Practical Radio Frequency Test & Measurement

A Technician's Handbook

JOSEPH J. CARR

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This book is dedicated to the fond memory of
Michael J. Shaffer, D.Sc.
Dr. Shaffer was Director, Bioelectronics Laboratory and
Professor of Anesthesiology (Emeritus),
George Washington University Medical Center
Apollo Space Capsule Life Support System
Flight Director, Biosatellite II
Radar and Telecommunications Engineer
Signals Officer, Royal Signal Corps
(British Army World War II)
Engineer, Mentor, Boss, and Friend

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PREFACE

This book is about making measurements on radio frequency devices and in radio frequency systems. "RF" is different from low frequencies for a number of reasons. The chief reason is that component sizes approximate wavelengths in many cases, so they become very important. Also, in certain cases stray inductance and capacitance have a much more profound effect on RF circuits than they do on lower frequency circuits. In this book you will find information about a number of topics, including the basic theories of RF and any other form of measurement, as well as some useful RF components for test and measurement. The Smith Chart is covered (it would be a contradiction to leave it out). Additional topics include: signal sources and signal generators; spectrum and network analyzers; RF power measurement; measurement of time, frequency, and period; radio receiver measurements; radio transmitter measurements; RF amplifier (including high-power amplifier) measurements; antenna gain and pattern measurement; antenna and transmission line measurements; L-C-R measurements at RF frequencies; and manual Time Domain Reflectometry (TDR) measurements.

My approach taken is to examine the instruments used in the various types of measurement and then to look at practical measurement methods. The latter is not always easily understood without knowledge of the former.

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CHAPTER EIGHT

Radio Receivers and Their Measurements

Radio receivers are at the heart of all communications systems. Wireless telecommunications, broadcasting, and radar could not exist without radio receivers. Navigation depends on radio receivers today. The Global Positioning System (GPS) receiver used in maritime, airborne, and land navigation systems uses a radio receiver at its heart. In this chapter we will cover several topics. We will discuss the different types of radio receivers that are on the market, as well as how to interpret receiver specifications. Finally, we will discuss the testing approaches.

WHAT IS A RADIO RECEIVER?

A radio receiver is an electronic device that must perform two basic functions: (a) *It must respond to, detect, and demodulate desired signals;* and (b) *It must not respond to, detect, or be adversely affected by undesired signals.* If it fails in either of these two functions, then it is a poorly performing design. Both functions are necessary. Weakness in either function makes a receiver a poor bargain, unless there is some mitigating circumstance. The receiver's performance specifications tell us the manufacturer's claims about how well their product performs these two functions. The tests associated with radio receivers are designed to measure either or both of these attributes.

Crystal Video Receivers

Crystal video receivers (Figure 8.1) grew out of primordial crystal sets, but are used in microwave bands even today. The original crystal sets used a naturally occurring PN junction "diode" made from a natural lead compound called galena crystal with an inductor-capacitor (L-C) tuned circuit. Later, crystal sets could be made using germanium or silicon diodes. When vacuum tubes became generally available, it was common to place an audio amplifier at the output of the crystal set. Modern crystal video receivers use silicon or gallium-arsenide microwave diodes

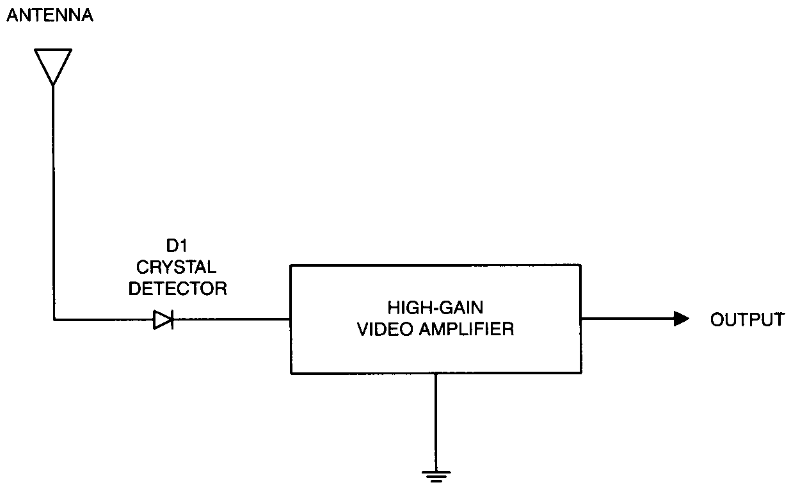

Figure 8.1 Crystal video receiver block diagram.

and a wideband video amplifier (rather than the audio amplifier). Applications include some speed radar receivers, aircraft radar warning receivers, and some very short-range communications receivers.

Tuned Radio Frequency (TRF) Receivers

The tuned radio frequency (TRF) radio receiver uses an L-C resonant circuit in the front-end, followed by one or more radio frequency amplifiers followed by a detector stage. Two varieties are shown in Figures 8.2 and 8.3. The version in Figure 8.2 is called a *tuned gain-block receiver.* It is commonly used in certain VLF scientific applications. Early TRF models (1920s) used independently tuned L-C circuits, but those proved to be very difficult to tune without creating an impromptu Miller oscillator circuit. Later versions mechanically linked ("ganged") the tuned circuits to operate from a single tuning knob, as in Figure 8.3.

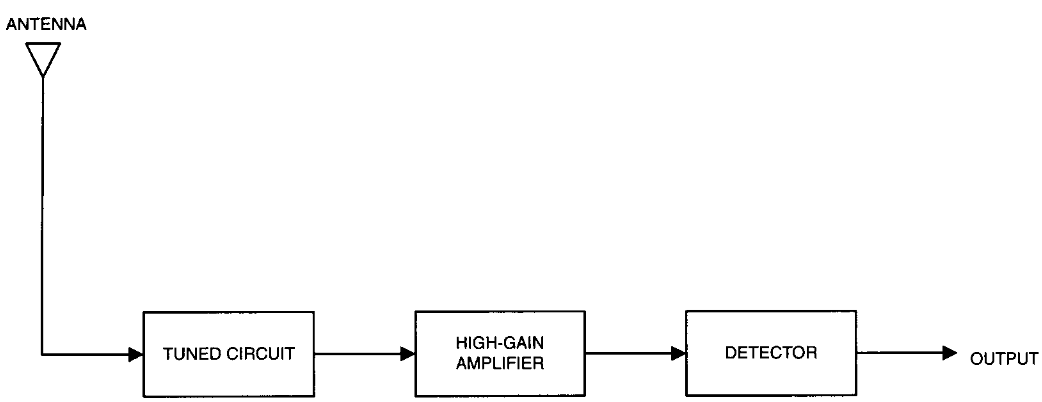

Figure 8.2 Simple-tuned radio frequency receiver block diagram.

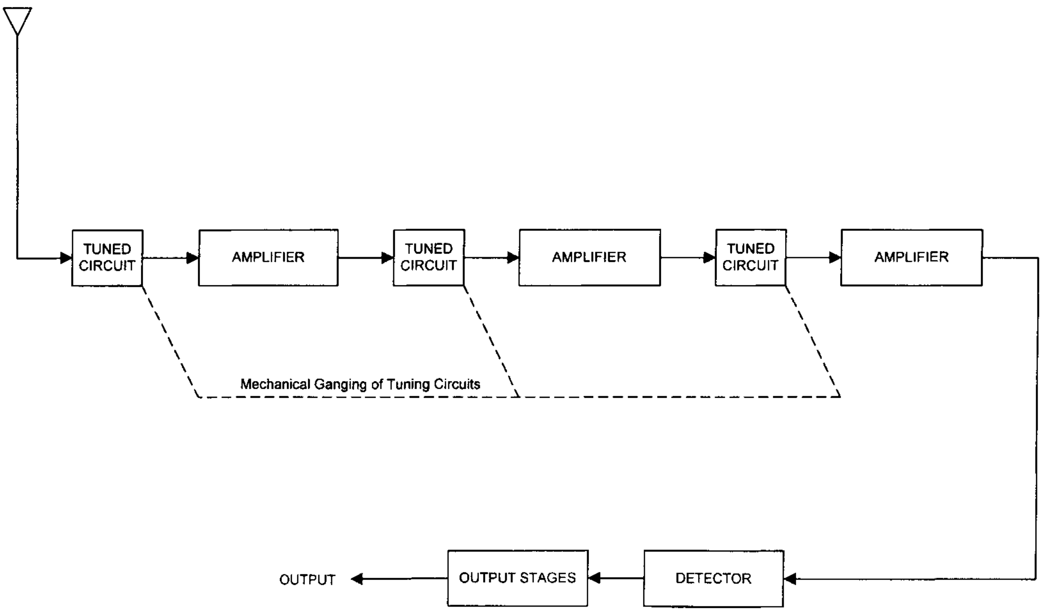

Figure 8.3 Multiply-tuned radio frequency receiver block diagram

Superheterodyne Receivers

Figure 8.4 shows the block diagram of a superheterodyne receiver. We will use this hypothetical receiver as the basic generic framework for evaluating receiver performance and the associated tests. The design in Figure 8.4 is called a *superheterodyne* receiver and represents the largest class of radio receivers in use today.

The superheterodyne receiver block diagram in Figure 8.4 is typical of many receivers. The purpose of a superheterodyne is to convert the incoming RF frequency to a single frequency where most of the signal processing takes place. The front-end section of the receiver consists of the *radio frequency* (RF) amplifier and any RF tuning circuits that may be used (A-B-C in Figure 8.4). In some cases, the RF tuning is very narrow, and basically tunes one frequency. In other cases, the RF front-end tuning is broadbanded. In that case, bandpass filters are used.

The frequency translator section (D and E) is also considered part of the front-end in most textbooks, but here we will label it as a separate entity. The translator consists of a frequency mixer and a local oscillator. This section does the heterodyning, which is discussed in more detail below. The output of the frequency translator is called the *intermediate frequency* (IF).

The translator stage is followed by the intermediate frequency (IF) amplifier. The IF amplifier (F-G-H) is basically a radio frequency amplifier tuned to a single frequency. The IF can be higher or lower than the RF frequency, but it will always be a single frequency.

A sample of the IF amplifier output signal is applied to an *automatic gain control* (AGC) section (L-M). The purpose of this section is to keep the signal level in the output more or less constant. The AGC circuit consists of a rectifier and ripple

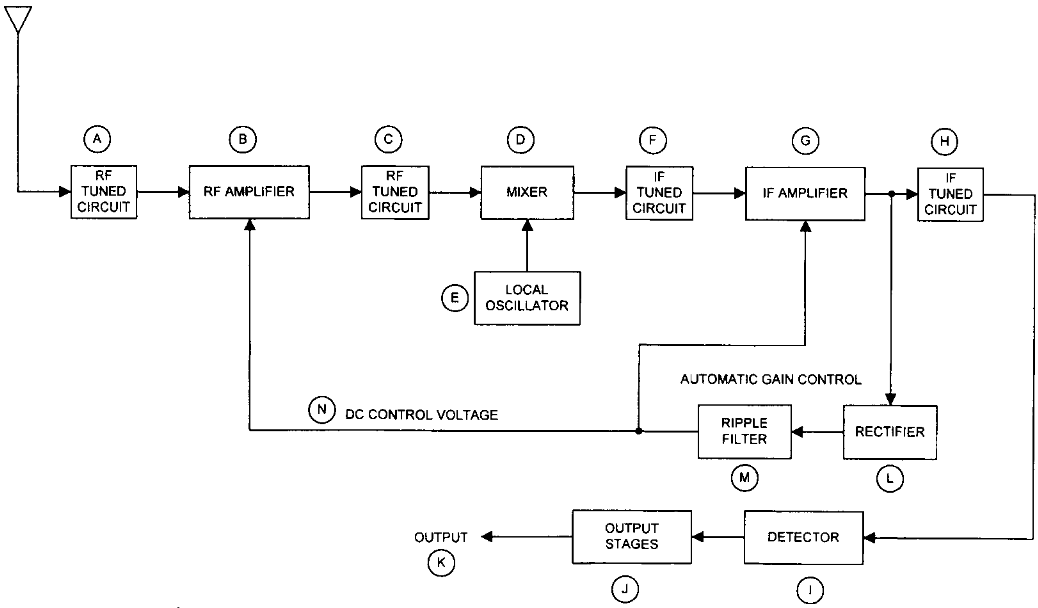

Figure 8.4 Superheterodyne receiver block diagram.

filter that produces a DC control voltage. The DC control voltage is proportional to the input RF signal level (N). It is applied to the IF and RF amplifiers to raise or lower the gain according to signal level. If the signal is weak, then the gain is forced higher, and if the signal is strong, the gain is lowered. The end result is to smooth out variations of the output level.

The detector stage (I) is used to recover any modulation on the input RF signal. The type of detector depends on the type of modulation used for the incoming signal. Amplitude modulation (AM) signals are generally handled in an *envelope detector.* In radio astronomy a special variant of the envelope detector called a *square law detector* is used. The difference is that the straight envelope detector is linear, while the square law detector is nonlinear to voltage. The square law detector takes advantage of the fact that the power level of the signal is related to the square of the applied signal voltage.

The output stages (J-K) are used to amplify and deliver the recovered modulation to the user. If the receiver is for broadcast use, then the output stages are audio amplifiers and loudspeakers. In some scientific receivers the output stages consist of integrator circuits and (sometimes) DC amplifiers.

Heterodyning

The main attribute of the superheterodyne receiver is that it converts the radio signal's RF frequency to a standard frequency for further processing. Although today the new frequency, called the *intermediate frequency* or *IF,* may be either higher or lower than the RF frequencies, early superheterodyne receivers always down-converted RF signal to a lower IF frequency (IF < RF). The reason was purely practical, for in those days higher frequencies were more difficult to process than lower frequencies. Even today, because variable-tuned circuits still tend to offer different performance over the band being tuned, converting to a single IF frequency, and obtaining most of the gain and selectivity functions at the IF, allows a more uniform overall performance over the entire range being tuned.

A superheterodyne receiver works by frequency converting (or simply "heterodyning"—the added "super" is vintage 1920s advertising hype) the RF signal. This occurs by nonlinearly mixing the incoming RF signal with a *local oscillator* (LO) signal. When this process is done, disregarding noise, the output spectrum will contain a large variety of signals according to:

$$F_O = mF_{RF} \pm nF_{LO} \qquad [8.1]$$

Where:

F_{RF} is the frequency of the RF signal

F_{LO} is the frequency of the local oscillator

m and n are either zero or integers (0, 1, 2, 3 . . . n)

Equation 8.1 means that there will be a large number of signals at the output of the mixer, although for the most part the only ones that are of immediate concern to understanding superheterodyne operation are those for which m and n are either 0 or 1. Thus, for our present purpose, the output of the mixer will be the fundamentals (F_{RF} and F_{LO}) and the second-order products ($F_{LO} - F_{RF}$ and $F_{LO} + F_{RF}$), as seen in Figure 8.5. Some mixers, notably those described as *double-balanced mixers* (DBM), suppress F_{RF} and F_{LO} in the mixer output, so only the second-order sum and difference frequencies exist with any appreciable amplitude. This case is simplistic and is used only for this present discussion. Later on, we will look at what happens when third-order ($2F1 \pm F2$ and $2F2 \pm F1$) and fifth-order ($3F1 \pm 2F2$ and $3F2 \pm 2F1$) become large.

