

Clean, Punchy, Competitive Contest Audio Without Splatter

I'm retired from a career in professional audio, and, although I designed a lot of music systems, one of my specialties was designing speech systems that allowed worshippers to understand the priest in big, very reverberant churches. To do that effectively, I had to learn a lot about the human perception of sound, and how our ear/brain translate what we hear into speech intelligibility. I also had to learn how to design sound systems that gave our ears what they need, while giving them as little as possible of what they don't need. In rooms with difficult acoustics, the enemy is reverberation. In radio systems, it's noise and interference.

As an active NCCC member, I'm often asked to help other members tweak their audio for "maximum smoke." Top operator Jeff, N6GQ, likes to travel to interesting places for contests, and when he does, we always make a sked to make sure he's sounding good. The 2015 CQ WW SSB found Jeff in EA8 (operating as EF8U), and we worked through setup of the FT-950s that were in the shack he was using. In his soapbox comments, Jeff said, "My goal was to be clean, punchy, and not splatter,

as the large EF8R MS effort was under way nearby, and there would be times when we'd be on the same band and likely S-9 + 60 looking at each other, so I didn't want to be a wide signal in their path. This paid off, as I was told over and over both how loud I sounded, as well as how punchy the audio was."

While sorting through the soapbox for that contest, CQ WW Contest Director K5ZD came across Jeff's comments and suggested that I write something for *NCJ*. So here it is. Let's start with fundamental concepts.

Frequency Content of Speech

Human speech has content from about 100 Hz to 8 kHz, but only the energy between about 400 Hz and 4 kHz contributes to speech intelligibility. Vocal content below 400 Hz provides "body" to the voice (great for singers and radio announcers), but that low-frequency output of the mic also contains breath pops, room noise, mic handling noise, wind noise, and reverberation. This low-frequency energy can easily be as much as half of the power picked up by the mic, but it contributes nothing to communication,

and it wastes transmitter power. Likewise, speech content above 3 kHz provides "presence" and helps communication a bit, but the added bandwidth adds noise (and QRM from other stations). Most SSB transmit filters are 2.7 kHz wide, so a well-adjusted rig will align those filters so that they pass audio between 400 Hz and 3.1 kHz. A few radios allow the user to tweak this setting in a menu. (These bandwidth limits for speech communication were established in the earliest days of long-distance telephony. They allow what's necessary, but nothing extra, and for more than a century, they have allowed more and more conversations to be crammed into the same bandwidth.)

Thus, our first rule is to *minimize any part of the audio signal below about 400 Hz, and to not waste bandwidth transmitting sound above 3 kHz*. We have several controls over this. First, we can choose a microphone without excess low-frequency response (see "Choosing a Microphone," below). Many rigs provide menu settings to tailor the audio frequency response. Study the manual for your radio to understand how to choose settings for your rig.

