Can You Hear Me Now?

Adjusting the

Transmit Audio Chain

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We all heard some terrible transmitter audio during the recent phone contests. Some signals occupied far more space than necessary. Background noises such as blowers and other shouting operators obscured the voices from other stations. Some signals were simply too distorted or muffled to understand.

Poor quality audio means fewer contacts. If other stations cannot understand your call or exchange, they may ask for repeats – slowing down your rate. Or they simply ignore your signal as too unintelligible or difficult, a waste of their time.

On the other hand, a signal with clean, crisp audio is easy to work, even when that signal is weaker than others around it. With an hour's work you can achieve that clean, crisp sound. This article systematically aligns the transmit audio chain – the sequence of equipment between the operator's voice and the antenna – for good results. An operating position at w1km serves as an example of steps to follow. The control settings appropriate for your voice and station equipment likely will differ, but the control adjustment process should be similar.

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1 Components of the transmit audio chain

w1km resembles many modern contest stations. A soundcard in the computer provides voice memory keyer functions. An interface box handles galvanic isolation, switching and other tasks necessary to interconnect the radio with the computer. This box can be homebrew or a commercial solution. At w1km we use a microKeyer manufactured by MicroHam.

Figure 1 outlines the flow of audio. A short study reveals over a dozen locations along the audio chain where adjustments affect the quality of ssb audio on the air. No wonder stations have poor audio – so many opportunities exist to contaminate the signal! Audio, once contaminated, cannot be restored.

The general strategy for setting all these adjustments proceeds as follows:
– Operator training.
– Controls affecting audio signal levels from the microphone through the recording of wave files (messages).
– Controls affecting audio signal levels from the playback of message to the radio's audio input jack.
– Controls used for live speech routed through the soundcard.
– Transmitter mic and speech processor level, and any other internal audio bandwidth, threshold, or equalizer settings.
– Transmitter power output level to maintain linear operation of an external power amplifier.

2 Operator

Good quality audio begins with the operator. Speech coaches teach that your best voice occurs while standing, not a convenient operating position for the contest. But sit up straight in your operating chair, with the legs apart. This posture allows you to inhale and exhale air more efficiently and uniformly with muscles around your belly. Not only does good posture in the chair result in better voice control, but it also conserves energy for the long contest.

Train yourself not to shout into the microphone. Shouting distorts your information. Operators shout out of desperation or excitement. But yelling does not make your signal any stronger in the pileup. Notice your shouting, and train yourself to relax again to speak with a normal voice. Not only will your signal sound better, you will finish the contest with less exhaustion. And, at a multi-op station, your fellow operators will thank you for not interfering with reception!

We are accustomed to hearing our voice at a particular volume in our ears during non-radio activities. Our voice reaches our own ears through the bones of the head (and other paths) as well as through the air. Headphones attenuate the path through the air, tricking the brain into cranking up the speaking volume. With training, we can learn to "whisper" when wearing headphones, speaking at a more normal volume. (You may also use the monitor signal from the transmitter to your headphones to replace the normal feedback to the ears. By increasing the monitor volume a bit, you may find yourself speaking at a more natural level. so2r operators often don't have this option available, since they tune another receiver while transmitting.) Practice will help break any habit of shouting.

When speaking into the microphone, aim for maximum signal (your voice) to noise (the surrounding environment) ratio. Usually this means close-talking: positioning the microphone about an inch (2-3 cm) from the lips. Modern microphones pick up the surrounding environment easily and, unfortunately, directional microphones don't fit into headsets. If your operating position includes significant external noises (e.g., blower fans), take steps to reduce this background as much as possible. Other contesters tuning across your signal will far more easily understand a voice speaking against a quiet background. The band contains plenty of other distracting signals – don't add any more from your own station!
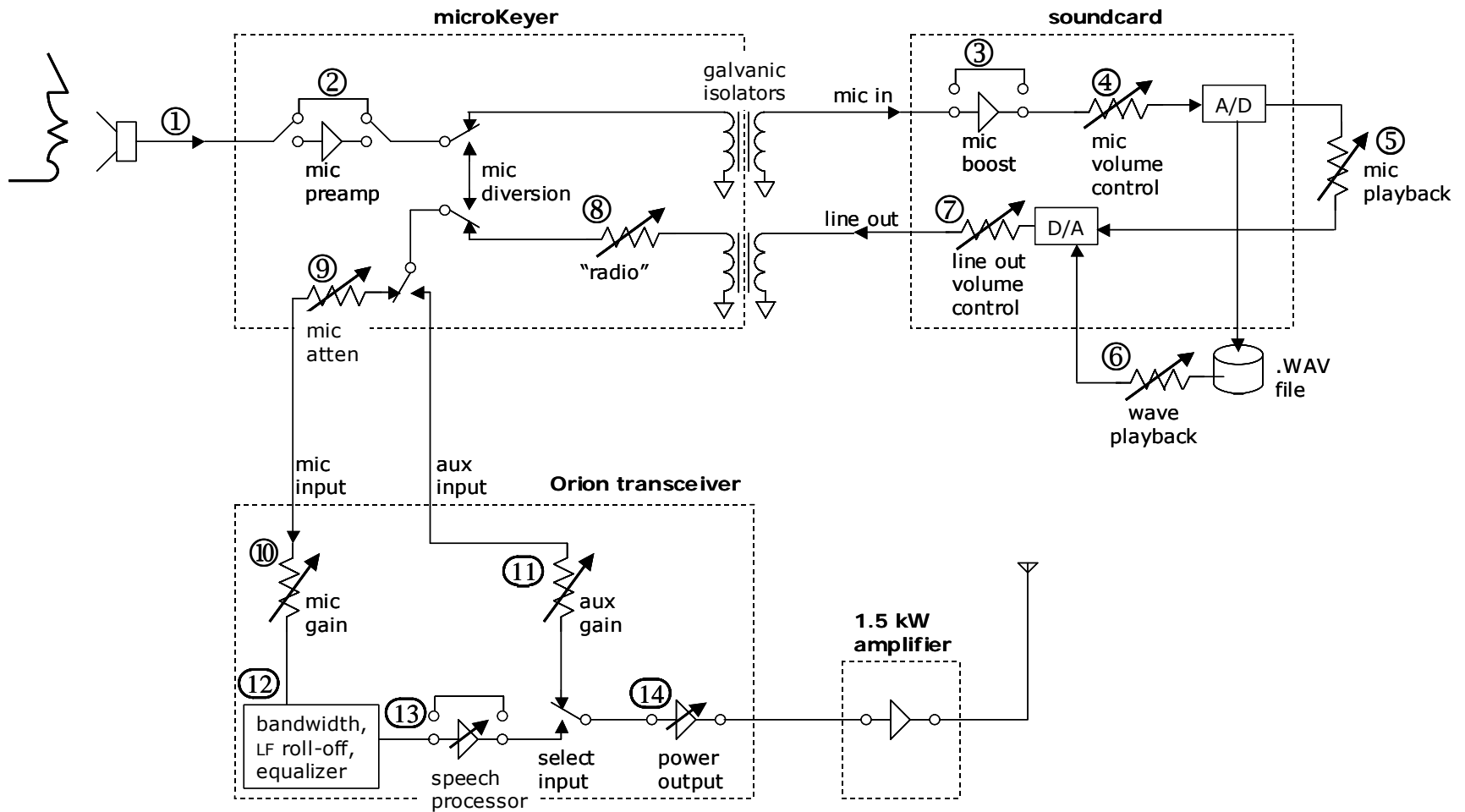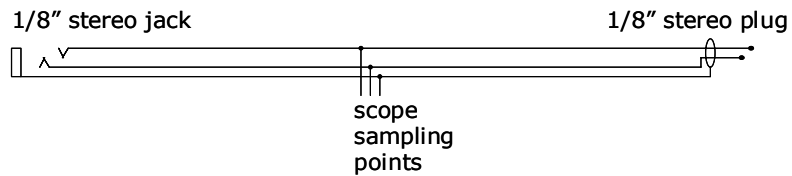
Figure 1: Flow of transmitted audio at W1KM. The numbered points identify key controls affecting transmission quality.

