Explanation of "Flat Audio"

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I'd like to start off with a quick "Fly-By" description of just how audio works in FM two-way radio. Here we will briefly examine pre-emphasis and de-emphasis, repeater audio, and then we will discuss Flat Audio, and how it differs from most repeater audio scenarios today. Later, we will delve into the details in more depth. So here we go with the "Fly-By."

Fly-By

When FM began to be used in the late 1930's and early 1940's it quickly became the preferred mode of radio communications, primarily because of its far superior signal-to-noise ratio, and its ability to provide noise-free communications. At that time it was a one-radio to one-radio service. The same as what we call 'Simplex' today. Repeaters were not even dreamed of yet.

All early FM transmitters used Phase Modulation, which means they automatically doubled the deviation of an FM signal with every doubling of the audio frequency. This effect is called Pre-Emphasis, and works as a 6 dB per octave increase in deviation, or a 'roll-up' in deviation. It provided for a much better signal-to-noise ratio, by making the higher frequencies have as much audio punch as the lower frequencies, as it evened out the audio. The receivers of the day all had a De-Emphasis, or a 'roll-down' circuit in them, to restore the audio back to normal.

This is the way we still do it today, even with FM transmitters. FM transmitters have a pre-emphasis circuit built in, to be compatible with the existing PM transmitters, and because all of the receivers out there are running de-emphasis circuits.

When repeaters came along, hooking up the audio between the repeater RX and TX became a hotly contested topic, with many variations on how to do it. Some repeater builders took the user's pre-emphasized audio and de-emphasized it in the repeater RX, then pre-emphasized it again in the repeater TX for delivery to the end user. Then the end user's receiver de-emphasized it again, thus returning the audio to normal. Whew! This could probably work out if all of the emphasis curves in the repeater are matched (a difficult task). Also, all the processing in the TX's speech amplifier affects the audio quality too.

Another scenario is when a mismatch occurs in a repeater because only one side of the repeater processes the audio. For instance, if there is an extra pre-emphasis stage in the repeater, the audio will sound very tinny, and high-pitched, all high's with the low's missing. If there is an extra de-emphasis stage in the repeater, the audio will sound very mushy and lifeless, all low's with the high's missing.

Many repeater builders have different approaches, for instance, taking speaker audio and cramming it in to microphone input. Ouch! Others try to run CTCSS on the repeater TX by just connecting the RX audio and the CTCSS encoder audio outputs together, thereby loading down both circuits, and making everything sound mushy. Then when we begin to link repeaters together, it gets even worse, with all those link and repeater receivers and transmitters. Double Ouch. There have been some really interesting lash-ups over the years, with varying degrees of success and failure.

Flat Audio

When most of us talk about 'flat audio' we're talking about passing audio through the repeater without any de-emphasis or pre-emphasis stages, or other audio processing in between.

Since the audio is already pre-emphasized by the user's transmitter, you're not dealing with 'flat audio' in the repeater anyway.

Taking the audio from the discriminator on the RX, where it is still pure and pre-emphasized by the user, before it is de-emphasized, we pass it through a controller, and then inject it in the transmitter's audio chain well past the microphone input, past the speech amplifier, past the pre-emphasis network, and directly into the modulator. This keeps the audio 'flat' through the repeater. Hence the term, 'flat audio.'

This way the audio is un-modified as it works its way through the repeater. The RX leaves it alone, the controller leaves it alone, and the TX leaves it alone. The only audio shaping, or processing, is done at the user's transmitter, and the end user's receiver, just like simplex.

There are two things that need to be addressed in this Flat Audio scenario however. That is:

1) a Clipper (or Limiter) in the TX audio chain, to limit the deviation of the transmitter; and

2) a Low Pass audio filter past the clipper in the audio chain, to eliminate the high-order harmonics produced in the clipping process (and to reduce the high-frequency noise, especially squelch bursts, generally up around 8000 Hz or so, that are very annoying to listen to in FM radio).