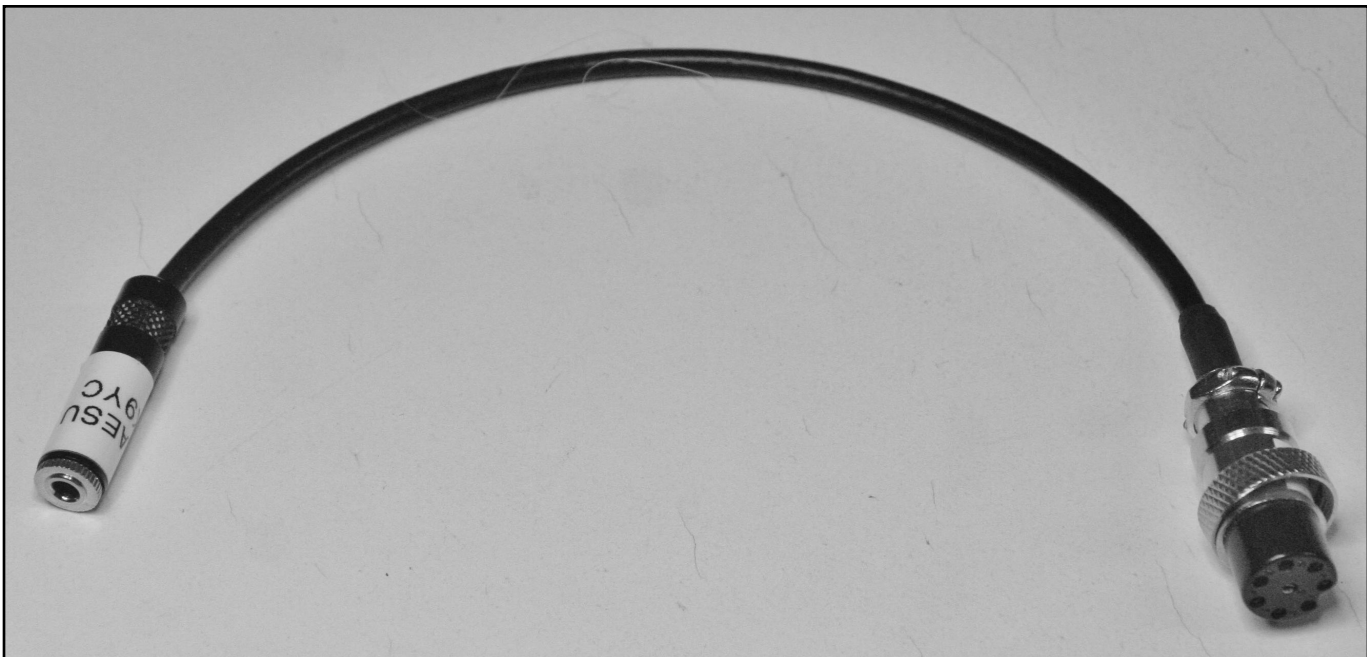


Figure 1 — Adapter to connect computer microphone to Yaesu (see text).

Some rigs, like the Elecraft K3, K3S, and KX3, make it even easier to tailor the frequency response. They have a built-in octave-band equalizer (called TXEQ) covering the speech range. Each band can be set for up to 18 dB of boost or cut in 1 dB steps. A good starting point for most mics and voices is maximum cut of the three lowest bands (50, 100, 200 Hz), and 3-6 dB cut of the fourth band (400 Hz) leaving all other bands set flat (no boost or cut). Some mics/voices may benefit from a bit more cut at 400 Hz, or from 3-6 dB of cut or boost in the two highest bands. Save these tweaks for when you have a trained listener to advise you.

Getting Audio Levels Right

This is the most critical part, and more than half of the stations I hear in a typical contest screw it up. The most common mistake is cranking these settings up too high, often way too high! The result is muffled, distorted audio that is hard to copy, often with lots of room noise. There are several adjustments that we must get right, and any one of them can make us sound bad.

In one common contesting setup, the mic feeds the mic input of the computer soundcard, and the soundcard feeds the rig (best to a line level input). For SO2R, the soundcard feeds both radios via L and R outputs, and the logging software controls switching. The advantage of this setup is that the operator's mic can easily be used to record new messages during the contest, especially important if you're running in split mode. In this setup, *the most critical settings* are (1) the mic gain of the computer soundcard, which must be set so that the mic *never* overloads the soundcard input, even when things get *really* exciting; (2) the output gain of the computer soundcard, which must be set so that the soundcard itself never overloads, and (3) the input gain of the radio, which must be set so that it never overloads. Any one of these overload points will turn your audio to mush!

The key to all of these adjustments is to listen with headphones as you adjust them. Adjust soundcard mic input and soundcard output by plugging headphones into the soundcard output jack and carefully listen for any distortion. If you have trouble hearing it, recruit a friend to help. Record one or more messages with the same mic that you'll use live, and play them back, again listening with headphones. Make sure there's no distortion. If there is, turn down the recording gain and do it again until it sounds clean. Once you have a good recording, adjust the soundcard gain settings so that the level (loudness) of the live mic is the same as the recording. Now

you're ready to remove the headphones and feed the soundcard to the radio(s).

If you are able to feed the computer soundcard to a line level input on your radio, you should be able to adjust the input gain of the radio for good modulation. Start with no compression (processing off or turned all the way down) and look for correct indicated power output on your rig's meter. If possible, listen on another radio (with its RX antenna disconnected and its IF bandwidth set wide (3-4 kHz), again making sure there is no distortion. Make all level adjustments to your rig with equalization set as described above.