1/8" stereo jack   1/8" stereo plug

scope
sampling
points

Figure 2: Short cable contains spliced-in wires for measuring signals on the two audio channels.

3 Microphone

Making measurements

I used an oscilloscope to explore signal levels all along the audio chain. Since the suppliers prefabricated the cables between the equipment with ⅛ inch stereo plugs and jacks, I built a sampling cable: a short section (few inches) of shielded cable with a stereo plug on one end and a stereo jack on the other. See Figure 2 for a diagram. Room inside the connectors was limited, so I opened up the center of the cable to splice in three colored wires. Heat shrink protected each splice from its neighbors, and a larger diameter heat shrink healed the incision into the cable. The scope probe clips to the shield and one of the two other colored wires in order to sample the voltages inside the cable.

I measured the microphone voltages for each headset at W1KM, a motley collection of various generations of Heil equipment. Initially I wanted to know what the normal range of mic voltages in the simplest configuration: a mic plugged directly into the front panel mic jack of the radio. With one exception, at a normal speaking voice, all of the headset mics provide about 200 mV$_{P-P}$ (peak to peak) on hard consonant sounds such as the "k" and "t" of "kilowatt". When aggressively shouting, the microphones produce up to 400 mV$_{P-P}$ on these hard consonants. Vowels generate significantly lower voltages: about one-quarter the peak levels associated with hard consonants and sibilants.

The exception was the Heil ProSet Plus headset. The boom contains two switch-selectable mic elements: "full range" and

"DX". The full range element produces voltages similar to the other headset, but the DX element generates no more than 100 mV$_{P-P}$. I only had one headset of this model to test, so I do not know if this lower voltage range is typical.

At W1KM we decided to use the Heil ProSet Plus' full range mic element for two reasons:
– In the final listening tests described later, the DX element's narrower frequency range to be less intelligible under some conditions.
– We wished to avoid changing control settings in the middle of a contest. Headsets follow their owner-operator from one operating position to another in our multi-op station. Thus, we preferred mic elements with similar voltage outputs.

I also measured the signal voltages with the mic plugged directly into the soundcard mic input jack. These voltages were the same as with the mic plugged into the radio, indicating the radio mic jack and soundcard mic jack load impedances do not differ enough to affect the signal voltage levels.

Choosing a target voltage level

Let's assume that the station's dynamic microphones, connected directly to the Orion mic input, generates not much more than 200 mV$_{P-P}$ with a normal speaking voice… and perhaps a bit more with an excited operator. Our goal now is to adjust the interconnect box and computer so that this equipment also delivers a clean signal of this same level to the radio's mic input jack.

Why stick to these levels? Examine the signal flow inside the microKeyer. Depending on configuration, when stepping on the footswitch the microKeyer may route the microphone signal directly to the radio via the path marked "mic diversion" in Figure 1. This provides a simple override a computer-generated voice message: stomp on the footswitch and get the computer out of

the way![1] It also permits continued contesting even if the logging software crashes and must be rebooted. Hence, real-time speaking over the air and pre-recorded message playback must arrive at the radio with identical levels.

4 Soundcard mic input audio chain

Now let's determine the proper settings for any microphone amplifiers (points 2 and 3 in Figure 1) and gain controls (point 4) so that this portion of the transmit audio chain delivers an appropriate signal level to the soundcard's A/D converter.

Soundcard level- setting considerations

"Appropriate signal level" here means delivering enough voltage to use most of the encoding range of the converter, while leaving some spare headroom to accommodate the occasional outburst of enthusiasm or desperate shout. We want to employ as much of the encoding range as practical, so that the encoder digitizes the voice more faithfully; i.e., with higher resolution.

