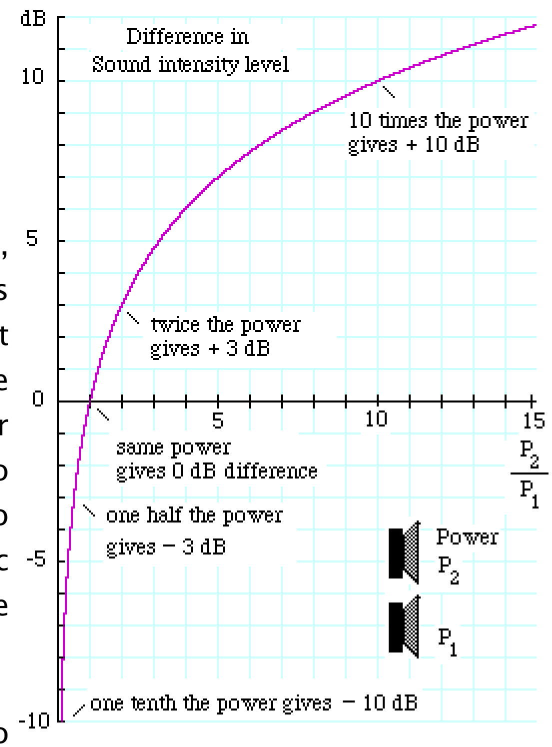
What is a decibel?

And what are the different types of decibel measurement: dB, dBA, dBC, dBV, dBm and dBi? How are they related to loudness? (A related page allows you to measure yourhearing response and to compare with standard hearing curves.)



Plot of 10 log (P_2/P_1)

Definition and examples

The decibel (**dB**) is used to measure sound level, but it is also widely used in electronics, signals and communication. The dB is a logarithmic unit used to describe a ratio. The ratio may be power, sound pressure, voltage or intensity or several other things. Later on we relate dB to the **phon** and the **sone** (other units related to loudness). But first, to get a taste for logarithmic units, let's look at some numbers. (If you have forgotten, go to What is a logarithm?)

For instance, suppose we have two loudspeakers, the first playing a sound with power P_1, and another playing a louder version of the same sound with power P_2, but everything else (how far away, frequency) kept the same.

The **difference in decibels** between the two is defined to be

$10 \log (P_2/P_1)$ dB where the log is to base 10.

If the second produces twice as much power than the first, the difference in dB is

$10 \log (P_2/P_1) = 10 \log 2 = 3$ dB.

If the second had 10 times the power of the first, the difference in dB would be

$10 \log (P_2/P_1) = 10 \log 10 = 10$ dB.

If the second had a million times the power of the first, the difference in dB would be

$10 \log (P_2/P_1) = 10 \log 1000000 = 60$ dB.

This example shows one feature of decibel scales that is useful in discussing sound: they can describe very big ratios using numbers of modest size. But note that the decibel describes a *ratio*: so far we have not said what power either of the speakers radiates, only the ratio of powers. (Note also the factor 10 in the definition, which puts the 'deci' in decibel).

Sound pressure, sound level and dB. Sound is usually measured with microphones and they respond (approximately) proportionally to the sound pressure, p. Now the power in a sound wave, all else equal, goes as the square of the pressure. (Similarly, electrical power in a resistor goes as the square of the voltage.) The log of the square of x is just 2 log x, so this introduces a factor of 2 when we convert to decibels for pressures. The difference in sound pressure level between two sounds with p_1 and p_2 is therefore:

$20 \log (p_2/p_1)$ dB = $10 \log (p_2^2/p_1^2)$ dB = $10 \log (P_2/P_1)$ dB where again the log is to base 10.

