SCI220 – Foundations of Musical Acoustics
Cogswell Polytechnical College
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Week 5 – Class Notes

Loudness (FMA)

Chapter 13: The Loudness of Single and Combined Sounds

13.1 Thresholds of Hearing and Pain for a 1000 Hz Sinusoid

The ear, and like most microphones, is a pressure measuring device in which the eardrum is alternately pressed inward and pulled outward in response to the oscillatory fluctuation of pressure above and below the normal atmospheric pressure of a room. Under the oscillatory change in the minimum auditory pressure at the threshold of hearing, the eardrum moves back and forth a distance that is much less than the diameter of the atoms that constitute air.

We can sum up the following for our hearing thresholds:

1. At 1000Hz the minimum audible sound pressure at the threshold of hearing is about 1/3,530,000,000 of atmospheric pressure.
2. A 100-fold increase in the amplitude above the reference value brings us to the lower region of musical loudness, while a 10,000-fold increase in amplitude puts sound into the middle of the musical loudness range.
3. The threshold of pain is found to lie at pressure amplitudes 1,000,000 times larger than the reference value if our reference is 1000Hz.
4. The extent of the amplitude range for music shows that the enormous physical variation in amplitude does not manifest itself as an equally enormous variation in perceived loudness.

13.2 The Decibel Notation and Its Application to Acoustical Signals

The *decibel* is a term originally used to describe the way with the gain or loss of wave energy as it is transmitted successively through one member after another of a communication chain of telephone cables, switchboards, equalizing networks, repeater amplifiers, and the like. The decibel is not a quantity of sound energy, rather it is a way of expressing an energy flow relationship between any two signals.

It is interesting to note that our hearing process itself does not give a measure of acoustic energy. The energy involved in any vibrational process is proportional to the square of the amplitude of the vibration.

For example, a 2-fold increase in amplitude involves a 2x2 = 4-fold increase in the energy. Tripling the amplitude yields a 9-fold increase in energy and a 10-fold energy increase means a 3.162-fold increase of amplitude ($\sqrt{(10)}$). Now, if we say that there is a gain at the output of 10dB there must be a 10-fold difference between input and output signal energy and as a result, the amplitude of the output signal is 3.162 times larger than the input signal amplitude. If two signals are cascaded the resulting amplitude will produce a 10x10 = 100-fold energy increase, with an energy gain on 20dB. We can sum up this relationship with the following formula:

$$dB = 20\log_{10}(a/a0)$$

where a/a0 is our signal amplitude ratio. For example, a = 2 and a0 =1. a/a0 = 2 (2:1 ratio) and thus we obtain a 6.02dB gain.

The reference sound pressure amplitude for all acoustical measurements in air is $\sqrt{(2)}$ x 0.0002 = 0.000283 dyne/cm². The factor of $\sqrt{(2)}$ refers to the *RMS amplitude* (root mean square). The standard SPL reference used is identical to the 1/3,530,000,000 of atmospheric pressure.

If we have distinct sources A, B, C,… in a room (having the same or different frequencies) which have various pressure amplitudes p_a, p_b, p_c,… as measured at some point in the room, then the statistically averaged net pressure amplitude p_{net} at that point turns out to be equal to the square root of the sum of the squares of different amplitudes, as shown by the formula:

$$p_{net} = \sqrt{(pa^2 + pb^2 + pc^2 + ...)}$$

One must first convert each SPL to its corresponding pressure amplitude, combine these by the formula, and then find the net SPL.

If a weaker signal happens to arrive earlier than a stronger one (based on precedence effect), we will attribute all the sound to the source for the weaker signal.

13.3 Hearing and Pain Thresholds at Various Frequencies

Based on the threshold of hearing curve (fig.13.3, p.229) we can observe that the overall sensitivity of our ears falls rapidly as we test them at lower and lower frequencies. Also, our ears are more sensitive at frequencies around 4000Hz than to those around 1000Hz. The ripples shown in the sensitivity curve at high frequencies are produced partly by natural-frequency resonance effects in the listener's ear canals and partly by perception issues in our neurological processing when two sound pressure signals interact when received at the two sides of the listener's head.