Almost every wave file editor provides a graphical picture of the recorded (incoming) waveform. I used Goldwave[2] to examine my soundcard input and recorded wave files. After running a number of tests, I decided to let the normal voice peaks (those hard consonants in "kilowatt") occupy about 80% of the encoder range.

I also wanted to keep the Windows volume control sliders around their mid-range. Many soundcards or motherboard-provided soundcard functions add distortion to signals with the volume control sliders set above 80% of their full-range. At the

opposite end of the scale, selecting a satisfactory level can be difficult if only a few increments at the bottom of the on-screen volume control produce radically different levels of signal during recording or playback.

Configuring Windows

Configure your soundcard's recording volume controls to employ only those soundcard inputs used during contesting. By default, after installation Windows includes other recording volume control sliders for devices such as a CD/DVD drive. Use the Windows volume control Options menu and select "Properties" to open up the dialog box where you select active (usable) inputs and outputs; see Figure 3. By selecting the "recording" button, I could uncheck every audio source except for the microphone input and the soundcard line input (used by my logging software to record receiver audio to disk for an archive of the contest, and also used for RTTY decoding).

Preserving signal to noise ratios

Mic ouput voltages are rather weak. We must minimize stray signals picked up by the cabling between the mic and the soundcard to preserve a high ratio between your voice's voltage (signal) and stray junk (noise). A quality cable with 100% shield coverage provides a good starting point.

At the start of this project, I reasoned that employing the microKeyer's mic preamp (point 2 in Figure 1) would further help to achieve that aim.[3] By boosting the mic signal levels close to the mic, any stray noise picked up in subsequent cabling would have little impact on the signal to noise ratio.

Without this preamp, the voice peaks on the cable between the microKeyer and the soundcard mic input jack reached $150\,\text{mV}_{\text{P-P}}$.

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[1] Some logging software packages automatically abort transmission of a recorded message when the operator steps on the footswitch. Typically the software detects footswitch status on the CTS lead of a serial port.

[2] You may download Goldwave from www.goldwave.come and use all of its features without charge. After a generous number of operations in a single work session, the software occasionally pops up a nagging dialog box, asking for a paid registration. Registration costs US$40 and is required only once.

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[3] I understand that Icom radios support electret mic elements; these elements require a DC bias voltage for proper operation. Soundcards provide this bias voltage routinely. The microKeyer mic preamp does not provide a bias voltage and cannot support electret mics.
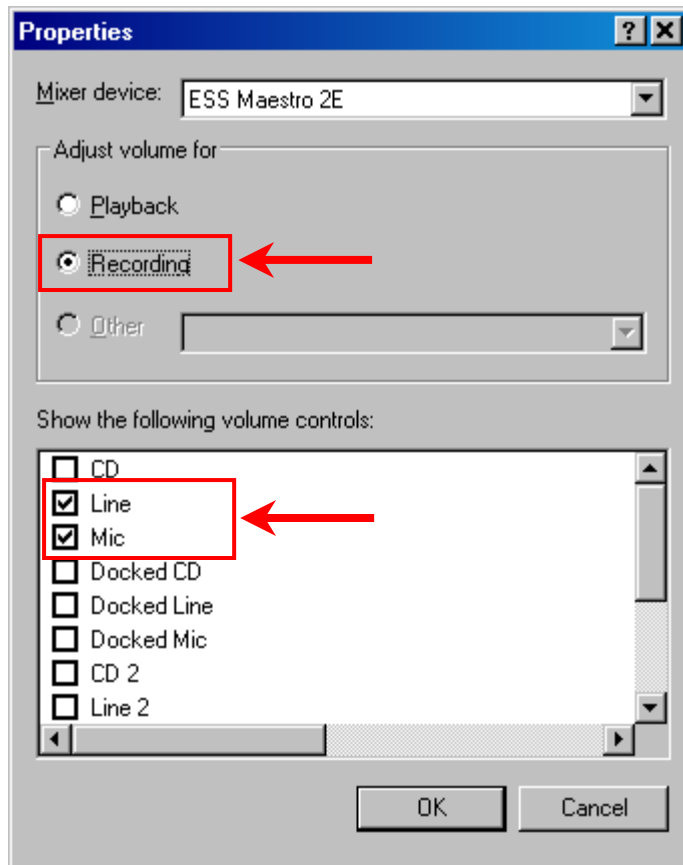
Figure 3: In the Windows volume control, use the Options\Properties menu to open this dialog box. By selecting only the controls needed by your station, you eliminate stray signals from other sources.

(Galvanic isolation, discussed below, accounts for the loss from the 200 mV$_\text{P-P}$ mic output levels.) Routing the signal through the preamp produced a very healthy 2 V$_\text{P-P}$ on voice peaks, about +20 dB gain.

However, this 2 V$_\text{P-P}$ signal completely overwhelmed my sound-card's D/A converter, resulting in very distorted recordings. Even setting the Windows volume control for the mic input (point 4 in Figure 1) at just one pixel above zero resulted in negative-going voice peaks hitting 100% of the converter's encoding range. (Negative-going voice peaks hit the encoder more heavily than positive-going peaks. Because soundcard mic jacks applied some DC voltage between tip and sleeve, it seems plausible the card's circuit did not fully block this DC voltage before delivering the signal to the encoder.)

(I could maintain the advantage of a larger voltage on the interconnecting cable by placing a passive external pad next to the soundcard mic input jack.4 But I preferred not to add more equipment to the audio chain and decided to evaluate the situation without the microKeyer's mic pre-amp and an external pad. As it turned out, we didn't need them.)

When I bypassed the preamp at point 2 and set the Windows volume control (point 4) at maximum, the results reversed. Even very loud speech into the microphone produced only enough oomph to hit 50% of the encoder range.

Most soundcards contain a "mic boost" control, accessible via the Windows volume control dialog. Click the "Advanced" button under the mic volume control's slider (Figure 4). If your volume control does not show an "Advanced" button, turn on the Advanced Controls option under the Options pull-down menu.

