Understanding Receiver Sensitivity

By Ian Poole

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Receiver sensitivity is one of the key specifications of any radio. The two main requirements of any set are that it should be able to separate one station from another, i.e. selectivity, and signals should be amplified so that they can be brought to a sufficient level to be heard. As a result receiver designers battle with many elements to make sure that these requirements are fulfilled

Noise

Today technology is such that there is little problem in being able to achieve very large levels of amplification. This is not the limiting factor. In any receiving station the limiting factor is noise - weak signals are not limited by the actual signal level, but by the noise masks them out. This noise can come from a variety of sources. It can be picked up by the antenna or it can be generated within the receiver.

It is found that the level of noise that is picked up by a receiver falls as the frequency increases. At HF and frequencies below this the combination of galactic, atmospheric and man-made noise is relatively high and this means that there is little point in making a receiver superbly sensitive. Normally receivers are designed such that the internally generated noise is much lower than any received noise, even for the quietest locations.

At frequencies above 30 MHz the levels of noise start to reach a point where the receiver noise becomes far more important. By improving the noise performance of the set, it becomes possible to hear much weaker signals.

In terms of the receiver noise performance it is always the first stages or front end that is most crucial. At the front end the signal levels are at their lowest and even very small amounts of noise can be comparable with the incoming signal. At later stages in the set the signal will have been amplified and will be much larger. The same levels of noise as are present at the front end will be a much smaller proportion of the signal and will not have the same effect. Accordingly it is important that the noise performance of the font end is optimised for its noise performance.

Measuring noise performance

There are a number of ways in which the noise performance, and hence the sensitivity of a receiver can be measured. The most obvious method is to compare the signal and noise levels for a known signal level. Obviously the greater the difference between the signal and the unwanted noise, the better the sensitivity performance.

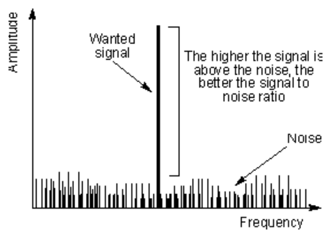

Figure 1 Signal to noise ratio

The difference is normally shown as a ratio between the signal and the noise (S/N) and it is normally expressed in decibels. As the signal input level obviously has an effect on this ratio, the input signal level must be given. This is usually expressed in microvolts. Typically a certain input level required to give a 10 dB signal to noise ratio is specified.

A number of other factors apart from the basic performance of the set can affect the specification. The first is the actual bandwidth of the receiver. As the noise spreads out over all frequencies it is found that the wider the bandwidth of the receiver, the greater the level of the noise. Accordingly the receiver bandwidth needs to be stated.

Additionally it is found that when using AM the level of modulation has an effect. The greater the level of modulation, the higher the audio output from the receiver. When measuring the noise performance the audio output from the receiver is measured and accordingly the modulation level of the AM has an effect. Usually a modulation level of 30% is chosen for this measurement.

This method of measuring the performance is most commonly used for HF communications receivers. Typically one might expect to see a figure in the region of 0.5 microvolts for a 10 dB S/N in a 3 kHz bandwidth for SSB or Morse. For AM a figure of 1.5 microvolts for a 10 dB S/N in a 6 kHz bandwidth at 30% modulation for AM might be seen.

SINAD

Whilst signal to noise specifications are often seen, another similar specification that is used is the SINAD measurement. This is most commonly used in conjunction with FM receivers, although there is no reason why it cannot be used for AM or even SSB. The measurement is similar to signal to noise ration, but includes distortion and is a ratio of signal plus noise plus distortion to noise plus distortion. To make the measurement a signal modulated with an audio tone is entered into the receiver. A measurement of the whole signal, i.e. the signal plus noise plus distortion is made. As the frequency of the tone is known and the regenerated audio is passed into a filter to remove the tone. The remaining noise and distortion is then measured.

