Selectivity: how a radio tells one signal from another

Selectivity is one measure of a good receiver, whether for CB, shortwave listening, ham, FM music or whatever. Here's an easy-to-understand explanation of what selectivity is all about and how to tell if you bought a good radio:

by George McCarthy

Have you ever wondered how your radio is able to select just one signal from among the millions that are being broadcast all over the world at any one moment? Or perhaps you have thought about what causes signals from adjacent CB channels to slop over onto the channel you are trying to use. What's inside your radio that determines how well it can tune in a particular signal, while rejecting others that are nearby?

That ability is generally referred to as a receiver's *selectivity*. We define selectivity as the ability to discriminate among many signals of varying strengths that are relatively close to the frequency of the station we want to hear. For ham and CB receivers it is one of the most important characteristics to look for. Of course, a good receiver also needs *sensitivity* to pick up weak signals, *frequency stability* so that it stays at the spot where it has been tuned, and resistance to *front end overload* where extremely strong signals cause the receiver's amplifying circuits to malfunction.

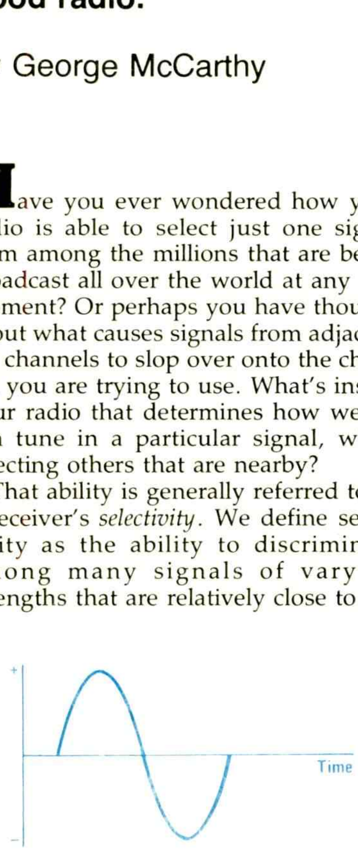

Figure 1 **One complete cycle of alternating current. The number of times this occurs per second determines the frequency.**

For the first two dozen years of the developing radio industry, receiver circuitry was fairly simple. Circuits responsible for selecting the received frequency were made up only of coils and capacitors. Tuning was accomplished either by varying the coil with taps or sliders, or by varying the capacitor mechanically by rotating it. The capacitor was *variable* in that a series of rotating plates could be moved to intersect a series of stationary plates, much like an egg slicer. You've seen such a variable capacitor if you've ever peeked inside a radio.

After the simple crystal sets using only a coil with sliders on it, came vacuum tube sets which had more tuned circuits. Each set of coil and capacitor allowed the receiver to tune sharper and pick out a particular signal from the group of signals in some portion of the frequency spectrum.

A problem developed as more and more stations came on the air. The first signals were all in Morse Code (cw) sent by spark gap transmitters, which transmitted very broad code signals that took up a lot of space. A *broad* signal occupies more space on the frequency spectrum than is necessary for the transmission it is sending.

A look at the spectrum

In general terms, *frequency* describes the number of times some particular event takes place. When we use the term to describe events in the audio and radio ranges, we are referring to *cycles*. A cycle describes a waveform that goes through changes consisting of rising to a maximum in one direction falling to zero and then rising to a maximum in the opposite direction and falling again to zero. Think of the 60 cycle per second alternating current of the electric energy supplied to your home.

All alternating cycles have a certain frequency. This is only a way of saying how often they complete one cycle on a time scale of one second. If the power from a Hi-Fi amplifier moves the speaker cone back and forth 40 times a second, we will hear a very low pitched sound. If the same amplifier moves the cone of a tweeter speaker back and forth 15,000 times a second, we will hear a very high pitched tone.

These alternating cycles don't stop at the end of our audio range. As they alternate more rapidly they move into that portion of the frequency spectrum we know as supersonic—beyond our hearing, but within the range of most animals.