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4 Web-based calculators such as www.mcsquared.com/dbframe.htm may help you quickly design an appropriate T-pad passive attenuator. You will have to make some assumptions about mic preamp source and soundcard sink impedances and choose a target level of attenuation. Check your headset and soundcard manuals or manufacturer websites for specifications. The Heil HC5 mic element used at W1KM runs about 2000 Ω. Soundcard mic input impedances often are in the 500–600 Ω range.
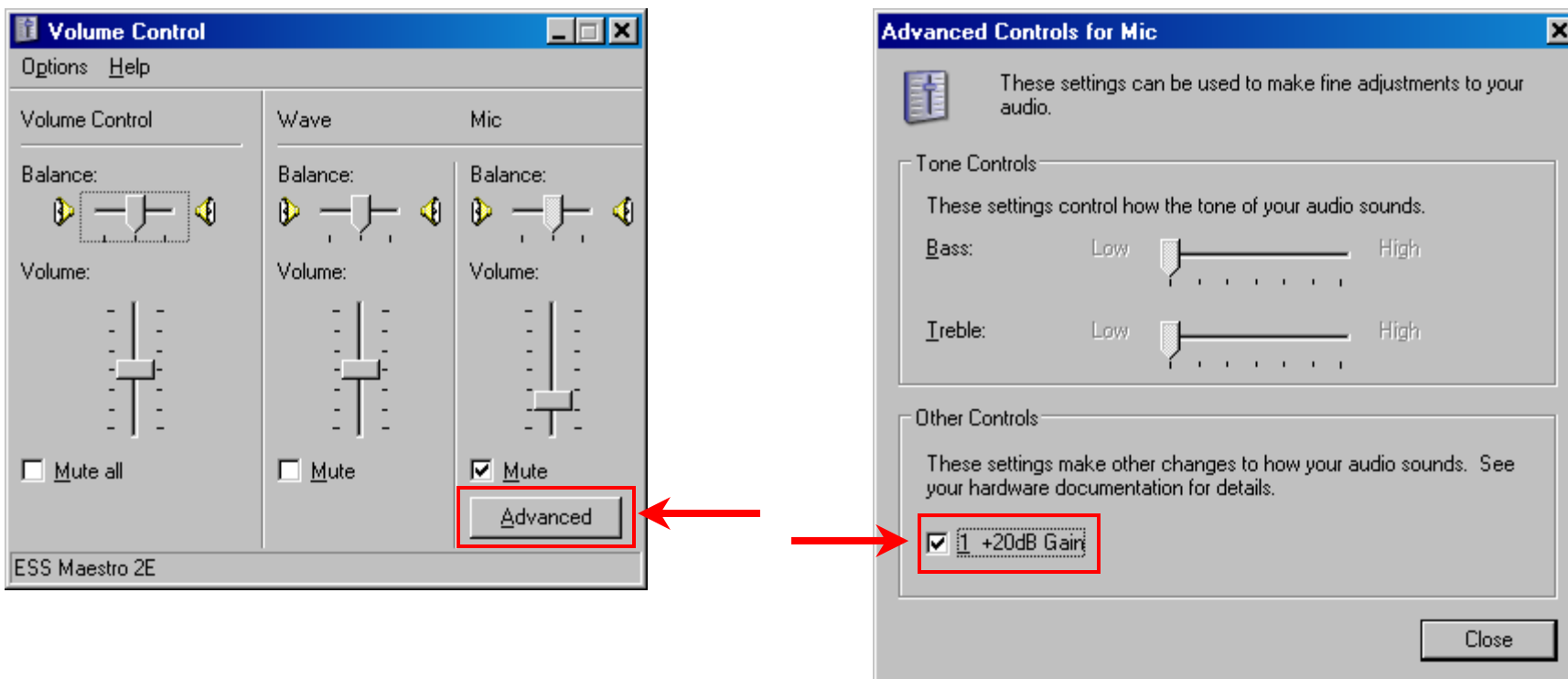
Figure 4: Click the mic volume control's "Advanced" button to reveal the mic boost control. For this particular soundcard in my Dell Inspiron laptop, the mic boost checkbox says +20 dB gain, but actual gain seems significantly less.
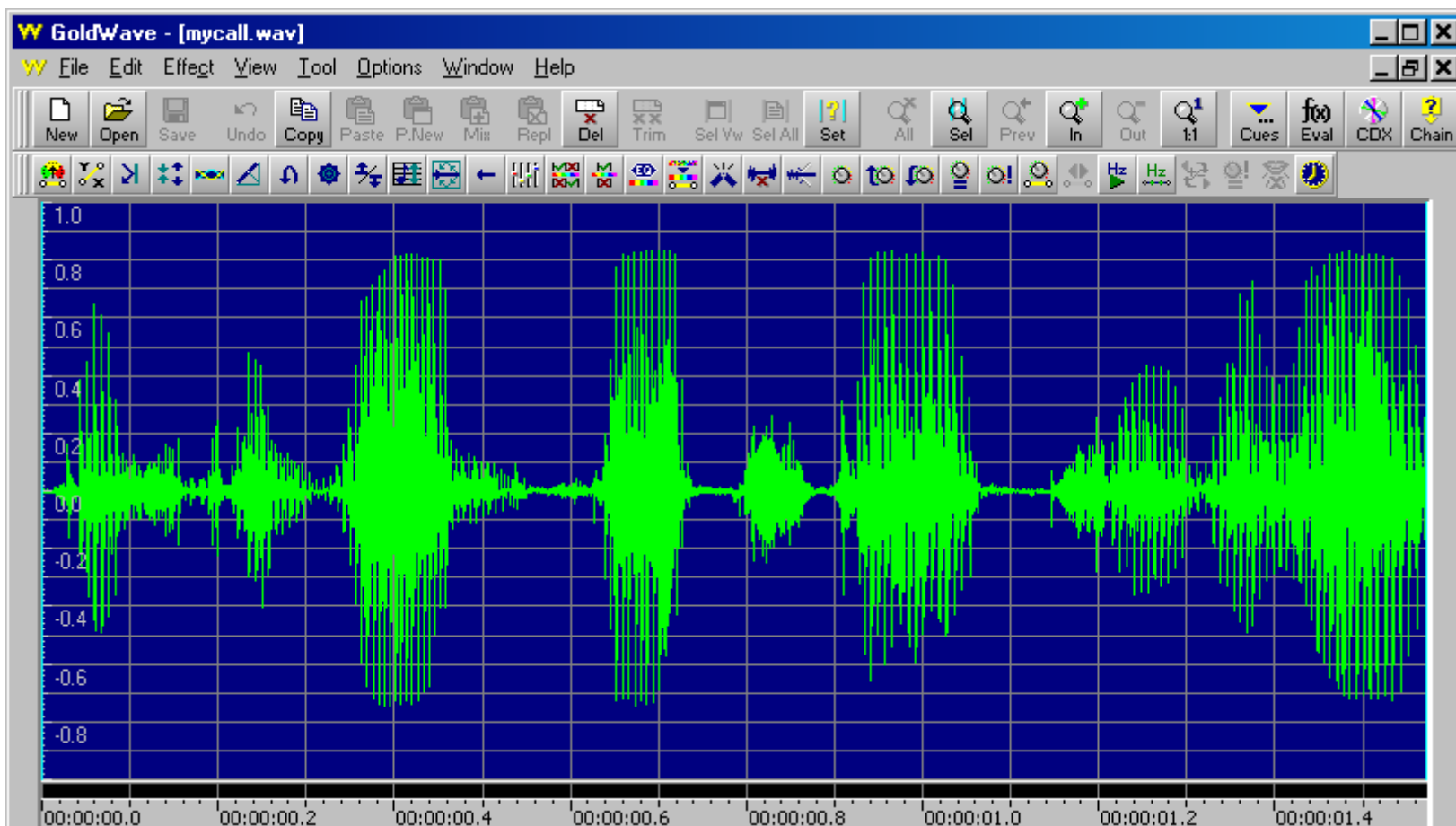
Figure 5: Typical recorded message with peaks of about 80% of the soundcard's A/D converter. This message says "Whiskey One Foxtrot Juliett".

