DEALING WITH NOISE

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The object of radio is to convey information from one place to another without wires. Some listeners may consider anything that interferes with this traffic to be 'noise', even though it may pass vital information to others. This article deals with the kinds of noise that nobody wants to hear.

Noise takes several forms, most of them dependent on the radio frequency. These include: atmospheric (mostly lightning and rain static); man-made (conducted/radiated hash and impulses like motors, dimmers, power lines); cosmic (hiss and thermal noise from distant space); and noise generated within the receiver (white noise, hum and distortion).

There are six primary ways to tackle noise: the filters in and between our ears; filters between the antenna and the detector; at the power line; audio filtering; grounding techniques, and optimum adjustment of the receiver.

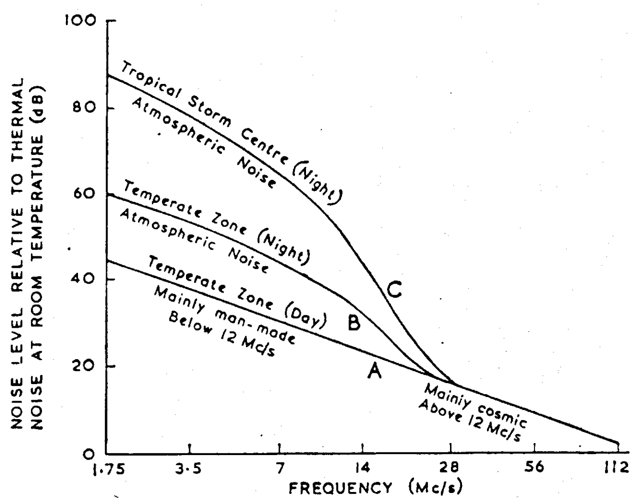


Fig. 15.6. Variation of external noise level with frequency. Curve A shows that during the day in temperate zones the noise is mainly man-made at frequencies above about 12 Mc/s. In these zones, atmospheric noise adds considerably to the total noise level at night (curve B). In tropical zones the atmospheric noise is relatively severe: curve C represents the worst conditions in these zones. The vertical scale indicates the number of decibels by which the noise level in a perfect receiver would increase if it were disconnected from a dummy aerial and fed from an efficient aerial of similar impedance.

Fig. 1. External Noise

THE FILTERS IN AND BETWEEN OUR EARS

The SWL listening post is a human-machine system. The human brain is the final receiving device, and everything else from the antenna through the receiver and peripheral devices to the ears is there to process the incoming information and make it more intelligible.

For the broadcast listener, pleasure and comfort are the primary aims, but for most SWLs the essential thing is INTELLIGIBILITY. Most of the time we are looking for the identity of a station, and the material of the program is of secondary interest. But the brain can be fatigued by noise and distortion, and the ears can be damaged by sudden loud crashes or clicks, so we have to provide for increased comfort as well. And of course, we do sometimes want to enjoy the program!

The ear has between 30 Hz and 20 kHz twentyfour audio sub-bands, within each of which a loud signal will 'mask' a weaker one[1]. The good news is that this enables a strong signal to mask weaker noise. The bad news is that noise can mask a weaker signal. Hence the importance of signal-to-noise ratio (more correctly expressed as 'signal to noise-plus-distortion ratio'). If the stronger signal is strong enough, the brain does not hear the weaker one within the same sub-band. At the lower end of the audio spectrum the sub-bands are of the order of 100 Hz wide, while at the upper end they are of the order of 2 kHz wide. This is one of the reasons why an operator can mentally separate two tones a given number of Hz apart when they are heterodyned to a lower part of the audio spectrum (listening to 300 Hz and 500 Hz rather than 1200 and 1400 for example).

Note that this 'masking' occurs only when one signal is stronger than the other. If the two signals are of equal strength the ear can separate two signals that are more than about 50 Hz apart. This explains why some morse code operators prefer to use a wide bandwidth filter and rely on the brain. Non-masking noise does not destroy the intelligibility of a signal, although it does introduce some stress.

The ear has a dynamic range of 100 dB or more (it can hear very quiet sounds and deal with very loud ones). It can hear the DIFFERENCE in the level of two sounds better if they are both quiet than if they are both loud. The sounds can be separated most easily if one is on the fringe of audibility and the other within the audible range. The telephone companies know this very well, so audio levels on the system are kept low enough to make hum and noise as unnoticeable as possible.