Note that the local oscillator frequency can be either higher than the RF frequency (*high-side injection*) or lower than the RF frequency (*low-side injection*). There is ordinarily no practical reason to prefer one over the other except that it will make a difference whether the main tuning dial reads high-to-low or low-to-high.

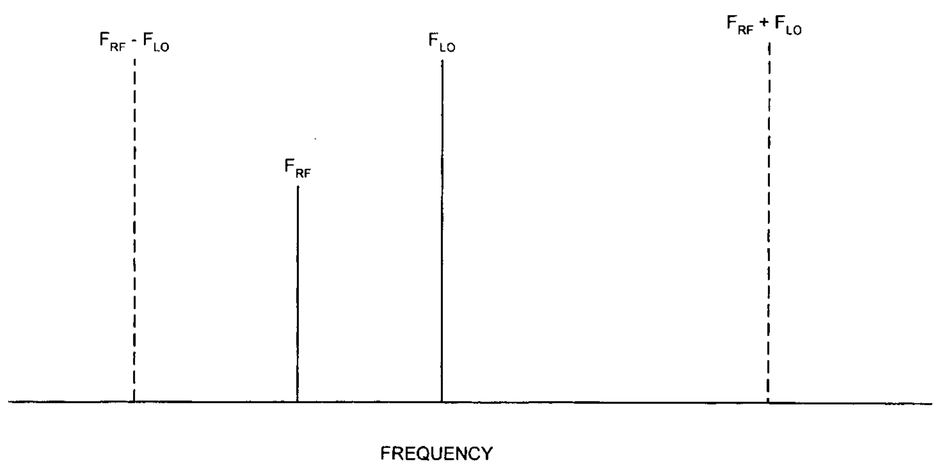

Figure 8.5 Simplified heterodyning mixer spectrum uses either sum or difference between LO and RF signals to produce an Intermediate Frequency (IF).

The candidates for IF include the sum (LO + RF) and difference (LO − RF) second-order products found at the output of the mixer. A high-Q tuned circuit following the mixer will select which of the two are used. Consider an example. Suppose an AM broadcast band superheterodyne radio has an IF frequency of 455 KHz, and the tuning range is 540 to 1,700 kHz. Because the IF is lower than any frequency within the tuning range, it will be the difference frequency that is selected for the IF. The local oscillator is set to be high-side injection, so it will tune from (540 + 455) = 995 kHz to (1,700 + 455) = 2,155 kHz.

Front-End Circuits

The principal task of the front-end and frequency translator sections of the receiver in Figure 8.4 is to select the signal and convert it to the IF frequency. But in many radio receivers there may be additional functions. In some cases (but not all), an RF amplifier will be used in front of the mixer. Typically, these amplifiers have a gain of 3 to 10 dB, with 5 to 6 dB being very common. The tuning for the RF amplifier is sometimes a broad bandpass-fixed frequency filter that admits an entire band. In other cases, it is a narrow-band, but variable-frequency, tuned circuit.

Intermediate Frequency (IF) Amplifier

The IF amplifier is responsible for providing most of the gain in the receiver, as well as the narrowest bandpass filtering. It is a high gain, usually multistaged, single-frequency tuned radio frequency amplifier. For example, one HF shortwave receiver block diagram lists 120 dB of gain from antenna terminals to audio output, of which 85 dB are provided in the 8.83 MHz IF amplifier chain. In the example of Figure 8.4, the receiver is a *single conversion* design, so there is only one IF amplifier section.

Detector

The detector demodulates the RF signal and recovers whatever audio (or other information) that will be heard by the listener. In a straight AM receiver, the detector will be an ordinary half-wave rectifier and ripple filter; it is called an *envelope detector.* In other detectors—notably double sideband suppressed carrier (DSBSC), single sideband suppressed carrier (SSBSC or SSB), or continuous wave (CW or morse telegraphy)—a second local oscillator, usually called a *beat frequency oscillator* (BFO), which is operating near the IF frequency, is heterodyned with the IF signal. The resultant difference signal is the recovered audio. That type of detector is called a *product detector.* Many AM receivers today have a sophisticated *synchronous detector,* rather than the simple envelope detector. Receivers that accept frequency modulation (FM) will provide some sort of frequency or phase-sensitive detector such as a ratio detector, discriminator, quadrature detector, phase-locked loop detector, and so forth.

Audio Amplifiers

The audio amplifiers are used to finish the signal processing. They also boost the output of the detector to a usable level to drive a loudspeaker or set of earphones. The audio amplifiers are sometimes used to provide additional filtering. It is quite common to find narrow-band filters to restrict audio bandwidth, or notch filters to eliminate interfering signals that make it through the IF amplifiers intact.

There are three basic areas of receiver performance that must be considered. Although interrelated, they are sufficiently different to merit individual consideration: *noise, static,* and *dynamic.* We will look at all of these areas, but first let's look at the units of measure that we will use in this series.

Units of Measure

Input Signal Voltage

Input signal level, when specified as a voltage, is typically stated in either *microvolts* (μV) or *nanovolts* (nV). The volt is simply too large a unit for practical use on radio receivers. Signal input voltage (or sometimes power level) is often used as part of the *sensitivity* specification, or as a test condition for measuring certain other performance parameters.

There are two forms of signal voltage that are used for input voltage specification: *source voltage* (V_{EMF}) and *potential difference* (V_{PD}), as illustrated in Figure 8.6. The source voltage (V_{EMF}) is the open terminal (no load) voltage of the signal generator or source, while the potential difference (V_{PD}) is the voltage that appears across the receiver antenna terminals with the load connected (the load is the receiver antenna input impedance, R_{in}). When $R_s = R_{in}$, the preferred "matched impedances" case in radio receiver systems, the value of V_{PD} is one-half V_{EMF}. This can be seen in Figure 8.6 by noting that R_S and R_{in} form a voltage divider network driven by V_{EMF}, with V_{PD} as the output.

dBm

These units refer to *decibels relative to one milliwatt (1 mW) dissipated in a 50-ohm resistive impedance* (defined as the 0-dBm reference level) and is calculated from:

$$dBm = 10\ LOG \left[\frac{P_{Watts}}{0.001}\right] \qquad [8.2]$$

or,

$$dBm = 10\ LOG\ (P_{MW}) \qquad [8.3]$$

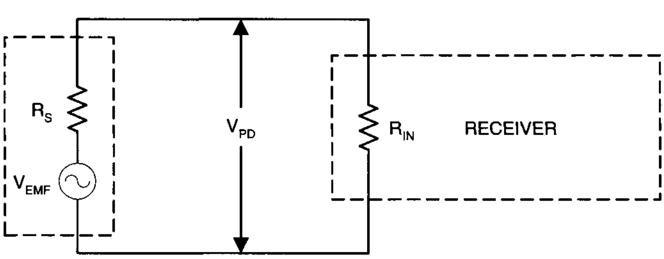

Figure 8.6
Input signal level definitions.

In the noise voltage case calculated above, 0.028 μV in 50 ohms, the power is $V^2/50$, or 5.6×10^{-10} watts, which is 5.6×10^{-7} mw. In dBm notation, this value is 10 LOG (5.6×10^{-7}), or −62.5 dBm.

dBmV

This unit is used in television receiver systems in which the system impedance is 75 ohms, rather than the 50 ohms normally used in other RF systems. It refers to the signal voltage, measured in decibels, with respect to a signal level of one millivolt (1 mV) across a 75-ohm resistance (0 dBmv). In many TV specs, 1 mV is the full quieting signal that produces no "snow" (i.e., noise) in the displayed picture. Note: 1 mV = 1,000 μV.

dBμV

This unit refers to a signal voltage, measured in decibels, relative to one microvolt (1 μV) developed across a 50-ohm resistive impedance (0 dBμV). For the case of our noise signal voltage, the level is 0.028 μV, which is the same as −31.1 dBμV. The voltage used for this measurement is usually the V_{EMF}, so to find V_{PD} divide it by two after converting dBμV to μV. To convert dBμV to dBm, merely subtract 113; that is, 100 dBμV = −13 dBm.

It requires only a little algebra to convert signal levels from one unit of measure to another. This job is sometimes necessary when a receiver manufacturer mixes methods in the same specifications sheet. In the case of dBm and dBμV, 0 dBμV is 1 μV V_{EMF}, or a V_{PD} of 0.5 μV, applied across 50 ohms, so the power dissipated is 5×10^{-15} watts, or −113 dBm.

NOISE

A radio receiver must detect signals in the presence of noise. The *signal-to-noise ratio* (SNR) is the key here, because a signal must be above the noise level before it can be successfully detected and used.

Noise comes from a number of different sources, but for the sake of this discussion we can divide them into two classes: *sources external to the receiver* and *sources internal to the receiver.* There is little one can do about the external noise sources, for they consist of natural and manmade electromagnetic signals that fall within the passband of the receiver. Figure 8.7 shows an approximation of the external noise situation from the middle of the AM broadcast band to the low end of the VHF region. One must select a receiver that can cope with external noise sources, especially if the noise sources are strong.

Some natural external noise sources are extraterrestrial. It is these signals that form the basis of radio astronomy. For example, if you aim a beam antenna at the eastern horizon prior to sunrise, a distinct rise of noise level occurs as the sun slips above the horizon, especially in the VHF region. The reverse occurs in the West at sunset, but is less dramatic, probably because atmospheric ionization decays much slower than it is generated. During World War II, it is reported that British radar operators noted an increase in received noise level any time the Milky Way was above the horizon, decreasing the range at which they could detect in-bound

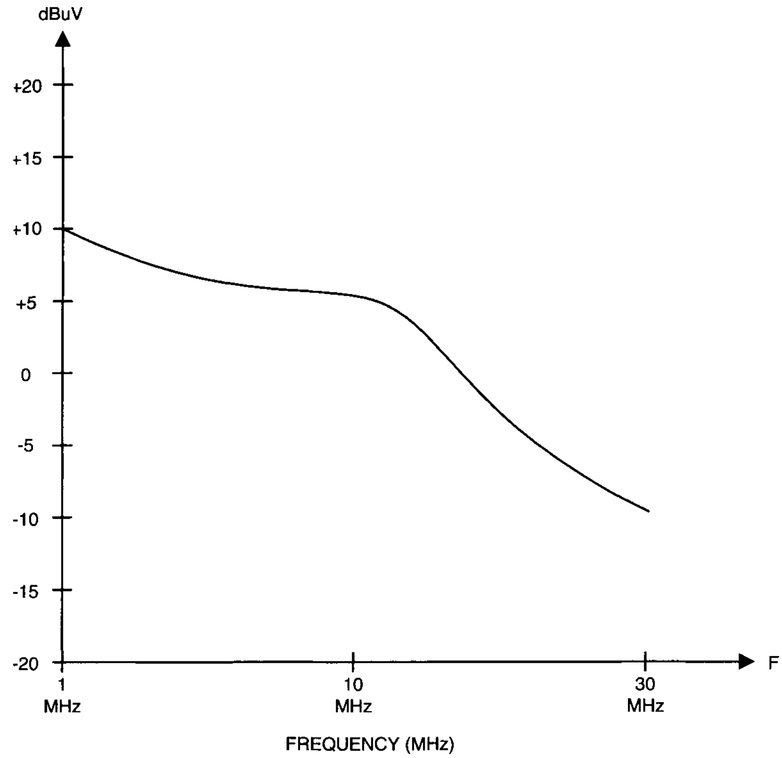

Figure 8.7 Atmospheric noise vs. frequency.

bombers. There is also some well-known, easily observed noise from the planet Jupiter in the 18- to 30-MHz band.

The receiver's internal noise sources are affected by the design of the receiver. Ideal receivers produce no noise of their own, so the output signal from the ideal receiver would contain only the noise that was present at the input along with the radio signal. But real receiver circuits produce a certain level of internal noise of their own. Even a simple fixed-value resistor is noisy. Figure 8.8A shows the equivalent circuit for an ideal, noise-free resistor, while Figure 8.8B shows a practical real-world resistor. The noise in the real-world resistor is represented in Figure 8.8B by a noise voltage source, V_n, in series with the ideal, noise-free resistance, R_i. At any temperature above *Absolute Zero* (0 °K, or about −273 °C) electrons in any material are in constant random motion. Because of the inherent randomness of that motion, however, there is no detectable current in any one direction. In other words, electron drift in any single direction is canceled over even short time periods by equal drift in the opposite direction. Electron motions are therefore statistically decorrelated. There is, however, a continuous series of random current pulses generated in the material, and those pulses are seen by the outside world as noise signals.

If a shielded 50-ohm resistor is connected across the antenna input terminals of a radio receiver, the noise level at the receiver output will increase by a predictable amount over the short-circuit noise level. Noise signals of this type are

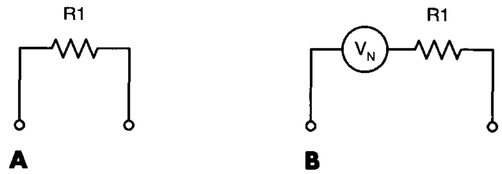

Figure 8.8
(A) Ideal input resistance is noiseless; (B) Real input resistance contributes thermal noise.

called by several names: *thermal agitation noise, thermal noise,* or *Johnson noise.* This type of noise is also called "white noise" because it has a very broadband (nearly gaussian) spectral density. The thermal noise spectrum is dominated by midfrequencies (10^4 to 10^5 Hz) and is essentially flat. The term "white noise" is a metaphor borrowed from white light, which is composed of all visible color frequencies. The expression for such noise is:

$$V_N = \sqrt{4\,K\,T\,B\,R} \qquad [8.4]$$

Where:

- V_n is the noise potential in volts (V)
- K is Boltzmann's constant (1.38×10^{-23} J/°K)
- T is the temperature in degrees Kelvin (°K), normally set to 290 or 300 °K by convention.
- R is the resistance in ohms (Ω)
- B is the bandwidth in hertz (Hz)

Table 8.1 and Figure 8.9 show noise values for a 50-ohm resistor at various bandwidths out to 5 and 10 kHz, respectively. Because different bandwidths are used for different reception modes, it is common practice to delete the bandwidth factor in Equation 8.4 and to write it in the form:

$$V_N = \sqrt{4\,K\,T\,R} \;\; V/\sqrt{Hz} \qquad [8.5]$$

With Equation 8.5 one can find the noise voltage for any particular bandwidth by taking its square root and multiplying it by the equation. This equation is essentially the solution of the previous equation normalized for a 1-Hz bandwidth.

Signal-to-Noise Ratio (SNR or S_n)

Receivers are evaluated for quality on the basis of *signal-to-noise ratio* (SNR or S/N), sometimes denoted S_n. The goal of the designer is to enhance the SNR as much as possible. Ultimately, the minimum signal level detectable at the output of an amplifier or radio receiver is that level that appears just above the noise floor

TABLE 8.1 Bandwidth and thermal noise.