Sound files to show the size of a decibel

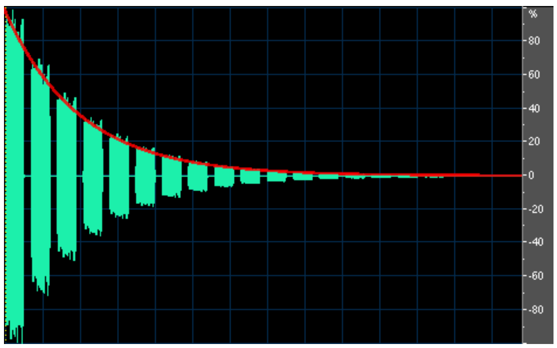
What happens when you halve the sound power? The log of 2 is 0.3, so the log of 1/2 is −0.3. So, if you halve the power, you reduce the power and the sound level by 3 dB. Halve it again (down to 1/4 of the original power) and you reduce the level by another 3 dB. That is exactly what we have done in the first graphic and sound file below.

How big is a decibel? In the next series, successive samples are reduced by just one decibel.

One decibel is close to the Just Noticeable Difference (JND) for sound level. As you listen to these files, you will notice that the last is quieter than the first, but it is rather less clear to the ear that the second of any pair is quieter than its predecessor. $10 * \log_{10}(1.26) = 1$, so to increase the sound level by 1 dB, the power must be increased by 26%, or the voltage by 12%.

What if the difference is less than a decibel? Sound levels are rarely given with decimal places. The reason is that sound levels that differ by less than 1 dB are hard to distinguish, as the next example shows.

You may notice that the last is quieter than the first, but it is difficult to notice the difference between successive pairs. $10*\log_{10}(1.07) = 0.3$, so to increase the sound level by 0.3 dB, the power must be increased by 7%, or the voltage by 3.5%.



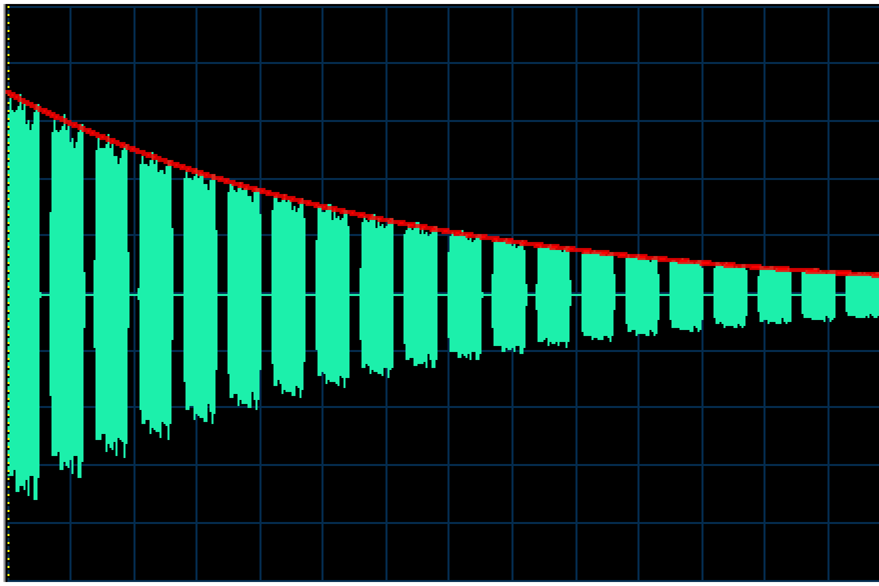
Broadband noise decreasing by 3 dB steps.

The first sample of sound is white noise (a mix of all audible frequencies, just as white light is a mix of all visible frequencies). The second sample is the same noise, with the voltage reduced by a factor of the square root of 2. 2-0.5 is approximately 0.7, so -3 dB corresponds to reducing the voltage or the pressure to 70% of its original value. The green line shows the voltage as a function of time. The red line shows a continuous exponential decay with time. Note that the voltage falls by 50% for every second sample.

Note, too, that a doubling of the power does not make a huge difference to the loudness. We'll discuss this further below, but it's a useful thing to remember when choosing sound reproduction equipment.

Sound files and graph by John Tann and George Hatsidimitris.

How big is a decibel? In the next series, successive samples are reduced by just one decibel.

