Specifications – Removing the Mystery
Chapter 3 – Sensitivity – It's Like Gain Only Different

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The concept of gain is fairly straightforward, but in order to measure or calculate gain, you need to have the same thing coming out as what went in – volts in and volts out, amps in, amps out, dollars in, dollars out. However, with many audio devices, what comes out is in a different from than what went in, even though in some (but not all) cases, both the input and output are electrical. A device that converts one form of energy to another is called a *transducer.*

Microphones convert air pressure to voltage. Loudspeakers do the opposite, using electrical power to move air. An analog-to-digital or digital-to-analog converter converts between volts and bits. A power amplifier converts volts to watts. So how do we express the relationship between a transducer's input and output in a way that's as meaningful and handy as gain? It's called *sensitivity*. Let's look at a few devices where it applies.

Microphones

A microphone's sensitivity is specified in terms of its output voltage for a given sound pressure level (SPL). Modern microphone manufacturers usually specify SPL in Pascals (Pa), the SI (International System of Units) unit of pressure. It's like pounds per square inch in your tires, but a more conveniently sized unit for dealing with small changes in pressure. 1 pascal is equivalent to 94 dB SPL, about the volume level of a tolerably loud concert.

A microphone's output is usually measured in millivolts RMS, unloaded (open circuit), though in the spec sheet it's often expressed as dBV or dBu. In real life, with the mic loaded by a mic preamp's typical input impedance of around 2,000 ohms, its output level will drop about 1 dB from the lab measurement. The measurement is made in an anechoic chamber or free field environment (no nearby reflective surfaces), with the source on axis to the mic. Directional mics are usually measured at 1 kHz while omni mics are sometimes measured at 250 Hz.

Modern mics specify sensitivity as dBV or mV per Pa, though when perusing vintage mic specs, you might find SPL in dynes/cm^2 (dynes per square centimeter) or µbars (microbars, from "barometric pressure"). 1 dyne/cm^2 = 1 µbar = 1/10 Pa = 74 dB SPL.

The sensitivity spec for the popular Shure SM57 is: "-56 dBV/Pa (1.6 mV)." The classic RCA 77-DX is specified as "Uni-directional: -50 dBm" with a footnote that the reference SPL is 10 dynes/cm^2. Since it states that the output is unloaded, hence delivering a negligible amount of power, I'm guessing that "dBm" is what we now call dBu (2.4 mV).

This handy on-line calculator will help you to compare mics specified in different units:

http://www.sengpielaudio.com/calculator-transferfactor.htm

A mic with high sensitivity isn't necessarily better than a low sensitivity one. Sensitivity is a characteristic that gives us guidance as to how the mic is best used, and much gain we'll need in order to make it work in a given application.

Loudspeakers – Passive and Active

Loudspeakers, which convert electrical energy to sound pressure, have a sensitivity specification similar to that of microphones, only in reverse. Loudspeaker sensitivity is usually specified as SPL for a given power, measured on axis at a distance of 1 meter or 1 foot. The specification is usually written as "91 dB, 1 W, 1 meter." SPL is usually measured at a single frequency, typically 1 kHz for a full range speaker, with more appropriate frequencies used when measuring limited-range speakers such as super-tweeters or subwoofers,

Two speakers that have similar sensitivity, driven by the same amplifier and playing the same program material can differ considerably in subjective loudness. Two factors that influence this are the speaker's directivity and how its impedance varies with frequency ("8 ohms" is only nominal).

In a powered (active) speaker system, the input is to an amplifier rather than a loudspeaker, so the sensitivity is specified as input voltage (dBu) for a given SPL. Powered speakers usually have an input sensitivity control to accommodate a range of inputs, so sensitivity is often specified at the maximum setting or at one of the de facto standard nominal operating levels of +4 dBu or –10 dBV. Another important specification is SPL with the amplifier(s) delivering their maximum power, which is usually limited to avoid damage to the amplifier or speakers. A typical powered speaker's spec might read:

Input sensitivity: +4 dBu, 94 dB SPL @ 1m
Maximum continuous SPL: 106 dB @ 1m
Peak SPL: 112 dB @ 1m

This tells you that with the input sensitivity control (if there's one) set for +4 dBu (there's often a detent at that position) it'll reach its rated SPL at an input level of +16 dBu [+4 dBu + (106-94) dB] and can tolerate occasional input peaks of +22 dBu. However, if your mixer's maximum output is +24 dBu (and we all run them up to the red line sometimes), backing off the speaker's input sensitivity control by a couple of dB will keep all of your mixer's headroom.

This analysis can be extended to power amplifiers, which are usually specified as maximum power into a specified impedance, as well as how many volts at the input are required for a given amount of power. Do the math and set up your system accordingly.

Earphones

I addressed earphone sensitivity in the July 2014 issue of Recording, so I won't repeat that treatise here other than with regard to published specs. Since we attach 'phones directly to our ears, the SPL is measured as close as possible to the diaphragm rather than 1 meter away. Also, because earphones are low power devices, their input specification is in milliwatts rather than watts.

When people complained that their older earphones weren't loud enough when driven by their low voltage phones and tablets, lower impedance earphones started coming to market, with sensitivity as the SPL at 1 volt rather than the power necessary to achieve a specified SPL.