Bandwith (Hz)	*Noise Voltage*
1000	2.83E-08
1500	3.46E-08
2000	4.00E-08
2500	4.47E-08
3000	4.90E-08
3500	5.29E-08
4000	5.66E-08
4500	6.00E-08
5000	6.33E-08
5500	6.63E-08
6000	6.93E-08
6500	7.21E-08
7000	7.49E-08
7500	7.75E-08
8000	8.00E-08
8500	8.25E-08
9000	8.49E-08
9500	8.72E-08
10000	8.95E-08

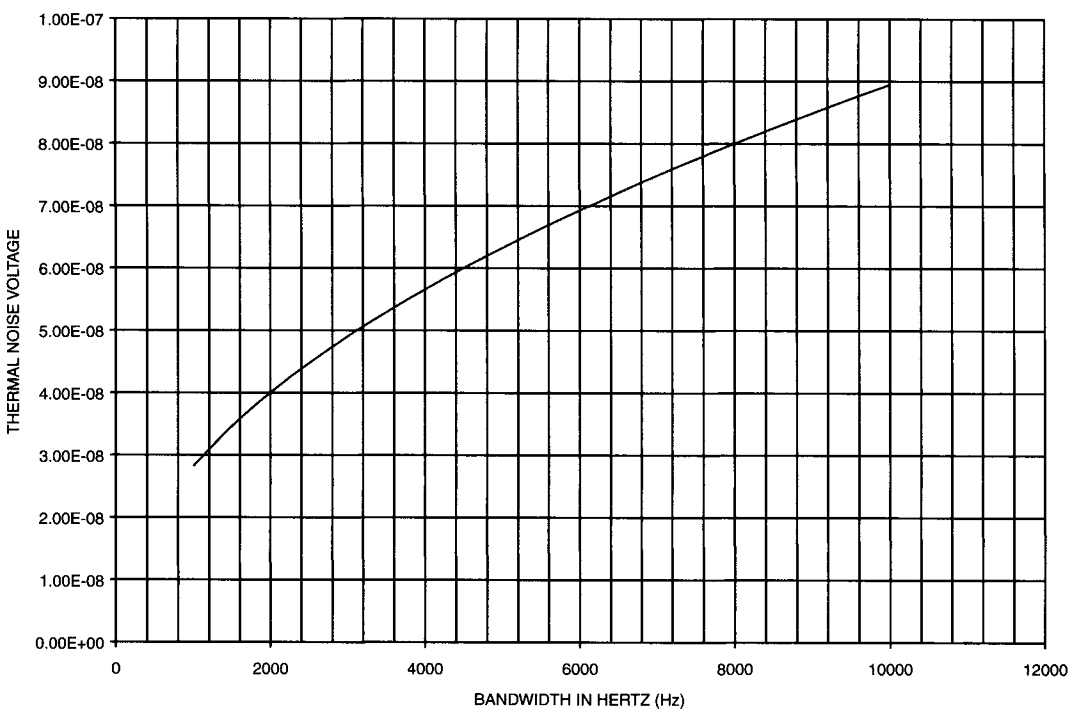

Figure 8.9 Thermal noise vs. receiver bandwidth.

level. Therefore, the lower the system noise floor, the smaller the *minimum allowable signal.*

Noise Factor, Noise Figure, and Noise Temperature

The noise performance of a receiver or amplifier can be defined in three different but related ways: *noise factor* (F_n), *noise figure* (NF), and *equivalent noise temperature* (T_e); these properties are definable as a simple ratio, decibel ratio, or Kelvin temperature, respectively.

Noise Factor (F_n)

For components such as resistors, the noise factor is the ratio of the noise produced by a real resistor to the simple thermal noise of an ideal resistor.

The noise factor of a radio receiver (or any system) is the ratio of output noise power (P_{no}) to input noise power (P_{ni}):

$$F_N = \left[\frac{P_{NO}}{P_{NI}}\right]_{T=290\,°K} \qquad [8.6]$$

In order to make comparisons easier, the noise factor is usually measured at the standard temperature (T_o) of 290 °K (standardized room temperature). In some countries, however, 299 °K or 300 °K are commonly used (the differences are negligible). It is also possible to define noise factor F_N in terms of the output and input signal-to-noise ratios:

$$F_N = \frac{S_{NI}}{S_{NO}} \qquad [8.7]$$

Where:

S_{NI} is the input signal-to-noise ratio
S_{NO} is the output signal-to-noise ratio

Noise Figure (NF)

The *noise figure* is frequently used to measure the receiver's "goodness," that is, its departure from "idealness." Thus, it is a *figure of merit.* The noise figure is the noise factor converted to decibel notation:

$$\text{N.F.} = 10\,\text{LOG}\,(F_N) \qquad [8.8]$$

Where:

N.F. is the noise figure in decibels (dB)
F_n is the noise factor
LOG refers to the system of base-10 logarithms

Noise Temperature (T_e)

The noise "temperature" is a means for specifying noise in terms of an equivalent temperature. That is, the noise level that would be produced by a resistor at that temperature (expressed in degrees Kelvin). Evaluating the noise equations shows that the noise power is directly proportional to temperature in degrees Kelvin, and also that noise power collapses to zero at the temperature of Absolute Zero (0 °K).

Note that the equivalent noise temperature T_e is *not* the physical temperature of the amplifier, but rather a theoretical construct that is an *equivalent* temperature that produces that amount of noise power in a resistor. The noise temperature is related to the noise factor by:

$$T_e = (F_N - 1)T_o \qquad [8.9]$$

and to noise figure by

$$T_e = 290\,[10^{(N.F./10)} - 1] \qquad [8.10]$$

Noise temperature is often specified for receivers and amplifiers in combination with, or in lieu of, the noise figure.

Noise in Cascade Amplifiers

A noise signal is seen by any amplifier following the noise source as a valid input signal. Each stage in the cascade chain (Figure 8.10) amplifies both signals and noise from previous stages and also contributes some additional noise of its own. Thus, in a cascade amplifier, the final stage sees an input signal that consists of the original signal and noise amplified by each successive stage, plus the noise contributed by earlier stages. The overall noise factor for a cascade amplifier can be calculated from *Friis' noise equation:*

$$F_N = F_1 + \frac{F_2 - 1}{G1} + \frac{F_3 - 1}{G1\ G2} + \cdots + \frac{F_N - 1}{G1\ G2 \ldots G_{N-1}} \qquad [8.11]$$

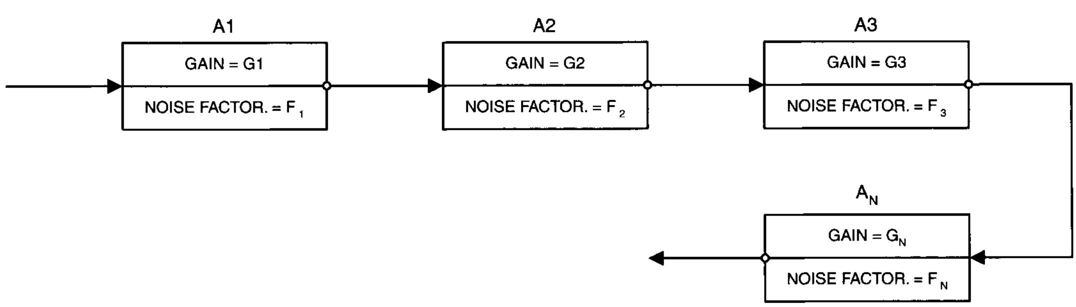

Figure 8.10 Three-stage amplifier for gain/noise figure calculations.

Where:

F_n is the overall noise factor of N stages in cascade
F_1 is the noise factor of stage-1
F_2 is the noise factor of stage-2
F_n is the noise factor of the nth stage
G1 is the gain of stage-1
G2 is the gain of stage-2
G_{n-1} is the gain of stage $(n - 1)$

As you can see from Friis' equation, the noise factor of the entire cascade chain is dominated by the noise contribution of the first stage or two. High-gain, low-noise radio astronomy RF amplifiers typically use low-noise amplifier (LNA) circuits for the first stage or two in the cascade chain. Thus, you will find an LNA at the feedpoint of a satellite receiver's dish antenna, and possibly another one at the input of the receiver module itself, but other amplifiers in the chain might be more modest (although their noise contribution cannot be ignored at radio astronomy signal levels).

The matter of signal-to-noise ratio (SNR) is sometimes treated in different ways that each attempt to crank some reality into the process. The signal-plus-noise-to-noise ratio $(S + N/N)$ is found quite often. As the ratios get higher, the SNR and $S + N/N$ converge (only about 0.5 dB difference at ratios as little as 10 dB). Still another variant is the SINAD (signal-plus-noise-plus-distortion-to-noise) ratio. The SINAD measurement takes into account most of the factors that can deteriorate reception.

Receiver Noise Floor

The *noise floor* of the receiver is a statement of the amount of noise produced by the receiver's internal circuitry and directly affects the *sensitivity* of the receiver. The noise floor is typically expressed in dBm. The noise floor specification is evaluated as follows: the more negative the better. The best receivers have noise floor numbers of less than −130 dBm, while some very good receivers offer numbers of −115 to −130 dBm.

The noise floor is directly dependent on the bandwidth used to make the measurement. Receiver advertisements usually specify the bandwidth, but note whether or not the bandwidth that produced the very good performance numbers is also the bandwidth that you will need for the mode of transmission you want to receive. If, for example, you are interested only in weak 6-kHz-wide AM signals, and the noise floor is specified for a 250 Hz CW filter, then the noise floor might be too high for your use.

Static Measures of Receiver Performance

The two principal static levels of performance for radio receivers are *sensitivity* and *selectivity*. The sensitivity refers to the level of input signal required to produce a usable output signal (variously defined). The selectivity refers to the ability of the receiver to reject adjacent channel signals (again, variously defined). Let's take a look at both of these factors. Keep in mind, however, that in modern high-performance radio receivers the static measures of performance may also be the least relevant compared with the dynamic measures.

Sensitivity

Sensitivity is a measure of the receiver's ability to pick up ("detect") signals, and is usually specified in microvolts (μV). A typical specification might be "0.5 μV sensitivity." The question to ask is: "relative to what?" The sensitivity number in microvolts is meaningless unless the test conditions are specified. For most commercial receivers, the usual test condition is the sensitivity required to produce a 10 dB signal-plus-noise-to-noise (S + N/N) ratio in the mode of interest. For example, if only one sensitivity figure is given, one must find out what bandwidth is being used. The normal bandwidth is 5 to 6 kHz for AM, 2.6 to 3 kHz for single sideband, 1.8 kHz for radioteletype, or 200 to 500 Hz for CW. Radio astronomy receiver bandwidths tend to be much wider, so sensitivity figures must be adjusted.

The amount of sensitivity improvement is seen by evaluating some simple numbers. Recall that a claim of "x − μV" sensitivity refers to some standard such as "x − μV to produce a 10 dB signal-to-noise ratio in y-Hz bandwidth." Consider the case where the main mode for a high frequency (HF) shortwave receiver is AM (for international broadcasting), the sensitivity is 1.9 μV for 10 dB SNR, and the bandwidth is 5 kHz. If the bandwidth were reduced to 2.8 kHz for SSB, then the sensitivity improves by the square root of the ratio, or $\sqrt{5/2.8}$. If the bandwidth is further reduced to 270 Hz (i.e., 0.27 kHz) for CW, then the sensitivity for 10 dB SNR is $\sqrt{5/0.27}$. The 1.9 μV AM sensitivity therefore translates to 1.42 μV for SSB and 0.44 μV for CW. If only the CW version is given, then the receiver might be made to look a whole lot better than it is, even though the typical user may never use the CW mode (see Figure 8.11).

The sensitivity differences also explain why weak SSB signals can be heard under conditions when AM signals of similar strength have disappeared into the noise, or why the CW mode has as much as a 20-dB advantage over SSB, *ceteris paribus*.

In some receivers, the difference in mode (AM, SSB, RTTY, CW, etc.) can conceivably result in sensitivity differences that are more than the differences in the bandwidths associated with the various modes. The reason is that there is sometimes a "processing gain" associated with the type of detector circuit used to demodulate the signal at the output of the IF amplifier. A simple AM envelope detector is lossy because it consists of a simple diode (1N60, 1N34, etc.) and an R-C filter (a passive circuit). Other detectors (e.g., product detector for SSB, or synchronous AM detectors) have their own signal gain, so they may produce better sensitivity numbers than the bandwidth suggests.

Another indication of sensitivity is *minimum detectable signal* (MDS), which is usually specified in dBm. This signal level is the signal power at the antenna input terminal of the receiver required to produce some standard S + N/N ratio, such as 3 dB (Figure 8.12). The MDS is actually misnamed, for there is good evidence that trained, experienced radio operators can filter out noise in their ears (or brains), and detect signals buried as much as 10 dB below the MDS. But such skills are not

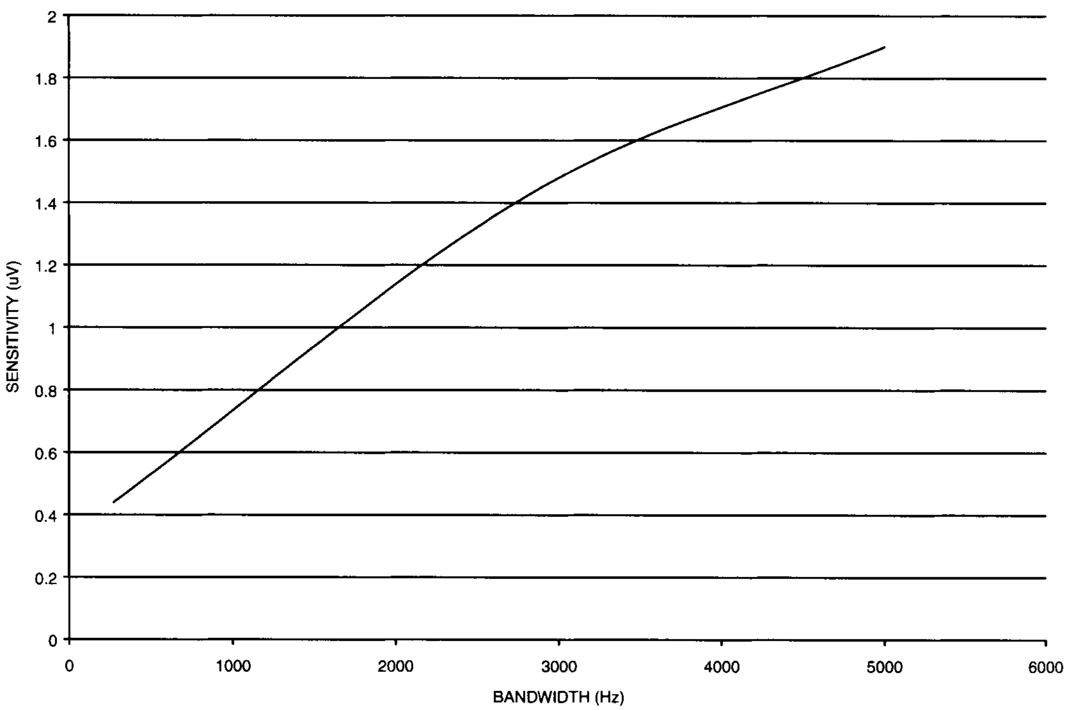

Figure 8.11 Sensitivity vs. bandwidth.

the norm. The MDS is an example of an *operational definition,* that is, a procedure that yields coherent and usable results: MDS is 3 dB above the noise floor.