If these two items are not addressed, the repeater system, while sounding flat, will pass high frequency noise, and can produce very wide over-deviated signals. It is important to stay within the 5 KHz deviation bandwidth with our repeaters, and not over-deviate.

So, that was our quick "Fly-By" of Flat Audio. Now, we will delve into the subject in more detail, with a thorough description of how it really works.

What is Modulation?

Just over 100 years ago, back in 1902, Fessenden developed a system to modulate a continuous wave with the human voice. Prior to that most voice transmissions were attempts at modulating spark transmitters, with generally poor results.

Modulation is a mixing process. When RF and Audio frequencies are combined in a standard AM transmitter (such as one used for commercial broadcasting) four output signals are generated: the original carrier or RF signal; the original audio signal; and two sidebands, whose frequencies are the sum and difference of the original RF and audio signals, and whose amplitudes are proportional to that of the original audio signal. The RF envelope (sum of the sidebands and carrier), as viewed on an oscilloscope, has the shape of the modulating waveform.

The bandwidth of an AM signal is twice the highest audio frequency component of the modulating wave. So, if the highest audio frequency is 3000 Hz, the occupied bandwidth of an AM signal will be double that, or 6 KHz wide.

Frequency Modulation (FM) was first technically addressed by John R. Carson in the February, 1922 issue of the *Proceedings of the IRE* Journal. By mathematical analysis, he “proved” FM inferior to AM on two counts: bandwidth requirements and distortion.

The Carson analysis held until May 1936, when another paper on FM appeared in the same Journal. In this work Major Edwin H. Armstrong set the stage for a new viable FM mode of communications. The basic theory behind his ideas is still in use today!

A Little History

I will start off with a little history of NBFM, or Narrow Band Frequency Modulation.

Methods of radiotelephone communication by Frequency Modulation were developed in the late 1930’s by Major Edwin Armstrong in an attempt to reduce the problems of static and noise associated with receiving AM broadcast transmissions of the day. The real advantage of FM, its ability to produce high quality signal-to-noise ratio audio when receiving a signal of only moderate strength, has made FM the preferred mode chosen for mobile communication services and quality broadcasting.

With AM, and SSB, the very process of demodulating audio causes the receiver to be looking for changes in amplitude, therefore any static or noise is recovered in the receiver along with the audio. When FM was first introduced, the main selling point for the new mode was that noise-free voice reception was finally possible. This is still very true today. FM inherently has a much better signal-to-noise ratio than AM. That is one of the reasons why FM sounds so good, compared to an equivalent AM or SSB signal.

The disadvantages of FM are few, most notably its wider bandwidth requirement. By way of example, a 5 KHz deviated FM signal with an audio (voice) frequency of 3000 Hz occupies about 16 KHz of radio spectrum. Compare this to AM, which occupies about 6 KHz of radio spectrum for the same 3000 Hz of audio, and less spectrum (about half) for an equivalent SSB signal.

This is one reason why FM is most popular in the VHF and UHF regions, where spectrum is more available. For the amateur radio service the FCC limits the low end for FM to 29.500 MHz. On the high frequency amateur bands, 80 through 10 meters, single sideband is the most widely used radiotelephony mode, partly because it occupies comparatively less spectrum which is important for frequencies that travel by skywave.

Audio Frequencies

We’ll leave the radio world for now, and enter the audio world. In today’s FM two-way radio communications, the audio frequencies we utilize are between 300 Hz and 3000 Hz for voice.

Another way of stating it is that frequencies between 300 Hz and 3000 Hz are the "audio range" or "voice band" of frequencies. The sub-audible, or CTCSS (PL) frequencies in FM are well below 300 Hz, and will be discussed in another section.

To understand just what the "audio band" of frequencies is, let's think about the piano for a moment. Most of us know what "Middle C" sounds like, that is the white key in the middle of the keyboard, just to the left of the two black keys. If you have a piano nearby, it may be helpful to sit at it, and try this little exercise. If not, try to imagine how the notes sound from memory.