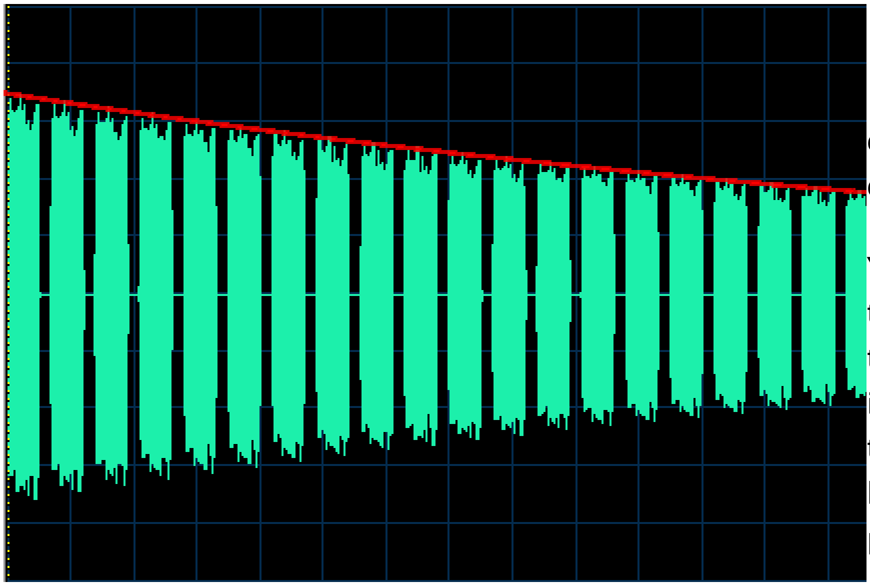


wav mp3

Broadband noise decreasing by 1 dB steps.

One decibel is close to the Just Noticeable Difference (JND) for sound level. As you listen to these files, you will notice that the last is quieter than the first, but it is rather less clear to the ear that the second of any pair is quieter than its predecessor. 10*log10(1.26) = 1, so to increase the sound level by 1 dB, the power must be increased by 26%, or the voltage by 12%.

What if the difference is less than a decibel? Sound levels are rarely given with decimal places. The reason is that sound levels that differ by less than 1 dB are hard to distinguish, as the next example shows.



wav mp3

Broadband noise decreasing by 0.3 dB steps.

You may notice that the last is quieter than the first, but it is difficult to notice the difference between successive pairs.
10*log10(1.07) = 0.3, so to increase the sound level by 0.3 dB, the power must be increased by 7%, or the voltage by 3.5%.

Standard reference levels ("absolute" sound level)

When the decibel is used to give the sound level for a single sound rather than a ratio, then a reference level must be chosen. For sound intensity, the reference level (for air) is usually chosen as 20 micropascals, or 0.02 mPa. (This is very low: it is 2 ten billionths of an atmosphere. Nevertheless, this is about the limit of sensitivity of the human ear, in its most sensitive range of frequency. Usually this sensitivity is only found in rather young people or in people who have not been exposed to loud music or other loud noises. Personal music systems with in-ear speakers ('walkmans') are capable of very high sound levels in the ear, and are believed by some to be responsible for much of the hearing loss in young adults in developed countries.)

So if you read of a sound pressure level of 86 dB, it means that

$20 \log (p_2/p_1) = 86$ dB

where p_1 is the sound pressure of the reference level, and p_2 that of the sound in question. Divide both sides by 20:

$\log (p_2/p_1) = 4.3$

$p_2/p_1 = 10^{4.3}$

4 is the log of 10 thousand, 0.3 is the log of 2, so this sound has a sound pressure 20 thousand times greater than that of the reference level ($p_2/p_1 = 20{,}000$). 86 dB is a loud but not dangerous level of sound, if it is not maintained for very long.

What does 0 dB mean? This level occurs when the measured intensity is equal to the reference level. i.e., it is the sound level corresponding to 0.02 mPa. In this case we have

sound level = $20 \log (p_{measured}/p_{reference}) = 20 \log 1 = 0$ dB

Remember that decibels measure a ratio. 0 dB occurs when you take the log of a ratio of 2. So 0 dB does not mean no sound, it means a sound level where the sound pressure is equal to that of the reference level. This is a small pressure, but not zero. It is also possible to have negative sound levels: - 20 dB would mean a sound with pressure 10 times smaller than the reference pressure, ie 2 micropascals.