A healthy ear is quite sensitive to frequencies ranging from 250Hz to 6000Hz; beyond these limits the sound pressure must be increased if anything is to be heard. However, for most adults a 10-fold signal amplification is needed at the signal amplitudes given by the curve. For younger people, the hearing curve corresponds quite well to that in fig.13.3. As one ages, the curve above 4000Hz rises more steeply so that the pressure amplitude needed for audibility at 10KHz may rise to values compared to those required at 20Hz.

The sound pressure amplitude that produces a feeling of pain and discomfort at 1000Hz is nearly the same as that level needed for all other frequencies. This amplitude is about 1,000,000 times the reference value, giving it an SPL of 120dB.

13.4 Variations in the Perceived Loudness of s Single-Component Sound: Sones

To measure human perception of sound we need to provide ourselves with a standardized unit of perceived loudness, the sone. If a listener with healthy hearing sits in a anechoic chamber facing a distant loudspeaker, he will hear a sound whose loudness is defined as 1 sone when a source having a frequency of 1000Hz produces an SPL of 40dB at his ear. In many situations loudness figures expressed in sones obey simple additive arithmetic (as opposed to decibels, which lie in a logarithmic scale).

Our perception of the loudness variations associated with uniform sound pressure excitation at our ears differs when the chosen sound pressure is high from that observed when it is low. Meaning that it will be barely able to perceive the loudness peaks and dips when the a test sound (with gradually changing frequency) is softly played than, where on the other hand when the test sound is considerably louder it will be possible to perceive those changes in loudness.

13.5 Loudness of Combined Single-Component or Narrow-Band Noise Signals Having Identical or Different Pitches

The single source and multiple-source curves (from fig 13.5) differ because of the statistical way in which the sound pressures from several sources combine (as shown by the square root formula in section 13.2). It is seen that loudness grows only slowly as more and more equal sources are brought in. For example, it takes 10 sources sounding simultaneously in order to double the overall loudness, and it will take 100 sources to quadruple it.

If we consider a signal composed by a large number of sinusoidal components whose frequencies were randomly chosen, but limited to 10% of a particular center frequency (narrow-band noise signal), we will notice that the total SPL level will be equal to that of a signal with a single sinusoid component that matches the center frequency of the noise signal.

When two signals have center frequencies that are two or more octaves apart, the loudness of each signal is arithmetically added. For example, a signal of 13 sones is played simultaneously with another of 5 sones will produce an overall loudness of 18 sones. However, if the center frequencies are spaced by an octave the total loudness will be decreased as prescribed by the curve in fig.13.6. For example, two signals with 13 sones each will produce a total loudness of 24 sones.

If two frequencies of the two groups are initially made identical and are then progressively separated, there will not be a change in loudness until they are more that 3 semitones (a minor 3rd) apart. The frequency range within which a pair of sinusoidal groups must lie if they are to show these lumping together effects is known as the *critical bandwidth*. This melting together of the loudness perception of signals having more or less the same frequency (and within a critical bandwidth) explains why we could replace a single sinusoid by a narrow-band noise signal of the same combined SPL.

Assuming that sound pressure measurements are already made of the various noise-partials, and that each one of the harmonically related narrow-band noise components have been converted into sones. In therm of these loudness S_1, S_2, S_3,... the total loudness S_{tnp} of the collection of noise partials can be calculated with the formula:

$$S_{tnp} = S_1 + 0.75S_2 + 0.50S_3 + 0.50S_4 + 0.30S_5 + 0.20(S_6 + S_7 + S_8)$$

If we were to have a collection of five 13-sone components we would have a total loudness of 39.65 sones. Meaning that the five equally loud components are predicted to be 3 times as loud as any one of its components. This formula assumes that the first fundamental frequency is the loudest of the set, and that the loudnesses of the second, third, fourth, fifth, and sixth noise-partials do not fall below the proportions of $S_1/4$, $S_1/6$, $S_1/8$, and $S_1/10$ when compared with the fundamental frequency component. If they decrease more rapidly, some of the will have little strength and will be masked by the lower frequencies. Finally, the loudness of the 6th and higher components are relatively small.

