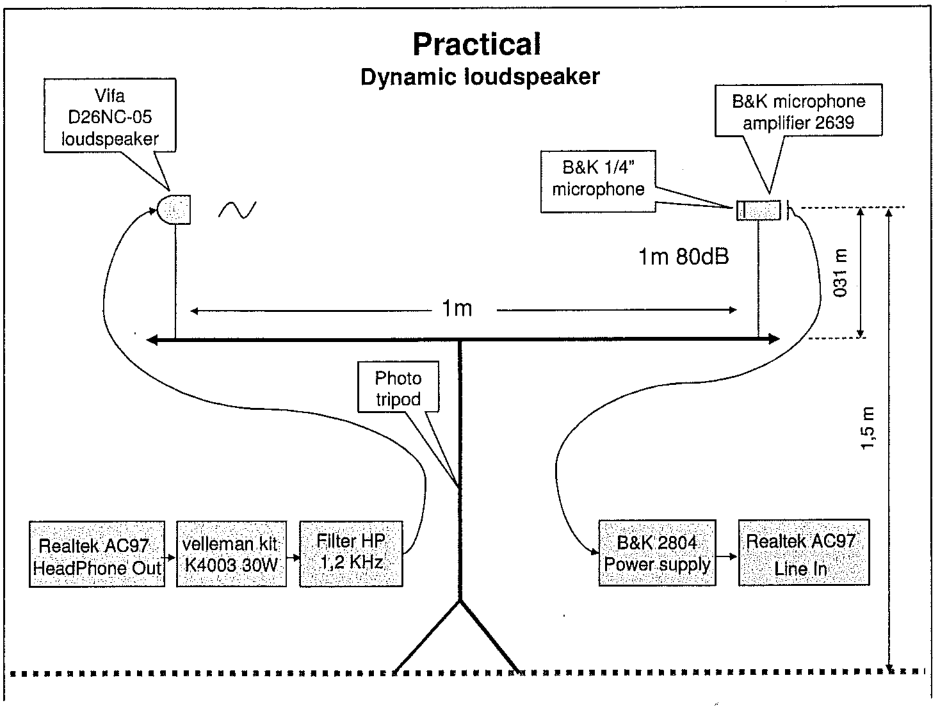
The lab. will start with an explanation of the setups, the problems and analysis of the data. **You are expected to use data sheets and note taking of you own design for these exercises**

A review of digital audio concepts and some practical aspects are found in the appendix of this write-up.

NOTE: The microphones are calibrated and the data (dB on scale and mv p-p) noted on the microphone amplifier. You can tally your results directly from the scale on the amplifier or measure the peak-to-peak value on the oscilloscope. If you prefer the latter you will need to convert your results to dB (peak equivalent RMS), that is 20 log (mv $pp_{measured}$/mv $pp_{calibration(equivalent\ RMS)}$). (You will, or have, calibrated microphones in another exercise.) Be **very careful with the microphones** since the instructor will remove the protective cap during your measurements to obtain a flat response up to 120 kHz. NB! A microphone costs several hundred dollars!

Divide your group into two mini-groups.

A) Dynamic loudspeaker



Setup – Note book SigmaNote Z500N - Sound card (Realtek AC97) – Power Amplifier (velleman kit K4003 30W) - speaker (Vifa D26NC -05-06) - B&K 4135 1/4' microphone– B&K 2633 preamplifier – B&K 2804 Microphone Power Supply

<u>Exercise A1:</u>

Transfer function (frequency response) of the speaker from 2 kHz to 20 kHz

Use the Build in sound card (Realtek AC97) in the note book (SigmaNote Z500N) as Sound Generator and as Sound Recorder:

- Control Panel I Sounds and Audio Devices I Audio.
- In Sound Playback and in Sound Recording chose Realtek AC97 Audio and click the Volume... button.
- Switch the mains to Off on the Power Amplifier (velleman kit K4003 30W).
- Connect the MiniJack – BNC cable from the Headphone Out on the Notebook to the Power Amplifier phono in.
- Connect the speaker (Vifa D26NC -05-06) to the HT out on the Power Amplifier.

Use the program Adobe Audition 1.5 as Sound Generator (Sweep 2 kHz – 20 kHz):

- Start Adobe Audition 1.5.
- Load the file 35dBSweep2K_20K.
- Switch the Power Amplifier mains to On
- Test the signal (click the Play button).

Use the program Adobe Audition 1.5 as Sound Recorder:

- Connect the microphone out to mic1 on the MONACOR MMX 24 mixer.
- Start **a new session** of Adobe Audition 1.5
- Click the red Record Button.
- Set New Waveform I Sample Rate 44100 I Channels Mono I Resolution I 16-bit
  Wait to click OK!