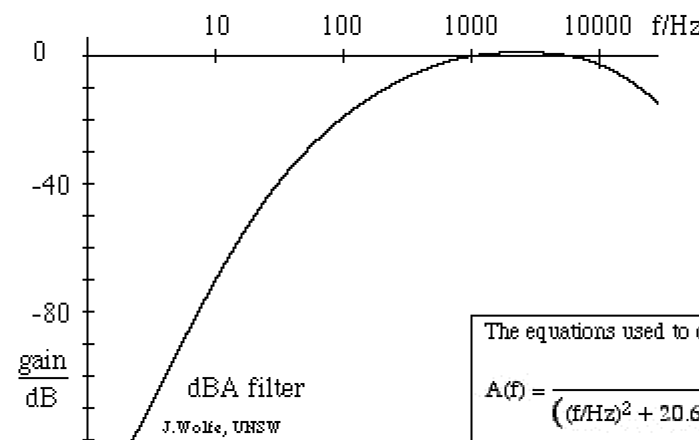
Not all sound pressures are equally loud. This is because the human ear does not respond equally to all frequencies: we are much more sensitive to sounds in the frequency range about 1 kHz to 4 kHz (1000 to 4000 vibrations per second) than to very low or high frequency sounds. For this reason, sound meters are usually fitted with a filter whose response to frequency is a bit like that of the human ear. (More about these filters below.) If the "A weighting filter" is used, the sound pressure level is given in units of **dB(A)** or **dBA**. Sound pressure level on the dBA scale is easy to measure and is therefore widely used. It is still different from**loudness**, however, because the filter does not respond in quite the same way as the ear. To determine the loudness of a sound, one needs to consult some curves representing the frequency response of the human ear, given below. (Alternatively, you can measure your own hearing response.

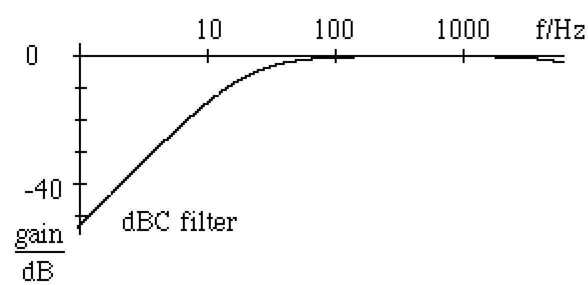
Logarithmic response, psychophysical measures, sones and phons

Why do we use decibels? The ear is capable of hearing a very large range of sounds: the ratio of the sound pressure that causes permanent damage from short exposure to the limit that (undamaged) ears can hear is more than a million. To deal with such a range, logarithmic units are useful: the log of a million is 6, so this ratio represents a difference of 120 dB. Psychologists also say that our sense of hearing is roughly logarithmic (see under sones below). In other words, they think that you have to increase the sound intensity by the same factor to have the same increase in loudness. Whether you agree or not is up to you, because this is a rather subjective question.

The filters used for dBA and dBC

The most widely used sound level filter is the A scale, which roughly corresponds to the inverse of the 40 dB (at 1 kHz) equal-loudness curve. Using this filter, the sound level meter is thus less sensitive to very high and very low frequencies. Measurements made on this scale are expressed as dBA. The C scale is practically linear over several octaves and is thus suitable for subjective measurements only for very high sound levels. Measurements made on this scale are expressed as dBC. There is also a (rarely used) B weighting scale, intermediate between A and C. The figure below shows the response of the A filter (left) and C filter, with gains in dB given with respect to 1 kHz. (For an introduction to filters, see RC filters, integrators and differentiators.)