If your rig lacks a line input, or the line input won't allow you to do what you need to do (eg, perhaps you can't apply EQ with it, or you can't use VOX with it), you'll need to make an adapter to feed the mic input. That adapter will need a 20 dB pad (voltage divider) between the soundcard and the rig. This requires resistors in a ratio of 10:1, with the smaller resistor wired in parallel with the mic input and the large resistor in series between the computer output and the mic input. Values aren't critical, if they're in the right range. 1000 Ω and 100 Ω , or 470 Ω and 47 Ω are good choices. Low wattage resistors are fine, so the pad can usually be fitted inside connectors.

Once you have good, clean-sounding modulation, set your rig so that its display shows you a bar graph for compression. Then turn on (or turn up) compression (processing). Speaking as you normally would during a contest, increase the compression until the display indicates 10 dB of compression on voice peaks. Most rigs sound good at 10 dB, and most get nasty when pushed beyond that — intelligibility degrades, room noise increases. Again, listen to yourself on another radio if you can (no antenna, wide IF bandwidth). Once you've made these adjustments, you're ready to recruit a trained listener.

Alternative Setups

When contesting from the West Coast, 40 meters almost never supports running to EU, so I never need to re-record messages on the fly. Instead, I record all my contest messages in advance of the contest using an audio application such as *Audacity*. There's a *WebEx* talk about this on the public section of the NCCC website (visit <http://nccc.cc/misc/RecordingVoiceMessages-K9YC.wmv>). During the contest, I feed the mic to my YCCC SO2R box, which feeds it to the mic inputs of left and right radios, and I feed computer outs to the line inputs of the two radios. Both are switched by *N1MM Logger*.

With this alternate setup, we adjust

the computer output level as before, then with compression turned off, adjust both mic gain and line gain for indication of desired output power on voice peaks. When properly set, the live mic and your recording should sound identical and equally loud (you should use the same mic for recording and for live talking). Now adjust compression as before for indicated 10 dB on voice peaks.

Your Radio's Power Supply

Most modern rigs are designed to operate from dc power supplies that provide 13.8 to 14 V dc. When operated at a lower voltage, though, the distortion produced by most of these rigs increases, often by 3-6 dB. This distortion produces harmonics and inter-modulation products (splatter).

Setting Up Your Power Amp

Once audio is well adjusted, the most common cause of splatter is a badly tuned or over-driven power amplifier. Power amps are cleanest when their load (the antenna) is closely matched to the output stage. Tube amps have output stages that must be tuned, either manually, automatically, or by the automatic recall of previous settings for the frequency in use. Most solid-state amplifiers have fixed output networks for both harmonic suppression and to transform a 50 Ω load to their designed load impedance. If the antenna in use does not provide a 50 Ω load, an antenna tuner *must* be used to (1) minimize distortion, and (2) so that the amplifier will not "fold back" (reduce power) to protect itself.

ALC or Not to ALC

ALC between amplifier and radio should *never* be used to set output power. Doing so is a recipe for very nasty splatter. My 35 year-old TEN-TEC Titans require only 50-60 W for full output; an ACOM 1010 requires 50-75 W, and the KPA500 only about 28 W. Always set output power by setting the drive level (power output) of your rig needed to reach that power level. It *is* good practice to use ALC to protect a power amplifier in the event of some failure in the antenna system. To do this, hook up the ALC as directed by the amplifier and radio manufacturers, but set power output by setting the level of drive from the rig to the amplifier.