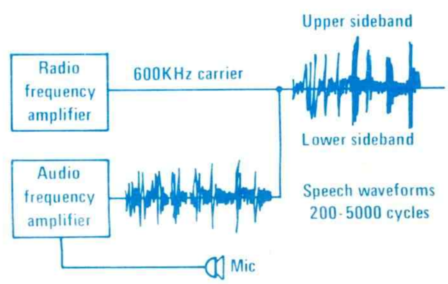

Figure 2 **When the steady radio frequency carrier at 600 kHz is modulated by audio tones up to 5,000 cycles, two sidebands are produced. They are identical. The upper sideband is the sum of the carrier and the audio, 600-605 kHz. The lower sideband is the difference between carrier and audio, 595-600 kHz. A total space of 10 kHz is required for the fully-modulated am signal from 595-605 kHz.**

In terms of the entire frequency spectrum we are still talking about very low frequencies since the high end of the frequency spectrum describes cycles that are occurring at the rate of 300,000,000 per second! The spectrum has been divided up into *bands* to describe certain segments. Each band within the total radio frequency spectrum has transmission characteristics which are unique. We will deal mainly with the *shortwave* bands and how your receiver functions at these frequencies.

To picture a band, think of the horizontal axis on a graph. Fortunately, many radio dials are made in just this fashion, with the lowest frequency on the left and gradations or numbers, rising to the highest frequency on the right. It is a scale that most of you have seen, so it will be easy to relate to it.

Obviously the vast range of the spectrum would be cumbersome to describe if the single standard of cycles per second were used. Multipliers are used that have universal acceptance. Kilocycles means 1,000 cycles.

So, for more than 50 years frequency was described in the radio field as kilocycles and megacycles and everyone understood what was meant. The general public knew just where their favorite stations were on the am broadcast band by tuning to a certain number of kilocycles. The term was changed to honor the German physicist Heinrich Hertz, who first proved the existence of radio waves in his laboratory. Now it is know as kilohertz and megahertz. Of course, the whole idea of radio waves was postulated by the English physicist, James Maxwell, 30 years before Hertz proved it, so we could have been faced with "kilomaxwells."

Now we hear music

In time those old spark gap code transmitters gave way to tube amplifiers, which transmitted narrow code signals and took up much less space in the spectrum. Then a new development came along. The radio *carrier* which had been only turned off and on to send Morse Code was now being *modulated* by having an audio signal superimposed on the radio signal. Voice and music could now be transmitted over the radio!

The very process of modulation created additional radio frequencies besides the one used to transmit the carrier. Although the audio range runs up to a high of 20 kHz, most tones of the human voice are within the 5 kHz range, and many musical tones are within that limit. In short, it's perfectly adequate to transmit audio signals only within the range of 200 to 5,000 Hz on the am broadcast band. At least 10 kHz of room was needed for both the sidebands that were generated by the modulation process.

The U.S. broadcast band for am stations covers the frequency spectrum from 550 kHz to 1650 kHz. That's only 1100 kHz of room to hold all the stations in the country! If we allow some room between stations, so they don't interfere with each other, that would allow only five dozen or so stations to broadcast at the same time.

Fortunately, the range on this band is quite limited most of the time, and station frequencies are allocated based on the region or area to be served by a particular station.

How about the ability of receivers to discriminate between stations on this band? Some old TRF receivers didn't do too badly, as long as they weren't too close to a powerful station. After all, their job was to select a station that was 10 kHz wide on a band that was only 550 to 1650 wide. As a ratio or percentage, that 10 kHz was pretty low when compared to the received frequency. Let's look at it this way. If there is a station on 600 kHz and another station on at 620 kHz, they are 20 kHz apart. Even with those 5 kHz sidebands, the separation is still 10 kHz between the "edges" of the two signals. All the receiver must do is be able to discriminate between the two frequencies, and yet be wide enough to allow both of those sidebands to come on through, be amplified, detected and come out of the speaker.

If we compare the 10 kHz separation to the total frequency involved, 600 kHz, we are looking at a selection ratio of 1 to 60, not too tough a job. What happens when we tune the radio to the *high* end of the broadcast band? Now we find ourselves discriminating between two stations separated by the same distance. Only now the ratio has increased to 10 kHz out of 1600 kHz, or a ratio of 1 to 160, a much tougher job of selection.