In my case, using the soundcard's mic boost control and setting the mic volume to ¼ of full-scale produced the desired encoding range. Figure 5 shows the resulting recording of our team's callsign in the most recent ARRL DX phone contest. The voice peaks at around 80% of the A/D converter encoding range. Note the low signal level between syllables, indicating a quiet background noise level; more about this later.

5 Soundcard line output audio chain

Having adjusted the controls affecting signal levels between the microphone and its corresponding digital bit stream inside the computer, now adjust the controls affecting the audio generated by the computer and delivered to the transceiver. These controls sit at points 5 through 9 in Figure 1.

More Windows configuration

Open the Settings\Control Panel\Sounds and Audio Devices properties dialog in Windows. Click the Sounds tab in this dialog, and select the "no sounds" sound scheme; see Figure 6. This prevents accidental transmission of Windows' sound effects over the air.

Depending on the capabilities of your soundcard, this properties dialog box may show other tabs or buttons used to control equalizers, quadraphonic (or more) sound channels, and other specialized functions. Try to disable as many of these special functions as possible. When you cannot disable a function, set its parameters for neutral or minimal effect on the output. For example, set an equalizer function to 0 dB across the spectrum.
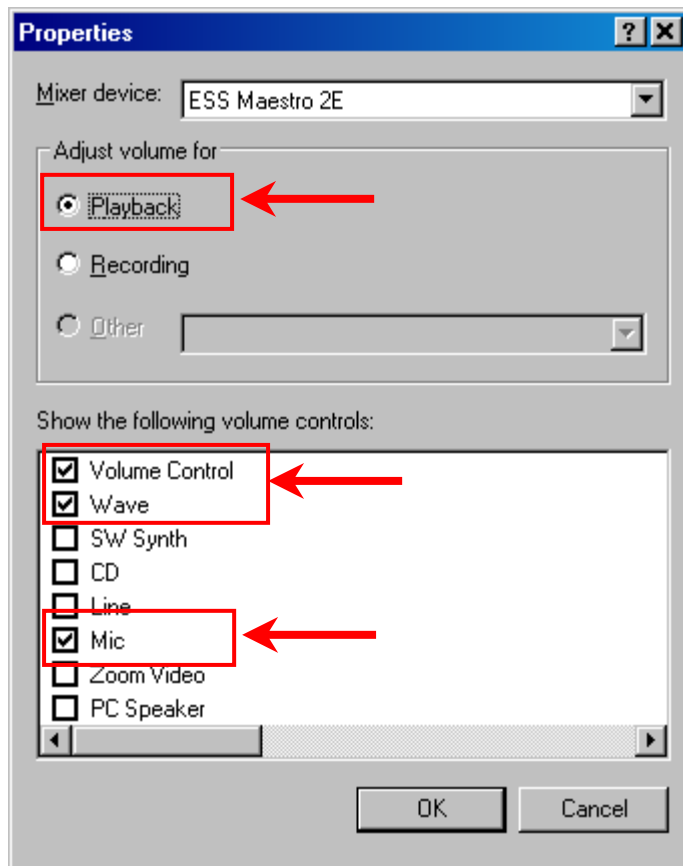
During these adjustments, we want the soundcard to deliver to the radio a faithful reproduction of what it received from the microphone. While you might be tempted to add special effects like reverberation to make your signal distinctive, such modifications will not aid the intelligibility of your signal on crowded bands or after transiting the polar auroral zones.