Triode Power Amplifiers

Triode power amplifiers must have their output networks tuned for maximum output as indicated by a good power meter. If that's more or less than the desired output level, simply change the drive (power out) level of the radio to get what you want. Whether operating CW, SSB, or digital modes, I tune in CW mode with a series

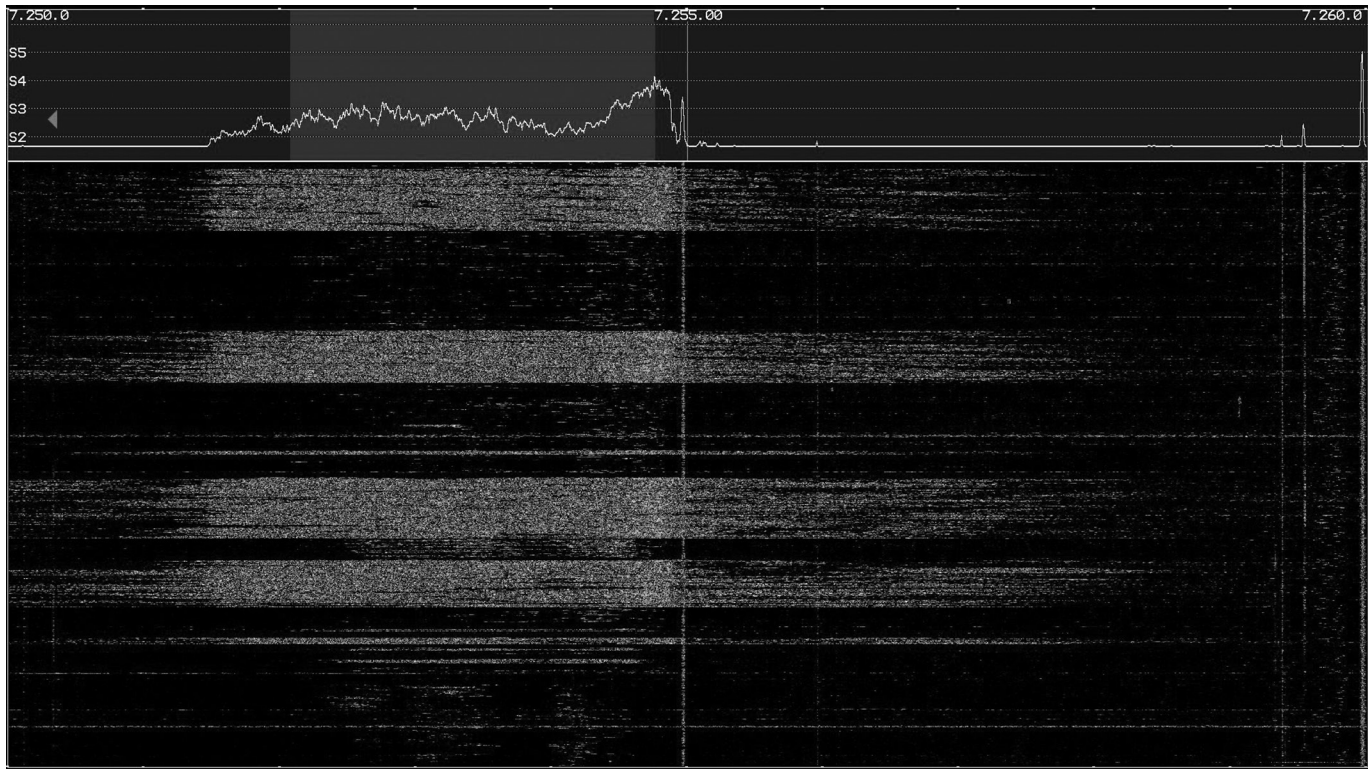


Figure 2 — Screen grab from P3/SVGA screen of a 40 meter rag chew. The shaded section of the top of the display is the 2.7 kHz IF of my radio. One station is much narrower and weaker. Horizontal lines extending mostly above the signal, but also below it is splatter on audio peaks. Splatter in the upper sideband of this LSB signal is only 15 dB or so below the signal, and extends more than 3 kHz above the signal. Note also the strong peak in the LSB (nearly 10 dB) just below the suppressed carrier. This is excessive LF energy (50-400 Hz) that this station is wasting (and that is creating most of the splatter). It's quite common to hear this trash in the suppressed sideband during contests, mostly wildly over-amplified room noise resulting from excessive processing (compression) and a mic that's turned up way too high.

of dits, carefully tuning both capacitors for maximum output. I start out several dB below full output, then increase drive and retune at the higher drive level. Using dits is easier on the tubes because it reduces the dissipation (and the grid current) by one-half, as compared to key down. Also, excessive grid current is a primary cause of triode tube failure; minimum grid current generally coincides with maximum output.