Play the "Middle C" key. Middle C is technically called C4, as it is the fourth "C" from the bottom (left-hand end) of the keyboard. Middle C, or C4, has a frequency of 261.63 Hz, if the piano is tuned properly. That's right, 261 Hz. It is even a little lower than our 300 Hz low-end cut-off limit of the FM audio spectrum described above. It is technically in the Sub-Audible frequency range because it is below the voice band. Although most of us can still hear it, can't we. That is why you don't see many PL's above 200 Hz… and that is because you can actually hear them. Most PL's are usually around 100 Hz or so. More on PL's later.

Now, let's get back to our piano exercise. If you go up one octave on the piano, to the C that is twelve keys higher (both black and white keys) and play that key, the frequency will be 523 Hz. This is called C5. Notice that it is twice the frequency of C4 (261 Hz x 2 = 523 Hz). This difference is called one "Octave."

An "Octave" is defined as follows: ***"a one-octave separation occurs when the higher frequency is twice the lower frequency."*** Thus, the octave ratio is 2:1. Every time you double the frequency, you go up one octave in frequency.

Moving to the next higher octave, or to the next 'C' which is called C6, play it, and the frequency will be 1046 Hz. This is twice the frequency of C5. Play the next higher octave, which is C7, and the frequency will be 2093 Hz. This is again twice the frequency of C6.

The upper limit of our "Audio Range" in FM is 3000 Hz, and the closest piano key for that frequency is the highest F# (F sharp) on the piano, which is the very left-hand black key of the last set of three black keys on the keyboard, or the seventh key from the top (right-hand end) of the keyboard. The frequency of F# is 2960 Hz.

So, when I say that the audio frequencies used in our modern-day FM radios are from 300 to 3000 Hz, I am saying that the frequency range is the same as playing the piano keys from Middle D, or D4 (296.66 Hz) to the highest F#, or F#7 (2960 Hz).

Speech

Let's talk a little about how this audio frequency business translates into communication. Human speech is an interesting subject. Speech is comprised of multiple audio frequencies, mostly in the 125 to 4000 Hz range. But, all frequencies are not created equal in human speech. Some frequencies are louder than others, and some carry more intelligence than others. Here are a couple of facts concerning audio frequencies and human speech:

Fact Number One: The lower frequencies, those between 125 and 500 Hz, contain about 55% of the speech energy. But, only contribute about 4% to speech intelligibility.

Fact Number Two: The higher frequencies, those between 1000 and 4000 Hz, contain only about 4% of the total speech energy, but contribute an amazing 50% to speech intelligibility.

However, to understand human speech, we actually need to hear the higher frequencies (above 1000 Hz) more than the lower frequencies, as that is where about half the speech intelligibility is contained. Interesting, isn't it?

So, when a radio manufacturer says he is producing a "communications quality" radio, what frequencies do you think he is enhancing? Why the high frequencies, of course. As that is where over 50% of the speech intelligibility is.

Frequency Modulation Defined

Now, let's get back to radio. We now understand that over half of the speech energy is contained below 500 Hz when we are transmitting FM, so it normally follows that the amplitude of the lower frequencies is going to be much greater than the amplitude of the higher frequencies.

Let's step back for a moment, and define what FM is. When a modulating voice signal is applied to an FM modulator, the carrier frequency is increased during one half-cycle of the modulating signal and decreased during the half-cycle of the opposing polarity. In other words, the carrier frequency of our transmitter varies at an audio rate above and below our carrier frequency according to what the amplitude of the audio voice signal is.

So, to define Frequency Modulation, or FM, we would say that:

> **The change in the carrier frequency is proportional to the instantaneous amplitude of the modulating signal.**

Amplitude, then, is what drives FM. So, it follows that if 55% of the amplitude energy is contained in the voice frequencies below 500 Hz, then a true FM transmitter is putting 55% its energy, or its power, into the lower audio frequencies. That is where only 4% of speech Intelligibility is.

Also, by the same logic, the same true FM transmitter is only putting 4% of its energy, or power, into the voice frequencies above 1000 Hz. And that is where over 50% of the speech intelligibility is, remember?