The equations used to calculate these plots are:

$$A(f) = \frac{12200^2\,(f/\mathrm{Hz})^4}{\left((f/\mathrm{Hz})^2 + 20.6^2\right)\left((f/\mathrm{Hz})^2 + 12200^2\right)\sqrt{(f/\mathrm{Hz})^2 + 107.7^2}\;\sqrt{(f/\mathrm{Hz})^2 + 737.9^2}}$$

$$C(f) = \frac{12200^2\,(f/\mathrm{Hz})^2}{\left((f/\mathrm{Hz})^2 + 20.6^2\right)\left((f/\mathrm{Hz})^2 + 12200^2\right)}$$

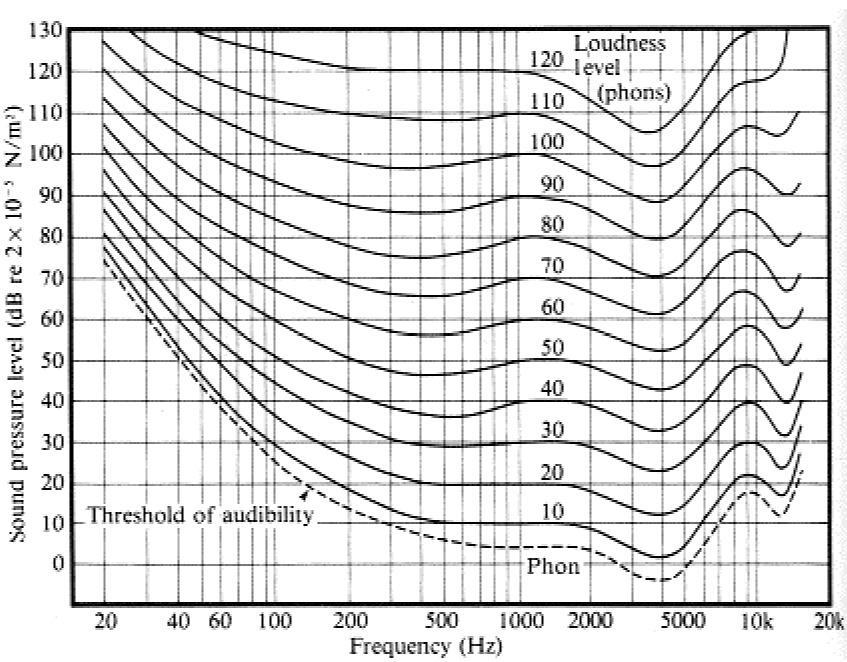
$$\beta_{dBA} = \beta_{dB} + 20\log_{10}\frac{A(f)}{A(1000\ \mathrm{Hz})} = \beta_{dB} + 20\log_{10}\frac{A(f)}{0.794}$$

$$\beta_{dBC} = \beta_{dB} + 20\log_{10}\frac{C(f)}{C(1000\ \mathrm{Hz})} = \beta_{dB} + 20\log_{10}\frac{C(f)}{0.993}$$

On the music acoustics and speech acoustics sites, we plot the sound spectra in dB. The reason for this common practice is that the range of measured sound pressures is large.

Loudness, phons and sones

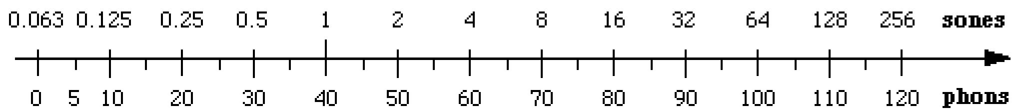
The **phon** is a unit that is related to dB by the *psychophysically measured* frequency response of the ear. At 1 kHz, readings in phons and dB are, by definition, the same. For all other frequencies, the phon scale is determined by the results of experiments in which volunteers were asked to adjust the loudness of a signal at a given frequency until they judged its loudness to equal that of a 1 kHz signal. To convert from dB to phons, you need a graph of such results. Such a graph depends on sound level: it becomes flatter at high sound levels.



Curves of equal loudness determined experimentally by Robinson & Dadson in 1956, following the original work of Fletcher & Munson (Fletcher, H. and Munson, W.A. (1933) J.Acoust.Soc.Am. 6:59; Robinson, D.W. and Dadson, R.S.(1956) Br. J. Appl. Phys. 7:166. Plots of equal loudness as a function of frequency are often generically called Fletcher-Munson curves.)