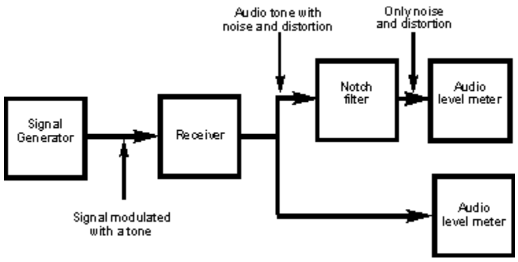

Figure 2 Making a SINAD measurement

Normally the specification takes the form of a certain input level required to achieve a given SINAD. Typically a SINAD of 12dB is taken because this corresponds to distortion factor of 25%. A typical specification might be that a receiver has a sensitivity of 0.25 uV [microvolts] for a 12 dB SINAD. Obviously the lower the input voltage needed to achieve the given level of SINAD, the better the receiver performance.

Whilst the measurement is most commonly associated with FM equipment there is no reason why it cannot be used for AM, and in deed it often is. It can also be used for SSB, but it is necessary to ensure that the receiver is tuned into exactly the right frequency so that the audio tone is reconstituted with exactly the right pitch so that it can be properly notched out in the measurement.

To make the measurement a signal modulated with an audio tone is entered into the receiver. The frequency of the tone is known and the regenerated audio is passed into a filter to remove the tone. The remaining noise and distortion is then measured.

Noise Figure

For equipment that is used above 30 MHz a system known as Noise Figure is more widely used. However there is no reason why it cannot be used at any frequency and sometimes it is. The system is very versatile and can be used to measure the noise performance of a whole receiver, or a small part of a system like a preamplifier.

Essentially the measurement assesses the amount of noise each part of the system or the system as a whole introduces. If the system were perfect then no noise would be added to the signal when it passed through the system and the signal to noise ratio would be the same at the output as at the input. As we all know this is not the case and some noise is always added. This means that the signal to noise ratio at the output is worse than the signal to noise ratio at the input.

A figure known as the noise factor can be derived simply by taking the signal to noise ratio at the input and dividing it by the signal to noise ratio at the output:

As the signal to noise ratio at the output will always be worse, this means that the noise factor is always greater than one.

The noise factor is rarely seen in specifications. Instead the noise figure is always seen. This is simply the noise factor expressed in decibels.

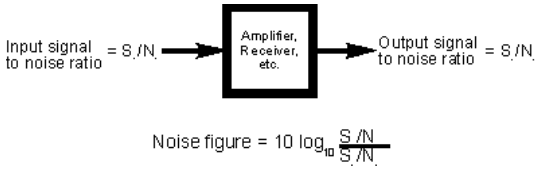

Figure 3 Noise figure

In the diagram S1 is the signal at the input, N1 is the noise at the input and S2 is the signal at the output and N2 the noise at the output

As an example if the signal to noise ratio at the input was 4:1, and it was 3:1 at the output then this would give a noise factor of 4/3 and a noise figure of 10 log (4/3) or 1.25 dB. Alternatively if the signal to noise ratios are expressed in decibels then it is quite easy to calculate the noise figure simply by subtracting one from another because two numbers are divided by subtracting their logarithms. In other words if the signal to noise ratio was 13 dB at the input and only 11 dB at the output then the circuit would have a noise figure of 13 - 11 or 2 dB.

Typical examples

The specifications of different pieces of equipment will vary quite widely. A typical HF receiver may have a noise figure of 15 dB of more and function quite satisfactorily. A better level of performance is not necessary because of the high level of atmospheric noise. However an amateur receiver used on Two metres, for example, might have a noise figure of 3 or 4 dB. Preamplifiers for this band often have a noise figure of around 1 dB. However it is interesting to note that even the best professional wide-band VHF UHF receivers may only have a noise figure of around 8 dB.

Summary

Receiver sensitivity is one of the vital specifications of any receiver. Whether measured as a signal to noise ratio, SINAD or noise figure it is essential that any receiver has a sufficient level of sensitivity. However this is not the complete story as other specifications are also important, as we shall see nest time when we look at dynamic range

Understanding Receiver Selectivity

By Ian Poole

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Selectivity is one of the major specifications of any receiver. Whilst the sensitivity is important to ensure that it can pick up the signals and receive them at a sufficient strength, the selectivity is also very important. It is this parameter that determines whether the receiver is able to pick out the wanted signal from all the other ones around it. The filters used in receivers these days have very high levels of performance and enable receivers to select out individual signals even on today's crowded bands.

Superhet principle

Most of the receivers that are used today are superhet radios. In these sets the incoming signal is converted down to a fixed intermediate frequency. It is within the IF stages that the main filters are to be found. It is the filter in the IF stages that defines the selectivity performance of the whole set, and as a result the receiver selectivity specification is virtually that of the filter itself.