A spec for classic earphones will read something like this:
"Sensitivity: 94 dB @ 50 mW, Impedance: 60 ohms"

For the newfangled 'phones tailored for phones:
"Sensitivity: 106 dB @ 1V"

Digital Audio Interfaces

The spec sheet for these devices should include sensitivity, but it rarely does. When the input is a mic, instrument, or line level analog source and the output is USB, Thunderbolt, Ethernet, S/PDIF, or AES3, you're not comparing apples in to apples out, and a conversion between signal formats is involved. The output of an A/D converter is a 16- or 24-bit digital word, with all bits on (for example: 11111111111111) representing the largest number it can handle. This is known as "full scale" and is the reference for digital level in dB. Unlike for analog devices where the reference level is +4 dBu and operating levels can go above or below that, for digital devices, 0 dBFS (dB referenced to full scale) is the highest level possible - levels below full scale are expressed as negative values below 0 dBFS. An A/D converter works in reverse. A recording at 0 dBFS produces the maximum analog output level.

So, how many volts in does it take to reach 0 dBFS? And conversely, how many volts do you get out when playing 0 dBFS recording? Here are three important things to understand about digital sensitivity:

- There is no headroom above 0 dBFS in a digital system.
- There is no standard for the relationship between volts and bits.
- Input sensitivity is fixed on some devices and adjustable on others.

The meter on early digital recorders (DATs, etc.), usually had a mark at -20, -18, -16, or even -12 dBFS. Setting the eyeball average level to that mark allowed its respective number of dB of headroom of headroom. But you could make as much or as little headroom as you wanted.

Today's computer audio interfaces are built around A/D and/or D/A converter chips that have a fixed sensitivity (that will be found on the spec sheet for the chip), with the manufacturer adding gain or attenuation to turn it into a useful product. Mic inputs have a variable gain preamp ahead of the A/D converter, so the preamp gain control doubles as the sensitivity adjustment. Sometimes you'll find a specification like: "Mic Input (maximum gain): -56 dBu = 0 dBFS" though more often than not, the mic preamp gain will be specified.

Fixed gain line inputs are usually specified relative to a nominal operating level: "+4 dBu = -16 dBFS" or are referenced to full scale: "+20 dBu = 0 dBFS" (these are equivalent). This is the genesis of the vague input specification of "-16 dBFS"

Interfaces nearly always have an output level control, with the maximum output level at 0 dBFS ("0 dBFS = +22 dBu") sometimes specified. This can be helpful when setting up your monitoring system.

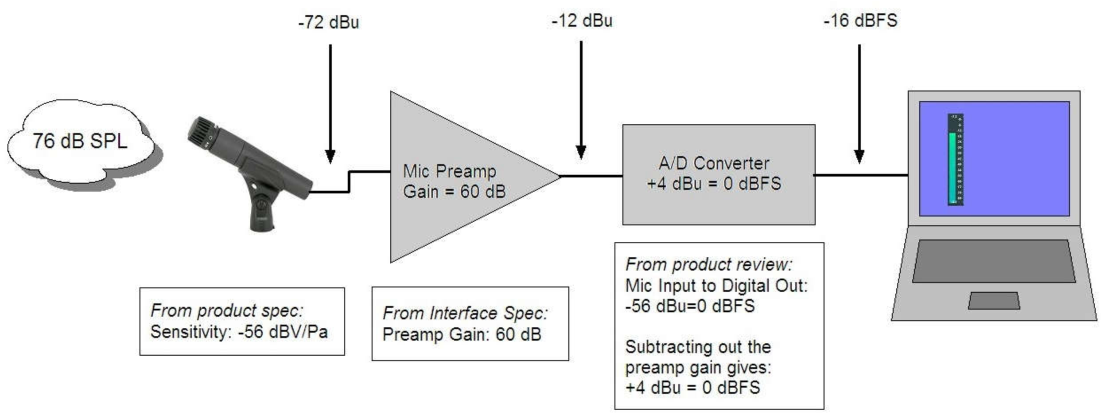
Putting It All Together

Let's look at a typical signal chain to see if we can get from here to there with the help of a spec sheet. In this example, our job is to make a spoken word recording using an SM57 mic connected to a computer through a typical USB interface.

Our first task is to figure out how many volts will come out of the mic when our narrator is talking. Using my best narrator voice, I became the "calibrated source" by speaking about 6 inches from my SPL meter. The eyeball average reading was 76 dB SPL. According to the spec sheet, the SM57 has a sensitivity of -56 dBV/Pa (94 dB SPL). Since my measured SPL is 18 dB (94 – 76) below the reference level, we can expect the mic's output to be 18 dB below that at the reference SPL, or –74 dBV (-56 – 18). This converts to -72 dBu.

The interface manufacturer specifies the mic preamp's gain to be 60 dB. An astute reviewer (that's me) wrote that at maximum gain, the sensitivity from mic input to digital output is -56 dBu = 0 dBFS. (thanks, Mike) Since the mic's output is 16 dB below the 0 dBFS reference, the record level will be around –16 dBFS.

In real life, the DAW's meter will probably show peaks in the –10 to –4 dBFS range, and here's why. The mic spec is based on the RMS voltage of a sine wave, and a sine wave isn't very much like human speech. The DAW meter responds to peak rather than RMS levels, so the level shown on the meter will appear higher than this theoretical exercise predicts.



If we're aiming for 16 dB of headroom, the DAW meter shows that we're occasionally using most of that. In this case, it took all of the preamp's gain to reach a decent record level. Replace the soft voiced narrator with a powerful rock singer and the preamp gain will need to come down by 20 dB or more in order to avoid overloading. Replace the SM57 with a hot condenser mic, and you might have to switch in a 20 dB pad. And that, folks, is how it works in the real world.