That means that if you listen to a true FM transmitter, with 55% of its power in the voice band below 500 Hz, it will sound very mushy, almost all bass or low frequencies with hardly any high frequency component at all.

Phase Modulation Defined

Next, let's define Phase Modulation, or PM. It is possible to convey intelligence by modulating any property of a carrier, including its frequency and phase. When the frequency of the carrier is varied in accordance with the amplitude variations in a modulating signal, the result is frequency modulation (FM).

Similarly, varying the phase of the carrier current is called phase modulation (PM). Frequency and phase modulation are not independent, since the frequency cannot be varied without also varying the phase, and vice versa. In other words, phase is the mathematical derivative of frequency.

If the phase of the current in a circuit shifts, there is an instantaneous frequency change during the time that the phase is shifting. The amount of frequency change, or deviation, is directly proportional to how rapidly the phase is shifting and the total amount of the phase shift. The rapidity of the phase shift is directly proportional to the frequency of the modulating signal. Further, in a properly operating PM system the amount of phase shift is proportional to the instantaneous amplitude of the modulating signal.

So, to define Phase Modulation, or PM, we would say that:

> **The change in the carrier frequency is proportional to both the instantaneous voltage and the frequency of the modulating signal.**

This is the outstanding difference between FM and PM, since in FM the frequency deviation is proportional only to the amplitude of the modulating signal.

By contrast, in Phase Modulation the deviation increases with both the instantaneous amplitude and the frequency of the modulating signal. That means that PM has a built-in Pre-Emphasis, where the deviation increases with modulation frequency. Apart from this difference, when receiving FM or PM, it is difficult to distinguish between the two.

Notice I used the word Pre-Emphasis above. This is probably a good place to explain how Pre-Emphasis and De-Emphasis works.

Pre-Emphasis and De-Emphasis

What is pre-emphasis? Pre-emphasis follows a 6 dB per octave boost rate. This means that as the audio frequency doubles, the amplitude (and deviation) doubles (by 6 dB). So, with pre-emphasis, the following examples are typical of pre-emphasis on an FM transmitter's deviation:

- a 500 Hz audio tone will make 1 KHz of deviation
- a 1000 Hz audio tone will make 2 KHz of deviation
- a 2000 Hz audio tone will make 4 KHz of deviation

Why do we even have pre-emphasis in NBFM communications? There are actually two reasons:

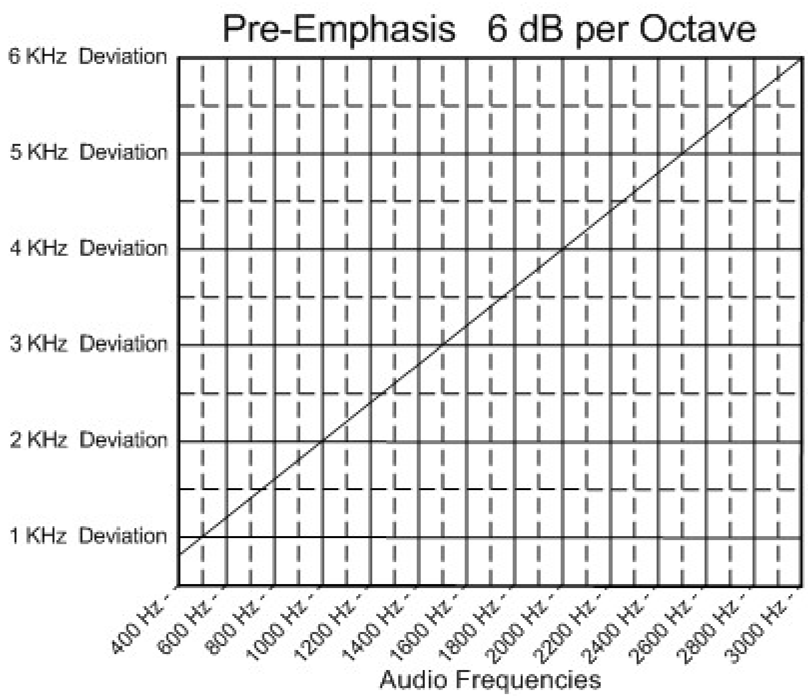
1) The early transmitters were really PM (Phase Modulated), not FM, so they naturally had a 6 dB/octave 'roll-up' or pre-emphasis. PM was the standard modulation method. When FM transmitters came along, their audio had to be intentionally pre-emphasized

