Clean, Punchy, Competitive Contest Audio Without Splatter

by Jim Brown K9YC

I'm retired from a career in pro audio, and although I designed a lot of music systems, one of my specialties was designing speech systems that allowed worshipers 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 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. 2015 CQWW SSB found Jeff in EA8 (operating as EF8U), and we worked through setup of the FT950s 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 M/S effort was underway nearby - and there would be times when we'd be on the same band and likely S9+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. It ran in the Mar/Apr 2016 issue. 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 communications – it wastes transmitter power. Likewise, speech content above 3 kHz provides "presence" and helps communications a bit, but the added bandwidth adds noise (and QRM from other stations). Most SSB TX 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 communications were established in the earliest days of long distance telephony – they allow what's necessary, but nothing extra. And over 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" later in this article. Many rigs provide menu settings to tailor the audio frequency response. Study the manual for your radio to understand and choose settings for your rig.

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 sound card, the sound card feeds the rig (best to a Line Level Input). For SO2R, the sound card 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* in the *computer sound card*, which must be set so that the mic ***never*** overloads the sound card input, even when things get ***real*** exciting; 2) the *Output Gain* of the *computer sound card*, which must be set so that the sound card 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 #1 (sound card mic input) and #2 (sound card output) by plugging headphones into the sound card 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 recording gain and do it again until it sounds clean. Once you have a good recording, adjust sound card 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 sound card to the radio(s).

If you are able to feed the computer sound card to a Line Level Input of 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 things you need to do (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 sound card 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. 1K and 100 ohms, or 470 and 47 ohms are good choices. Low watt resistors are fine, so 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 and turn on (or turn up) Compression (processing). Talking as you normally would during the 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 than 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: Contesting from W6, 40M 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 like Audacity. There's a WebEx talk about this on the public section of the NCCC website. During the contest, I feed the mic to my YCCC SO2R box, which switches it between the mic inputs of left and right radios, and I feed computer outs to Line Inputs of the two radios. Both are switched by N1MM Logger Plus. http://nccc.cc/misc/RecordingVoiceMessages-K9YC.wmv

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 (hopefully you used the same mic for recording and for live talking). Now adjust compression as before for indicated 10 dB on voice peaks.

Power Supply For Your Rig: Most modern rigs are designed to operate from DC power supplies that provide 13.8 – 14 VDC. When operated at a lower voltage, the distortion produced by most of these rigs increases, often by 3-6 dB. That distortion produces harmonics and intermodulation products (splatter).

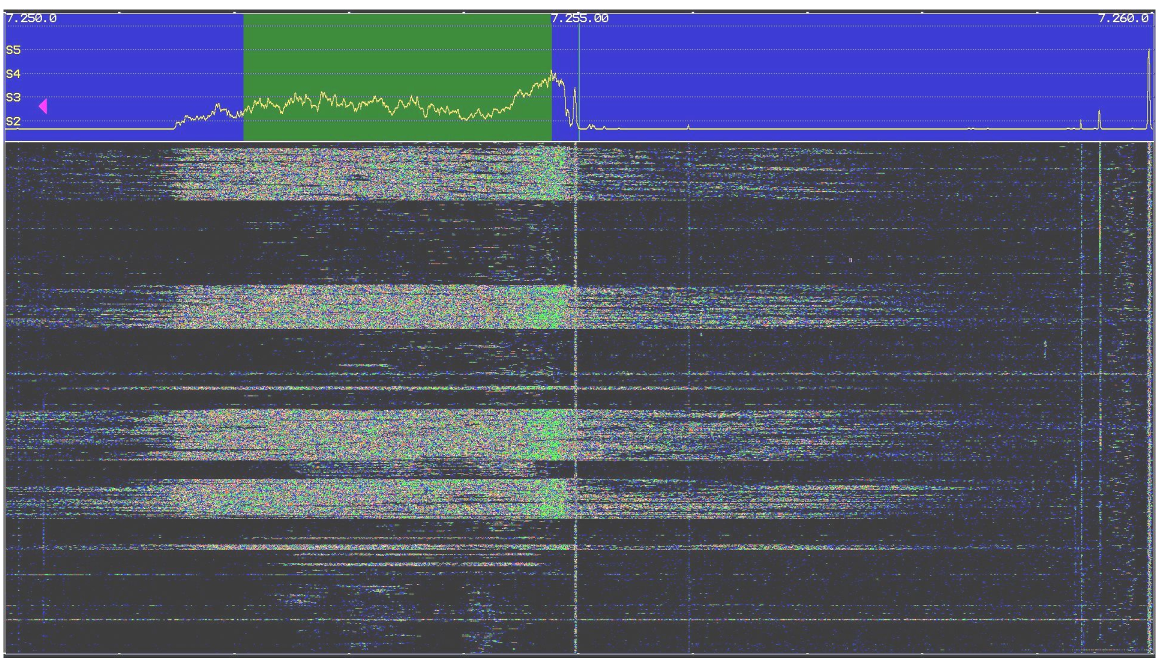

Fig 1 – Splatter On 40M

This screen grab from a P3/SVGA screen shows about 1 minute of a 40M ragchew. Display bandwidth is 10 kHz. The shaded section of the top of the display shows the 2.7 kHz-wide RX bandwidth setting of my radio. One station is much narrower and weaker. Horizontal lines extending mostly above the signal but also below it are 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, indicating a badly tuned or overdriven power amp, or ALC between amp and rig to set power, or both. Note the broad peak near the suppressed carrier – this is wasted low frequency energy, and is probably contributing most of the splatter. The straight vertical lines are local noise, not part of the transmitted signals.