So the superheterodyne receiver was invented. It sounds complicated, but it operates on a simple principle. Musicians had long known that if two notes were struck on a piano at least four distinct tones could be heard. And they were in exact mathematical relationship to each other, being the sum and the difference between the two original notes, expressed in Hertz per second in the audio range.

The sum and the difference

The same thing can be done in the radio frequency range. Each signal you want to hear is being transmitted on a certain frequency. If you generate within your receiver another signal that is separated from the received signal, they will *beat* together just like those two piano notes. And they will produce two new frequencies, the *sum* and *the difference* between the transmitted frequency and your own internal frequency. Now you can choose to deal only with the beat or *heterodyne* frequency that represents the difference between the received signal and your own local signal.

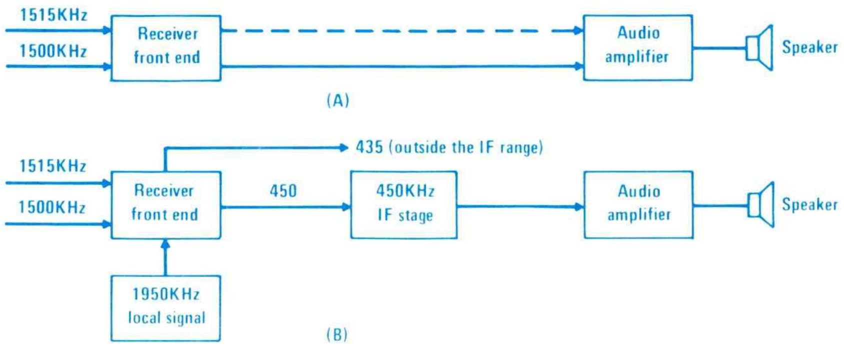

Figure 3 **In tuned radio frequency (TRF) receiver A, the difference between the two signals is only 15 kHz out of 1500 kHz, a ratio of 100:1. In the superheterodyne circuit B, the local signal has been used to beat with the incoming signals. The difference produces two signals, one at 450 kHz and one at 435 kHz. The 15 kHz difference is now out of 450 kHz, a ratio of 30:1. The signal at 1515 kHz will fall outside the bandpass of the i.f. stage and not be amplified.**

You see, the higher we go in frequency, the less the signal width there is in relation to the received frequency. In spite of putting several tuned circuits in the amplifying section of the receiver, the gate through which signals will come is still quite broad in relation to one signal. Unless some modification is made, it is likely that such a receiver will simply receive several stations at the same time. Now you see why kids' walkie-talkies supposed to work on channel 14 seem to pick up signals from channels 10 through 18 with no trouble at all. It is basically a TRF receiver. Operating in the 27 mHz band and trying to select a signal 8 kHz wide and reject others is an impossibility! The selection ratio now is about 3.3 million to 1—and the gate for such a receiver is wide enough to let in signals many kHz away.

Let's take an example from our am broadcast receiver to see how the heterodyne principle works. In our TRF receiver if we wanted to hear a station at 1500 kHz, we tuned the set to that frequency. If there was another strong station at 1515 kHz, it was likely that we would hear it, too, or at least the top part of its lower sideband. In fact, we could pick up the beat of the music being played just from the interference peaks, the squawk squawk sounds of voice or music peaks.

Suppose we had an *internal* signal being generated inside our receiver that was tuned to 1,950 kHz. That's going to beat or heterodyne with the incoming signal. It will produce two new frequencies, one at 3,450 being the sum of the two, and one at 450 kHz, being the difference between the two.

Now, suppose we had some *fixed tuned* circuits tuned to 450 kHz. They would pick up that 450 kHz signal and treat it just as if it were the original signal being transmitted. That's right! The entire signal would come on through, carrier frequency and both sidebands. Only the entire bit would have been transferred to this *intermediate frequency*, known as the IF stage. We've now got that 10 kHz wide signal back down to a ratio of 1 to 45. That's a lot better than a ratio of 1 to 150.