In radar receivers, the MDS is usually described in terms of a single-pulse return and a specified $S + N/N$ ratio. Also, in radar and other pulse receivers, the sensitivity can be improved by integrating multiple pulses. If N return pulses are integrated, then the sensitivity is improved by a factor of N if coherent detection is used, and $\sqrt{N}$ if noncoherent detection is used.

The noise floor cannot ever be zero because of thermal agitation in resistances and impedances. If the receiver antenna input impedance is 50 ohms, then there will be a −174 dBm/Hz thermal noise level present. As bandwidth increases, then the noise increases proportionally. For example, in an SSB receiver that has a 3,000 Hz IF bandwidth, the noise level will be 10 LOG (3,000) = 34.8 dB higher, or (−174 dBm) + 34.8 dB = −139 dBm.

Modulated signals represent a special case. For those sensitivities, it is common to specify the conditions under which the measurement is made. For example, in AM receivers the sensitivity to achieve 10 dB SNR is measured with the input signal modulated 30% by a 400 or 1,000 Hz sinusoidal tone.

An alternate method is sometimes used for AM sensitivity measurements, especially in servicing consumer radio receivers (where SNR may be a little hard to measure with the equipment normally available to technicians who work on those radios). This is the "standard output conditions" method. Some manuals will specify the audio signal power or audio signal voltage at some critical point, when the

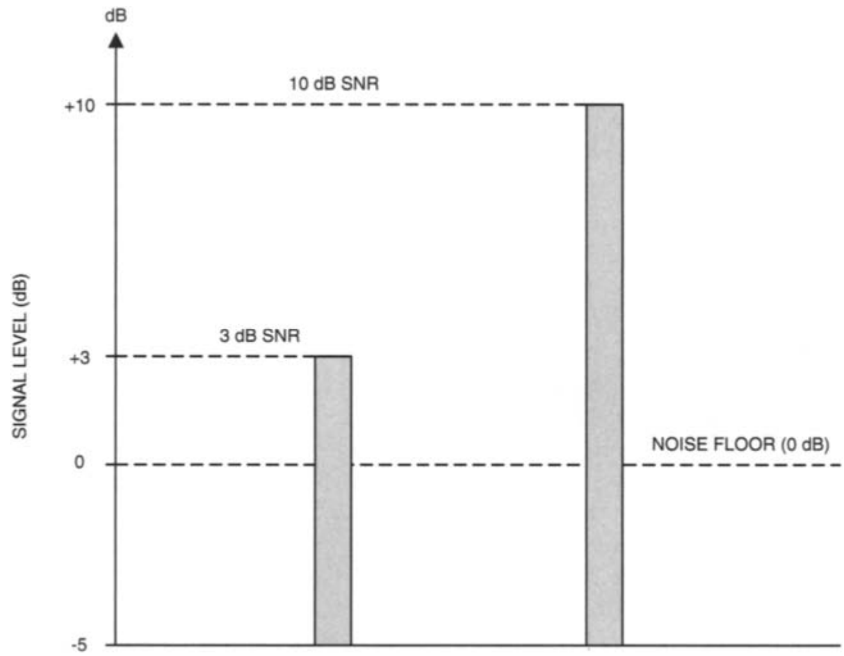

Figure 8.12 Definitions of sensitivity.

30% modulated RF carrier is present. In one automobile radio receiver, the sensitivity was specified as "X μV to produce 400 mW across 8 ohm resistive load substituted for the loudspeaker when the signal generator is modulated 30% with a 400 Hz audio tone." The cryptic note on the schematic showed an output sine wave across the loudspeaker with the label "400 mW in 8Ω (1.79 volts), @30% mod. 400 Hz, 1 μV RF."

The sensitivity is sometimes measured essentially the same way, but the signal levels will specify the voltage level that will appear at the top of the volume control, or output of the detector-filter, when the standard signal is applied. Thus, there are two ways seen for specifying AM sensitivity: *10 dB SNR* and *standard output conditions.*

There are also two ways to specify FM receiver sensitivity. The first is the 10 dB SNR method discussed above, that is, the number of microvolts of signal at the input terminals required to produce a 10 dB SNR when the carrier is modulated by a standard amount. The measure of FM modulation is *deviation* expressed in kilohertz. Sometimes, the full deviation for that class of receiver is used, while for others a value that is 25 to 35% of full deviation is specified.

The second way to measure FM sensitivity is the level of signal required to reduce the no-signal noise level by 20 dB. This is the "20-dB quieting sensitivity of the receiver." If you tune between signals on an FM receiver, you will hear a loud "hiss" signal, especially in the VHF/UHF bands. Some of that noise is externally generated, while some is internally generated. When an FM signal appears in the passband, that hiss is suppressed, even if the FM carrier is unmodulated. The *quieting sensitivity* of an FM receiver is a statement of the number of microvolts required to produce some standard quieting level, usually 20 dB.

Pulse receivers, such as radar and pulse communications units, often use the *tangential sensitivity* as the measure of performance, which is the amplitude of pulse signal required to raise the noise level by its own RMS amplitude (Figure 8.13).

Selectivity

Although no receiver specification is unimportant, if one had to choose between sensitivity and selectivity, the proper choice most of the time would be to take selectivity.

Selectivity is the measure of a receiver's ability to reject adjacent channel interference. Or put another way, it's the ability to reject interference from signals on frequencies close to the desired signal frequency.

In order to understand selectivity requirements, one must first understand a little bit of the nature of radio signals. An unmodulated radio carrier theoretically has an infinitesimal (near-zero) bandwidth (although all real unmodulated carriers have a very narrow, but nonzero, bandwidth because they are modulated by noise and other artifacts). As soon as the radio signal is modulated to carry information, however, the bandwidth spreads. Even an on/off telegraphy (CW) or pulse signal spreads on either side of the carrier frequency an amount that is dependent on the sending speed and the shape of the keying waveform.

An AM signal spreads out an amount equal to twice the highest audio modulating frequencies. For example, a communications AM transmitter will have audio components from 300 to 3,000 Hz, so the AM waveform will occupy a spectrum that is equal to the carrier frequency (F) plus or minus the audio bandwidth (F ± 3,000 Hz in the case cited). An FM carrier spreads out according to the *deviation*. For example, a narrow-band FM landmobile transmitter with 5 kHz deviation spreads out ±5 kHz, while FM broadcast transmitters spread out ±75 kHz

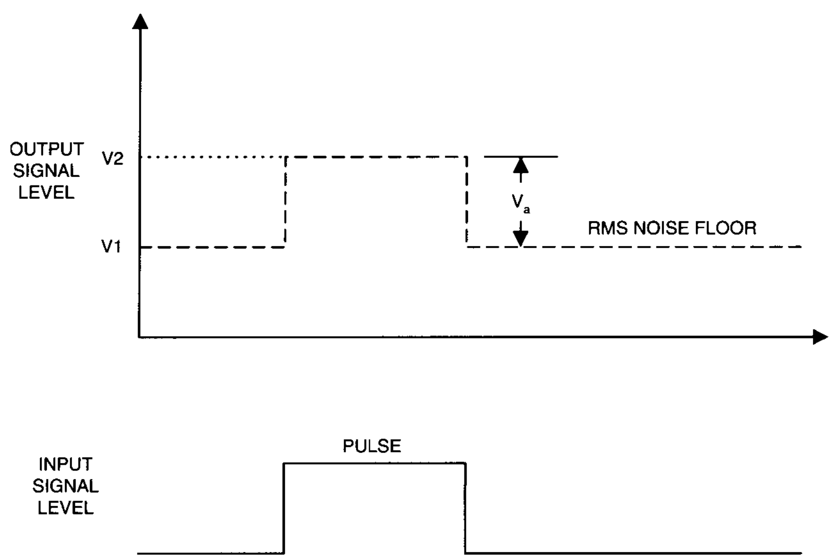

Figure 8.13 Tangential sensitivity.

An implication of the fact that radio signals have bandwidth is that the receiver must have sufficient bandwidth to recover the entire signal. Otherwise, information may be lost and the output is distorted. On the other hand, allowing too much bandwidth increases the noise picked up by the receiver and thereby deteriorates the SNR. The goal of the selectivity system of the receiver is to match the bandwidth of the receiver to that of the signal. That is why receivers will use 270 or 500 Hz bandwidth for CW, 2 to 3 kHz for SSB, and 4 to 6 kHz for AM signals. They allow you to match the receiver bandwidth to the transmission type.

The selectivity of a receiver has a number of aspects that must be considered: *front-end bandwidth, IF bandwidth, IF shape factor,* and the *ultimate* (distant frequency) *rejection.*

Front-End Bandwidth

The "front-end" of a modern superheterodyne radio receiver is the circuitry between the antenna input terminal and the output of the first mixer stage. The reason why front-end selectivity is important is to keep out-of-band signals from afflicting the receiver. For example, AM broadcast band transmitters located nearby can easily overload a poorly designed shortwave or VLF/LF receiver. Even if these signals are not heard by the operator (as they often are), they can desensitize a receiver, or create harmonics and intermodulation products that show up as "birdies" or other types of interference on the receiver. Strong local signals can take a lot of the receiver's dynamic range and thereby make it harder to hear weak signals.

In some "crystal video" microwave receivers, that front-end might be wide open without any selectivity at all, but in nearly all other receivers there will be some form of frequency selection present.

Two forms of frequency selection are typically found. A designer may choose to use only one of them in a design. Alternatively, both might be used in the design, but separately (operator selection). Or finally, both might be used together. These forms can be called the *resonant frequency filter* (Figure 8.14A) and *bandpass filter* (Figure 8.14B) approaches.

The resonant frequency approach uses L-C elements tuned to the desired frequency to select which RF signals reach the mixer. In some receivers, these L-C elements are designed to track with the local oscillator that sets the operating frequency. That is why you see two-section variable capacitors for AM broadcast receivers with two different capacitance ranges for the two sections. One section tunes the LO and the other section tunes the tracking RF input. In other designs, a separate tuning knob ("preselector" or "antenna") is used.

The other approach uses a suboctave bandpass filter to admit only a portion of the RF spectrum into the front-end. For example, a shortwave receiver that is designed to take the HF spectrum in 1-MHz pieces may have an array of RF input bandpass filters that are each 1 MHz wide (e.g., 9 to 10 MHz).

In addition to the reasons cited above, front-end selectivity also helps improve a receiver's *image rejection* and *1st IF Rejection* capabilities.

Image Rejection

An *image* in a superheterodyne receiver is a signal that appears at twice the IF distant from the desired RF signal; it is also located on the opposite side of the LO frequency from the desired RF signal. In Figure 8.15, a superheterodyne operates with

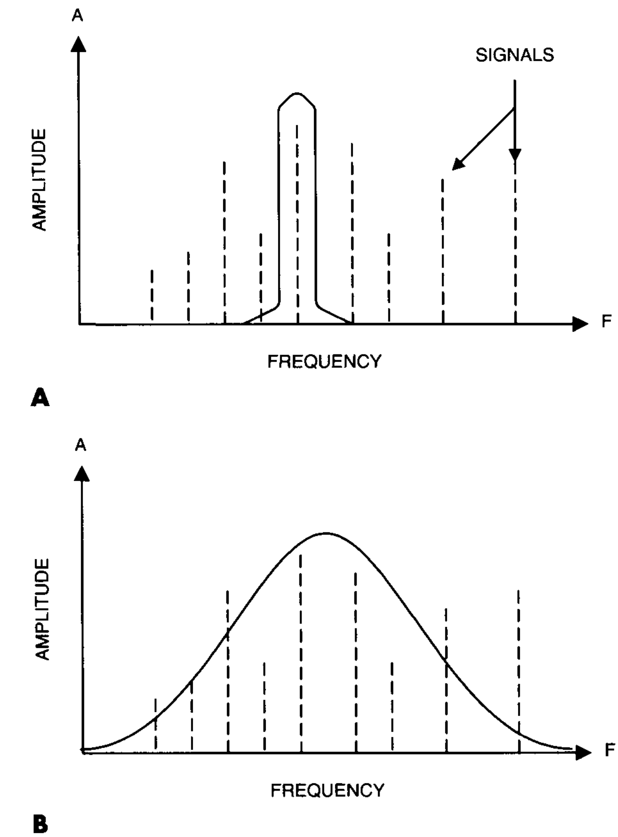

Figure 8.14
Selectivity defined: (A) Proper selectivity eliminates stray signals; (B) Poor selectivity admits unwanted stray signals.

a 455 kHz (i.e., 0.455 MHz) IF, and is turned to 24.0 MHz (F_{RF}). Because this receiver uses low-side LO injection, the LO frequency F_{LO} is 24.0 − 0.455, or 23.545 MHz. If a signal appears at twice the IF below the RF (i.e., 910 kHz below F_{RF}), and reaches the mixer, then it too has a difference frequency of 455 kHz, so it will pass right through the IF filtering as a valid signal. The image rejection specification tells how well this image frequency is suppressed. Normally, anything over about 70 dB is considered good.

Tactics to reduce image response vary with the design of the receiver. The best approach, at design time, is to select an IF frequency that is high enough that the image frequency will fall outside the passband of the receiver front-end. Some modern HF receivers use an IF of 8.83 MHz, 9 MHz, 10.7 MHz, or something similar, and for image rejection these frequencies are considerably better than 455 kHz receivers in the higher HF bands. However, a common trend is to do *double conversion.* In most such designs, the first IF frequency is considerably higher than the RF, being in the range 35 to 60 MHz (50 MHz is common in HF receivers, 70 MHz in microwave receivers).

The high IF makes it possible to suppress the VHF images with a simple low-pass filter. If the 24.0 MHz signal (above) were first up, converted to 50 MHz (74 MHz LO), for example, the image would be at 124 MHz. The second conversion brings the IF down to one of the frequencies mentioned above, or even

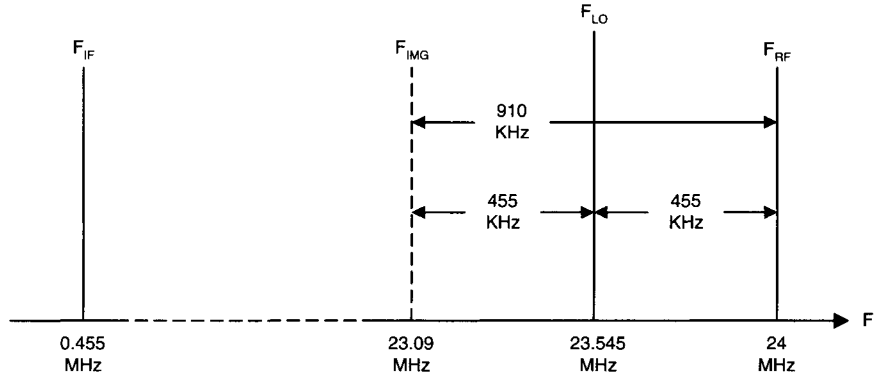

Figure 8.15 Image frequency defined.