Another reason for keeping the soundcard's reproduction of your voice as neutral as possible hinges on the use of the footswitch-activated mic diversion function shown in Figure 1. During
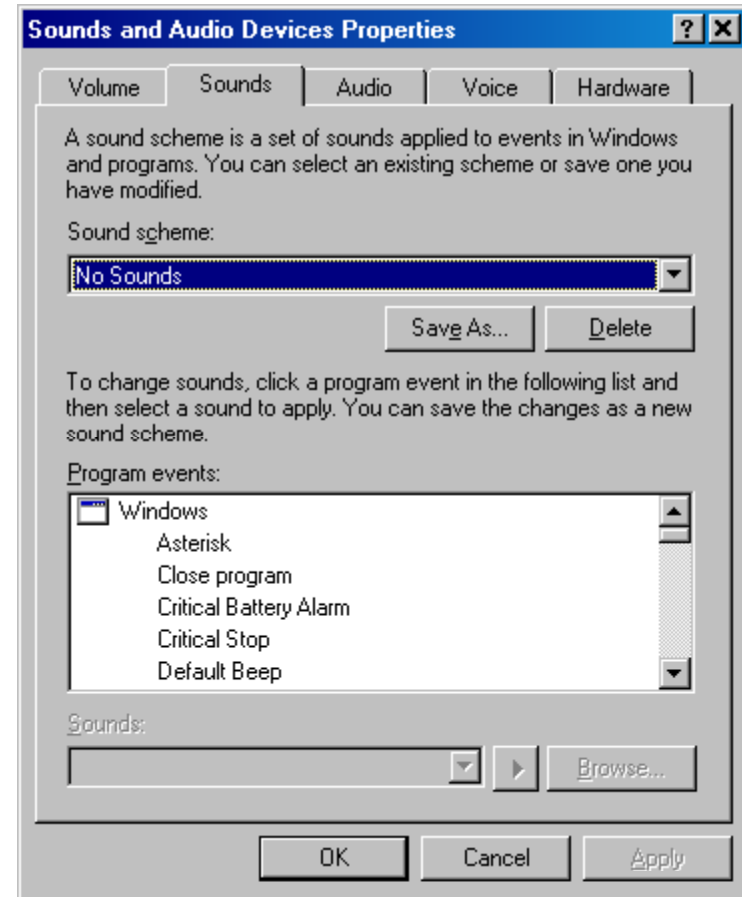
Figure 6: Use the Windows Settings\Control Panel\Sounds and Audio Devices dialog to turn off all sounds generated by the operating system. This step prevents operating system sound effects from going out over the air.

diversion your microphone connects directly to the transmitter without passing through soundcard signal processing. You want your voice to sound the same regardless of the path it takes from microphone to transmitter.

phone" and "wave". As was done for recording, you want to eliminate signals from any unnecessary audio sources.

Setting controls

The goal for this part of the audio chain is to deliver to the radio a signal at the same level occurs as when the microphone connects directly to the radio; i.e., 200 mV$_{P-P}$ for the equipment at W1KM.

I began by attaching the scope to the soundcard line output and was startled to discover a 100 mV$_{P-P}$ signal. At the time no one spoke into the microphone and the computer was not playing a recorded message. Investigation revealed this signal contained a sine wave with a period of 2 µs, equivalent to a frequency of 500 kHz. Since the transmitter's audio stage rejects signals this far outside the audio passband, I could ignore it.

After configuration as described above, the Windows volume control for playback now contains three controls affecting levels for contest operations; I set each to mid-scale as an initial guess:

– The Mic slider (point 5 in Figure 1) governs signal levels when the operator speaks live on-air, and the interconnection box configuration routes the operator's voice into the soundcard and back out again. This slider is not the same as the recording mic volume control, so be sure to adjust the playback sliders, not the recording sliders. In this part of the procedure you will set this control last.

– The Wave slider (point 6) governs signal levels during wave file playback of your recorded messages.

– The Volume control slider (point 7) represents a master gain control for all forms of output from the soundcard's line output jack.

The microKeyer includes:

– A front panel control labeled "radio" (point 8) that adjusts the signal level delivered to the radio. I set this control to 12 o'clock as an initial guess.

– An attenuator trim pot (point 9) for signals delivered to the radio's front panel mic jack. Adjust this pot with a small Philips

Figure 7: In the Windows volume control, use the Options\Properties menu to open this dialog box. Select playback, and check only the three controls listed here. Your contesting software does not require any other sources for transmitting.

Finally, use the Windows volume control program's Options menu to open the Properties dialog. In that dialog, after selecting "playback", uncheck every playback source except for "micro-

screwdriver through a hole on the right side of the cabinet. The full counterclockwise setting represents no attenuation, and I began with this setting. (The box arrives from the factory with this control set in mid-range.)

To provide a consistent audio source, I used my logging software to play a previously recorded CQ message whose peak levels hovered around the recording goal of 80% of the A/D encoder range. The scope probe, when attached to the output of the soundcard, revealed a vigorous 700 mV$_{P-P}$ on voice peaks. The "radio" control in the microKeyer and losses in the galvanic isolator brought this level down to 200 mV$_{P-P}$ at the transmitter's mic input jack – right on target.

The Windows playback mic slider remained as the final adjustment in this phase. First I configured the microKeyer to route the microphone through the soundcard for live on-air speaking; i.e., not using the mic diversion shown in Figure 1. By speaking the same CQ message in a normal speaking voice, I could adjust the mic playback slider setting for the same 700 mV$_{P-P}$ soundcard line output audio signal level on the scope as for the recorded messages. The initial mid-scale setting turned out to be perfect.

6 Transmitter adjustment

Transmitter adjustments vary by model and manufacturer. The description below for the Orion radios at W1KM illustrates the general approach. You should consult your own radio's operating manual for the corresponding adjustment instructions.

The Orion contains seven relevant adjustments:

– Selection of the audio source: front panel jack, AUX rear panel connector, or both. For this article we assumed delivery of the transmit audio to front panel mic jack.[5]

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5 Most modern transceivers provide a second transmit audio connection on the rear panel, usually designed for 600 Ω line level input. On some transmitters the design assumes this signal contains AFSK or PSK tone signals, and bypasses the audio equalization and speed processing stages.

