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# Setting Your TNC's Audio Drive Level

## Why it's important, and how to do it...

### by John Ackermann N8UR

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I'd lay reasonable odds that most 1200 baud packet radio stations are getting less range than they should, and generating more retries than they need to.

Why?

Because their TNCs are overdriving the microphone input on their radios. The audio output control on many TNCs is very touchy, and the difference between proper audio and too much may be only a few degrees on the control. If you've just plugged your TNC into your radio and gone on the air without adjusting the level, the odds are very good that you are transmitting a distorted signal, and your packet radio performance is suffering as a result.

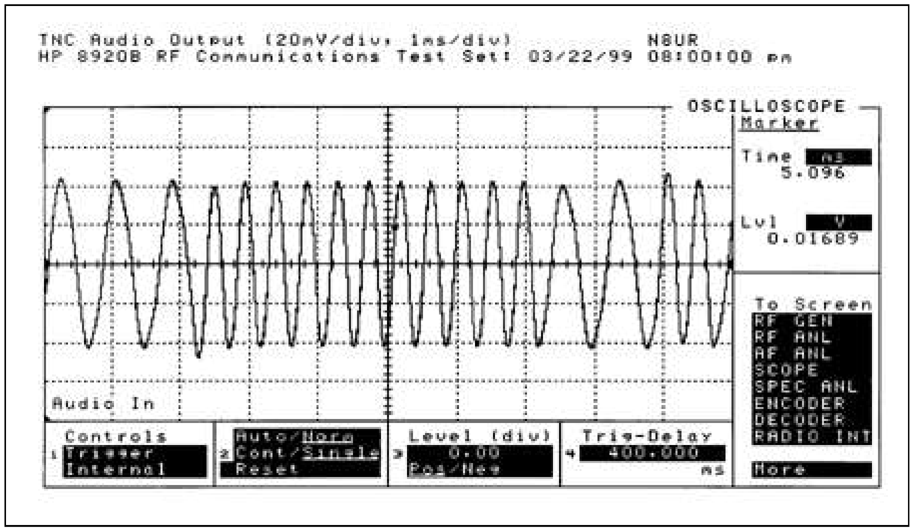
The purpose of this page is to show why this is a problem, and how you can fix it -- whether or not you have a bunch of fancy test equipment.

If you're anxious to learn how to adjust your TNC without worrying about why you need to, you can go directly to Fixing the Problem, but I don't recommend it.

## Understanding the Problem -- The Packet Radio Audio Path

1200 baud TNCs generate two tones, usually 1200 and 2200 Hz, to represent the one's and zero's of the binary data they are fed by their host computer. Those tones are fed to the input of an FM transmitter, which transmits them just as if they were voice. This form of modulation is called Audio Frequency Shift Keying, or "AFSK".

Here's what an oscilloscope display of the audio coming out of a TNC looks like. The closely spaced waves are cycles of the 2200 Hz high tone, and the more widely spaced waves are the 1200 Hz low tone (by the way, each cycle of 1200 Hz represents one bit of 1200 baud data; just under two cycles of 2200 Hz represents one bit). Note that both tones are at the same amplitude.