Start Sound Generator and Sound Recorder:

- Switch to the session Sound Generator (use ALT + TAB).
- Click the Play Looped Button.
- Adjust the Volume Play Back Control.
- Click the Minimize Button.
- Switch to the session Sound Recorder.
- **Click OK** in the New Waveform on the Sound Recorder session.
- Adjust the Volume of the Recording Control.
- Click on the Stop Button.
- Click on the Untitled I Press the Del button I Save Changes I No
- Start a new record by clicking on the red Record Button.
- Record a Sweep I Stop I File I Save As ......

Analyze:

- **Filter:** Effects I Filters I Scientific Filters... I High Pass I Cutoff 1500 Hz
- **Zoom:** Keep Right Click down I drag on the time or dB scale.
- **Cut:** Keep Left Click down I drag on the curve I Click Del Button.
- **Frequency Analysis:** Alt + Z or Analyze I Show Frequency Analysis.

I) Questions to consider:

- Why is the transfer function so nonlinear? (The loudspeaker and the microphone are linear from 2 kHz to 20 kHz).

Exercise A2:

Clicks and reflections (echoes)

Use the program Adobe Audition 1.5 as Sound Generator):

- File I Close.
  File I Load the file Singleclick.
- Try to make a recording by using the same technique as described above.

II) Questions to consider:

- Is it possible to locate the reflective subjects based on the speed of sound in air (340 m/sec)?
- Suggestions to avoid echoes.

Exercise A3:

Directional characteristics

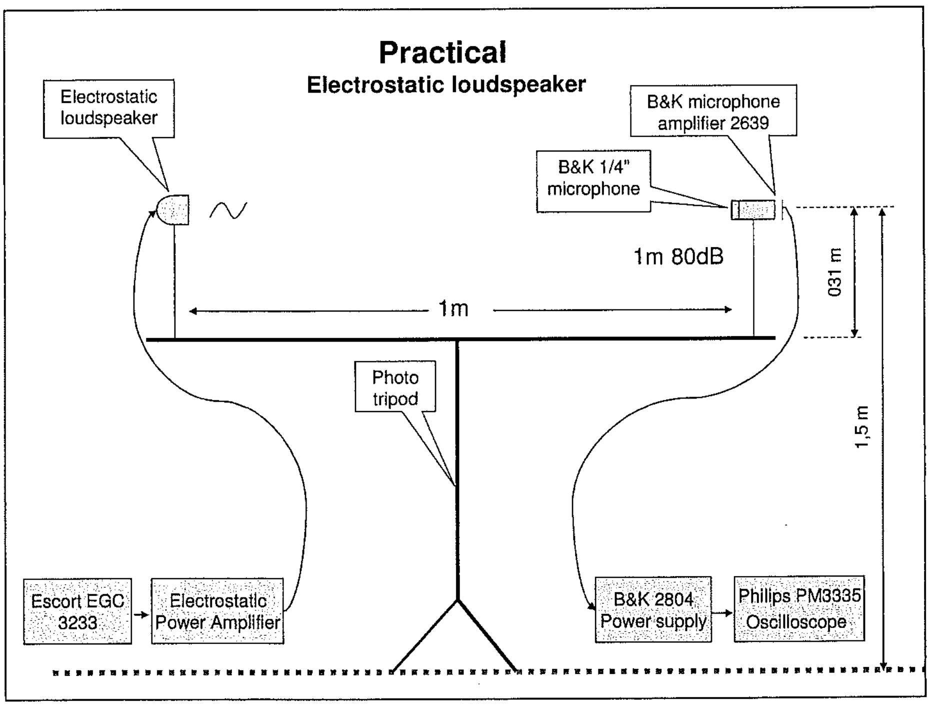
- File I Close.
  File I Load the file 15KzSinus.
- Try to make a Directional recording from 0 to 60 deg, in relevant steps.

Exercise A4:

Near field/far field boundary at 15 kHz

- Start at 1m and note the dB or pp value then halve the distance (be very careful doing this). Note the new value and proceed halving the distance until come within a few cm of the loudspeaker. Graph your results (See the B&K handbook pg. 19).

B) Electrostatic loudspeaker



Setup – Escort EGC-3233 Function Generator – Electrostatic Power Amplifier - Electrostatic Loudspeaker (60-1: -9dB re 1 Pa(RMS) at 20 kHz and 145 kHz, d = 1m) - B&K 4135 1/4' microphone – B&K 2639 preamplifier – B&K 2804 Microphone Power Supply –Philips PM3335 Oscilloscope.