<u>Setting Up Your Power Amp</u>: 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 previously settings for the frequency in use. Most solid state amplifiers have fixed output networks for both harmonic suppression and to transform a 50 ohm load to their designed load impedance. If the antenna in use does not provide a 50 ohm load, an antenna tuner <u>must</u> be used to 1) minimize distortion, and 2) so that the amplifier will not "fold back" (reduce power) to protect itself.

<u>ALC Between Amplifier and Rig</u> 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 required 50 – 75 W and KPA500 only about 28W. Always set output power by setting drive level (power out) 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 manufacturers of amplifier and rig, but set power out by setting drive from the rig to the amplifier.

<u>Triode Power Amplifiers</u> 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) of the rig to get what you want. Whether operating CW, SSB, or digital modes, I tune in CW mode with a series 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 keydown. Also, excessive grid current is a primary cause of triode tube failure; minimum grid current generally coincides with maximum output.

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

<u>A power amplifier that automatically recalls previous settings</u> 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 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.

<u>Tips for Listening On The Air</u> Start by listening with your IF filters set as wide as you can. This way, you're listening to what is being transmitted, not what is being limited by the bandwidth of your receive IF. Make sure that your radio is not being overloaded – turn preamp off and attenuator on if needed to keep the S-meter in a middle range. And 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 and RTTY, 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). And, of course, 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 omni-directional 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 being fixed). The source impedance is quite high (megohms), 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. 8VDC through 5.6K 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 though 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 breathe 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 proximity effect. Although cardioids reduce room noise pickup, proximity effect generally makes them a poor choice in the ham shack.

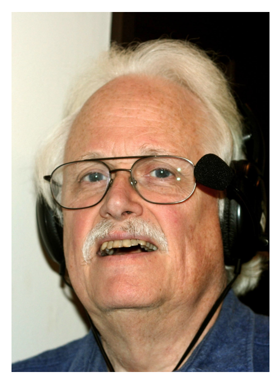

Fig 2 –Mic Positioning

Cardioids work on the principle of acoustic cancellation between sound impinging on front and rear openings of the microphone housing. Proxmity 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, so buying used from a trustworthy source is a good option.

An omnidirectional mic, whether dynamic or an 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 CM500 about two inches above and an 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: Pro 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 six years ago, W6XU, an EE working for an audio equipment manufacturer, discovered the Yamaha CM500 boom mic headset, which at the time was selling for about $45 (current cost is 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-in TRS plugs, one for the headphones and 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-in 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 8V on a pin of the mic connector, so all it takes to provide bias is a 5.6K resistor between 8V and the mic input pin. This can be a very low watt 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 like Full Compass and Sweetwater. You want Neutrik part number NYS240BG.

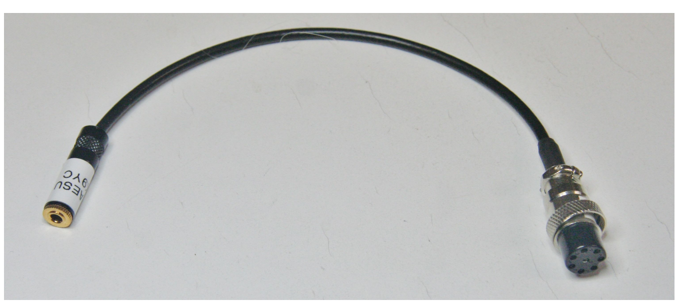

Fig 3 – Adapter For Computer Mic to Yaesu FT1000MP

K6LL recently bought a Koss SB-45, which is quite similar to the CM500 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 XYL 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 for audio and power amp setup for SSB Contesting:

1) Set your rig to minimize audio content below 400 Hz and above 3.2 kHz.

2) 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.

3) Set processing for an indicated 10 dB on voice peaks.

4) Resist the urge to turn Mic Gain or Compression up louder – once you have levels set as described here, turning it up louder makes you sound WORSE, not better.

5) Tune your power amp carefully.

6) Don't overdrive your power amp, and don't use ALC to set TX power.

7) If your rig runs on nominal 13.8VDC, run it from a supply as close to 14V as possible.

Being a good neighbor on the bands isn't just politeness. In major contests, it's required. When in S&P mode, I'll tune right past signals that are too distorted to copy easily, and when running, I won't waste my time trying to work those stations calling me. In an email exchange, Jeff 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 does speak for me. Take these steps and you will make SSB contesting more enjoyable for all of us, including yourself. And you'll boost your score!

Acknowledgement: Thanks to Bob Wolbert, K6XX, for his extensive contributions to the discussion of the causes and cures of distortion in rigs and power amplifiers. There's much more of this in the slides for his tutorial talk on the subject to the Northern California Contest Club several years ago. http://k9yc.com/K6XXAmpTalk.pdf