455 kHz. The lower frequencies are preferable to 50 MHz for bandwidth selectivity reasons because good quality crystal, ceramic, or mechanical filters in those ranges filters are easily available.

First IF Rejection

The first IF rejection specification refers to how well a receiver rejects radio signals operating on the receiver's first IF frequency. For example, if your receiver has a first IF of 70 MHz, it must be able to reject radio signals operating on that frequency when the receiver is tuned to a different frequency. Although the shielding of the receiver is also an issue with respect to this performance, the front-end selectivity affects how well the receiver performs against first IF signals.

If there is no front-end selectivity to discriminate against signals at the IF frequency, then they arrive at the input of the mixer unimpeded. Depending on the design of the mixer, they then may pass directly through to the high-gain IF amplifiers and be heard in the receiver output.

IF Bandwidth

Most of the selectivity of the receiver is provided by the filtering in the IF amplifier section. The filtering might be L-C filters (especially if the principal IF is a low frequency like 50 kHz), a ceramic resonator, a crystal filter, or a mechanical filter. Of these, the mechanical filter is usually regarded as best for narrow bandwidths, with the crystal filter and ceramic filter coming in next.

The IF bandwidth is expressed in kilohertz (kHz), and is measured from the points on the IF frequency response curve where gain drops off −3 dB from the mid-band value (Figure 8.16). This is why you will sometimes see selectivity referred to in terms such as "6 kHz between −3 dB points."

The IF bandwidth must be matched to the bandwidth of the received signal for best performance. If a too-wide bandwidth is selected, then the received signal will be noisy, and SNR deteriorates. If the bandwidth is too narrow, then you might experience difficulties recovering all of the information that was transmitted. For example, an AM broadcast band radio signal has audio components to 5 kHz, so

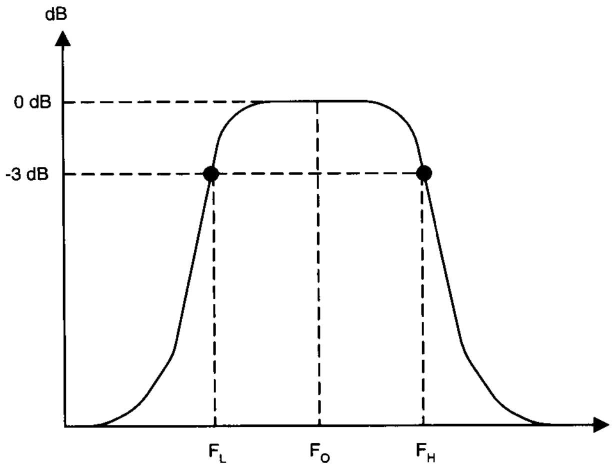

Figure 8.16
Selectivity bandwidth defined.

the signal occupies up to 10 kHz of spectrum space (F ± 5 kHz). If a 2.8 kHz SSB IF filter is selected, then it will tend to sound "mushy" and distorted.

IF Passband Shape Factor

The shape factor is a measure of the steepness of the receiver's IF passband and is taken by measuring the ratio of the bandwidth at −6 dB to the bandwidth at −60 dB. The general rule is that the closer these numbers are to each other, the better the receiver. Anything in the 1:1.5 to 1:1.9 region can be considered high quality, while anything worse than 1:3 is not worth looking at for "serious" receiver uses. If the numbers are between 1:1.9 and 1:3, then the receiver could be regarded as being middling, but useful.

The importance of shape factor is that it modifies the notion of bandwidth. The cited bandwidth (e.g., 2.8 kHz for SSB) does not take into account the effects of strong signals that are just beyond those limits. Such signals can easily "punch through" the IF selectivity if the IF passband "skirts" are not steep. After all, the steeper they are, the closer a strong signal can be without messing up the receiver's operation. Thus, selecting a receiver with a shape factor as close to the 1:1 ideal as possible will result in a more usable radio.

Distant Frequency ("Ultimate") Rejection

This specification tells something about the receiver's ability to reject very strong signals that are located well outside the receiver's IF passband. This number is stated in negative decibels (−dB), and the higher the number the better. An excellent receiver will have values in the −60 to −90 dB range, a middling receiver will have numbers in the −45 to −60 dB range, and a terrible receiver will be −44 or worse.

Stability

The *stability* specification measures how much the receiver frequency drifts as time elapses or temperature changes. The LO drift sets the overall stability of the receiver. This specification is usually given in terms of *short-term drift* and *long-term* *drift* (e.g., from crystal aging). The short-term drift is important in daily operation, while the long-term drift ultimately affects general dial calibration.

If the receiver is VFO controlled, or uses partial frequency synthesis (which combines VFO with crystal oscillators), then the stability is dominated by the VFO stability. In fully synthesized receivers, the stability is governed by the master reference crystal oscillator. If either an *oven-controlled crystal oscillator* (OCXO) or a *temperature-compensated crystal oscillator* (TCXO) is used for the master reference, then stability on the order of 1 part in $10^8/°C$ are achievable. For most users, the short-term stability is what is most important, especially when tuning SSB, ECSS, or RTTY signals. A common spec for a good receiver will be 50 Hz/hour after a 3-hour warm-up, or 100 Hz/hour after a 15-minute warm-up. The smaller the drift the better the receiver.

The foundation of good stability is established at design time. The local oscillator, or VFO portion of a synthesizer, must be operated in a cool, temperature-stable location within the equipment, and it must have the correct type of components. Capacitor temperature coefficients are often selected in order to cancel out temperature related drift in inductance values.

Post-design time changes can also help, but these are less likely to be possible today than in the past. The chief cause of drift problems is heat. In the days of valve oscillators, the heat of the valves produced lots of heat that created drift.

A related phenomenon seen on low-cost receivers, or certain home-brew receivers of doubtful merit, is mechanical frequency shifts. Although not seen on most modern receivers (even some very cheap designs), it was once a serious problem on less costly models. This problem is usually seen on VFO-controlled receivers in which vibration to the receiver cabinet imparts movement to either the inductor (L) or capacitor (C) element in an L-C VFO. Mechanically stabilizing these components will work wonders.

AGC Range and Threshold

Modern communications receivers must be able to handle signals over the range of about 1,000,000:1. Tuning across a band occupied by signals of wildly varying strengths is hard on the ears and hard on the receiver's performance. As a result, most modern receivers have an *automatic gain control* (AGC) circuit that smoothes out these changes. The AGC will reduce gain for strong signals and increase it for weak signals (AGC can be turned off on most HF communications receivers). The AGC range is the change of input signal (in dBμV) from some reference level (e.g., 1 μV_{EMF}) to the input level that produces a 2 dB change in output level. Ranges of 90 to 110 dB are commonly seen.

The AGC threshold is the signal level at which the AGC begins to operate. If set too low, then the receiver gain will respond to noise and irritate the user. If set too high, then the user will experience irritating shifts of output level as the band is tuned. AGC thresholds of 0.7 to 2.5 μV are common on decent receivers, with the better receivers being in the 0.7 to 1 μV range.

Another AGC specification sometimes seen deals with the speed of the AGC. Although sometimes specified in milliseconds, it is also frequently specified in subjective terms like "fast" and "slow." This specification refers to how fast the AGC responds to changes in signal strength. If set too fast, then rapidly keyed signals (e.g., CW) or noise transients will cause unnervingly large shifts in receiver gain. If set too slow, then the receiver might as well not have an AGC. Many receivers provide two or more selections in order to accommodate different types of signals.

Dynamic Performance

The dynamic performance specifications of a radio receiver are those that deal with how the receiver performs in the presence of very strong signals, either co-channel or adjacent channel. Until about the 1960s, dynamic performance was somewhat less important than static performance for most users. However, today the role of dynamic performance is probably far more critical than simplistic static performance because of crowded band conditions.

There are at least two reasons for this change in outlook. First, in the 1960s receiver designs evolved from valves to solid-state. The new solid-state amplifiers were somewhat easier to drive into nonlinearity than tube designs. Second, there has been a tremendous increase in radio frequency signals on the air. There are far more transmitting stations than ever before, and there are far more sources of electromagnetic interference (EMI—pollution of the airwaves) than in prior decades. With the advent of new and expanded wireless services available to an ever-widening market, the situation can only worsen. For this reason, it is now necessary to pay more attention to the dynamic performance of receivers than in the past.

Intermodulation Products

Understanding the dynamic performance of the receiver requires knowledge of *intermodulation products* (IP) and how they affect receiver operation. Whenever two signals are mixed together in a nonlinear circuit, a number of products are created according to $mF1 \pm nF2$, where m and n are either integers or zero. Mixing can occur in either the mixer stage of a receiver front-end, or in the RF amplifier (or any outboard preamplifiers used ahead of the receiver), if the RF amplifier is overdriven by a strong signal.

It is also theoretically possible for corrosion on antenna connections, or even rusted antenna screw terminals, to create IPs under certain circumstances. One even hears of alleged cases where a rusty downspout on a house rain gutter caused reradiated mixed signals. However, all such cases that I've heard of have that distant third- or fourth-party quality ("I know a guy whose best friend's brother-in-law saw . . . ") that suggests the profound apocryphal quality of these reports. I know of no first-hand accounts verified by a technically competent person.

The spurious IP signals are shown graphically in Figure 8.17. Given input signal frequencies of F1 and F2, the main IPs are:

Second-order:	$F1 \pm F2$
Third-order:	$2F1 \pm F2$
	$2F2 \pm F1$
Fifth-order:	$3F1 \pm 2F2$
	$3F2 \pm 2F1$

The second-order and third-order products are those normally specified in a receiver because they tend to be the strongest. In general, even-order IMD prod-

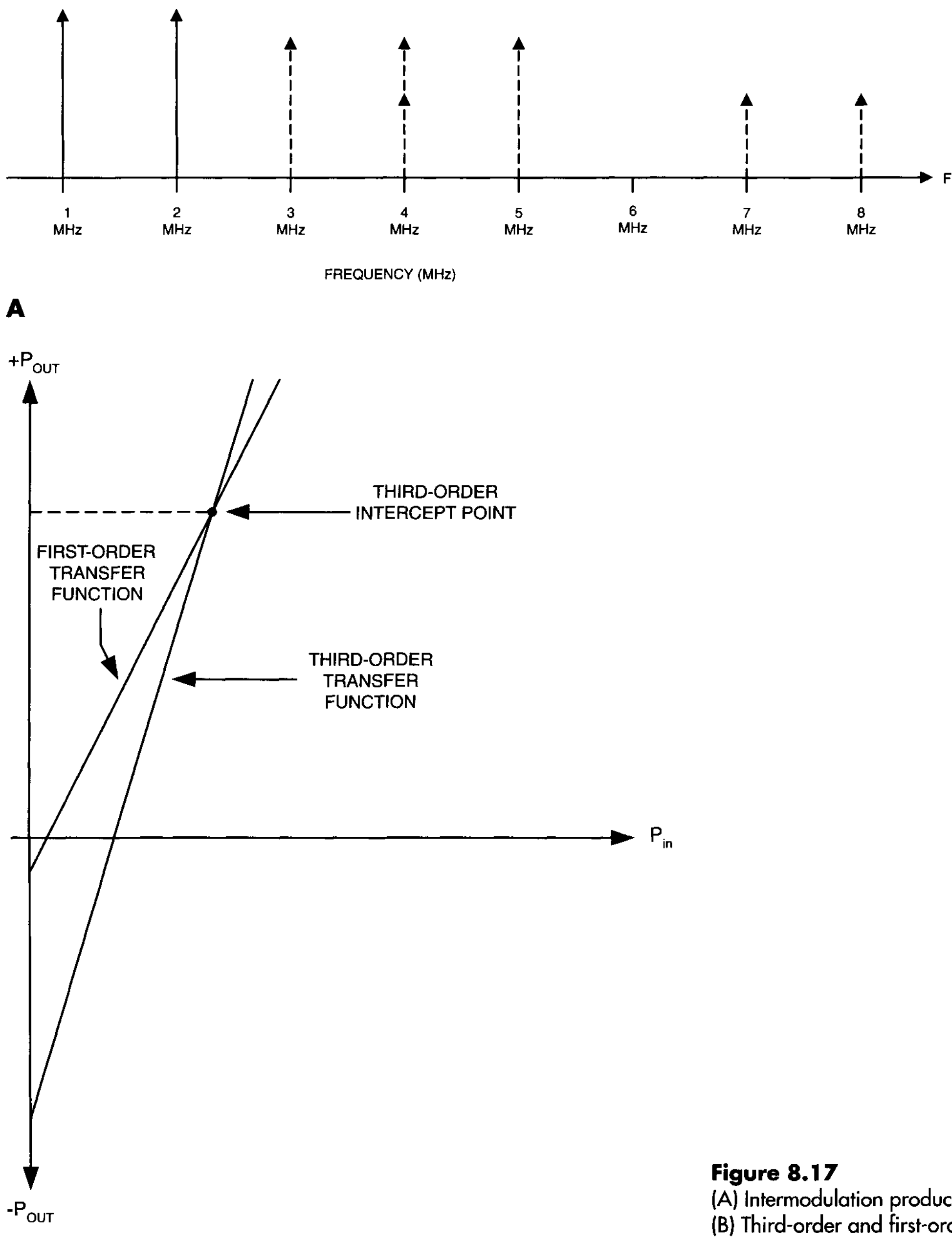

Figure 8.17
(A) Intermodulation products;
(B) Third-order and first-order products compared.

ucts (2, 4, etc.) tend to be less of a problem because they can often be ameliorated by using external filtering ahead of the receiver's antenna input. Prefiltering ("preselection") tends to reduce the amplitude of out-of-channel interfering signals, reducing the second-order products within the channel. Third-order IMD products are more important because they tend to reflect on the receiver's dynamic range, as well as its ability to handle strong signals. The third-order products are usually not easily influenced by external filtering.

When an amplifier or receiver is overdriven, the second-order content of the output signal increases as the square of the input signal level, while the third-order responses increase as the cube of the input signal level. When expressed in dB the third-order transfer function has a slope three times that of the first-order transfer function (Figure 8.17B).

Consider the case where two HF signals, F1 = 10 MHz and F2 = 15 MHz, are mixed together. The second-order IPs are 5 and 25 MHz; the third-order IPs are 5, 20, 35, and 40 MHz; and the fifth-order IPs are 0, 25, 60, and 65 MHz. If any of these are inside the passband of the receiver, then they can cause problems. One such problem is the emersion of "phantom" signals at the IP frequencies. This effect is seen often when two strong signals (F1 and F2) exist and can affect the front-end of the receiver, and one of the IPs falls close to a desired signal frequency, F_d. If the receiver were tuned to 5 MHz, for example, a spurious signal would be found from the F1-F2 pair given above.

Another example is seen from strong in-band, adjacent channel signals. Consider a case where the receiver is tuned to a station at 9610 kHz, and there are also very strong signals at 9600 kHz and 9605 kHz. The near (in-band) IP products are:

Third-order:	9595 kHz ($\Delta F = 15$ kHz)
	9610 kHz ($\Delta F = 0$ kHz)[ON CHANNEL!]
Fifth-order:	9590 kHz ($\Delta F = 20$ kHz)
	9615 kHz ($\Delta F = 5$ kHz)

Note that one third-order product is on the same frequency as the desired signal and could easily cause interference if the amplitude is sufficiently high. Other third- and fifth-order products may be within the range where interference could occur, especially on receivers with wide bandwidths.