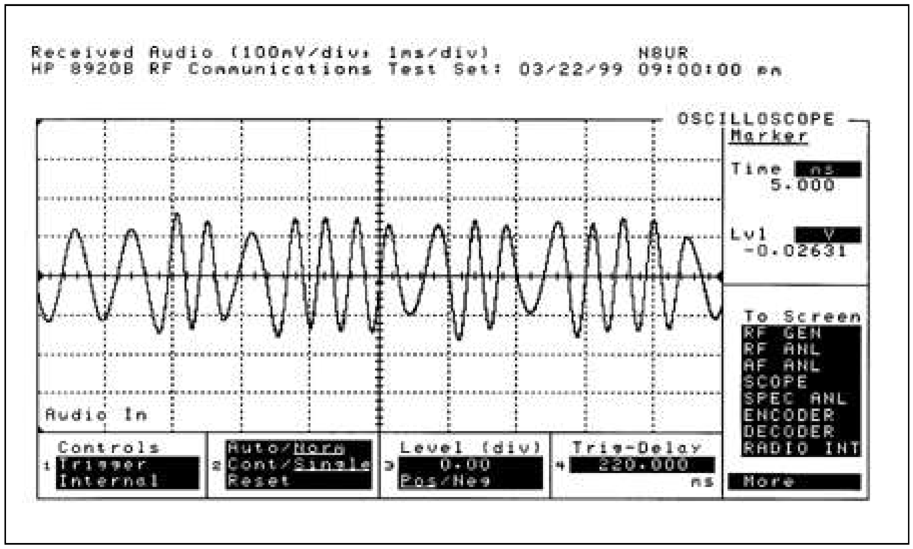



The receiver at the other end of the link recovers these tones, and sends them via its speaker jack to the TNC, where a demodulator (the "dem" part of a modem) converts the tones back to binary form for further processing. For best performance, the tones fed into the demodulator should be at equal amplitudes, or biased a bit toward the high tone. If there's a lot of imbalance (or "twist") in the tone levels, demodulation performance is likely to suffer.

More important, if the low tone coming out of the speaker is stronger than the high tone, many TNCs based on the TAPR TNC-2 design, including the very popular MFJ-1270 series, will have great difficulty demodulating packets. The demodulator chip in these TNCs, the XR-2211, is particularly unhappy if it sees a low tone that is louder than the high tone. Because XR-2211 demodulators are so common, it's very important that we transmit signals that these units can decode, and that means signals that do not have a a twist favoring the low tone.

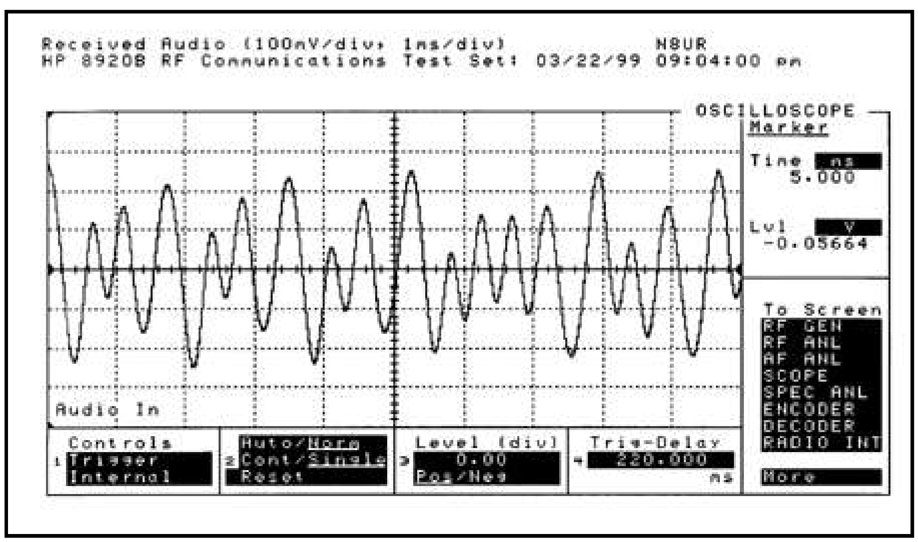
The key message underlying this page is that *packet tones should come out of the receiver speaker at equal amplitudes, or with the high tone a bit louder than the low one.* ***Under no circumstances should the low frequency tone be louder than the high tone***. A properly adjusted packet station meets this requirement.

Here's an oscilloscope picture of packet tones at the speaker jack of a receiver. The TNC and transmitter generating these tones were adjusted to provide proper deviation (more on that later), and the tones at the transmitter's input were of equal amplitude.




In a perfect world, the two received tones would also be at the same amplitude. Here, you can see that the high tone is somewhat louder than the low one; the transmitter I used for this test boosts the high frequencies a bit more than it should. Within reason, that's not a problem; a twist in the opposite direction, though, is a different matter. By the way, you can click here for details of the test setup I used for these pictures.

Here's an oscilloscope picture of the tones that come out of the speaker when the TNC is over-driving the transmitter.




You can see that now, the lower tone is much louder than the high tone. That's the opposite of what we want, and a TNC will have a lot of trouble decoding this packet. Unfortunately, if you've never paid attention to the transmit audio level setting on your TNC, there's a good chance that this is what signals from your station look like.

Why does this happen?

The FM radios we use were designed to transmit the human voice, not digital tones, and there are a couple of circuits in these radios designed to improve voice transmission. Unfortunately, these circuits do very bad things to packet tones if the audio input level is too high.

The first circuit (actually, two circuits -- one in the transmiter and a complementary one in the receiver) is made up of the **preemphasis** and **deemphasis** networks, and the second is the transmitter's **audio clipper**. Each serves an important purpose in voice communications, but when combined with excessive transmit audio levels from the TNC, they can distort packet data beyond recognition.