to maintain compatibility with the PM transmitters already in service. In very early narrowband literature, you won't even find the terms "pre-emphasis" and "de-emphasis." Engineers simply 'rolled-off' the audio in the receiver with a single-pole filter to reverse the PM transmitter's 'roll-up' characteristic and restore the transmitted audio back to normal;

2) Pre-emphasis is needed in FM to maintain a good signal-to-noise ratio across the entire voice band. Theory tells us that white noise increases with frequency at a receiver discriminator. When de-emphasis is added to a receiver this noise is attenuated, thus improving the signal-to-noise ratio.

Pre-emphasis is used to shape the voice signals with the increased level of the higher frequencies being applied to the modulator, which results in a better transmitted audio signal-to-noise ratio due to the highs being above the noise as much or more than the lows.

The following graph illustrates the Pre-Emphasis Curve, audio frequencies along the bottom.



We must recognize that early narrowband FM radio was intended for one transmitter – one receiver applications. This business of repeaters, and linking repeaters came much later. Virtually all FM radios today, including commercial broadcast, use pre-emphasized and de-emphasized audio. When you talk with someone on simplex, their TX pre-emphasizes their audio. If you could listen to 'raw' pre-emphasized audio it would sound very tinny, mostly highs. However, your receiver de-emphasizes the audio, returning it back to normal.

By contrast, if you could listen to 'raw' FM audio it would sound almost all bass, or low frequencies, as most of the amplitude, or energy, is in the low frequency range, and hardly any highs, or treble at all, as the higher frequencies would be more in the noise, as their amplitude would be much less, and would virtually not be readable.

Signal-to-Noise Ratio

This is a ratio which defines the ability to demodulate, or recover the audio, of a radio signal. This is one of the best advantages of FM over any other form of modulation.

Noise is an ever-present part of radio communications. If you aren't aware of this phenomenon, tune to the Low Bands with a communications receiver, from say 160 to 10 meters, and simply tune across the band. What you hear is noise. Lots of noise. Hopefully, if you're lucky (if the band is "in"), as you tune across say 20 meters, 14.000 to 14.300 MHz, the noise will be interrupted occasionally by signals. If you're really lucky, some of the signals will be loud enough to overcome most of the noise. You are experiencing Signal-to-Noise at its finest. This is a low Signal-to-Noise ratio situation.

In FM, when a sufficient signal arrives at the receiver, the signal quiets – that is, because of the high RF limiter, the background noise disappears. Completely. This differs greatly from AM on the Low Bands, because most natural occurring noise is amplitude modulated, noise is ever-present. In FM, unless the station you're talking to is very weak and barely making it, there will be no noise at all on his signal. This is a high Signal-to-Noise ratio situation.

In fact, the sensitivity of an FM receiver is rated in terms of the amount of input signal required to produce a given amount of quieting, usually 20 dB. The use of solid-state devices allows modern FM receivers to achieve 20 dB quieting with only 0.15 to 0.10 *u*v of input signal.

So, the big difference between the AM or SSB on the Low Bands and FM communications is the amount of noise you must listen to while communicating. The lower the amount of noise on a signal, the better the Signal-to-Noise ratio is.

Speech Clipper

A Clipper, or Limiter, is a safety valve that limits the peak deviation of an FM or PM transmitter. In broadcast, it is sometimes referred to as a Safety Clipper. It keeps the deviation down to a pre-set level, usually 5 KHz for land mobile communications.