The IP orders are theoretically infinite because there are no bounds on either m or n. However, in practical terms, because each successively higher order IP is reduced in amplitude compared with its next lower-order mate, only the second-order, third-order, and fifth-order products usually assume any importance. Indeed, only the third-order is normally used in receiver specification sheets.

−1 dB Compression Point

An amplifier produces an output signal that has a higher amplitude than the input signal. The transfer function of the amplifier (indeed, any circuit with output and input) is the ratio OUT/IN, so for the power amplification of a receiver RF amplifier it is P_o/P_{in} (or, in terms of voltage, V_o/V_{in}). Any real amplifier will saturate given a strong enough input signal (see Figure 8.18). The sloping dotted line in Figure 8.18 represents the theoretical output level for all values of input signal (the slope of the line represents the gain of the amplifier). As the amplifier saturates (solid line), however, the actual gain begins to depart from the theoretical at some level of input signal (P_{in1}). The −1 dB compression point is that output level at which the actual gain departs from the theoretical gain by −1 dB.

The −1 dB compression point is important when considering either the RF amplifier ahead of the mixer (if any), or any outboard preamplifiers that are used.

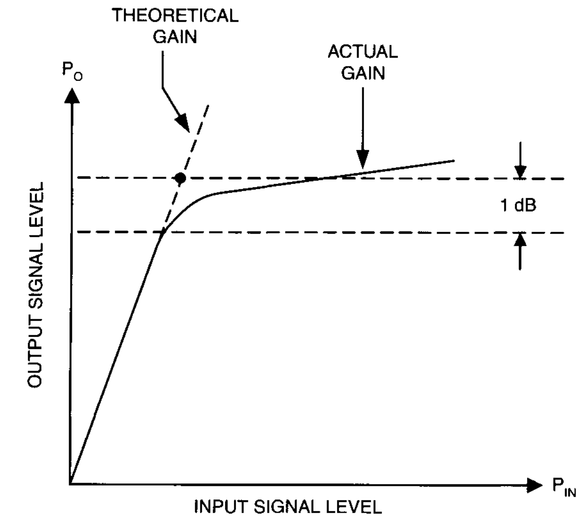

Figure 8.18
1-dB compression defined.

The −1 dB compression point is the point at which intermodulation products begin to emerge as a serious problem. It is also the case that harmonics are generated when an amplifier goes into compression. A sine wave is a "pure" signal because it has no harmonics (all other waveshapes have a fundamental frequency plus harmonic frequencies). When a sine wave is distorted, however, harmonics arise. The effect of the compression phenomenon is to distort the signal by clipping the peaks, thus raising the harmonics and intermodulation distortion products.

Third-Order Intercept Point

It can be claimed that the third-order intercept point (TOIP) is the single most important specification of a receiver's dynamic performance because it predicts the performance as regards intermodulation, crossmodulation, and blocking desensitization.

Third-order (and higher) intermodulation products (IP) are normally very weak and do not exceed the receiver noise floor when the receiver is operating in the linear region. But as input-signal levels increase, forcing the front-end of the receiver toward the saturated nonlinear region, the IP emerges from the noise (Figure 8.19) and begins to cause problems. When this happens, new spurious signals appear on the band and self-generated interference begins to arise.

Figure 8.20 shows a plot of the output signal versus fundamental input signal. Note the output compression effect that was seen earlier in Figure 8.18. The sloping dotted gain line continuing above the saturation region shows the theoretical output that would be produced if the gain did not clip. It is the nature of third-order products in the output signal to emerge from the noise at a certain input level, and to increase as the cube of the input level. Thus, the slope of the third-order line increases 3 dB for every 1-dB increase in response to the fundamental signal. Although the output response of the third-order line saturates similarly to that of the fundamental signal, the gain line can be continued to a point where it

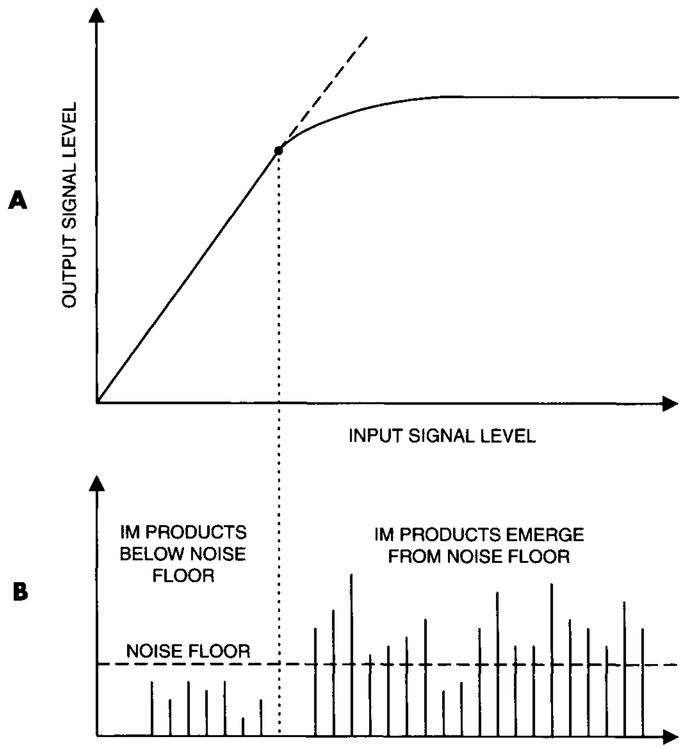

Figure 8.19
Operating above compression point causes IM products to emerge from the noise floor. (A) Output-vs.-input signal levels; (B) Distortion products and noise floor.

intersects the gain line of the fundamental signal. This point is the *third-order intercept point* (TOIP).

Interestingly enough, one receiver feature that can help reduce IP levels back down under the noise is the use of a *front-end attenuator,* also known as an *input attenuator.* Even a few dB of input attenuation is often enough to cause the IPs to drop back into the noise, while afflicting the desired signals only a small amount.

Other effects that reduce the overload caused by a strong signal also help. The apparent third-order performance of a receiver can improve dramatically when either a lower-gain antenna or front-end attenuator is used. Inserting a 6-dB barrel attenuator in the input ("antenna") line can eliminate or reduce the IP products, showing just the actual signals. Rotating a directional antenna away from the direction of the interfering signal will also accomplish this effect in many cases.

Preamplifiers are popular receiver accessories, but they can often reduce rather than enhance performance. Two problems commonly occur (assuming the preamp is a low-noise device; if not, there are three). The best known problem is that the preamp amplifies noise as much as signals, and while it makes the signal louder it also makes the noise louder by the same amount. Since it is the signal-to-noise ratio that is important, one does not improve the situation. Indeed, if the preamplifier is itself noisy, it will deteriorate the SNR. The other problem is less well known, but potentially more devastating. If the increased signal levels applied to the receiver drive the receiver in a nonlinearly way, then IPs begin to emerge. One source reported an event where transmissions were being heard at several spots on

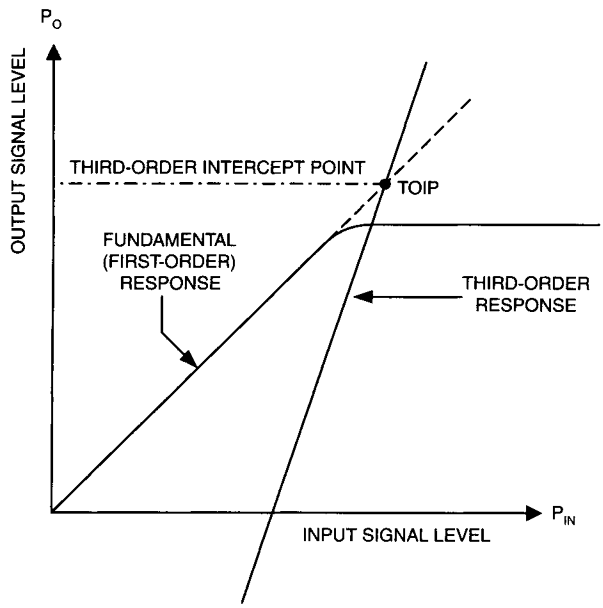

Figure 8.20
Third-order intercept point (TOIP or IP3) defined.

the band. It was discovered that the complainant was using two preamplifiers in cascade to achieve higher gain, and that disconnecting them caused the spurs to evaporate. This was clearly a case of a preamplifier deteriorating, rather than improving, a receiver's performance.

When evaluating receivers, a TOIP of +5 to +20 dBm is excellent performance, while up to +27 dBm is relatively easily achievable, and +35 dBm has been achieved with good design; anything greater than +50 dBm is close to miraculous (but attainable). Receivers are still regarded as good performers in the 0 to +5 dBm range, and middling performers in the −10 to 0 dBm range. Anything below −10 dBm is not a very wonderful a machine to own. A general rule is to buy the best third-order intercept performance that you can afford, especially if there are strong signal sources in your vicinity.

Dynamic Range

The *dynamic range* of a radio receiver is the range from the minimum discernible signal to the maximum allowable signal (measured in decibels, dB). While this simplistic definition is easy to understand conceptually, in the concrete it is a little more complex. Several definitions of dynamic range are used.

One definition of dynamic range is that it is the input signal difference between the sensitivity figure (e.g., 0.5 μV for 10 dB S + N/N) and the level that drives the receiver far enough into saturation to create a certain amount of distortion in the output. This definition was common on consumer broadcast band receivers at one time (especially automobile radios, where dynamic range was somewhat more important due to mobility). A related definition takes the range as the distance in dB from the sensitivity level and the −1 dB compression point. Still another definition, the *blocking dynamic range*, is the range of signals from the sensitivity level to the blocking level (see below).

A problem with the above definitions is that they represent single-signal cases, so they do not address the receiver's dynamic characteristics. Dye (1993) provides both a "loose" and a more formal definition that is somewhat more useful and is at least standardized. The loose version is that dynamic range is the range of signals over which dynamic effects (e.g., intermodulation) do not exceed the noise floor of the receiver. Dye's recommendation for HF receivers is that the dynamic range is two-thirds the difference between the noise floor and the third-order intercept point in a 3-kHz bandwidth. Dye also states an alternative: dynamic range is the difference between the fundamental response input signal level and the third-order intercept point along the noise floor, measured with a 3-kHz bandwidth. For practical reasons, this measurement is sometimes made not at the actual noise floor (which is sometimes hard to ascertain), but rather at 3 dB above the noise floor.

There is a measurement procedure that produces similar results (the same method is used for many amateur radio magazine product reviews). Two equal-strength signals are input to the receiver at the same time. The frequency difference has traditionally been 20 kHz for HF and 30 to 50 kHz for VHF receivers (modern band crowding may indicate a need for a specification at 5 kHz separation on HF). The amplitudes of these signals are raised until the third-order distortion products are raised to the noise floor level.

For 20-kHz spacing, using the two-signal approach, anything over 90 dB is an excellent receiver, while anything over 80 dB is at least decent.

The difference between the single-signal and two-signal (dynamic) performance is not merely an academic exercise. Besides the fact that the same receiver can show as much as a 40-dB difference between the two measures (favoring the single-signal measurement), the most severe effects of poor dynamic range show up most in the dynamic performance.

Blocking

The blocking specification refers to the ability of the receiver to withstand very strong off-tune signals that are at least 20 kHz away from the desired signal, although some use 100-kHz separation. Very strong signals appearing at the input terminals of a receiver may desensitize the receiver, that is, reduce the apparent strength of desired signals over what they would be if the interfering signal were not present.

Figure 8.21 shows the blocking behavior. When a strong signal is present, it takes up more of the receiver's resources than normal, so there is not enough of the output power budget to accommodate the weaker desired signals. But if the strong undesired signal is turned off, then the weaker signals receive a full measure of the unit's power budget.

The usual way to measure blocking behavior is to input two signals: a desired signal at 60 dBμV, and another signal 20 (or 100) kHz away at a much stronger level. The strong signal is increased to the point where blocking desensitization causes a 3-dB drop in the output level of the desired signal. A good receiver will show $\geq$90 dBμV, with many being considerably better. An interesting note about

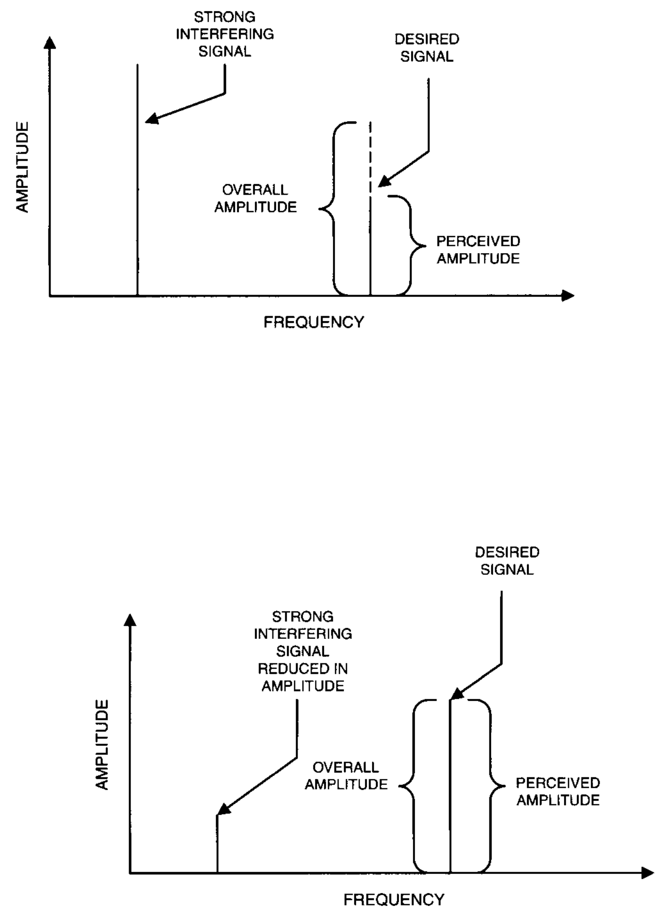

Figure 8.21
Desensitization due to strong interfering signal can be overcome if the signal levels are reduced below compression point.

modern receivers is that their blocking performance is so good that it is often necessary to specify the input level difference (dB) that causes a 1-dB drop, rather than a 3-dB drop, of the desired signal's amplitude.

The phenomenon of blocking leads us to an effect that is often seen as paradoxical on first blush. Many receivers are equipped with front-end attenuators that permit fixed attenuation values of 1 dB, 3 dB, 6 dB, 12 dB, or 20 dB (or some subset) to be inserted into the signal path ahead of the active stages. When a strong signal that is capable of causing desensitization is present, *adding attenuation often increases the level of the desired signals in the output,* even though overall gain is reduced. This occurs because the overall signal that the receiver front-end is asked to handle is below the threshold where desensitization occurs.