## Preemphasis and Deemphasis

Weak FM signals have an annoying hiss that reduces intelligibility. Preemphasis and deemphasis are a tool to reduce that noise.

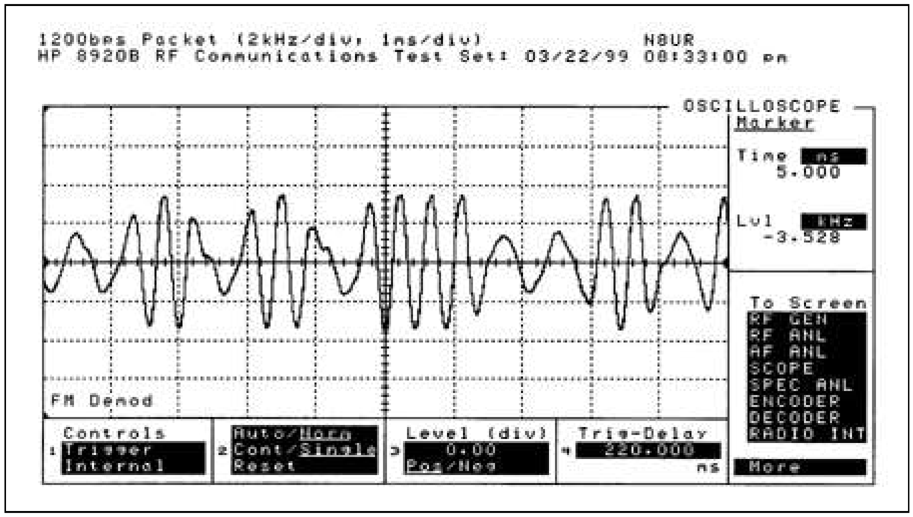
The problem is that most of the audio noise in a radio circuit is at high frequencies, not low ones. Unfortunately, so is most of the information in your voice (low frequencies make voice sound good, but high frequencies make it intelligible). And to make things worse, most of the power in our voices is in the lower frequencies. As a result, it's hard to understand someone whose voice is competing with high-frequency noise or hiss.

To get around this problem, someone had the bright idea of boosting the high frequencies in the transmitted audio (like turning up the treble control on a stereo), so that they are louder than they should be, and then cutting the highs back by the same amount at the receiver -- in effect, turning the treble control down. That brings the voice signal back to its proper tonal range, but also knocks down any noise that was introduced along the way.

The end result is that voices sound normal, but hiss is reduced -- the signal-to-noise ratio is improved. It turns out that this works really well for voice signals, and it's standard practice in all FM voice transmitters today. (And, by the way, this is how Dolby noise reduction removes hiss from cassette tapes.)

The boost that's given at the transmitter is called **preemphasis**; it's a high-pass filter that emphasizes the high frequencies in the audio spectrum (hence the name). Preemphasis increases the level of high frequencies by 6dB per octave -- in other words, the voltage fed to the modulator for a tone of 2000 Hz is twice as high as the voltage for a tone of 1000 Hz. Preemphasis should cause the 2200Hz high packet tone to be about 5dB louder than the 1200 Hz tone. (Note: this discussion is about voltage and not power ratios, so 6dB represents a doubling, not 3dB.)

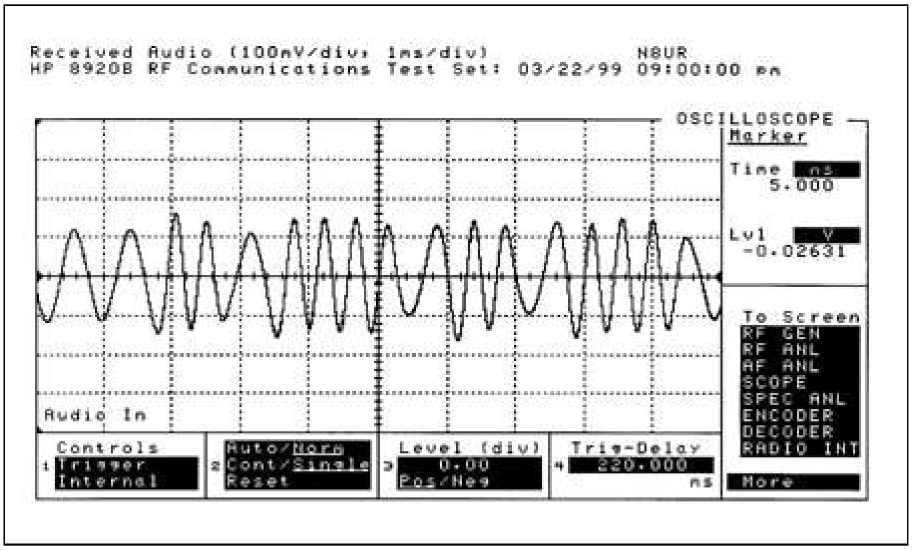
Here's what a properly deviated packet radio signal, with preemphasis applied, looks like coming out of the transmitter.