Tetrode Power Amplifiers

Tuning tetrode power amps is a bit more complicated. Veteran amplifier designer Tom Rauch, W8JI, has written an excellent tutorial and applications note on the topic (www.w8ji.com/loading_amplifier.htm). It should be considered required reading for anyone with a tetrode or pentode power amplifier.

A power amplifier that automatically recalls previous settings must have been tuned properly in the first place, if the recalled settings are to minimize distortion (splatter). If multiple antennas are used on a band, settings that are recalled for the tuning of one antenna may be wrong for another. Setups like this may require the

use of an outboard tuner, ideally one that can switch between multiple outputs for the different antennas.

Tips for Listening On the Air

Start by listening with your IF filters set as wide as possible. This way, you're listening to what is being transmitted, not what is being limited by the bandwidth of your receiver's IF. Make sure that your radio is not being overloaded. Turn off the preamp and turn on the attenuator, if needed to keep the S meter in a mid range. Make sure that your noise blanker, noise reduction, notch filter, and any "audio EFX" are turned off. Now that you know what that sounds like, narrow your IF to a normal contesting bandwidth and see if it still sounds good.

When listening to yourself on a second radio at your own station, disconnect the antenna, turn off the preamp, turn on the attenuator, and turn up the audio gain to the point of being comfortably loud. What you're looking for is any audible distortion.

Whether listening to your own signal or to a friend's, after you've listened wide with a wide filter, switch to a narrow SSB filter

setting and tune both sides of the signal. Listen carefully for any splatter (on CW, listen for phase noise and clicks).

If you have a modern spectrum display (P3, LP-Pan, or SDR), set it fairly narrow (25-50 kHz wide is good for SSB, 5-10 kHz for CW). Make sure it isn't being overloaded. Check the settings just as for the receiver itself. Again, look for splatter, which will show up as short horizontal lines in the waterfall on audio peaks. Splatter is a sign of problems in the output stage — a linear amplifier, or the rig itself if you're running barefoot. Look for overdrive, mistuning, the use of ALC between amp and rig. Study the section on setting up amplifiers again.

Choosing a Microphone

Beginning in the late 1950s, Shure introduced the model 440, the first microphone designed specifically for SSB transmission. The modern version of that mic is the 444D. Both are omnidirectional mics with low frequency response falling below 400 Hz and with a pronounced peak around 3 kHz that compensates for some of the loss in the SSB transmit filter.

These are excellent-sounding mics, but they're tabletop designs, not well suited to modern contesting. Most mics in the Heil line are more practical applications of the same concepts.

Mics come in several basic forms. *Dynamic* mics operate on the same principle as a loudspeaker, (a coil moving in a magnetic field) but in reverse. A loudspeaker works pretty well as a microphone, and has been used that way for half a century in intercom systems. *Electret condenser* mics are very different. The diaphragm is one plate of a capacitor; a voltage is applied between the two plates, the other plate being fixed. The source impedance is quite high (in the megohm range), and must be transformed to a lower impedance by a FET follower built into the mic, so that what it feeds doesn't load down the mic. The *electret* capsule is pre-polarized, but the FET follower needs a small positive voltage fed through a load resistor to operate. This voltage is called bias. A value of 8 V dc through 5.6 k Ω is typical.

Both types of mics are built with an omnidirectional pattern (picks up equally in all directions) or a cardioid pattern (picks up better in one direction) and can be thought of as "half space" mics.

Cardioid mics have an important characteristic called proximity effect, which is a very strong bass boost for sound sources very close to the mic. In addition to making voices "bass heavy," proximity effect magnifies breath pops, wind noise, and handling noise. Virtually all mics used in live sound are cardioids, and those intended for use by singers have a strong low-frequency rolloff that partially compensates for proximity effect. Although cardioids reduce room noise pickup, proximity effect generally makes them a poor choice in the ham shack.