13.6 The Combined Loudness of Two or More Sinusoids; Relationships Advertised by Beats

Beating is defined as the alternate increase and decrease of the amplitude of motion of a single object that is acted upon by a pair of sinusoidal forces having somewhat different frequencies. The rate at which the resulting oscillatory amplitude changes is called the beat frequency.

Whenever two of the sinusoids are in-phase, the ear is momentarily provided with an acoustical signal whose pressure amplitude is double that which either component can produce by itself, hence there is an increase of loudness of almost 50%. When the two signals fall progressively out-of-phase, the loudness will diminish until they cancel each other out and produce a sound level far below the hearing range; meaning we would have momentary periods of silence interspersed by swelling of sound.

Beats within signals of equally large magnitudes will prove to be more obtrusive than those between of weaker signals, simply because the range between the maximum loudness and silence is far greater in the first case than in the other. Near the threshold of hearing the beating may again be more clearly heard because of the altered way in which our ears relate loudness to sound pressure in this region of soft sounds.

In a room, the fluctuation phenomena will increase its loudness, both because the sounds irregularities serve to call attention to them and because various aspects of the sounds are made manifest one after the other in some random sequence. Meaning that because of the presence of room-caused fluctuations in the sound pressure amplitude of all components the phenomenon of masking is greatly reduced.

If we were to ignore the effects of masking, we find a new formula that helps us calculate the loudness of a sound that is made up of sinusoidal components whose frequencies are wide enough separated to avoid beating.

The formula follows:

$$S_{tnp} = S_1 + S_2 + S_3 + S_4 + S_5 + \dots$$

The difference between this formula an our previous one, is that the total loudness arises by giving equal weight to all component contributions rather than decreasing the importance of the higher frequency components.

13.8 The Sound Level Meter

The sound level meter is a compact device having the acoustical function somewhat analogous to that of a light meter. The dial reading is expressed in decibels.

Sound meters do not give a true measure of loudness, because they have no way to take care of masking or beats. Even if its weighting networks were able to take proper account of the way in which various sound components interact to determine the loudness of a sound, the use of the decibel scale causes problems. This is due to the fact that equal increases in decibel readings do not at all produce equal increases in perceived loudness.

In the simplest case, the numerical measure of loudness is obtained by the meter reading and then using the curves from fig.13.4 to convert the readings into sones. In fact, this method does work quite well for single sinusoids and/or their equivalent noise-partials.

In the more complex case, many sounds (for example, environmental and speech) are composed by thousands of randomly arranged sinusoidal partials whose measured amplitudes do not vary abruptly as we shift one frequency setting to another one. For *broad-band continuous-spectrum* sounds a sound meter can obtain reasonable results as a loudness measuring device is the weighting curves and used in conjunction with the curves in fig.13.10.

APPENDIX

Energy and Intensity

Power: $P = E/t$, where E is the energy received by a sound detector and t is time.
Measured in Watts, 1 W = 1 J/s

Intensity: $I = P/S = E/St$, where S is surface area.

$(I1/I2) = (A1/A2)^2$, A is pressure amplitude. For sine waves, $I = A^2/2\varrho c$. Where ϱ is the mass density, and c is the speed of sound.

Sound Intensity Level (SIL, in dB), $SIL = 10\log(I/I0)$. $I0 = 10^{-12}\ W/m^2$

Sound Pressure Level (SPL, in dB), $SPL = 20\log(p/p0)$. $Po = 0.0002\ N/m^2$ (RMS)

The Inverse Square Law

In a free field, the sound level will decrease with increasing distance from the source, as described by:

$$I1/I2 = (r1/r2)^2 \text{ or } SIL_2 - SIL_1 = 20\log(r1/r2)$$

where r1 is our initial position with respect to the sound source, and r2 the final distance from the source. Both distances are seen as the radius of a sphere.