Is there anything else that happened? Remember that interfering signal up at 1515 kHz? It's going to have to beat with our local signal also. But the difference between 1515 kHz and 1950 kHz is 435 kHz. Our IF stages are tuned to 450 kHz and that heterodyne signal from the station at 1515 kHz is 15 kHz away. And that's 15 kHz out of only 450, not out of 1500. As a ratio of frequency, the interfering signal is 30 to 1 instead of 100 to 1.

Remember the lower in frequency we go, the fewer number of kilohertz that can be accommodated in a tuned circuit. It's not difficult to construct a tuned circuit at 450 kHz that is only 10 or 15 kHz wide. To obtain the same width at much higher frequencies is much more difficult.

The superheterodyne principle is used universally in all radio receivers, even in that tiny transistor job you can stick in your pocket. It provides for excellent selectivity even on fairly crowded bands—such as the top part of the am broadcast band.

Not quite solved

So the problem with receiver selectivity was solved, right? Wrong. What was good enough down on the am broadcast band, simply wouldn't do the trick when we got up to those shortwave frequencies. And, don't forget, the stations on the broadcast band sit sedately on their assigned frequencies and don't try to elbow each other out of the way. Such a benign condition hardly exists in the assigned shortwave broadcasting bands, where there's intense competition for air space. And if that's bad, how about the crowded ham bands, or, even worse, those CB channels?

So far we've been talking about selectivity as if only one dimension mattered, the width of the frequency response of the IF stage. Unfortunately, we must concern ourselves with another dimension that is shown on the vertical scale because it is not enough to have selectivity if the rejection characteristic can be overcome by very strong signals that are close to the desired frequency. Our IF selection process must incorporate circuitry that *filters* out unwanted signals, even extremely strong ones. Thus it will look on a chart much more like a window than just a simple width of frequency.

It is very important to get a handle on the concept of filter depth as well as width. Basically, any device that allows certain frequencies to pass through, while rejecting other frequencies is a *filter*. Many of you already are familiar with the hi-pass filter which you put on your tv set to let high frequencies pass

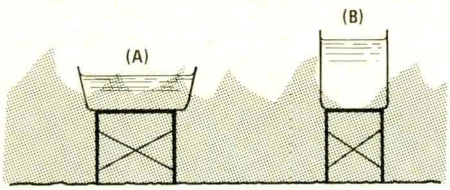

Figure 4 **Two aqueducts carry an equal amount of water across the surface of a lake. A has lower, sloping sides and is more likely to have lake waves slop over into it. B is more immune to wave action. A bandpass filter with steep sides, likewise, is less prone to strong, nearby signals.**

through, while discriminating against frequencies lower than the tv spectrum that might contain interfering signals. Or, perhaps you put a lo-pass filter on your ham or CB set to prevent high frequency signals from getting out and into someone's tv set.

Another type of filter is called a *band-pass filter*. This filter allows a specific band of frequencies to pass through it, while rejecting all frequencies above and below the specific band. This type of filter, installed in the IF stage of your ham, CB or communications grade receiver allows the kind of selectivity needed to pick out that one signal from many. Think of it like a "window" in your receiver.

The process of tuning your radio involves sliding this window along a radio frequency band if you have continuous tuning, or placing the window in the middle of a certain channel, if you tune by switching. If you have a *slider* in your CB radio, it allows you to shift the window back and forth a bit, in order to let the desired signal through.

The width of a band-pass filter is usually described in kiloHertz. On the horizontal scale of a graph it shows the frequencies that will be passed through by the filter. You might think of it in terms of a raised aqueduct with the radio frequencies as the water moving on through. Just how wide the band-pass filter should be depends on the specific job it must do.

Band-pass filters for use in the IF stage can be made to be very narrow, less than 500 Hz, or fairly broad, say 10 or 12 kHz. To copy a code (cw) signal, we need only pass through a very small increment of frequencies—just enough to create the tone we need to hear to decode those *dits* and *dahs*. In this case it's best to have a very narrow filter, say only 500 Hertz wide, to let us discriminate between very close together signals.

Telephone quality

When we need to copy the sounds of the human voice with enough clarity to understand what is being said, the filter must be much broader. In tests many years ago, the Bell Telephone Laboratory found that 90 percent of intelligibility is contained between 400 and 2700 Hertz and 95 percent within 200 to 3000 Hz. That was an important finding because in any crowded frequency spectrum there is no purpose in using any more room than necessary.