– Audio input gain control for the chosen source.
– Speech processor level.
– Audio equalization.
– Audio low frequency rolloff.
– Transmitted audio bandwidth.
– Power output.

Following the Orion instruction manual and using the CQ message recorded by the logging software, I first adjusted the audio input's gain (mic gain, in this example) until the ALC light winked on voice peaks.

After setting the mic gain, I faced a collection of settings for which an oscilloscope pattern or meter reading provided no clues for correctness. What I needed was the ability to hear the over-the-air signal as any other station would hear it.

To do this, I used another receiver at the station as a *monitor receiver*. At W1KM another Orion radio served this role, but you can use any other good quality receiver for this task.

The signal strength into the monitor receiver must remain at a level typical for a normal signal; e.g., S-9 or thereabouts. Many receivers (including the Orion) will deliver a distorted copy of the incoming signal if that signal drives the S-meter close to its limits. Use these techniques to reduce the signal strength of your 1.5 kW transmitter into the monitor receiver:

– If available, connect the transmitter to a dummy load with sufficient power capability. Enough leakage will occur locally for your receiver to pick up a usable signal.
– Disconnect the antennas from the monitor receiver. If this is insufficient to reduce the signal strength, disconnect the coax cables from the receiver's rear panel. Even a short length of coax to an antenna switch will add some signal strength. If a beverage or other specialized receiving antenna connects to the monitor receiver, leakage in the antenna selection relays

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For the Orion radio, the audio signal paths appear to be identical for both the front and rear panel inputs.

within the receiver can provide more signal strength than desired.

– Crank up the receiver's attenuation setting.

– Attach a shorted connector to the monitor receiver's antenna terminals.

One or more of the above steps will deliver a signal of modest strength to the monitor receiver. At w1km, disconnecting the antennas and setting 18 dB of attenuation did the job.

If possible, begin by setting the monitor receiver's bandwidth to a wide value. Play the recorded message from the computer while adjusting the speech processor level and, if available, bandwidth, equalization, rolloff, etc. These settings may interact, so go through them in turn multiple until they converge on a collection that sounds crisp, authoritative, and clear. Write down the positions or values for each control.

Many phone contesters operate with reduced receiver bandwidths and shifted passband tuning on the more crowded bands. You want your signal to be easy to understand under these circumstances as well. Change the monitor receiver's bandwidth and passband tuning controls. I cranked the monitor receiver down to the 1.8 kHz roofing filter, a DSP filter of 2.1 kHz, and +100 to +200 Hz of passband offset.

Now listen to your recorded messages again. Are they still easy to understand? Try some adjustments in either direction to the speech processing and other transmitter controls to see if further improvements in intelligibility result. Try some other settings on the monitor receiver that might be used by contesters listening on crowded bands, and check again to see if the intelligibility of your signal remains good over a wide range of monitor receiver settings.

During these steps, check that the transmitter power output remains appropriate for your class of entry. If using a follow-on power amplifier, check that the amplifier remains properly tuned and the transmitter delivers the correct output power to the amplifier input.

After a few minutes' work, I settled on these values for the transmitting Orion: 2700 Hz bandwidth, 200 Hz low frequency rolloff, and –5 dB equalization (slightly favoring the higher end of the voice).

Don't try to make these adjustments while talking yourself, or having another operator talk into the microphone. The sound of your voice or the other person's voice will overwhelm what you hear in the monitor receiver, preventing you from forming a good opinion of the transmitted signal quality.

Similarly, don't rely on the transmitter's "monitor" function for a quality check. The transmitter monitor does not accurately emulate a real receiver listening to the on-air signal.

7 Final checks

I used a special feature in my wave file editing software to simplify the following tests. Goldwave can create sounds by formulae or by selecting some standard sounds (sine wave, DTMF tone pairs, etc). I created a few wave files for experimentation and testing purposes:

– A 1500 Hz sine wave at 100% and 85% encoding range.

– White noise at 100% and 85% encoding.

– A sequence of DTMF tones at 100% and 85% encoding.

Each file lasted about 20 seconds. By assigning these files temporarily to some of the spare message playback keys in my logging software, I could play any of these test signals conveniently.

Amplifier linearity check

Since you have made the transmitter adjustments at the full power level appropriate for your class of entry, the monitor receiver's audio has included any impairment introduced by the power amplifier (if used).

An oscilloscope can provide a quick check for linear operation of an amplifier. Wind a length of hook-up wire around the coax line connected to the amplifier input and another length around the output coax. Increasing the number of turns will increase the signal picked off the coax; I find this simple approach picks up

enough signal to deflect the scope beam. Connect one wire to the scope's x input via a probe, and the other wire similarly to the y input. Adjust the x and y gain controls so that, while transmitting the recorded message, a diagonal line of around 45° appears on the scope.

Examine this line, focusing particularly on modulation peaks. If the diagonal line remains straight and the audio quality is good in the monitor receiver, the amplifier is operating linearly (good!). If the tip of the diagonal line starts to deviate away from straight, even just a little amount, then the amplifier has moved into non-linear operation. You're definitely transmitting some garbage on the bands then!

The 100% encoding level test tone files are particularly helpful here, as we want to be sure the amplifier remains in linear operation even if the operator speaks louder than normal.

Signal width checks

Use the recorded messages and the test tone files as source material. Tune the monitor receiver away from your transmit frequency, and listen critically nearby. Since the transmitter's signal strength in the monitor receiver remains at moderate levels, listening on an adjacent frequency reveals how your signal sounds to other contesters working next to you. If your monitor receiver registered s-9 when on-frequency, but 3 kHz away still sees your transmitter lifting the s-meter off the pin, then your signal is too wide: check again that the transmitter bandwidth is set correctly and the final amplifier stages (transmitter and external power amp) are properly tuned and not overdriven.