The high tone has nearly twice the amplitude of the low tone; that makes sense since the preemphasis boost is 6dB/octave -- 2200Hz is nearly double 1200Hz. (There's a little bit of distortion visible in the low tones in this picture; I'm not sure where that is coming from -- it may be an artifact of the measurement setup.)

The complementary circuit in the receiver provides **deemphasis** of 6dB per octave, but in the opposite direction -- the receiver circuit acts as a low-pass, instead of a high-pass, filter. The combination of the transmitter's preemphasis and the receiver's deemphasis results in the received signal matching what was at the microphone input, with the benefit of reduced hiss.

This is what a properly deviated signal looks like at the receiver's speaker terminals; the deemphasis circuit pretty much (though not exactly) cancels out the boost that preemphasis gives to the high tone.




## Audio Clipping

By themselves, pre- and de- emphasis don't cause problems for our AFSK tones. However, when they interact with another circuit that's included in communications transmitters, trouble lies ahead.

That circuit is the audio clipper. It's a handful of components designed to make sure that no matter how loud the audio input signal, the transmitter doesn't deviate more than a set amount (usually 5kHz for 2-way radios).

If the audio input is below the clipping point, the signal passes through the clipper untouched. If it is above that point, the signal is chopped off (or clipped) at the magic level. This makes sure that voice peaks don't cause overdeviation.

Because the clipper is a guardian that makes the transmitter stay within its legal deviation limits, it is usually the last part of the audio chain, right before the modulator. Most important, it follows the preemphasis circuit. That's worth repeating -- *the clipper works on preemphasized audio*.

Why is it worth repeating? Because if the clipper operates on a preemphasized signal, ***the high tones are more likely to be clipped than the low ones***. And that, my friends, is why this discussion is relevant to packet radio.

## Emphasis Plus Clipping Equals Trouble

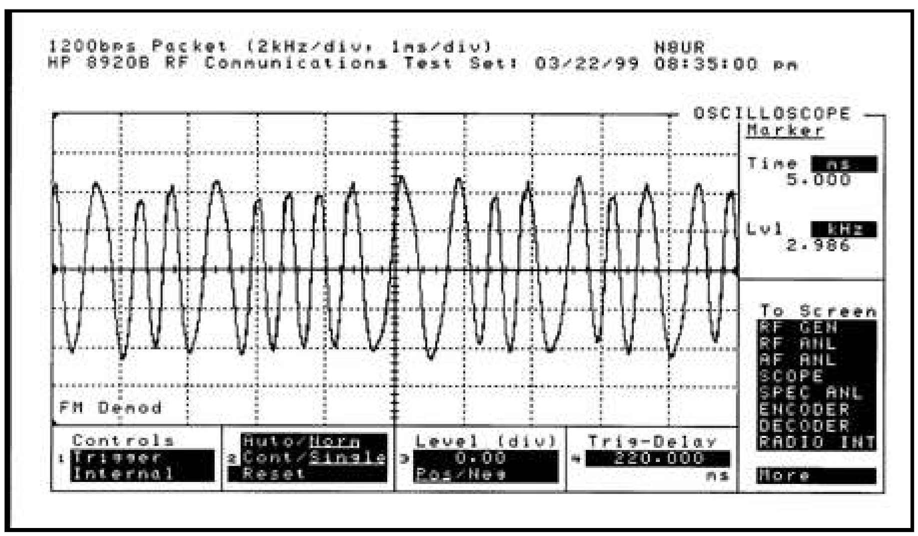
If the audio level the TNC feeds to the microphone jack is too high, the preemphasis circuit will still work normally, and boost the level of the high frequency tone by 6dB/octave -- making our 2200 Hz tone nearly twice as strong as the 1200

Hz one. However, the clipper will knock both tones down to the maximum allowed voltage. The result: the level difference between the two tones will decrease, and if the overdrive is great enough, will disappear entirely. In that case, the 1200 and 2200 Hz tones will be transmitted with the same deviation level. In addition, because the clipping process flattens off the top and bottom of the input wave, it adds harmonic distortion to the transmitted signal.