Your telephone transmits and receives only within these boundaries. The expression telephone quality is commonly used in the communications business to describe voice response characteristics which are entirely readable, but lacking the fullness of the human voice in an unfiltered environment.

Even if the voice frequency response were fairly restricted, say the typical 4 kHz in CB equipment, the modulation process is going to create those sidebands, the sum and difference of the carrier frequency, which will occupy a space a little greater than 8 kHz. For an am receiver to properly detect and

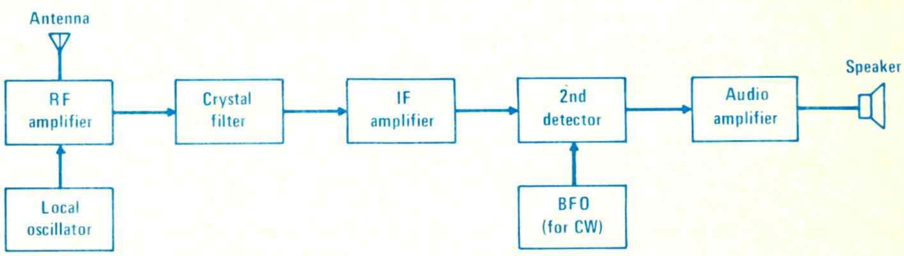

Figure 5 **The superheterodyne receiver uses a local oscillator to produce a signal to beat against the incoming signal and produce an intermediate signal at a different frequency which then is sent through a crystal bandpass filter which passes only a narrow band of frequencies, rejecting all others. To copy International Morse code, for instance, another local oscillator is needed. It is a beat frequency oscillator (BFO) which is offset just enough in frequency to produce a beat note with the carrier coming in through the i.f. amplifier.**

amplify that signal so it will come out of the speaker in about the same way it left the transmitter, the band-pass filter must be able to pass the entire signal through—all 8 kHz of it.

If, however, the signal being transmitted were of the single sideband variety (SSB) only *one* sideband is needed to pass through the filter in order to come out of the speaker in the same way it left the transmitter. Obviously, the filter needs only to be half as wide, 4 kHz in this case.

The width of the band-pass filter depends largely on the job it does. If we try to run a voice signal through the cw filter, so many of the voice frequencies are rejected by the filter that we can't make any intelligence out of the sounds we hear.

For many years filters were made up of coil and capacitor combinations contained in IF *transformers* operating in the frequency range of 200 to 500 kHz. The use of *crystal filters* became predominant when really good selectivity was needed because they have the characteristic of being sharper than the coil-capacitor types.

It is this sharpness that is directly related to the depth of the filter. Thinking of the analogy we made of a filter to a raised aqueduct or canal, we can see that the slope of the sides is an important determination in the volume of water that can be passed through in relation to the width of the canal or aqueduct. The steeper the sides of the aqueduct or canal, the higher it will be off of the bottom point. If we visualize such an aqueduct running along the surface of a lake, it is clear that waves from the lake will have less chance of slopping over into the aqueduct if the sides are high than if it were a wide aqueduct with gently sloping sides.

Keeping strong signals out

That's just the way it is with a band-pass filter. Only in this case the waves are radio signals from nearby frequencies. There are two important considerations involved in their ability to slop over into our filter. One is the height of the wave (strength of signal) and the other is the distance away from the aqueduct (number of kilohertz away from center frequency).

How can we measure the height of our filter's walls? Perhaps the best way would be to figure what strength a signal would need to get over the wall. We can do this in two ways. We can use the ratios of received radio voltage that would be needed to scale the wall in relation to the center frequency of the filter, showing just how much of an increase is required to make it over as the wall moves away from the middle or center frequency. Or we can use the power ratio of decibels (db) which will indicate the increase in power necessary to scale the wall at different frequency points. We will plot both the voltage (actually microvolts or millionths of a volt) and decibels at a zero level at the bottom of our aqueduct—the center frequency of our band-pass filter.