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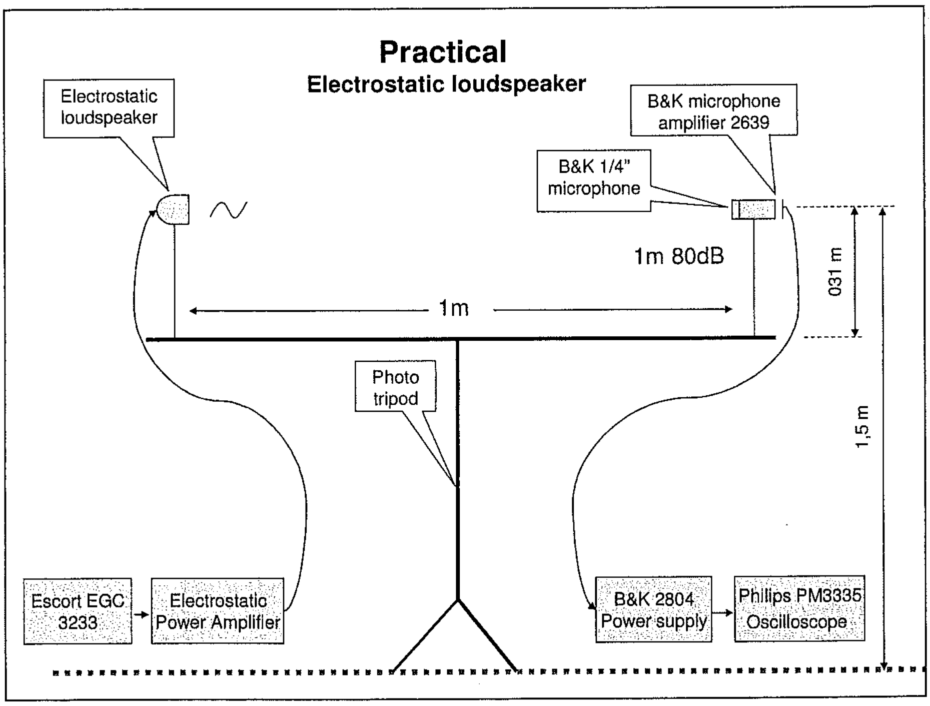

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Exercise B1:

Transfer function (frequency response) of the speaker from 20 kHz to 140 kHz

- Transfer function (frequency response) of the speaker from 20 kHz to 140 kHz in 10 kHz steps at 0° and 10° (if time permits). d = 1m. Plot your results.

Exercise B2:

- Directional characteristics at 25 kHz and 70 kHz in relevant steps to about 20°. For 70 kHz you will see narrow side lobes of less than 5°, but just note these, you don't need to measure them. Plot your results as polar plots or Cartesian coordinate plots.

Exercise B3:

- Near field/far field boundary at 30 kHz and 60 kHz: Start at 1m and note the dB or pp value then halve the distance (be very careful doing this). Note the new value and proceed halving the distance until come within a few cm of the loudspeaker. Graph your results (See the B&K handbook pg. 19).

Exercise B4:

Clicks

Setup – Change the signal generator to Electrostatic Spike Generator.

- Click generator: Measure the speed of sound (d = 1m) note the temperature and refer to your B&K handbook pg. 17. Use a click to examine the temporal characteristics of the received signal at 0° and 45°. The latter will illustrate a model of the sperm whale sound generator (see elsewhere in the course). Make a tracing of the temporal signals at 0° and 45°. If we sampled the signals on a high-speed A/D system we could also describe the spectral differences, but these are obvious from the temporal signal.

III) Questions to consider: from the individual setups and combined.

1) How do the frequency characteristics of the two types of loudspeakers compare when using the same technique? How do you explain the differences? (Take the diameters of the loudspeakers into consideration.) How do you account for the low frequency cutoff point (-3dB point) of the two types of loudspeakers?

2) How do you explain the differences between the directional characteristics of the electrostatic loudspeaker and the dynamic loudspeaker at the same frequency?

3) How does the physical radius of the sound source compare to that calculated from the formula: $\mathbf{r = 2\pi a^2 / \lambda}$? How do you explain the pressure variations measured close to the loudspeaker? How do you explain the difference in the distance of the near field:far field border for the electrostatic and dynamic loudspeakers?

IV) References:

Au, W.W.L. (1993) The sonar of dolphins. Springer-verlag, N.Y.
Beranek, L.L. (1954) Acoustics. McGraw-Hill, N.Y.
Borwick,J. (ed.) (1994) Loudspeaker and headphone handbook, 2nd. ed. Butterworth, Oxford

Brüel & Kjær (1985) Noise and vibration pocket handbook (reprinted with permission)
Frederiksen, E. (1977) Microphones used as sound sources. B&K Technical Review nr. 3, 1977

Bent Bach Andersen and Lee A. Miller July 2006