When the receiver picks up a signal that's clipped enough to equalize the deviation of the two tones, it will still deemphasize them, dropping the high tone by almost half compared to the low one.

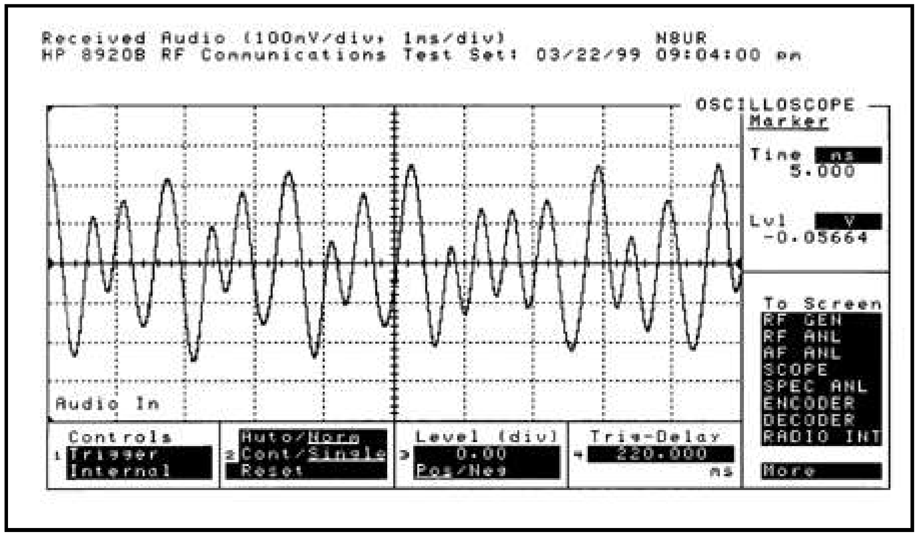
What was a good idea is now a bad one; due to clipping, the deemphasis circuit is no longer acting on a preemphasized signal. Instead of seeing two tones at the same level, the TNC's demodulator will see a 2200 Hz tone that's significantly weaker than the 1200 Hz tone. And, of course, that's exactly what the TNC **doesn't** want to see.

Here is the signal an over-driven transmitter puts out after the tones have been subjected to preemphasis and clipping.




The clipper has chopped both the high and low tones at the same level; there's no trace of preemphasis left. In fact, the low tone is actually a little bit louder than the high tone -- that's probably caused by the other distortion that "hard" clipping introduces.

You may think that this signal doesn't look too bad, but remember that this picture is *before* the receiver's deemphasis acts on it. Here's what the signal looks like by the time it gets to the receiver's speaker.




This picture is guaranteed to give a TNC fits. Recall that the demodulator chip used in the most popular TNCs particularly hates this condition -- it wants the high tone to be stronger than the low one. Even if the signal is very strong, the TNC will have trouble decoding it. If the signal is weak, and there's additional noise riding along with the tones, the chances of successfully decoding a packet become very slim.

***That's*** why setting your TNC's audio level properly is important. You can more than double the effective range of your station by sending packets that are easy for other stations to decode.

## Fixing the Problem

Now that we know what the problem is, how do we fix it? I'll describe three different methods, starting with a quick-and-dirty approach that will get you into the ball park even if you have no test equipment available.

The main goal is to get the audio level low enough that the preemphasized high tone gets through the transmitter without being clipped. Then, we'd like to optimize the signal a bit further to get the best performance we can.

Before we get started, here's one tip that may make the adjustments easier. You might find that there's only a tiny bit of difference in the audio level setting between no tone at all, and full clipping. That was the case with the TNC/radio combination I used for these tests. You can improve that situation by putting a resistor in series with the audio lead to the microphone. Finding the best value may take some experimentation, but something in the range of 4.7kohm to 47kohm will probably let you get the level setting control up into a range where adjustment is much easier.

## The No-Test-Equipment Packet Adjustment System

There's no excuse to transmit an over-driven packet signal. You can get your TNC's audio level set "close enough for government work" with nothing more than another receiver -- anything that will receive the transmitted frequency will do.