So what? Well, armed with this knowledge we know how to deal with each of two conditions:

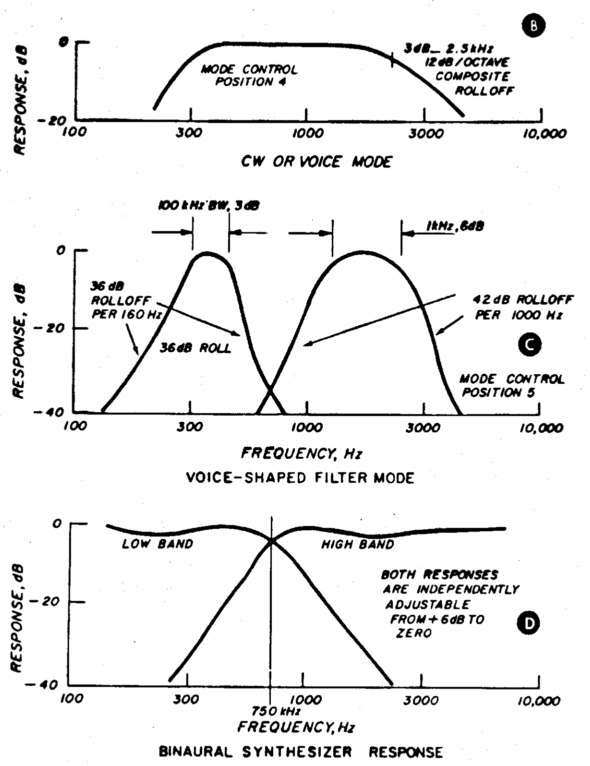


Fig. 2. Hildreth double peak filter

1. If the signal is stronger than the noise, turn down the volume until the noise is not troublesome.

2. If the noise is stronger than the signal, use a clipper to reduce the difference in strength, giving your brain a better chance to sort them out.

AUDIO FILTERING: SELECTIVE ATTENUATION

Best signal-to-noise ratio is obtained when the occupied bandwidth equals the necessary bandwidth. If the bandwidth of your filters is too wide, the amount of noise will be increased but not the signal. If the bandwidth is too narrow the signal will be reduced more rapidly than the noise. The necessary bandwidth does not have to accommodate all the frequencies produced by the human voice. You are seeking intelligibility not high fidelity.

Don Hildreth W6NRW produced a very effective 'Comm Audio Processor'[2] (CAP), based on the principle that almost all the necessary information in the voice is between 300 and 400 Hz (vowel sounds) and 1.5 and 2.5 kHz (intelligibility sounds). Anything below, above and between those bands does more harm than good.

The CAP takes the audio from the receiver, passes it through parallel 300-400 and 1500-2500 kHz filters (Figure 2), then through lowpass and highpass filters that split the audio at 750 Hz. The two signals are routed through separate adjustable-gain amplifiers to output terminals designed to feed loudspeakers or binaural headphones. I was fortunate to buy one before Don ceased production several years ago and it has been in daily use ever since. If you can find one, buy it.

One remarkable feature of the filter is that it makes grossly over-compressed speech sound quite reasonable. This is because much of the distortion caused by over-compression is in-band intermodulation appearing between the two humps of the filter combination. Because of the variable gain on each amplifier excessive bass or treble components of the voice can be compensated for (and it is surprising how often the bass end can be removed almost entirely without harming intelligibility). Because of the essential difference between stereo and binaural, it is sometimes possible to understand a voice better on speakers than on headphones. True stereo, and the "cocktail party effect" (the human ear/brain function that enables you to pick out the interesting conversation from the babble all around) depends on the ability of the ear and the brain to respond to differences in loudness, phase, and time of arrival of sounds. Sensing the apparent direction of arrival helps the brain separate the respective sounds. It seems complicated, but it is sure fun to experiment with it.

The CAP also incorporates a white noise generator (white noise is energy distributed evenly over the audio-frequency band, and is the hiss you should hear if you disconnect the antenna and turn up the gain with the receiver on wide bandwidth). White noise is used in offices to mask the tapping of typewriters and other distractions. It can be introduced into the receiver audio to replace a low level unpleasant noise with a designed level of white noise, or remove ringing from cascaded narrow filters, or mask the weak but audible and intelligible speech you don't want to get interested in. One has to be careful though, because the masking effect is greater when the masking frequency is the higher, and white noise is also used as an anesthetic and a sleep-inducer!