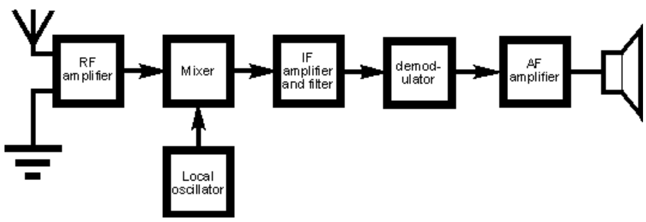

Figure 1 Block diagram of a basic superhet receiver

In some receivers simple LC filters may be used, although ceramic filters are better and are used more widely nowadays. For the highest performance crystal or mechanical filters may be used, although they are naturally more costly and this means they are only found in high performance sets.

Filter parameters

There are two main areas of interest for a filter, the pass band where it accepts signals and allows them through, and the stop band where it rejects them. In an ideal world a filter would have a response something like that shown in Figure 2. Here it can be seen that there is an immediate transition between the pass band and the stop band. Also in the pass band the filter does not introduce any loss and in the stop band no signal is allowed through.

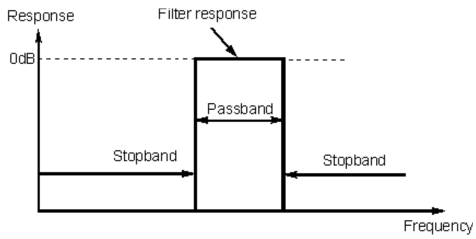

Figure 2 The response of an ideal filter

In reality it is not possible to realise a filter with these characteristics and a typical response more like that shown in Figure 3. It is fairly obvious from the diagram that there are a number of differences. The first is that there is some loss in the pass band. Secondly the response does not fall away infinitely fast. Thirdly the stop band attenuation is not infinite, even though it is very large. Finally it will be noticed that there is some in band ripple.

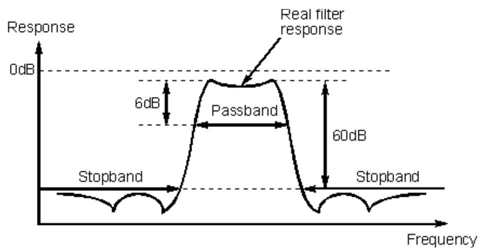

Figure 3 Typical response of a real filter

In most filters the attenuation in the pass band is normally relatively small. For a typical crystal filter figures of 2 - 3 dB are fairly typical. However it is found that very narrow band filters like those used for Morse reception may be higher than this. Fortunately it is quite easy to counteract this loss simply by adding a little extra amplification in the intermediate frequency stages and this factor is not quoted as part of the receiver specification.

It can be seen that the filter response does not fall away infinitely fast, and it is necessary to define the points between which the pass band lies. For receivers the pass band is taken to be the bandwidth between the points where the response has fallen by 6 dB, i.e. where it is 6 dB down or -6 dB.

A stop band is also defined. For most receiver filters this is taken to start at the point where the response has fallen by 60 dB, although the specification for the filter should be checked this as some filters may not be as good. Sometimes a filter may have the stop band defined for a 50 dB attenuation rather than 60 dB.

Shape factor

It can be seen that it is very important for the filter to achieve its final level of rejection as quickly as possible once outside the pass band. In other words the response should fall as quickly as possible. To put a measure on this, a figure known as the shape factor is used. This is simply a ratio of the bandwidths of the pass band and the stop band. Thus a filter with a pass band of 3 kHz at -6dB and a figure of 6 kHz at -60 dB for the stop band would have a shape factor of 2:1. For this figure to have real meaning the two attenuation figures should also be quoted. As a result the full shape factor specification should be 2:1 at 6/60 dB.

Filter types

There is a variety of different types of filter that can be used in a receiver. The older broadcast sets used LC filters. The IF transformers in the receiver were tuned and it was possible to adjust the resonant frequency of each transformer using an adjustable ferrite core.