Crossmodulation

Crossmodulation is an effect in which amplitude modulation (AM) from a strong undesired signal is transferred to a weaker desired signal. Testing is usually done (in HF receivers) with a 20-kHz spacing between the desired and undesired signals, a 3-kHz IF bandwidth on the receiver, and the desired signal set to 1,000 μV_{EMF} (−53 dBm). The undesired signal (20 kHz away) is amplitude modulated to the 30% level. This undesired AM signal is increased in strength until an unwanted AM output 20 dB below the desired signal is produced. A crossmodulation specification ≥100 dB would be considered decent performance. This figure is often not given for modern HF receivers, but if the receiver has a good third-order intercept point, then it is likely also to have good crossmodulation performance.

Crossmodulation is also said to occur naturally, especially in transpolar and North Atlantic radio paths where the effects of the aurora are strong. According to one (possibly apocryphal) legend, there was something called the "Radio Luxembourg Effect" discovered in the 1930s. Modulation from very strong broadcasters appeared on the Radio Luxembourg signal received in North America. This effect was said to be an Ionospheric crossmodulation phenomenon. If you or anyone you know has any direct experience with this effect, or a literature citation, I would be interested in hearing from you.

Reciprocal Mixing

Reciprocal mixing occurs when noise sidebands from the local oscillator (LO) signal in a superheterodyne receiver mix with a strong undesired signal that is close to the desired signal. Every oscillator signal produces noise, and that noise tends to amplitude modulate the oscillator's output signal. It will thus form sidebands on either side of the LO signal. The production of phase noise in all LOs is well known, but in more recent designs the digitally produced synthesized LOs are prone to additional noise elements as well. The noise is usually measured in -dBc (decibels below carrier, or, in this case, dB below the LO output level).

In a superheterodyne receiver, the LO beats with the desired signal to produce an intermediate frequency (IF) equal to either the sum (LO + RF) or difference (LO − RF). If a strong unwanted signal is present, then it might mix with the noise sidebands of the LO, to reproduce the noise spectrum at the IF frequency (see Figure 8.22). In the usual test scenario, the reciprocal mixing is defined as the level of the unwanted signal (dB) at 20 kHz required to produce a noise sidebands 20 down from the desired IF signal in a specified bandwidth (usually 3 kHz on HF receivers). Figures of −90 dBc or better are considered good.

The importance of the reciprocal mixing specification is that it can seriously deteriorate the observed selectivity of the receiver, yet it is not detected in the nor-

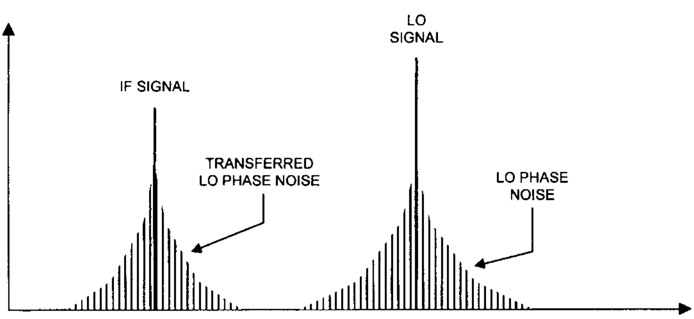

Figure 8.22
LO phase noise can be transferred to IF signal by reciprocal mixing.

mal static measurements made of selectivity (it is a "dynamic selectivity" problem). When the LO noise sidebands appear in the IF, the distant frequency attenuation (>20 kHz off-center of a 3-kHz bandwidth filter) can deteriorate 20 to 40 dB.

The reciprocal mixing performance of receivers can be improved by eliminating the noise from the oscillator signal. Although this sounds simple, in practice it is often quite difficult. A tactic that will work well, at least for those designing their own receiver, is to add high-Q filtering between the LO output and the mixer input. The narrow bandwidth of the high-Q filter prevents excessive noise sidebands from getting to the mixer. Although this sounds like quite an easy solution, as they say, "the devil's in the details."

IF Notch Rejection

If two signals fall within the passband of a receiver they will both compete to be heard. They will also heterodyne together in the detector stage, producing an audio tone equal to their carrier frequency difference. For example, suppose we have an AM receiver with a 5-kHz bandwidth and a 455-kHz IF. If two signals appear on the band such that one appears at an IF of 456 kHz and the other is at 454 kHz, then both are within the receiver passband and both will be heard in the output. However, the 2-kHz difference in their carrier frequency will produce a 2-kHz heterodyne audio-tone difference signal in the output of the AM detector.

In some receivers, a tunable, high-Q (narrow and deep) notch filter is in the IF amplifier circuit. This tunable filter can be turned on and then adjusted to attenuate the unwanted interfering signal, reducing the irritating heterodyne. Attenuation figures for good receivers vary from −35 to −65 dB, or so (the more negative the better).

There are some trade-offs in notch filter design. First, the notch filter Q is more easily achieved at low IF frequencies (such as 50 to 500 kHz) than at high IF frequencies (e.g., 9 MHz and up). Also, the higher the Q the better the attenuation of the undesired squeal, but the touchier it is to tune. Some happy middle ground between the irritating squeal and the touchy tune is mandated here.

Some receivers use audio filters rather than IF filters to help reduce the heterodyne squeal. In the AM broadcast band, channel spacing is typically 10 kHz and the transmitted audio bandwidth (hence the sidebands) are 5 kHz. Designers of AM BCB receivers usually insert an R-C low-pass filter with a −3 dB point just above 4 or 5 kHz right after the detector in order to suppress the 10 kHz heterodyne. This R-C filter is called a "tweet filter" in the slang of the electronic service/repair trade.

Another audio approach is to sharply limit the bandpass of the audio amplifiers. For AM BCB reception, a 5-kHz bandpass is sufficient, so the higher frequencies can be rolled off at a fast rate in order to produce only a small response an octave higher (10 kHz). In shortwave receivers, this option is weaker because the station channels are typically 5 kHz, and many don't bother to honor the official channels anyway. On the amateur radio bands, frequency selection is a perpetually changing "ad-hocracy" at best. Although the shortwave bands typically only need 3-kHz bandwidths for communications, and 5 kHz for broadcast, the tweet filter and audio roll-off might not be sufficient. In receivers that lack an effective IF notch filter, an audio notch filter can be provided. This accessory can even be added after the fact (as an outboard accessory) once you own the receiver.

Internal Spurii

All receivers produce a number of internal spurious signals that sometimes interfere with the operation. Both old and modern receivers have spurious signals from assorted high-order mixer products, from power supply harmonics, parasitic oscillations, and a host of other sources. Newer receivers with either (or both) synthesized local oscillators and digital frequency readouts produce noise and spurious signals in abundance. (Note: low-power digital chips with slower rise times—CMOS, NMOS, etc.—are generally much cleaner than higher-power, fast-rise time chips like TTL devices.)

With appropriate filtering and shielding, it is possible to hold the "spurs" down to −100 dB relative to the main maximum signal output, or within about 3 dB or the noise floor, whichever is lower.

Harry Helms, a writer of shortwave books (1994) has several high-quality receivers, including valve models and modern synthesized models. His comparisons of basic spur/noise levels were something of a surprise. A high-quality valve receiver from the 1960s appeared to have a lower noise floor than the modern receivers. Helms attributed the difference to the internal spurii from the digital circuitry used in the modern receivers.

RECEIVER SENSITIVITY MEASUREMENTS

Radio receiver specifications can be verified using test equipment and a few simple procedures. Such tests are made to evaluate receivers, to troubleshoot problems, and to verify performance. A number of receiver parameters are important, but perhaps the one that is most commonly discussed is *sensitivity.* Let's take a look at how these tests are done.

Sensitivity

Receiver sensitivity is a measure of how well the receiver will pick up very weak signals. As with most engineering measurements, the notion of sensitivity is an operational definition. In other words, there are standard procedures that will yield coherent results by which different receivers (or the same receiver before and after repairs) can be compared.

Sensitivity is basically a game of *signal-to-noise ratio* (SNR), or more properly, the signal-plus-noise-to-noise ratio $(S + N/N)$. For every receiver or amplifier there is a basic noise level consisting of the noise produced external to the receiver and noise produced inside the receiver. Even a receiver with its antenna input terminated in a shielded matching resistor, rather than an antenna or signal generator, will show a certain amount of thermal noise.

One important consideration when making sensitivity measurements (or comparing receiver sensitivity specifications) is bandwidth. Thermal and other forms of noise are gaussian and distributed over all possible bandwidths. The value of the noise at any given instant is dependent on the bandwidth of the channel. For most receivers this means the IF selectivity bandwidth, although in some cases the audio bandwidth is less than the IF so that number would dominate.

Figure 8.23 depicts two different definitions of SNR. Basically, you can't hear signals down in the noise. The minimum discernible signal (MDS) is operationally defined as the signal level that is the same as the noise floor, or the signal level that is 3 dB above the receiver's noise floor. But that sensitivity is not all that useful for most applications. There may be people out there who can listen to a signal that is only 3 dB above the noise floor. Most people, however, require a higher SNR to be practical. Although some definitions of SNR use 6, 12, or 20 dB, the standard for practical sensitivity is that it is the signal level that produces a 10-dB SNR. This definition is found on most CW, AM, and SSB receivers.

Signal Generator

The signal generator selected to make sensitivity measurements must have very high isolation figures. Most "service grade" signal generators are useful for doing troubleshooting, but are not satisfactory for making sensitivity measurements. The reason is that signal escapes around the cabinet flanges and control bushings. If you have a sensitive receiver or spectrum analyzer, then you can detect this signal.

Want to give it a try? Connect a shielded dummy load to the output of the signal generator, and turn the signal generator's output down to zero. Connect a whip or wire antenna to the receiver's antenna input, and then tune the receiver across the signal generator frequency with the RF gain cranked all the way up.

The signal generator should also have a calibrated output control. The correct calibrations are either *dBm* (power decibels relative to one milliwatt in a 50-ohm load), or microvolts (μV). Some signal generators have an output meter that can set relative output, but can become "calibrated" if a calibrated step-attenuator is connected between the output of the signal generator and the receiver under test.

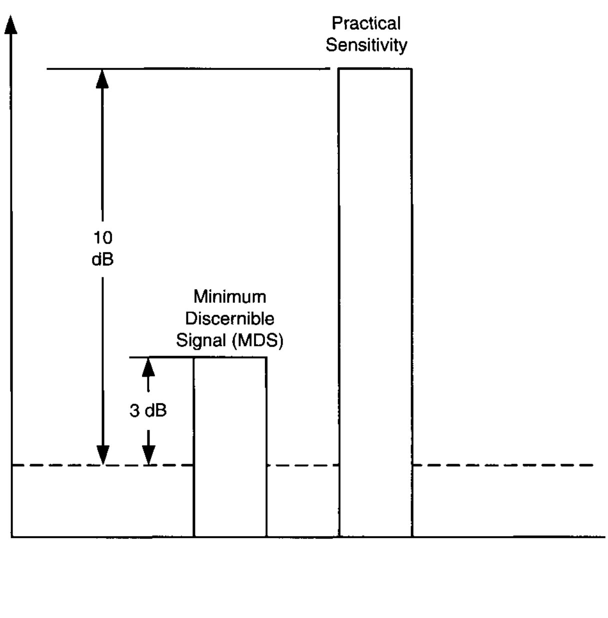

Figure 8.23
Practical sensitivity compared with minimum discernible signal (MDS).

You can find the exact level if you can measure the high level output of the signal generator.

"Laboratory grade" signal generators may be beyond the means of many people, but there is a relatively vigorous market in used or surplus equipment. There are a number of sources of such equipment listed on the World Wide Web. If you don't need the latest digitally synthesized signal generators, then you will be able to find good signal generators at low cost.

Test Set-up

Figure 8.24 shows the test set-up for most receiver sensitivity measurements. The attenuator is optional and may not be needed if the signal generator is adequately equipped with a good-quality calibrated output attenuator. When measuring an AM receiver, set the signal generator modulation for 30% depth and 1,000 Hz.

The receiver output level is measured using an audio AC voltmeter. Ideally, the instrument should be calibrated in decibels as well as volts and should have RMS reading capability.

The receiver must be correctly set up or the measurement will be in error. In most test set-ups, the receiver's RF and AF gain controls are turned to maximum, and the squelch is turned off. Furthermore, the *automatic gain control* (AGC) must be either turned off, or in the case of some models, clamped with a DC level according to the manufacturer's directions.

Minimum Discernible Signal (MDS) Sensitivity

To make the MDS measurement we need to find the signal level in dBm or μV that is 3 dB above the receiver noise floor.

1. Connect the equipment as in Figure 8.24, and set the receiver and signal generator to the same frequency.
2. Turn the signal generator output all the way down to zero.
3. Crank the RF gain and AF controls all the way up (you may want to set the audio output control to a convenient level if you don't have a dummy speaker load).
4. To make the measurement, you first measure the RMS value of the noise ("hiss") output on the AC voltmeter, and then increase the signal-generator output level until the receiver output level increases 3 dB.

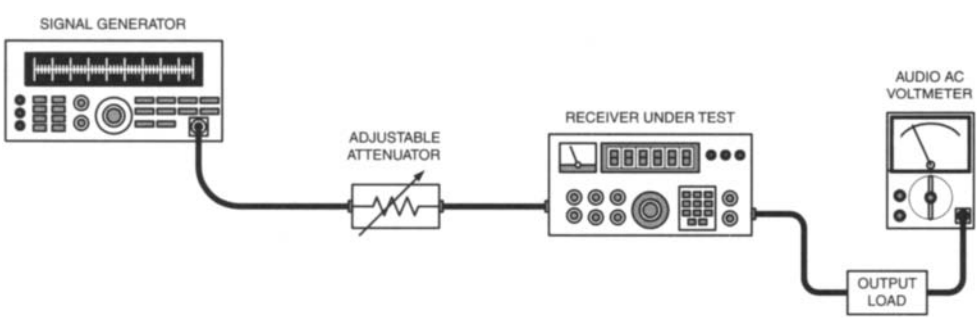

Figure 8.24 Standard test set-up for sensitivity measurements.

You can also determine the numerical value of the receiver noise floor by the same approach. Measure the output noise level, and then find the MDS by the procedure above. The receiver noise floor level will be the same as the signal generator output level (less any attenuation in line).

Standard Output Conditions

A sensitivity specification used for consumer radio receivers uses a standard output approach. A typical receiver sensitivity spec might read "xx μV for 400-mW in an 8-ohm load when modulated 30% by 1-kHz." The same equipment set up (Figure 8.24) can be used for this measurement. A power of 400 mW (0.4-watt) in an 8-ohms load is the same as 1.789 volts RMS, which can be read on the AC voltmeter. Use an 8-ohms noninductive resistor for the load rather than the loudspeaker-otherwise the sound levels are pretty annoying. Adjust the signal generator output level for an RMS output voltage of 1.789 volts, and read the output level from the signal-generator controls.

Full-Power Sensitivity

Some older receivers use the full-power sensitivity figure. This is the signal level that will produce the full-rated audio output power. Set the signal generator for 1-kHz modulation with 30% depth. Tune the radio and signal generator to the same frequency, and crank up the output until the audio output power is at the full-rated power level (e.g., 400 mW, 1 watt, etc.). The signal level that produces this condition is the "sensitivity" of the receiver.