Because the modulation index, or bandwidth, of the transmitted pre-emphasized FM signal increases with the modulation audio frequency, it can easily exceed 3000 Hz. For example, if a long squelch tail is re-transmitted through a repeater, that audio frequency is usually somewhere up around 8000 Hz. That means that the maximum deviation of 5 KHz can easily be exceeded. In the case of the 8000 Hz squelch noise, a deviation of 8 KHz is possible. Deviation that wide would surely ingress into adjacent channel repeaters.

Not only do these wider deviated signals go outside the normal bandwidth of most receivers and cut off the audio (as they over-deviate past the capability of the receiver to demodulate the audio) but more importantly, these wider sidebands can interfere with other adjacent channel repeaters and radios.

Therefore it is necessary that some sort of frequency clipping or limiting be placed between the audio source and the modulator. This clipper or limiter should provide clipping at about 5 KHz of transmitter deviation.

Low-Pass Audio Filter

The clipping process produces high-order harmonics which, if allowed to pass through to the modulator stage, would create unwanted sidebands. Therefore, an audio low-pass filter with a cut-off frequency between 2600 Hz and 3000 Hz is needed at the output of the clipper. This keeps the harmonics, and other high-frequency noise (such as squelch noise) out of the transmitter. This also keeps the occasional long squelch tail, or blast of white noise, from being heard so loudly. Oh you can still hear these noises, but their amplitude is far lower than without a low-pass filter.

Sub-Audible Frequencies

Sub-audible tones are frequently used to limit access, and are commonly called PL, a Motorola term standing for Private Line, or CG, a General Electric term for Channel Guard, but whatever name you apply to the sub-audible tone, it is simply describing CTCSS, or Continuous Tone Coded Squelch Systems.

This is a continuous tone that is sent along with the transmitted signal, at a much lower volume (I usually have it at .4 KHz deviation) and at a lower (sub-audible) frequency that you hopefully will not be able to hear really well. This is called ‘encoding.’ It simply means that you are transmitting this continuous tone along with your normal speech in your transmitter.

Decoding is where your receiver hears the continuous CTCSS tone, and ‘decodes’ the tone, thereby opening your receiver’s squelch so it can hear your transmitted audio.

These frequencies are much lower in frequency than normal, usually around 100.0 Hz. The lowest CTCSS frequency on most radios is about 67.0 Hz and the highest is about 254.1 Hz.

Flat Audio

Now that you are becoming experts in the audio and modulation arenas, I will now explain “Flat Audio.”

As we have discussed previously, early FM radios were designed to be one-radio to one-radio devices. And, to improve the signal-to-noise ratio, they were all PM transmitters, with the built-in audio ‘roll-up’ or pre-emphasis of 6 dB per octave. This made the signal-to-noise ratio better for reception, making the high frequencies usually as strong as the lows. The receivers all had the audio ‘roll-down’ or de-emphasis network, to restore the audio back to its original state. Everything worked great.

Enter repeaters. Repeaters extended the range of one-radio to one-radio communications. If they were high enough above average terrain, they greatly extended the range. However, because repeaters are duplex, and the RX repeats the audio it hears to the TX, there are issues with audio quality, and the processing of the audio.

There are many different ways to hook up the audio in repeaters. Some repeater builders take speaker audio out of the receiver, and connect it to the microphone input on the transmitter. This has several ramifications:

1. The RX speaker audio will be de-emphasized (again) by the repeater RX.
2. The RX speaker audio is almost always shaped or processed somewhat to match the speaker that the manufacturer specified to be used with the RX.
3. The speaker audio is almost always amplified well beyond what the microphone requires, usually somewhere between 3 to 5 watts of audio power. The microphone needs milli-watts of audio power. Some audio distortion will most likely be present in the amplification, typically 10%, and noise is amplified as well.
4. What is the impedance of the average speaker? 4 ohms? 8 ohms?
5. The squelch crash after a user stops transmitting will always be passed along through the audio chain from the repeater RX speaker. You are at the mercy of whatever squelch circuit the repeater RX is using.
6. The TX microphone input audio is always shaped by the manufacturer to match whatever kind of microphone element they are using, be it crystal, or dynamic, or cardioid, or ceramic. This is usually called the speech amplifier. The bottom line is that the audio is shaped, or processed, to match the microphone element. If the mic input is used, this shaping has an effect on the audio coming out of the repeater TX.
7. The audio is pre-emphasized (again) in the repeater TX.
8. The audio input impedance is usually several thousand ohms, not the 4 ohms, or 8 ohms of a speaker. If a controller is used, or an emitter follower, or a cathode follower, the impedance mismatch can probably be overcome however.
9. Once you set the volume, squelch, and modulation level knobs on the repeater, you'd better not touch them again, or your repeat audio levels will go crazy. If someone accidentally bumps one while near the repeater, you'll know it right away.