Today ceramic filters are more widely used. Their operation is based on the piezoelectric effect. The incoming electrical signal is converted into mechanical vibrations by the piezoelectric effect. These vibrations are then affected by the mechanical resonances of the ceramic crystal. As the mechanical vibrations are then linked back to the electric signal, the overall effect is that the mechanical resonances of the ceramic crystal affect the electrical signal. The mechanical resonances of the ceramic exhibit a high level of Q and this is reflected in its performance as an electrical filter. In this way a high Q filter can be manufactured very easily.

Ceramic filters can be very cheap, some costing only a few cents. However higher performance ones are also available, and these are likely to be found in scanners and many other receivers.

For really high levels of filter performance crystal filters are used. Crystals are made from quartz, a naturally occurring form of silicon, although today's components are made from synthetically grown quartz. These crystals also use the piezoelectric effect and operate in the same way as ceramic filters but they exhibit much higher levels of Q and offer far superior degrees of selectivity. Being a resonant element they are used in many areas where an LC resonant element might be found. They are used in oscillators - many computers have crystal oscillators in them, but they are also widely used in high performance filters.

Normally crystal filters are made from a number of individual crystals. The most commonly used configuration is called the half lattice filter as shown in Figure 4. Further sections can be added to the filter to improve the performance. Often a filter will be quoted as having a certain number of poles. There is one pole per crystal, so a six pole crystal filter would contain six crystals and so forth. Many filters used in amateur communications receivers will contain either six or eight poles.

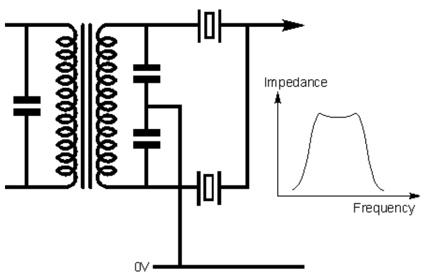

Figure 4 A basic half lattice crystal filter section

Choosing the right bandwidth

It is important to choose the correct bandwidth for a give type of signal. It is obviously necessary to ensure that it is not too wide, otherwise unwanted off-channel signals will be able to pass though the filter. Conversely if the filter is too narrow then some of the wanted signal will be rejected and distortion will occur. As different types of transmission occupy different amounts of spectrum bandwidth it is necessary to tailor the filter bandwidth to the type of transmission being received. As a result many receivers switch in different filters for different types of transmission. This may be done either automatically as part of a mode switch, or using a separate filter switch. Typically a filter for AM reception on the short wave bands will have a bandwidth of around 6 kHz, and one for SSB will be approximately 2.5 kHz. For Morse reception 500 and 250 Hz filters are often used.

Summary

Selectivity is particularly important on today's crowded bands, and it is necessary to ensure that any receiver is able to select the wanted signal as well as it can. Obviously when signals occupy the same frequency there is little that can be done, but by having a good filter it is possible to ensure that you have the best chance or receiving and being able to copy the signal you want.

Receiver Dynamic Range

By Ian Poole, <u>www.radio-electronics.com</u>

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Sensitivity is one of the main specifications people look at when buying a receiver. However the sensitivity of a set is by no means the whole story. The specification for a set may show it to have an exceedingly good level of sensitivity, but when it is connected to an antenna its performance may be very disappointing because it is easily overloaded when strong signals are present, and this may impair its ability to receive weak signals.

The overall dynamic range of the receiver is very important. It is just as important for a set to be able to handle strong signals well as it is to be able to pick up weak ones. This becomes very important when trying to pick up weak signals in the presence of nearby strong ones. Under these circumstances a set with a poor dynamic range may not be able to hear the weak stations picked up by a less sensitive set with a better dynamic range. Problems like blocking, inter-modulation distortion and the like within the receiver may mask out the weak signals, despite the set having a very good level of sensitivity.

What is dynamic range?
The dynamic range of a receiver is essentially the range of signal levels over which it can operate. The low end of the range is governed by its sensitivity whilst at the high end it is governed by its overload or strong signal handling performance. Specifications generally use figures based on either the inter-modulation performance or the blocking performance. Unfortunately it is not always possible to compare one set with another because dynamic range like many other parameters can be quoted in a number of ways. However to gain an idea of exactly what the dynamic range of a receiver means it is worth looking at the ways in which the measurements are made to determine the range of the receiver.