Here's what you do:

1. Hook up the TNC to the radio, set the radio to low power and (ideally) connect the antenna jack to a dummy load.

2. Tune the second receiver to the same frequency and remove the antenna; ideally, also connect its antenna jack to a dummy load (the idea is to keep the receiver from being overloaded by the transmitter's signal).

3. Following the instructions for your TNC, put it into "Cal" mode and key the transmitter. Here are the commands to do that on an MFJ-1270 TNC:

    a. At the command prompt, enter CAL.
    b. Press K to key the transmitter. (The TNC's timeout timer may limit the keyup time to ten seconds or so; if the transmitter cuts out before you're finished, unkey and then key again.
    c. Press the space bar to toggle between tones until you hear the high tone in the receiver's speaker.
    d. When finished, press K again to unkey the transmitter, and Q to quit the calibration routine.

4. Make sure that the TNC is sending the high tone.

5. Increase the TNC's TX Audio potentiometer (the instruction manual will tell you where it is) until the tone you hear in the receiver doesn't get any louder -- which indicates that the transmitter is clipping -- and then bring it back down again until you hear the tone get noticeably softer.

6. Turn the pot down a little bit more for good measure, and then toggle the tones to make sure that both are still coming through. It's better to have the deviation a little too low, than too high.

That's all there is to it. You've now set your TNC so that both tones will pass through the transmitter without clipping, and with preemphasis intact. You are transmitting a signal that's pretty close to what it should be.

## The Homebrewer's Method

The steps above will get you in the ballpark, but if you're interested in knowing your signal's deviation with a bit more precision, there's a simple way to do it that requires nothing more than some ingenuity, the willingness to open up a receiver and stick a wire in its guts, and an oscilloscope. The receiver can be a cheap scanner, and an old 'scope will work fine -- all it needs to do is display audio frequencies.

The end result will be a setup that can measure the deviation of an FM transmitter with surprising accuracy, and will let you adjust the audio level of your TNC to the deviation level you want. It can also be used to measure the relative frequency error of a transmitter, and to set deviation levels for voice radios.

## But What is Deviation, Anyway?

Before designing our deviation meter, we'd better know what we're planning to measure.

**Deviation** is the term used to describe modulation of an FM signal. The audio fed into the microphone input causes the transmitter's frequency to shift up and down from the center carrier frequency; the louder the input, the greater the shift. That instantaneous frequency shift is the transmitter's deviation. Deviation is usually measured as the peak positive or negative shift from the carrier (in other words, it's "center to peak", not "peak to peak").

The oscilloscope pictures shown on this page with the words "FM Demod" in the lower left corner are calibrated to show 2kHz of deviation per vertical division of the display. Because we're looking at an audio signal after it's been demodulated, the frequency shift around the carrier has been converted to amplitude on the 'scope -- the larger the displayed signal, the greater the devation. The goal of this project is to display a similar calibrated signal on your oscilloscope's display.

The radios that we normally use for ham radio FM, and which we are likely to use for 1200 baud packet, are designed for a maximum deviation of 5kHz, and their clipper circuits usually kick in at that point or a little lower.

What should the deviation of a packet signal be? We know that the high tone needs to be less than the clipping value, so you might think that "louder is better" and that we should run the deviation right up to the point where clipping begins. That's not necessary; it makes your transmitted bandwidth wider than it needs to be, and it can actually reduce performance in some cases.

Everyone has their own pet value for deviation, but most of the recommendations I've seen are to set the high tone to around 3kHz deviation, and let the low tone fall where it may. The range of settings I've seen published goes from 2.8kHz to 3.5kHz. I usually aim for 3.25kHz or so myself, but anything in this range is close to optimum.

If the transmitter's preemphasis is working correctly, 3kHz deviation of the high tone will yield low tone deviation of about 1.65kHz -- since the preemphasis is 6dB (double the amplitude) per octave (double the frequency), the low tone will be deviated 1200/2200, or 0.545, as much as the high tone.

## Building the Deviation Scope

Since this is a homebrewer's project, I won't provide detailed instructions, but this should get you going.

1. Find the discriminator output in the receiver, and attach one end of a 47kohm to 100kohm resistor to this point -- you may have to experiment to find a value that will provide enough signal without loading down the discriminator. Make sure that you tap into the discriminator *before* any coupling capacitor -- you need to have DC coupling to make this work. Attach the other end of the resistor to a piece of shielded wire, and route it outside the receiver. Attach the shield to ground in the receiver.