The **sone** is derived from psychophysical measurements which involved volunteers adjusting sounds until they judge them to be twice as loud. This allows one to relate perceived loudness to phons. A sone is defined to be equal to

40 phons. **Experimentally it was found that a 10 dB increase in sound level corresponds approximately to a perceived doubling of loudness.** So that approximation is used in the definition of the phon: 0.5 sone = 30 phon, 1 sone = 40 phon, 2 sone = 50 phon, 4 sone = 60 phon, etc.



Wouldn't it be great to be able to convert from dB (which can be measured by an instrument) to sones (which approximate loudness as perceived by people)? This is usually done using tables that you can find in acoustics handbooks. However, if you don't mind a rather crude approximation, you can say that the A weighting curve approximates the human frequency response at low to moderate sound levels, so dBA is *very roughly* the same as phons. Then use the logarithmic relation between sones and phons described above.

Recording level and decibels

Meters measuring recording or output level on audio electronic gear (mixing consoles etc) are almost always recording the AC rms voltage. For a given resistor R, the power P is V^2/R, so

difference in voltage level = $20 \log (V_2/V_1)$ dB = $10 \log (V_2^2/V_1^2)$ dB = $10 \log (P_2/P_1)$ dB, or

absolute voltage level = $20 \log (V/V_{ref})$

where V_{ref} is a reference voltage. So what is the reference voltage?

The obvious level to choose is one volt rms, and in this case the level is written as **dBV.** This is rational, and also convenient with modern analog-digital cards whose maximum range is often about one volt rms. So one has to remember to the keep the level in negative dBV (less than one volt) to avoid clipping the peaks of the signal, but not too negative (so your signal is still much bigger than the background noise).

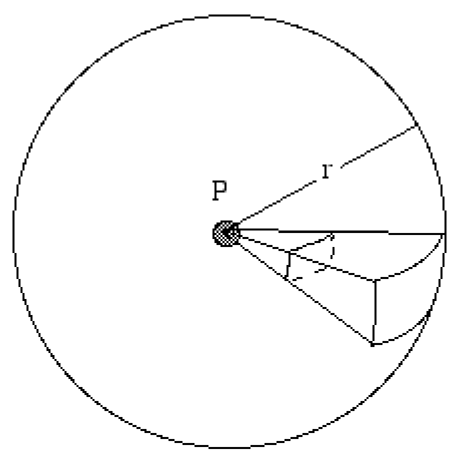
Sometimes you will see **dBm.** This used to mean decibels of electrical power, with respect to one milliwatt, and sometimes it still does. However, it's complicated for historical reasons. In the mid twentieth century, many audio lines had a nominal impedance of 600 W. If the impedance is purely resisitive, and if you set $V^2/600$ W =

1 mW, then you get V = 0.775 volts. So, providing you were using a 600 W load, 1 mW of power was 0 dBm was 0.775 V, and so you calibrated your level meters thus. The problem arose because, once a level meter that measures voltage is calibrated like this, it will read 0 dBm at 0.775 V even if it is not connected to 600 W So, perhaps illogically, dBm will sometimes mean dB with respect to 0.775 V. (When I was a boy, calculators were expensive so I used dad's old slide rule, which had the factor 0.775 marked on the cursor window to facilitate such calculations.)

How to convert dBV or dBm into dB of sound level? There is no simple way. It depends on how you convert the electrical power into sound power. Even if your electrical signal is connected directly to a loudspeaker, the conversion will depend on the efficiency and impedance of your loudspeaker. And of course there may be a power amplifier, and various acoustic complications between where you measure the dBV on the mixing desk and where your ears are in the sound field.

Intensity, radiation and dB

How does sound level (or radio signal level, etc) depend on distance from the source?



A source tht emits radiation equally in all directions is called **isotropic**. Consider an isolated source of sound, far from any reflecting surfaces -- perhaps a bird singing high in the air. Imagine a sphere with radius r, centred on the source. The source outputs a total power P, continuously. This sound power spreads out and is passing through the surface of the sphere. If the source is isotropic, the intensity I is the same everywhere on this surface, by definition. The **intensity I is defined as the power per unit area.** The surface area of the sphere is $4pr^2$, so the power (in our example, the sound power) passing through each square metre of surface is, by definition:

$$I = P/4pr^2.$$

So we see that, for an isotropic source, intensity is inversely proportional to the square of the distance away from the source:

$I_2/I_1 = r_1^2/r_2^2$.