Sensitivity
The first specification to investigate is the sensitivity of a set. The main limiting factor in any receiver is the noise generated. For most applications either the signal to noise ratio or the noise figure is used as described in a previous issue of MT. However for dynamic range specifications a figure called the minimum discernible signal (MDS) is often used. This is normally taken as a signal equal in strength to the noise level. As the noise level is dependent upon the bandwidth used, this also has to be mentioned in the specification. Normally the level of the level of the MDS is given in dBm i.e. dB relative to a milliwatt and typical values are around -135 dBm in a 3 kHz bandwidth.

Strong signal handling
Although the sensitivity is important the way in which a receiver handles strong signals is also very important. Here the overload performance governs how well the receiver performance.
In the ideal world the output of an amplifier would be proportional to the input for all signal levels. However amplifiers only have a limited output capability and it is found that beyond a certain level the output falls below the required level because it cannot handle the large levels required of it. This gives a characteristic like that shown in Fig. 1. From this it can be seen that amplifiers are linear for the lower part of the characteristic, but as the output stages are unable to handle the higher power levels the signals starts to become compressed as seen by the curve in the characteristic.

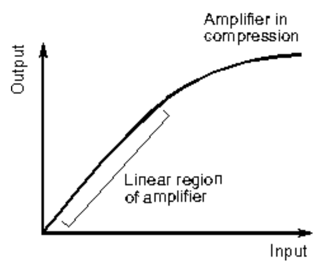

Fig. 1 A typical amplifier characteristic

The fact that the amplifier is non-linear does not create a major problem in itself. However the side effects do. When a signal is passed through a non-linear element there are two main effects which are noticed. The first is that harmonics are generated. Fortunately these are unlikely to cause a major problem. For a harmonic to fall near the frequency being received, a signal at half the received frequency must enter the amplifier. The front end tuning should reduce this by a sufficient degree for it not to be a noticeable problem under most circumstances.
The other problem that can be noticed is that signals mix together to form unwanted products. These again are unlikely to

cause a problem because any signals which could mix together should be removed sufficiently by the front end tuning. Instead problems occur when harmonics of in-band signals mix together.

Third order products
Problems occur when harmonics of in-band signals mix together. It is found that a comb of signals can be produced as shown in Figure 2, and these may just fall on the same frequency as a weak and intersting station, thereby masking it out so it cannot be heard.

It is simple to calculate the frequencies where the spurious signals will fall. If the input frequencies are f_1 and f_2, then the new frequencies produced will be at $2f_1 - f_2$, $3f_1 - 2f_2$, $4f_1 - 3f_2$ and so forth. On the other side of the two main or original signals products are produced at $2f_2 - f_1$, $3f_2 - 2f_2$, $4f_2 - 3f_1$ and so forth as shown in the diagram. These are known as odd order inter-modulation products. Two times one signal plus one times another makes a third order product, three times one plus two times another is a fifth order product and so forth. It can be seen from the diagram that the signals either side of the main signals are first the third order product, then fifth, seventh and so forth.

To take an example with some real figures. If large signals appear at frequencies of 30.0 MHz and 30.01 MHz, then the inter-modulation products will appear at 30.02, 30.03, 30.4 ...MHz and 29.99, 29.98, 29.97 ..... MHz.

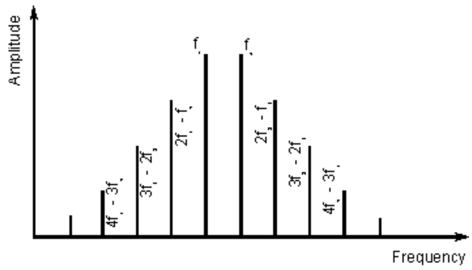

Fig. 2 Inter-modulation products

Blocking
Another problem that can occur when a strong signal is present is known as blocking. As the name implies it is possible for a strong signal to block or at least reduce the sensitivity of a receiver. The effect can be noticed when listening to a relatively weak station and a nearby transmitter starts to radiate, and the wanted signal reduces in strength. The effect is caused when the front-end amplifier starts to run into compression. When this occurs the strongest signal tends to "capture" the amplifier reducing the strength of the other signals. The effect is the same as the capture effect associated with FM signals.
The amount of blocking is obviously dependent upon the level of the signal. It also depends on how far off channel the strong signal is. The further away, the more it will be reduced by the front end tuning and the less the effect will be. Normally blocking is quoted as the level of the unwanted signal at a given offset (normally 20 kHz) to give a 3 dB reduction in gain.