2. Set the scope for DC coupling, and attach the probe to the wire. Turn the receiver on, and adjust the scope's vertical sensitivity until the noise just about fills the screen.

3. With the transmitter set to low power and feeding a dummy load, and the receiver tuned to the transmitter's frequency and also hooked to a dummy load, key the radio with no modulation applied.

4. The trace should now be a single horizontal line. Adjust the scope's vertical position control to center the trace on the screen.

5. Either adjust the transmitter up 5kHz in frequency, or the receiver down 5kHz (do one but not both!) and transmit again. The horizontal trace on the scope should now be displaced either up or down from the center point. If you can, adjust the vertical sensitivity control so that the trace is exactly 5 divisions from the center line. If you can't do that, just note as accurately as you

can the number of divisions of displacement, and in what direction. From this point on, *do not touch the vertical sensitivity control*.

6. Shift the frequency of the transmitter or receiver 5kHz to the opposite side of the desired frequency, key the transmitter, and once again note the amount and direction of displacement. The amount should be very similar to the result of the previous step, but in the opposite direction; if it is not, the receiver is probably out of alignment and you should take care of that before attempting to measure deviation.
7. Divide the displacement you measured in each of the previous two steps by five. The result is the divisions per kiloHertz of your measurement system. For example, if the 5kHz frequency change caused the trace to move exactly five divisions, each division equals 1kHz of deviation.

You've calibrated your system to measure deviation. You can transmit cal tones from the TNC as described in the previous section, and measure the amplitude of the recovered audio. Applying the kHz/division value you calculated above, you can easily determine the deviation of your transmitter, and adjust the TNC's audio level to get the proper value.

Since this system is DC coupled, it will also show the frequency offset of the transmitter as compared to the receiver. This is another valuable piece of test information, but it can make measuring deviation more difficult. If you want to look only at deviation and not frequency offset, you can switch the scope input to AC coupling when you're making measurements.

## The Expensive Way

Earlier, I said I'd describe three ways to adjust your packet station's transmitted audio. With two down, the third method is very simple. If you have access to a deviation meter or service monitor, you can use that. The procedure is the same as in the previous steps, except you won't need to go through the calibration steps.

Although these instruments may provide an analog or digital meter that displays deviation directly, I think it's still very valuable to display the deviation on a scope if one is available (most service monitors can do this). You can learn a lot about the overall quality of the signal from the scope display.

You may be surprised to find that a deviation meter isn't out of your, or your club's price range. There have been a number of units sold for the ham market for a couple of hundred dollars or less (including a Heathkit unit that was very popular and can still be found in flea markets). A few years ago, TAPR sold a deviation meter kit that interfaced to a scanner and provided a digital display. If you can find one, it works very well. AEA and MFJ have also sold deviation meters at ham prices. And, older communication service monitors can sometimes be found for surprisingly low prices, but beware -- some of these units can be frightfully expensive to fix if they're broken.

## An Idea for the Adventurous

Here's an idea you may want to consider if you're not afraid to take the cover off your packet radio transceiver.

Consider some of the facts we've been dealing with:

- The TNC demodulator doesn't mind if the high tone is somewhat louder than the low tone.
- The TNC hates it when the low tone is louder than the high tone.
- A clipped packet signal doesn't look too bad before deemphasis is applied; the two tones will be at about the same level; it's only after deemphasis that the twist is really bad.
- You're likely to be copying a lot of clipped signals.

Putting all this together, it might make sense to feed un-deemphasized audio to your TNC. Properly adjusted signals will have too much high tone, but the TNC can deal with that. Clipped signals will actually be pretty well balanced, and will decode much more successfully than if they had been deemphasized.

I've been doing this for years -- transmitting preemphasized audio, but driving the TNC with audio directly from the discriminator, before the deemphasis circuits. It seems to work very well, and I recommend that you think about this, particularly for nodes sites, BBSs, or other stations that will be communicated with lots of different stations.

I'll be honest, though, and say that I haven't done enough bench testing to determine how well this works. These tests are on my list of things to do, and will be reported here when they're finished.

## Conclusion

I hope this page has provided you with an understanding of why proper packet station setup is important, and the ability to adjust your station for best performance. As simple as the "quick and dirty" method of setup is, it's very surprising how few people have used it. Now, you don't have any excuse...

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