The principle of the Hildreth filter can be demonstrated quite well if you have a notch filter on your system. Just put a notch at about 1000 Hz in the middle of a noise-affected voice. If you have also a bandpass filter that attenuates below about 300 and above about 2000 Hz so much the better. An Autek QF1-A will do it quite well, except for the limitation to single channel output. Operation of the QF1-A can be improved by the addition of input and output level controls[4]. An excellent review of the QF1-A by Jerry Strawman appeared in *Proceedings 1988*[8].

An audio equalizer with sliders at 100, 300, 1000 and 3000 Hz will do a fair job, and several filters have been reviewed in past *Proceedings,* as listed in the bibliography.

If you are interested in listening primarily to voice, find a headset designed for voice frequencies. Guy Atkins bought a Roanwell aviation headset from Fair Radio Sales for $20 that he says is really outstanding for voice. I checked with Fair Radio in May 94 and they had no aviation headsets of any type left, but with defense cuts the way they are there will be more coming available. They are 600 ohms impedance, but Guy says they sound fine with his Drake R7 and R8. If you use phones of radically different impedance make sure the mismatch does not harm the intelligibility. Good reproduction is essential for this last link in the chain. If one of your ears is better than the other there is great merit in having a volume control on each earphone. The brain likes to get equal response from each ear.

AUDIO FILTERING: CLIPPING

An electric fence, a flashing electric sign, a succession of lightning discharges cause sharp spikes of noise to be superimposed on the audio. A clipper like that in Figure 3 may not work if it is connected across the low impedance of headphones or speaker. It may be necessary to step up the impedance to several hundred ohms as in Figure 4. The impedance ratio of the transformers is not important, but they should be designed to carry audio. You need something of the order of 600 ohms impedance across the clipper circuitry, taking care not to produce more audio voltage than the diodes or transistors can handle.

Germanium diodes are more useful than silicon, as they will pass current at a lower forward voltage. Figure 4 performs the same function as Figure 3 using a PNP and an NPN transistor self-powered from the audio. The spikes will still be present within the receiver, so use fast or no AGC to prevent the receiver from being paralyzed before the audio gets to the clipper.

The simple clippers shown here have to be used with caution, as hard clipping without compensating lowpass filters can produce distortion that is as damaging to intelligibility as the noise itself. There are many interesting series and/or parallel clipper circuits in the older receivers and this is a subject all to itself.

Digital signal processors are "in" as I write. The JPS NIR-10[6] was the first on the market. It is excellent for removing heterodynes and white noise, and has very sharp bandpass filters, but like all digital signal processors it has to have an intelligible signal to work on, so cannot dig an unreadable one out of noise. Revision 3.0 PROMS provide a

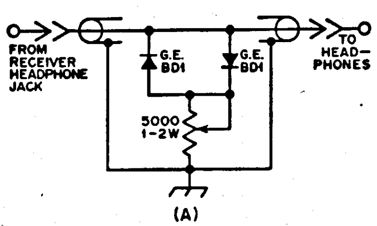


Fig. 3. audio clipper for high impedance input

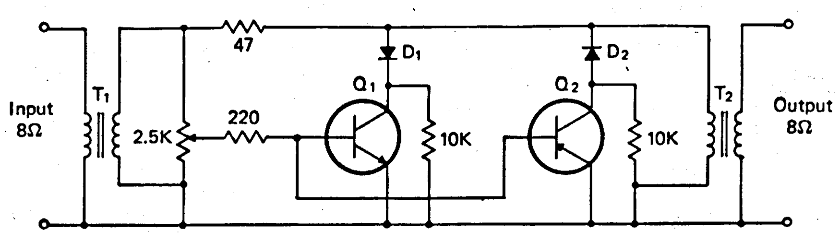


Fig. 4. audio clipper for low impedance input

useful automatically adjustable bandwidth feature, whereby the processor dynamically varies the bandwidth to fit the width of the desired signal. JPS provides new PROMS for $25, easily fitted into sockets. There are PROMS specifically for SWL use.

Several other DSP units have come on the market, including a very versatile one from MFJ and a barebones one from Radio Shack. A potential user would be wise to look carefully at the features of all of them before deciding which to buy.