As you can see, this method leaves a lot to be desired. Why do folks do it? Because it's easy! There is not much work involved to strap a speaker to a microphone. However, if you're willing to invest the time, there are better ways to get the audio from the receiver to the transmitter. Some folks have become very clever at this.

One more point. As we've discussed before, the audio from the users transmitter is already pre-emphasized, isn't it. It enters the repeater RX pre-emphasized. If the repeater RX de-emphasizes it, and then the repeater TX pre-emphasizes it again, in theory it should sound about the same coming out of the repeater TX. But it usually doesn't. One reason is because the de-emphasis and pre-emphasis curves in the repeater RX and TX are usually not exactly the same. They don't track one another perfectly. There is usually a difference.

It is much better to leave the audio alone when going through a repeater, and not process it, keeping the repeater audio path linear. The originating audio is pre-emphasized by the user's transmitter, and we let it be de-emphasized in the end user's receiver. This is the way it is done on simplex, without a repeater in-between. This is what Flat Audio is.

Another way of saying it is: When the repeater RX hears a signal, it leaves the audio alone, or keeps it flat, without any changes. The controller should likewise leave it alone, with no changes. And finally the repeater TX should leave it alone, without adding anything to it. That way, as far as the audio is concerned, the repeater was never there. Because the audio path is flat through the repeater. It has not been de-emphasized, shaped, pre-emphasized, and shaped again. It is flat. No processing. No changes.

How one accomplishes this Flat Audio is pretty simple, really. Here is a quick run-down:

1. Pick off the audio at the discriminator of the receiver. Usually the top (high side) of the squelch pot is a good starting point. That is where you can find discriminator noise with an oscilloscope.
2. Use a fast-acting, noise-free squelch circuit, like the Motorola MICOR squelch chip or a microprocessor controlled digital squelch Board. More on this digital squelch in a subsequent article.
3. If your controller gives you a choice, set the controller's input for Flat Audio.
4. Inject the audio well past the microphone input, past the speech amplifier circuitry, and past the pre-emphasis network in the repeater TX. This point is usually at the PL, or CTCSS injection point. This is usually past the TX's built-in Clipper and Low-Pass filter too, as they are typically in the low-level audio stages of the exciter.

If proper impedance techniques are used, then this will result in Flat Audio. It will sound great. However, you really need to install your own Clipper, to keep your transmitter from over-deviating. And, you need to install your own Low-Pass filter to keep the harmonics and high-frequency noise out of your transmitted audio. Finally, and very important, all stages in the repeat audio path must be designed with adequate headroom, or dynamic range.

Also, if you plan on running CTCSS on the transmitter output, you need an isolated input for the CTCSS encoder, so it won't load down your audio circuitry making it sound mushy.

And, if you are using a Phase Modulated transmitter, say a GE or most Japanese radios, you need to be able to compensate for its natural pre-emphasis, or 'roll-up' characteristics. That means you need to shape the audio going to the TX to reverse the pre-emphasis roll-up.

Finally, when adjusting audio levels in a repeater, quite often the RX has too little audio level available at the discriminator, or too much level. Similarly, some transmitters need more audio drive than others at the CTCSS input location. Many controllers cannot handle big variations in audio input and output levels, so you must build op-amps to balance the levels.

Thanks for taking the time to read my article on quality audio. It is my hope that more repeater builders will take the time to make their audio sound fantastic. It can easily be done. Please direct any question or comments to the author.

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