Figure 6 shows the typical selectivity curve (slope of sides) of a very superior band-pass filter. This is the shape of the window that your receiver will place at various points in the radio frequency band. If it is a steep-sided window as shown for this filter, it will present quite an obstacle to any signal that is not within its pass band. It would take a tremendous wave very close at hand to slop over this type of wall.

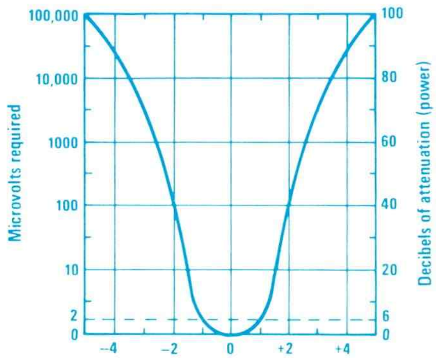

***Figure 6* The curve of a very sharp band-pass filter. Too sharp for am, it would have to be used for single-sideband reception. At the 60 db point a signal would have to be a million times stronger than one at the center to be heard, although it would be only 2500 cycles away in frequency.**

Just what kind of power ratios are we talking about when we think of the filter in Figure 6 as operating inside a radio receiver? Looking at the chart, we can see that the width of the filter at the 6 db point is 2.5 kHz. This is the point at which the width of a filter is normally first measured. It means that a signal four times as strong as one right in the middle of the band-pass will go through the filter, passing 2.5 kHz of frequency spectrum. The next point commonly measured is at 60 dbs. This is a power ratio one million times as strong as the zero db point in the middle of the filter. It would take a signal that strong only 2.5 kHz on either side of the center to get through at the same level as a signal right on center! That's a pass band of 5 kHz, only twice as wide as the window down at the 6 db point. In filter design this is known as the *shape factor* which represents the bandwidth at 6 db divided into the bandwidth at 60 db. The shape factor of the filter shown in Figure 6 is 2:1, a very, very sharp filter curve. If you look at the 80 db point, you can see that the width is 7.5 kHz, but 80 dbs is a power ratio of 100 million compared to that zero db point in the middle. That means that the signal of only 2 microvolts that goes through the filter easily, must be increased to 10,000 microvolts to make it through the filter only 3.75 kHz away from the center frequency.

Ideal Filter

What is an ideal band-pass filter to give you the best window for your needs? It should be wide enough to pass through all the frequencies you want to hear. Its rejection of unwanted frequencies should be as great as possible, so that signals on either side (high or low) of the one you want to hear will not interfere.

That's a pretty tall order for any receiver. Of course, that Hi-Fi fm receiver must have a very wide filter to pass all of those audio tones through. Your am CB radio will have a filter that is really put to the test, considering the sheer number of stations on adjacent channels, not to mention those that are running illegal power. In fact, the best band-pass filter in the world can't stop signals that are inside of its pass band. Such is the case with splatter that occurs from mis-used or mis-adjusted ham or CB transmitters that put out spurious signals up and down the band many kilohertz away from where they are supposed to be.

The band-pass filter is really the heart of any good receiver. In selecting a radio receiver for shortwave reception, CB or ham use, it is probably the single most important characteristic you should look for.

It would have been possible to make the band-pass filters much sharper for CB am radios if the frequency tolerance specifications required by the FCC of radio manufacturers had been tighter. The plus or minus .005 percent from center channel frequency actually allows two sets to be as much as 2.7 kHz apart from each other, and yet still meet specs. Add that figure to the eight kHz needed to accommodate both am sidebands and you begin to see that the six db point on a good CB filter has to be about 10 kHz wide to allow all possible signals on the channel to come through.

Very few manufacturers give their selectivity information in a manner that can be accurately related to typical filter curves. Knowing that a 2:1 shape factor is difficult and expensive to obtain, we can be fairly sure that very few manufacturers would even come close to it. But even a 2:1 factor would mean a bandwidth of 20 kHz at 60 db for our am filter.

The majority of CB channels are spaced every 10 kHz, right next to each other! So, even with a really good filter, part of the pass band will be in adjacent channels. A very strong, or very local, signal can easily force its way through the window and out of your loudspeaker.