Dynamic range definition
When looking at dynamic range specifications, care must be taken when interpreting them. The MDS at the low signal end should be viewed carefully, but the limiting factors at the top end show a much greater variation tin the way they are specified. Where blocking is used a reduction of 3 dB sensitivity is normally specified, but in some cases may be 1 dB used. Where the inter-modulation products are chosen as the limiting point the input signal level for them to be the same as the MDS is often taken. However whatever specification is given, care should be taken to interpret the figures as they may be subtlety different in the way they are measured from one receiver to the next.

To gain a feel for the figures which may be obtained where inter-modulation is the limiting factor figures of between 80 and 90 dB range are typical, and where blocking is the limiting factor figures around 115 dB are generally achieved in a good receiver.

Designing for optimum performance
It is not an easy task to design a highly sensitive receiver that also has a wide dynamic range. To achieve this performance a number of methods can be used. The front-end stage is the most critical in terms of noise performance. It should be optimised for noise performance rather than gain. Input impedance matching is critical for this. It is interesting to note that the optimum match does not correspond exactly with the best noise performance. The amplifier should also have a relatively high output capability to ensure it does not overload. The mixer is also critical to the overload performance. To ensure the mixer is not overloaded there should not be excessive gain preceding it. A high level mixer should also be used (i.e. one designed to accept a high-level local oscillator signal). In this way it can tolerate high input signals without degradation in performance. Care should be taken in the later stages of the receiver to ensure that they

can tolerate the level of signals likely to be encountered. A good AGC system also helps prevent overloading and the generation of unwanted spurious signals.
A receiver with a good dynamic range will be able to give a far better account of itself under exacting conditions than one designed purely for optimum sensitivity.

What do Those Specs Really Mean?

By Bob Grove W8JHD

Audio Output Power - Dynamic Range - Frequency Range - Keypad Frequency Entry - Modes - Noise Blankers - Notch Filter - Passband Tuning and IF Shift - Preamplifiers and Attenuators - Scannable Memory - Selectivity - Sensitivity - Tuning Steps

Everyone knows that specifications are important, but not everyone knows why. Oh, sure, we can generalize: "A sensitive shortwave receiver is better for DX." Maybe. Let's take a look at some of the more important specifications for shortwave receivers and try to make sense out of what they are telling us.

Frequency Range

While the shortwave spectrum is officially 1.8-30 MHz, we have to keep in mind that all receivers currently manufactured include the medium wave broadcast band as well (540-1700 kHz, the same as 0.54-1.7 MHz). But there's more.

Since virtually all portables are made and marketed overseas, the foreign domestic broadcast band (150- 300 kHz) is included as well. There are no voice transmissions below this, only some Navy digital communications; most tabletop receivers go down to 100 kHz.

Keypad Frequency Entry

Often called "Direct Entry," keypads are far more convenient for selecting discrete frequencies than rocking a dial back and forth, fine-tuning the desired frequency. Until digital synthesis of receiver oscillators, such exact control was impossible.

Tuning Steps

In the days of analog tuning, precise tuning of a signal to within a few hertz was easily obtainable, but with digital synthesis, such accuracy is expensive. Realistically, it becomes more of an issue with the reception of digital modes and single sideband than AM, where being off by hundreds of hertz is no problem.

Voice single-sideband stations, to sound natural, must be tuned within better than 25 Hz or so, while music, because of its absolute pitch intervals, must be even tighter.

Some receivers employ "direct digital synthesis," enabling increments as small as 1 Hz; in fact, 10 Hz is probably plenty good for virtually any hobby application.

Modes

Amplitude modulation (AM) is still the preferred mode for domestic and international broadcasting even though it does waste spectrum. It is sometimes called "full carrier double sideband," and the same audio information is duplicated in both sidebands (upper and lower). Synchronous detection (AM-Synch) is a receiving mode which locks onto the station's signal frequency without drifting. By choosing the stronger of the two sidebands, the reception remains stable during fades, and eliminates distortion produced by unequal sidebands.

Single sideband (SSB) actually transmits one sideband, eliminating both the carrier and the opposite sideband, making it inherently more spectrum-efficient, and immune from selective fading distortion. Virtually all two-way voice communications heard in the shortwave spectrum are in upper sideband (USB). Exceptions include amateur radio voice comms in the 160, 75, and 40 meter bands which are lower sideband (LSB).