FILTERING BEFORE THE DETECTOR

A built-in noise blanker senses noise pulses at a wideband point in the receiver, turns them upside down, and reintroduces them to the signal to cancel out the pulse or even punch a hole in the audio. A hole is less disturbing than a spike both to the later stages in the receiver and to the ear.

If you have a variable control for the blanker, don't advance it more than you have to or you may hear some noises-off from strong signals right outside the normal selectivity passband.

Blanking somewhere before the detector is much better than clipping at the audio level, because it removes the pulses before they have a chance to upset the AGC action or drive the detector into distortion. R-f and i-f bandpass filters have the same advantage. If the noise can be prevented from reaching the detector, it cannot affect the purity of the demodulation or the action of the AGC. The problem for the designer is that it is more difficult to achieve a precise shape of the bandpass curve in the higher frequency stages.

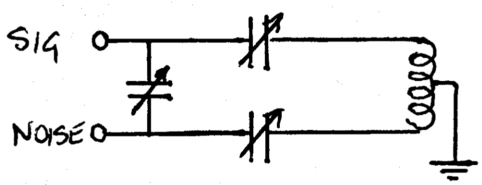


Fig. 5: Jones Circuit

ANTIPHASING AT THE ANTENNA TERMINAL

The most troublesome noise is usually that from the power lines. If you can pick it up on a separate antenna and feed it to a device that inverts the phase, noise can sometimes be cancelled before it ever reaches the receiver. The 'Jones noise-balancing circuit' Figure 5 was popular many years ago, but was not easy to adjust.

A much better antiphaser is the 'S.E.M. QRM Eliminator' that I reviewed in the *1992/93 Proceedings*[7] and still have in use. It works fine for me, but a friend in Vancouver found it was of no use at all in his downtown location. He relies on a small rotatable directional loop, pointing the null at the main source of the noise.

GROUNDING

Nick Hall-Patch covered this well in the *1992/93 Proceedings*[9], so I will only add a note about the possibilities of Bentonite. This is a water-attracting clay material that swells dramatically when wet. You drill a hole of much greater diameter than your ground rod, suspend the rod in the middle of it, backfill with Bentonite and pour on lots of water. The Bentonite swells and swells, gripping the rod tightly, assuring continuous contact with the rod and with the surrounding earth. It continues to attract any moisture that is around, so stays tight. A Bentonite product especially for ground enhancement is GEM-25A, produced by Erico Inc.[5].

Bentonite is a natural product, used widely as a slurry for well drilling, and also for sealing basement walls against moisture and lining water reservoirs. A good supply source is a welldriller supplies store. I am told it is not itself very conductive, so has to be mixed with something that is, to reduce the resistance of the ground connection.

As Nick commented, sometimes connecting the receiver to ground actually introduces noise. If noise is being conducted to ground from the utility neutral to the utility ground stake, the ground around it will have gradients of noise voltage. If an r-f grounding system intercepts some of those gradients, the noise may be heard in the receiver. The r-f ground should be as far from the utility ground as practicable.

POWER LINE NOISE

Much of the noise conducted or radiated by the power line is generated on the premises of power utility customers, and usually the best you can do about it is be sure it is not coming from within your own house!

