Newcomers and Elmers Net: Getting Good Audio On and Off the Air
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I want to explore what makes for good audio both in receiving and in transmitting.
-- While I am by no means an expert audiologist, I certainly appreciate good audio practices when I hear them, and I want to make my station sound the best I can without causing interference to others.
-- Having good audio on receive aids in copying a signal (less repeats needed) and reduces strain, and good transmit audio increases one's chances of being heard under difficult conditions.
-- Bad audio does just the opposite, and worse, harms the signals others can receive.

Receiving Audio
-- most radio speakers are not good quality
-- they are on the top or on the bottom; best speakers are forward facing
-- external speakers should be used whenever possible
-- don't think in terms of stereo speakers! Highs and lows are far less important on the radio than the mid-tones
-- DSP speakers can be used to clean up audio significantly if your radio does not have that option built in
-- bigger is usually better; there is a fuller sound, less "tinny"
Headphones are a great option, particularly for pulling weak signals out or for distinguishing between two or more strong signals
-- learn to use your RF gain and your AF gain; it is a lost art
-- when your RF gain is at full blast it is raising everything up including unwanted noise
-- by using less RF and more AF, you are boosting more or what you want to hear and less of what you don't
-- it is not perfect, but it will make an improvement for HF signals
-- preferably get someone to test your audio whose voice you know well – this will be a good marker
-- same thing is true for transmitting – get someone to help who knows what you sound like in person – they can give you the best advice for making adjustments
-- avoid distortion; it is better to turn down surrounding noise levels than to turn up audio on a radio – distortion makes everything hard to hear

Modulating in Moderation
Hams usually fall into one of two categories when it comes to station audio: they are either perfectionists or minimalists. There are not a lot of middle-of-the-road people when it comes to audio. Perfectionists are those folks who have mixing boards, filters, boom mikes and audio processors to refine

their sound, and they often like to work AM for the increased bandwidth. If limited to 3kHz or smaller on SSB, they will try to make use of every last Hz to boost their quality.

Minimalists, on the other hand, tend to use default settings for audio on their rigs (if any adjustment is even allowed), and stock microphones or inexpensive headsets. There is nothing wrong with this minimalist approach, especially in terms of saving money, but the minimum audio performance may not make the best use of one's signal efforts. I would like to suggest the best solution may be the old adage *moderation in all things*. Being aware of one's audio and adjusting it for best performance is important regardless of whether you are an audio aficionado or an audio minimalist.

Back to Basics

Getting caught up in good audio is easy to do because sounding good only makes sense. Sometimes the desire for rich-sounding audio stems from the AM transmissions of yesteryear—before the days of SSB voice transmissions were almost 6 kHz wide, with each sideband carrying the same information. This produced a very high quality audio signal (given the usual AM limitations).

Moving to SSB increased available bandwidth and boosted power, but at the expense of signal bandwidth. A SSB has a maximum width of about 3 kHz, with no adjacent sideband to add fullness and body to the audio. This 3 kHz limit is very real, however, and poses a strict limit to our bandwidth due to the design of an SSB transceiver.

Spurious Signals, Noise and Splatter

When transmitting, any signal not intentionally created can be termed *spurious*. Harmonics, intermodulation, splatter (the non-linear mixing of signals) are all potential problems for both audio and RF signals. While many operators believe that if their audio sounds clean to someone on the other end of the signal they are fine, this is not necessarily so.

There are many possible problems with one's signal that even an oscilloscope will not show because the problems can be outside the range of the scope's current reading. This unintentional mixing of signals can happen at non-harmonic points, including the sum/difference and product points of these multiple frequencies. The total bandwidth occupied by a SSB signal, including third-order intermodulation products, is approximately three times the audio bandwidth of the system.   We will start, however, with possible issues nearest the transmit/receive frequency and work outward from there.

Mixing Products

One frequency (f_1) in a non-linear system has harmonics ($2f_1$, $3f_1$ . . . etc.). When two signals are present, not only is there the product of their mixing, but also the sums and differences ($f_1 + f_2$, $f_2 - f_1$) of their combined mixing (and their harmonics), and the addition of the fundamental frequencies, and so on. That is a lot of noise!

The odd third-order intermodulation components are usually the ones we are most concerned about because they are closest to our original frequency, and therefore what we are likely to hear as we monitor our transmissions. However, all intermodulation components are of concern because we do not want to have our signals affect others regardless of the frequency or bandwidth.

even slight irregularities in the transfer process can produce distortion products in-band and out-of-band. In-band or intermodulation distortion products cannot be filtered out because they are contained within the desired signal frequencies at various intervals. It's a bit like recording a symphony performance where all the right instruments are present, but interspersed within those fine instruments are several people playing kazoos and clanging pots and pans. Like it or not they are part of the recording.

When voice tones are introduced into a mixer or amplifier, this represents a series of frequencies, each distinct, and each capable of distortion. Audio frequencies are converted up into RF ranges, transmitted, received, and converted back down to audio ranges. Any distortion or non-linearity along the way and the signal is polluted.

The wider the spread between audio tones (more bass and more treble) means wider bandwidth in actual tones, and therefore more tones for IMD to enter in beyond what you or others can hear when tuned into the same frequency. Unfortunately others **will** hear it outside the desired frequencies.

Basically the way to prevent interference with others (and with one's own receiver) is to make sure the transmit bandwidth is well within a 2k limit, meaning reduced bass and treble. As shown above, a reduction from 2k to 1k produces IM products which are only slightly out of band at third-order points.