But intensity is proportional to the square of the sound pressure, so we could equally write:

$p_2/p_1 = r_1/r_2$.

So, if we double the distance, we reduce the sound pressure by a factor of 2 and the intensity by a factor of 4: in other words, we reduce the sound level by 6 dB. If we increase r by a factor of 10, we decrease the level by 20 dB, etc.

Be warned, however, that many sources are not isotropic, especially if the wavelength is smaller than, or of a size comparable with the source. Further, reflections are often quite important, especially if the ground is nearby, or if you are indoors.

dBi and radiation that varies with direction

Radiation that varies in direction is called **anisotropic.** For many cases in communication, isotropic radiation is wasteful: why emit a substantial fraction of power upwards if the receiver is, like you, relatively close to ground level. For sound of short wavelength (including most of the important range for speech), a megaphone can help make your voice more anisotropic. For radio, a wide range of designs allows antennae to be highly anisotropic for both transmission and reception.

So, when you interested in emission in (or reception from) a particular direction, you want the ratio of intensity measured in that direction, at a given distance, to be higher than that measured at the same distance from an isotropic radiator (or received by an isotropic receiver). This ratio is called the **gain**; express the ratio in dB and you have the gain in **dBi** for that radiator. This unit is mainly used for antennae, either transmitting and receiving, but it is sometimes used for sound sources (and directional microphones).

Example problems

A few people have written asking for examples in using dB in calculations. So...

- All else equal, how much louder is loudspeaker driven (in its linear range) by a 100 W amplifier than by a 10 W amplifier?

The powers differ by a factor of ten, which, as we saw above, is 10 dB. All else equal here means that the frequency responses are equal and that the same input signal is used, etc. So the frequency dependence should be the same. 10 dB corresponds to 10 phons. To get a perceived doubling of loudness, you need an increase of 10 phons. So the speaker driven by the 100 W amplifier is twice as loud as when driven by the 10 W, assuming you stay in the linear range and don't distort or destroy the speaker. (The 100 W amplifier produces twice as many sones as does the 10 W.)

- If, in ideal quiet conditions, a young person can hear a 1 kHz tone at 0 dB emitted by a loudspeaker (perhaps a softspeaker?), by how much must the power of the loudspeaker be increased to raise the sound to 110 dB (a dangerously loud but survivable level)?

  The difference in decibels between the two signals of power P_2 and P_1 is defined above to be

  > DL = 10 log (P_2/P_1) dB   so, raising 10 to the power of these two equal quantities:
  > $10^{L/10} = P_2/P_1$ so:
  > $P_2/P_1 = 10^{110/10} = 10^{11}$ = one hundred thousand million.

  which is a demonstration that the human ear has a remarkably large dynamic range, perhaps 100 times greater than that of the eye.

- An amplifier has an input of 10 mV and and output of 2 V. What is its voltage gain in dB?

  Voltage, like pressure, appears squared in expressions for power or intensity. (The power dissipated in a resistor R is V^2/R.) So, by convention, we define:

  $$\begin{aligned} \text{gain} &= 20 \log (V_{out}/V_{in}) \\ &= 20 \log (2\text{V}/10\text{mV}) \\ &= 46 \text{ dB} \end{aligned}$$

  (In the acoustic cases given above, we saw that the pressure ratio, expressed in dB, was the same as the power ratio: that was the reason for the factor 20 when defining dB for pressure. It is worth noting that, in the voltage gain example, the power gain of the ampifier is unlikely to equal the voltage gain. The power is proportional to the square of the voltage in a given resistor. However, the input and output impedances of amplifiers are often quite different. For instance, a buffer amplifier or emitter follower has a voltage gain of about 1, but a large current gain.)

Source: http://www.co-bw.com/Audio_what_is_decible.htm