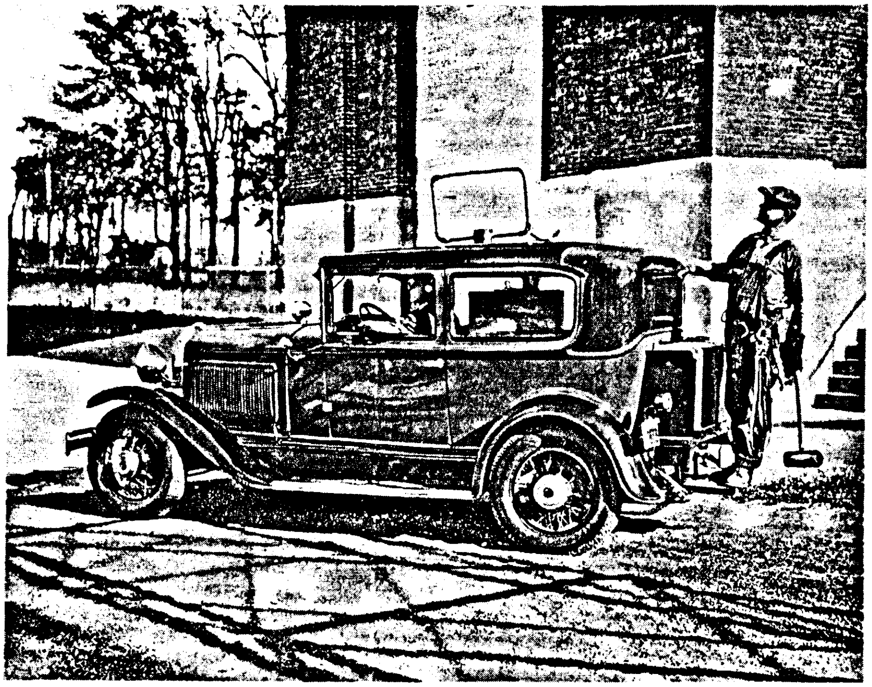
Sometimes the local distribution line (the single 7000, 14000 volts or more conductor carried at the top of the pole line) is the culprit. Frying noises from loose connections and tie points, irregular ticks and pops from loose pole hardware, raspy buzzing from faulty insulators, often quieter when the weather is wet. Expansion and contraction of wooden poles with climatic changes loosens things up, and power companies seldom go round tightening them unless someone complains about the noise.

If the noise is loud at the end of the line, it may not be originating there. A standing wave of noise may occur, and peaks and nulls will occur along the line. As you approach the source of the noise the peaks and nulls become less pronounced but the noise gets louder. You can often trace the noise to an offending pole by walking around with a portable radio, or even driving around in the car. Bonk the pole with a sledgehammer. If the noise changes immediately you probably have the pole. If there is a delayed reaction it is probably another pole. The experienced bonker gets to be able to estimate how far away the trouble is (and doesn't mention that he has been bonking).

About twenty years ago a ferric oxide semi-conducting glaze was developed for insulators, and found to produce no r-f interference on NEMA Class 56-3 pin-type insulators after a year of exposure on 44 kV line. The cost of these insulators is about the same as that of normal Q-glaze, but I have no idea how many have been put into use or where.

The grounding wire that comes down the pole and is wrapped around under the ground may not be in good contact with the ground. IT IS DANGEROUS, and is something the power company will fix in a hurry. While there the workers may be persuaded to check and tighten other things, especially if you can show them the interference affects broadcasting. Broadcasting is much more important than short wave or amateur radio.

A lot of noise is conducted into the receiver along the power cord, and ferrite beads or snap-on ferrite chokes are often useful. Or the power cord can be wound around a ferrite rod. I found there was some noise radiated from the keyboard of my computer, and to my surprise a ferrite rod slipped inside the coils of the coiled connector cord reduced it quite a lot. I don't know whether the noise is generated within the keyboard or conducted to it, but the rod fixed it and I settled for that.



Government Car equipped for the Investigation of Radio Inductive Interference.

OPTIMUM ADJUSTMENT OF THE RECEIVER

"What is the best setting for the r-f gain control?" Turn down the r-f gain until you can just hear 'band noise' in the absence of a signal. When you tune for a signal ANYTHING THAT IS STRONGER THAN THE NOISE WILL BE AUDIBLE.

If your automatic gain control (AGC) is audio-derived, and your receiver uses a product detector, this advice is 100% valid. If it is derived from a diode detector, as in most of the older receivers, some experimenting is necessary to find the best r-f gain setting for each type of signal. This is because when you are using a beat frequency oscillator (BFO) and a diode detector the ratio between the signal strength and BFO injection is important. With a product detector there is a lot more latitude.

An aside: about 30 years ago I bought a Drake 2B receiver. It has a slide switch to select either diode or product detector. After the first comparison between the two, the diode position was never used again. Listening to short wave AM broadcasting stations became so much more pleasant when we could select either the upper or lower sideband and reinsert a carrier. There are theoretical disadvantages to this - the audio is reduced by 6 dB and the noise by 3 dB when one sideband is removed. But the absence of selective fading distortion and sometimes removal of interference affecting only one sideband compensates. Whether your receiver is superb, average or poor, you can improve signal-to-noise ratio just by careful adjustment of the controls. Noise generated within the receiver is a fact of life, and for best reception the gain at various points within the set has to be 'balanced'. When you are struggling to hear a very weak signal every little step towards perfection helps.