Cardioids work on the principle of acoustic cancellation between sound impinging on front and rear openings of the cardioid housing. Proximity effect is the result of that process, and the fact that there is a single rear opening. An important variation of cardioids is built with extra openings in the handle, which greatly reduces proximity effect. The ElectroVoice 664 and 666 were the first popular mics of this type, which are called "variable-D" for the variable distant openings, as opposed to "single-D" cardioids with a single rear

opening. If you're looking for a good pro mic for your ham station, the variable-D EV RE10, 11, 15, 16, 18, 20, and 27, and the Shure SM53 and SM54 are great choices. All but the RE16, 20, and 27 are long discontinued, but dynamic mics last forever.

An omnidirectional mic, whether dynamic or electret, is the best choice for ham radio. It has no problem with proximity effect, so it can be worked close. I position the omni mic of my Yamaha CM500 about 2 inches above and 1 inch to the left of my mouth. This gives me room to munch and drink coffee while the CQ recording is playing, and it minimizes any sign of breath pops, while still being close enough to minimize room noise.

Using a Pro Mic in the Shack:

Professional electret mics cannot be used with ham gear because of how they are powered, but pro dynamic mics work well and are easy to wire. Their 3-pin XL-connector comes wired for balanced circuits — pin 1 is the shield, pins 2 and 3 carry the signal. To connect them to your ham rig, wire the shield to the shell of the Foster plug, and connect the signal pair to mic and mic return. Or wire both the shield and one side of audio to the rig connector shell, the other side of audio to the pin for mic in.

About 6 years ago, W6XU, an electrical engineer working for an audio equipment manufacturer, discovered the Yamaha CM500 boom mic headset, which, at the time, was selling for about \$45 (they're now about \$60). Josh arranged a group purchase for NCCC members, and many of us quickly became fans of the headset and the mic. The CM500 has an electret mic and nice cushy headphones. Both sound great, and the headset is easy to wear for a long contest weekend. It comes with two 1/8-inch TRS plugs, one for the headphones, the other for the mic. Both plug straight into the rear panel of a K3.

For other rigs, you'll need to make up a cable adapter for the mic. You'll need a cable-mount Foster plug to match your radio and a female 1/8 inch TRS jack to mate with the TRS plug on the headset. Check the manual for your radio for pinout of the mic connector. To connect the mic to the radio, run a single-conductor shielded cable (I use mini-coax) from the tip of the TRS jack to the mic input pin of the Foster plug, connecting the cable shield to the sleeve of the TRS jack and the shell of the

Foster plug. Virtually all modern radios have V+ in the range of 8 V on a pin of the mic connector, so all it takes to provide bias is a 5.6 k Ω resistor between the 8 V source and the mic input pin. This can be a very low-wattage resistor, so it's usually possible to fit it inside the Foster plug. Buy Foster plugs from your ham suppliers; female TRS jacks can be bought from pro audio vendors such as Full Compass and Sweetwater. You want Neutrik part number NYS240BG.

K6LL recently bought a Koss SB-45, which is quite similar to the CM500 but at half the cost. After using it for two 12 hour contest sessions, Dave reported that it is equal to or better than the CM500 in performance and comfort (and he likes the CM500). GM3SEK also likes the SB-45 (he can't buy CM500 in the UK), and reports that his spouse prefers the lighter weight headphones of the Koss CS-100, which he also recommends. As he puts it, it's a matter of which style of headphones you like.

Summarizing the Steps

- Set your rig to minimize audio content below 400 Hz.
- Get audio gains set right, from the mic input of the rig (or of the computer), the output gain of the computer, the line input of the rig.
- Set processing for an indicated 10 dB on voice peaks.
- Resist the urge to turn up your mic gain or compression controls. Once you have levels set as described here, turning either one up makes you sound worse, not better.
- Tune your power amp carefully.
- Don't overdrive your power amp, and don't use ALC to set transmit power.
- If your rig runs on nominal 13.8 V dc, run it from a supply as close to 14V as possible.

Being a good neighbor on the bands isn't just politeness. In contests like the CQ WW, it's required. In an e-mail exchange, Jeff, N6GQ, said, "Personally, I'm loathe to do many more SSB contests due to so many poor-quality signals on the air. It just so happens that the last good number of trips I've done have coincided over SSB contests." I think Jeff speaks for many of us; he certainly speaks for me. Take the steps I've outlined, and you will make SSB contesting more enjoyable for all of us.