Sensitivity

The measurement of a receiver's ability to respond to weak signals is its sensitivity. Since shortwave radio signals are detected as minute voltages, the measurement is made in microvolts (millionths of a volt).

Years ago, less sensitive vacuum-tube receivers required significantly larger antennas to capture enough signal energy to overcome their own noisy circuitry, the result of the hot filaments and cathodes producing electrical noise ("thermionic emission"). Modern solid-state electronics makes high sensitivity practical, with half-microvolt (0.5 uV) ratings, and smaller antennas commonplace.

Dynamic Range

But high sensitivity is only half the story. The ability of a receiver to respond faithfully and equally to weak and strong signals is a measure of its dynamic range, expressed in decibels (dB). Overly-sensitive receivers often become overloaded by strong signals, producing spurious, phantom signals which interfere with reception. Most common is intermodulation ("intermod"), but desensitization ("desense") which lowers the weak-signal capability of a receiver in the presence of strong signals.

Preamplifiers and Attenuators

During weak signal conditions, it is often an advantage to boost signal levels before they come into the receiver. Preamps are wide-bandwidth devices that amplify all signals over the entire frequency range at one time (with the possible exception of the medium-wave broadcast band to avoid strong local signal overload).

And if signal levels are generally excessive, an attenuator may be invoked to reduce all signal strengths to make them more manageable or the receiver's tuning and detecting circuitry.

Selectivity

Single-signal reception is the goal; we want it audible and without interference. There is little we can do to separate two signals on the same frequency, but there is plenty we can do to separate two adjacent-frequency signals.

Filters are frequency-selective components used in receivers to decrease the amount of spectrum being detected at any one time. While it may seem prudent to make filters as narrow ("sharp") as possible, in fact different modes require different bandwidths, as we noted before.

Since the human voice occupies approximately 3 kHz of audio spectrum, and AM signals double the amount of bandwidth, a conventional AM signal is about 6 kHz wide. If we narrow it down much below 4 kHz, we reduce its high frequency components considerably and it sounds muffled.

SSB is already narrower, so selectivity on the order of 2.1-2.4 kHz is common. Even narrower are digital modes; Morse code (continuous wave or "CW") is the narrowest of all, with bandwidths of less than 0.5 kHz adequate in most cases.

Passband Tuning and IF Shift

These two techniques allow the operator to manipulate a receiver's filtering circuitry to favor one of two close-spaced signals without simply narrowing the passband, which would produce muffling of the audio. Instead, the unwanted signal is rejected and the desired signal's bandwidth is preserved.

Notch Filter

A filter which can be invoked and adjusted to remove single tones ("heterodynes") from the desired signal is quite useful. Some advanced receivers use digital signal processing (DSP) to do this automatically and instantly without the listener having to turn a knob until the irritating pitch disappears.

Noise Blankers

Years ago, crackly electrical noise interference was reduced by an audio noise limiter (ANL). This was basically a voltage "clipper" which allowed an adjustable amount of normal audio to pass to the amplifier, but would clip off any sharp bursts of noise. These characteristically caused some distortion to the sound.

More modern receivers employ noise blankers which sense the arrival of the noise spike and momentarily shut off the circuitry for the duration of the interference spike. While they do result in less distortion, they are effective over a narrower range of interference than the old ANL.

Scannable Memory

The ability to store a favorite frequency and mode into a memory channel is certainly a benefit; switch the radio on, push a button, and there it is! Most shortwave sets now have memory, and often offer the ability to scan as well, allowing an automated hunt for active stations among the memorized channels.

Audio Output Power

In a home stereo system reserve audio powers in the 100-200 watt range are common. But we seldom crank the volume up that loud! In actual practice, as little as 3 watts into a decent-size speaker can provide room filling sound. Engineers often provide this specification along with another parameter: 10% total harmonic distortion (THD). This is the maximum audio power the receiver can deliver to a matched speaker without audibly distorting the sound. These definitions are admittedly simplified.

For more elaborate explanations of some of the often ignored or misunderstood specifications, return to the technical section to see additional online articles. However, the above summary should provide a guide to understanding the various circuit design characteristics which make up a receiver's specifications. After reading them over, you'll have a better idea of which specs are more important for your listening requirements!