If your monitor receiver can be set for a wide bandwidth, and you have a second computer available, then you may be able to examine the audio frequency response curve for the transmitting soundcard, interconnection box, transmitter and amplifier. Set the monitor receiver to the transmit frequency, connect its output to the second computer and start a spectrum display program such as Spectran. Transmit the white noise wave file. The spectrum display software, when set to the frequency domain, will show an audio response curve for the combination of the transmitting equipment and the monitor receiver. If the monitor receiver, with its receive bandwidth set wider than the transmitter, has a reasonably flat response, and the connection to the second computer's soundcard has not been overdriven or distorted, then the overall curve shape should be largely that of the transmitter.

Background noise

Use the monitor receiver to listen to the recorded contest messages. Can you hear anything in the background between syllables? Extraneous background noises can reduce the intelligibility of your signal, so try to identify and eliminate sources of noise in your shack.

A wave file editor also can help you judge background noise levels. Figure 5 shows very little signal between the syllables recorded in this wave file. Expanding the vertical scale in the editor reveals that the quiet parts of this message use just ½% of the encoder range, compared to 80% at peaks. This ratio represents 44 dB, which seems like a lot. When watching the power output meter of the amplifier, the transmitted signal drops from 1.5 kW to just a minimal indication. But when a DX station hears the w1km site's signal at s-9 plus 20 dB (and I hope we often exceeded that, hi!), the background noise level hovers at s-4. We have hardly achieved the tomb-like quiet of a broadcast studio – and, for a contest, probably don't need to. Forty dB seems plenty to allow the message to stand clearly above the residual noise.

We grow accustomed to the sounds within our own stations. Sometimes listening to a recorded empty message, made with a live mic but no one speaking, and then transmitted over the air and received with the monitor receiver, reveals the nature of a background noise source that eluded earlier notice.

Make notes!

Finally, draw a diagram, annotate it with measured voltages, and write down all the settings for future reference.

8 Other aspects

Galvanic isolation

The audio connection between computer soundcard and radio must contain galvanic isolation, typically achieved with 1:1 transformers. Without isolation, small short-term differences in currents circulating on the radio and computer chassis impose noise on the audio leads. Not only does this noise contaminate the transmit audio recorded messages and on-air signals, but frequently travels across the radio chassis to appear in the operator's headphones. Concentrating on accurate reception in crowded bands during the long hours of a contest is hard enough work – we need not make this task more difficult with additional locally-generated noise!

Many radio-computer interconnection articles suggest the inexpensive Radio Shack 1:1 isolation transformer part number 273-1374 for use as a galvanic isolator. This transformer performs poorly at audio frequencies. The small core saturates relatively easily, causing audio frequencies to mix non-linearly and generating distortion products. If you have an interest to explore this effect, try feeding two low- or mid-range audio tones into the transformer and examining the result at your soundcard in the frequency domain with audio spectrum display software. As the strength of the signal increases, you will see spurious mixing products at other frequencies appear on the display.

Radio Shack's website also lists a "ground loop isolator" for two audio channels, part number 270-054, for about $17. While the product description seems promising, the website specifications remain sketchy. I haven't tested this device.

Isolation transformers in the broadcast industry are expensive ($50 and up), large, heavy, and perhaps overkill for the audio frequency range used in voice or RTTY contesting. Perhaps readers can suggest other sources for appropriate isolation transformers.

External audio amplifiers

The microKeyers used at W1KM conveniently include both galvanic isolation transformers and a mic amplifier. If you build your own interconnection box, you may wish to include one or more audio amplifiers. Many on-line and retail electronics suppliers carry a simple, inexpensive (around $7) audio amplifier kit. The kit employs an LM386 audio amplifier chip, includes a small circuit board and a handful of parts including a gain-adjustment trim pot. Search for "lm386 audio kit" on the Internet to find suppliers. The kits run off a wide range of supply voltages; use your station's +12 Vdc supply or a 9 Vdc battery. The tiny current demand means one battery lasts for many contests.

Instrument characteristics

I used an oscilloscope to measure voltages, in part because I was curious about the peak voltages. A common voltmeter will not provide accurate RMS readings on anything other than a sine wave, and can not capture peak voltage values. During another project I attempted to use an inexpensive digital AC voltmeter to measure sine wave audio tones, but discovered that this meter read accurately only at the power line frequency of 60 Hz! The fine print in the Specifications section of the owner's manual confirmed my suspicions: the sensitivity of the meter declined dramatically as frequency increased across the audio spectrum. Even at 500 Hz the RMS AC voltage measurement were far from reality. If you wish to use a digital AC voltmeter for audio measurements, check the specifications first! And be aware that a voltmeter cannot reveal distortion on voice peaks.

Wave file encoding algorithm

Contest messages are short. Little justification exists for using compressed encoding algorithms, each of which introduces additional processor burdens and delays. Stick to a simple algorithm with high resolution; e.g., PCM signed 32-bit encoding, mono, at a speed of at least 11 kilo-samples per second.

AFSK and PSK operations

This article focuses on adjustments for voice operations. Modes such as AFSK and PSK have additional considerations that may require different control settings for the transmit audio chain. Other articles published recently in the NCJ and on websites devoted to these modes cover these considerations. Again, take notes on your configurations and settings so that you can switch modes quickly with confidence.

9 Conclusion

Although the procedures outlined above may seem rather lengthy and daunting at first, the task actually goes quickly. Check your results with some other stations on the air, asking them to listen not only to your signal, but also above and below for any splatter or excessive bandwidth. You should find that these steps result in excellent reports.

For us at the W1KM contesting site, the confirmation came Friday night during the ARRL DX phone contest in a 40 m multiplier pileup. A friend happened to be on that contest expedition. Although I was using a different callsign and my friend had no advanced knowledge of my location, he answered with the remark "Gee, that sounds like Eric. Nice audio, Eric! Can you work us on another band?"

A follow-on article will look at the receive audio chain. While the receive audio chain much simpler than transmit, you may be surprised at what we can do to improve reception.

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