The manufacturer of the receiver does not know how sensitive your headphones are, so there may be more gain than you need in the audio amplifier. Turn the audio volume control almost to zero (not quite against the counterclockwise stop). If you hear the gentlest of hiss, everything is fine. If you hear some hum or hash you should try to do something about it, knowing the trouble is in the audio stages or the power supply. If you hear more than a smidgeon of hiss, consider putting some attenuation between the output stage and the headphones - a series resistor may be all you need. If you have a volume control on each earphone, turn them down until you only just hear the hiss. If one of your ears is better than the other, adjust the two controls individually. This is quite important, as your brain always prefers two ears rather than one. The threshold of inaudibility is about 10 dB lower if you are listening with both ears.

If you are using an active filter unit between the set and the phones, make sure it is not contributing any hiss or hum in the absence of signal. One filter I have is specified as requiring 12-15V to operate. I noticed I could hear some background hiss when the receiver was switched off. I found that the filter unit was quiet when fed with only 9 volts and operation was as good as with 12 (in fact I believe it now has some accidental gain compression because of earlier saturation, which is welcome). Reducing the voltage was easier than introducing some attenuation.

In modern receivers the so-called r-f gain actually controls the gain of the intermediate-frequency stages. In older receivers it usually controls both i-f and r-f stages. By manipulation of this knob you have control over the amplification contributed by the i-f department. This is a very important part of the set as far as noise is concerned, because any noise generated there is passed on to the detector (where the demodulation process gives it the best chance of disrupting the desired signal) and is then further amplified by the later stages. Sometimes the effectiveness of i-f filters or the noise blanker is affected by the gain. Setting the r-f gain control so that 'band noise' is just audible is almost always the right place for it, and this also ensures that you will not hear surges of noise when the signal fades. When you find a signal adjust the audio control for comfortable level. Then if the signal is marginal you can jockey the two controls up and down a little for best signal to noise ratio.

I don't know why it is, but on one of my receivers there is usually one setting of the audio volume control that gives best intelligibility, often resulting in a lower volume than I would choose if the signal were 'armchair copy'. If you want an S-meter reading you probably have to advance the r-f gain control temporarily, and the signal may very well change from Readability 5 Strength 3 to Readability 3 Strength 9.

On a receiver that has r-f derived (as opposed to audio derived) AGC, a useful trick is to turn down the r-f gain control until the S-meter reading stays constant at the peak level of the signal. Turning down the gain introduces the same kind of negative bias as that produced by the AGC circuit, preventing noise from welling up when the signal fades down. I knew one SWL who did this all the time as a routine, and we seldom found a better setting on his receiver, which I believe was an HQ-129.

If noise coming in from the antenna is really bad, and you have an r-f attenuator switch, it is probably a good idea to use it, to prevent overloading of the first mixer. If more than one signal is getting into the front end, introducing 10 dB of attenuation will reduce any cross-modulation or intermodulation by more than 10 dB.

Noise power is related to bandwidth, so anything that provides selectivity in front of the receiver, like a sharply tuned antenna or an antenna coupler with a bandpass characteristic, will help. Some hash coming from the power line or fluorescent lamps centers on one part of the r-f spectrum, so a selective filter in front of the receiver will be especially effective for rejecting it.

For good reception your receiver sensitivity must be 'noise-limited', which means simply that noise generated within the receiver must be low compared with that picked up by the antenna. A simple test will show whether this condition is met. Set up your receiver for normal operation with medium or wide bandwidth and tune to a spot where there is no signal. Measure the 'band noise' at the audio output with a VOM using an a-c voltage range that gives a reading well up on the scale. Remove the antenna and replace it with a resistor (50/75 ohms or so if your receiver is designed for coaxial cable input, 400 ohms or so if designed for single wire antenna). The reading should drop to a lower level. A 6 dB drop on a quiet band (half the former voltage) is excellent.

Of course the difference will depend on how much noise is being picked up by the antenna, and one thing the test makes clear is that if you have lots of local noise you don't need a sensitive receiver. Sensitivity probably varies from one band to another, and this makes sense, because so does the average noise level. See Figure 1.

CONCLUSION

Every location is different, receivers vary widely, and your particular needs may be unique. If you find something new to think about here, that's good. If you find it so-so, congratulations! You have already solved the problem.

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