10-dB (S + N)/N Test

The 10-dB test method is the same as the 3-dB MDS method, except that the signal generator level is increased until the output is 10 dB above the noise floor.

An alternate method is sometimes used on AM receivers:

1. Set up the signal generator and receiver as discussed above.
2. Set the output of the receiver to produce at least 0.5 watts audio output, or if the rated output power is lower than 1 watt set it for at least 50-mW audio output power.
3. Turn off the modulation. If the audio output drops at least 10-dB (or more), then the signal generator setting is at the 10-dB S + N/N level. If the level drops less than 10-dB, then readjust the signal-generator output level upward a small amount and try again.

On-site Effective Sensitivity Test

This test is only done on-site where the receiver is installed. It is intended to get some idea of how well the receiver performs in its actual installed environment. Figure 8.25 shows the test set-up.

1. Measure the 10-dB S + N/N sensitivity as discussed above (see Figure 8.24 for set-up), and write down the number.
2. Connect the hybrid combiner, two-position coaxial switch, antenna, and dummy load into the circuit.
3. Set the switch to the dummy load and measure the sensitivity. It will be considerably worse than the 10-dB sensitivity.
4. Set the coaxial switch to the antenna, and again measure the 10-dB sensitivity. It should be lower still.

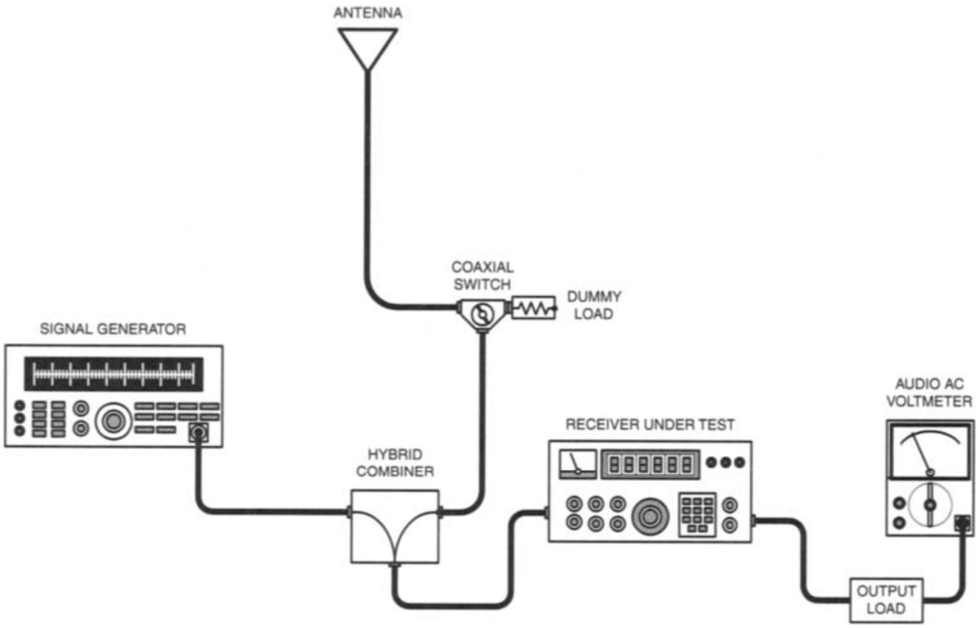

Figure 8.25 On-site effective sensitivity test.

The effective sensitivity is $SNR_{10dB} - (SNR_{Load} - SNR_{ANT})$. The figure $SNR_{Load} - SNR_{ANT}$ is the degradation factor. For example, suppose the 10-dB SNR is −122 dBm, the SNR when the load is connected is −77 dBm, and when the antenna is connected it is −70 dBm. The effective on-site SNR is:

$$SNR_{EFF} = SNR_{10dB} - (SNR_{Load} - SNR_{ANT})$$
$$SNR_{EFF} = -122 \text{ dBm} - [(-77 \text{ dBm}) - (-70 \text{ dBm})]$$
$$SNR_{EFF} = -122 \text{ dBm} - [-7 \text{ dBm}] = -115 \text{ dBm}$$

The effective sensitivity is only valid for the given site and conditions present when the test is performed. If the site is changed, or if the noise generators and other signals present change, then the test must be repeated.

FM Receiver Sensitivity

There are two basic methods for measuring the sensitivity of FM receivers: *20-dB quieting* and *12-dB SINAD.* The 20-dB quieting method is typically used on FM broadcast band receivers, and was once popular for communications receivers as well. More recently, the 12-dB SINAD method is preferred.

20-dB Quieting Method

This method relies on the fact that the FM detector will suppress noise once the limiting signal level is reached. The well-known capability of FM to eliminate noise relies on the fact that most noise amplitude modulates the carrier. If the amplitude can be clamped below the level where the noise is effective, then the frequency variations can be detected to recover the audio. This effect is called "quieting," which is the reduction of noise as the signal level increases.

To measure the 20-dB quieting sensitivity:

1. Connect the receiver and signal generator as in Figure 8.24. Keep the signal-generator output at zero. The modulation (deviation) should be set to whatever is appropriate for the class of receiver being measured.
2. Turn the RF gain all the way up. Set the audio output to produce a convenient reading in the high end of the AC voltmeter scale.
3. Measure the output noise level and write it down.
4. Turn the signal generator output level up until the reading on the AC voltmeter drops 20 dB. The signal generator output level that accomplishes this is the 20-dB quieting sensitivity (typically less than 1 μV).

SINAD Sensitivity

The sensitivity of FM receivers is often expressed in terms of SINAD. This approach to signal-to-noise ratio recognizes that the problem of detection depends on not simply signal and noise level, but also distortion. The SINAD (signal-noise-distortion) method is described by Equation 8.12.

$$SINAD = \frac{Signal + Noise + Distortion}{Noise + Distortion} \qquad [8.12]$$

In terms of decibels, the following equation is used:

$$SINAD\ (dB) = 20\ LOG \left[\frac{V_{Signal} + V_{Noise} + V_{Distortion}}{V_{Noise} + V_{Distortion}} \right] \qquad [8.13]$$

Where:

SINAD(dB) is the SINAD sensitivity expressed in decibels (dB)
V_{Signal} is the output voltage due to signal
V_{Noise} is the output voltage due to noise
$V_{Distortion}$ is the output voltage due to distortion

The standard 12-dB SINAD sensitivity corresponds to a 4:1 SNR ratio, in which the sum or noise and distortion is 25% of the signal voltage. As signal levels get higher, the SINAD and 10-dB S/N values tend to converge.

Figure 8.26 shows a typical test set-up for the SINAD measurement. The output AC voltmeter is augmented by a Total Harmonic Distortion (THD) analyzer,

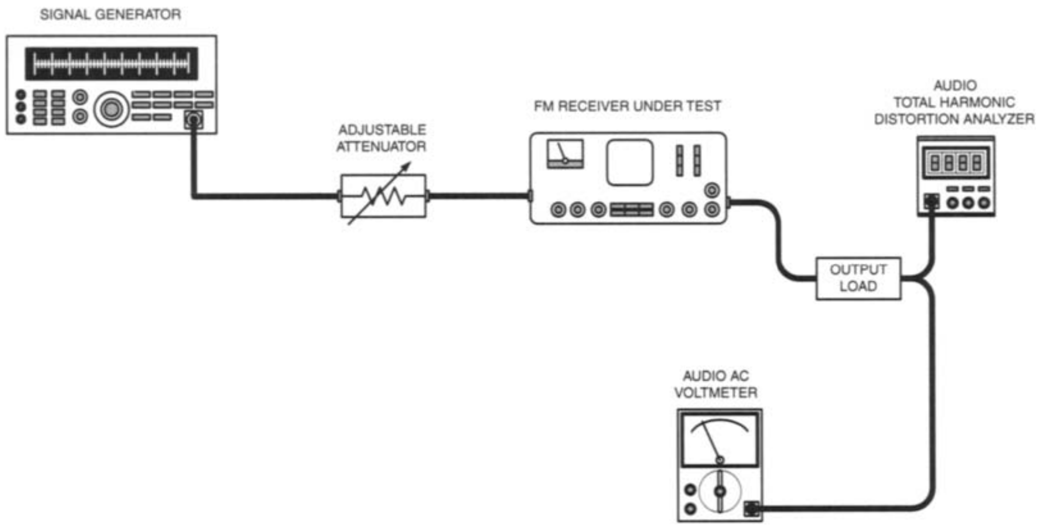

Figure 8.26 SINAD sensitivity test set-up.

both of which measure the output signal across the audio load (speaker, load resistor, etc.)

1. Set the signal generator frequency and receiver frequency to the same value.
2. Set standard conditions: modulating frequency 1-kHz sine wave; deviation set to 60% of the peak deviation used for that service. For example, for an FM BCB receiver, deviation is ±75 kHz, so set the signal generator deviation to $0.6 \times \pm 75$ kHz = ±45 kHz. For a communications receiver designed for ±5 kHz deviation, set deviation to $0.6 \times \pm 5$ kHz = ±3 kHz.
3. Adjust the receiver audio output to approximately 50% of the receiver's rated audio output.
4. Adjust the signal generator output until the input signal is high enough to produce 25% distortion. This is the 12-dB SINAD sensitivity.

Special SINAD sensitivity meters are available that combine the THD analyzer and audio voltmeter functions in one instrument.

RECEIVER NOISE MEASUREMENT

Radio reception is a game of signal-to-noise ratio. For that reason, it is sometimes necessary to measure noise levels and the noise performance of receivers.

Noise Floor

The noise floor of a receiver is the same as the minimum detectable signal (MDS) and is measured in exactly the same way. When you measure the MDS, it is defined as the signal level that causes an output 3 dB above the noise floor. Hence, when you measure the MDS, you have also measured the noise floor.

Setting standard conditions is necessary to make this measurement properly. For example, one caution is to use a signal level that is at least 10 dB above the expected sensitivity of the receiver, and then work down to find the 3 dB increase level.

One of the standard conditions is to ensure that the noise floor measurements are made in a standard bandwidth. It is often the case that receivers with multiple bandwidths will specify noise floor and sensitivity in the narrowest available bandwidth. However, that may not be useful if the normal mode for that receiver requires a wider bandwidth. For example, an H.F. communications receiver may have filters for AM mode (BW = 6 kHz), single-sideband (BW = 2.8 kHz), and CW (BW = 500 Hz). If the mode required for your application is SSB, then do not rely on sensitivity and noise floor measurements made on the 500-Hz bandwidth for the CW mode.

1-dB Compression Point

The 1-dB compression point is the input signal level at which receiver gain drops 1-dB (Figure 8.27). The gain of the system is depicted by the P_{out}/P_{in} line. This characteristic is linear up to a point, that is, a 1-dB increase in input signal level causes a proportionally scaled output level change. At some point, however, the receiver is saturated and cannot accommodate any further input signal. The operational definition of where this occurs is the 1-dB compression point.

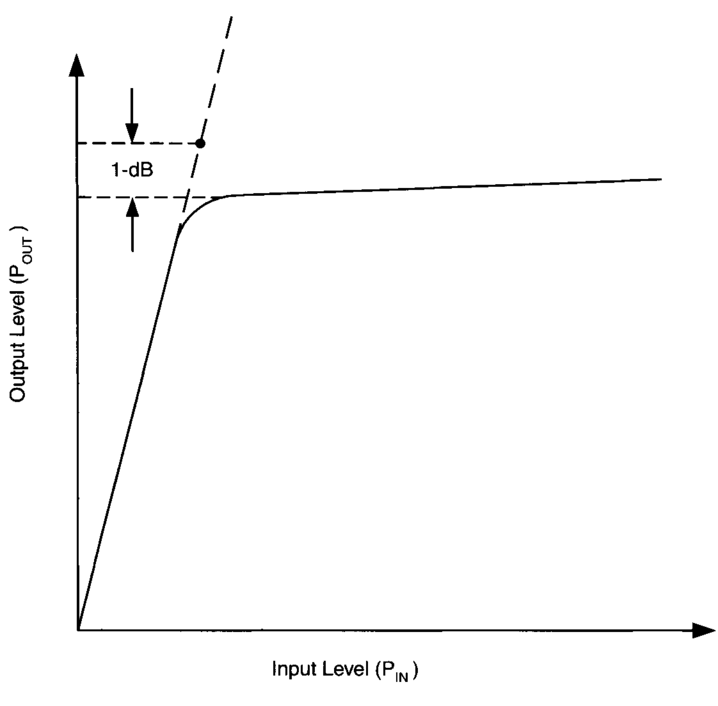

Figure 8.27
1-dB compression point.

To measure this point, use the standard test set-up of Figure 8.24. Bring the input power level applied to the receiver antenna terminals up from some low value in 1-dB steps until the receiver output level drops 1 dB. The input level at which this occurs is the 1-dB compression point.

Dynamic Range

Dynamic range represents the total range of input signal levels that can be accommodated. The classical dynamic range measure is the *blocking dynamic range* (BDR). There is also the *Third-Order IMD Dynamic Range* (TOIMDDR).

Blocking Dynamic Range

The BDR measures the difference between the receiver noise floor and the level of an off-channel undesired signal to reduce the sensitivity to on-channel signals by a specified amount (Figure 8.28). In other words, it is a measure of the range from MDS to a specified *desensitization* of the receiver.

Connect the test set-up of Figure 8.29. Two signal generators are coupled through an isolating hybrid to a step attenuator and then to the receiver. An AC audio voltmeter is used to measure the output level of the receiver. Figure 8.28 shows the signal situation for the receiver input. Frequency F1 is the desired signal, while F2 is the interfering signal.

The amplitude of F1 should be set such that it is above the receiver MDS by at least the amount of the required minimum S/N ratio (e.g., 10 dB), but below the point where it begins to increase IMD products. Using a higher level minimizes the noise error contribution. The exact level is somewhat test-procedure dependent. If the receiver automatic gain control (AGC) can be disabled, then set the signal level about 10 dB below the 1-dB compression point. If the AGC cannot be disabled, then the signal level should be lower, such as about 20 dB above the noise floor.

Frequencies F1 and F2 must have some specified spacing (ΔF). In most cases, an H.F. receiver will call for a 20-kHz spacing, while a VHF/UHF receiver will call for a 20-kHz or 100-kHz spacing. Some special-purpose microwave receivers sometimes look at considerably larger spacing. Whatever the case, the same spac-

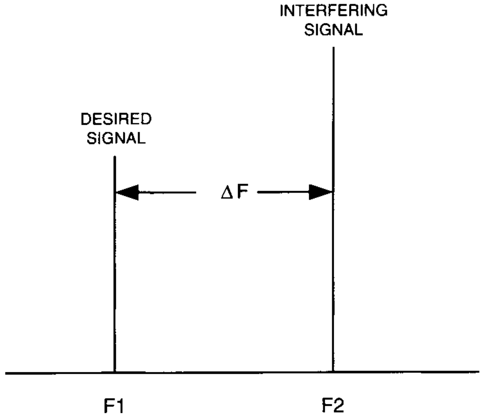

Figure 8.28
ΔF is the difference between a desired signal and an undesired signal.

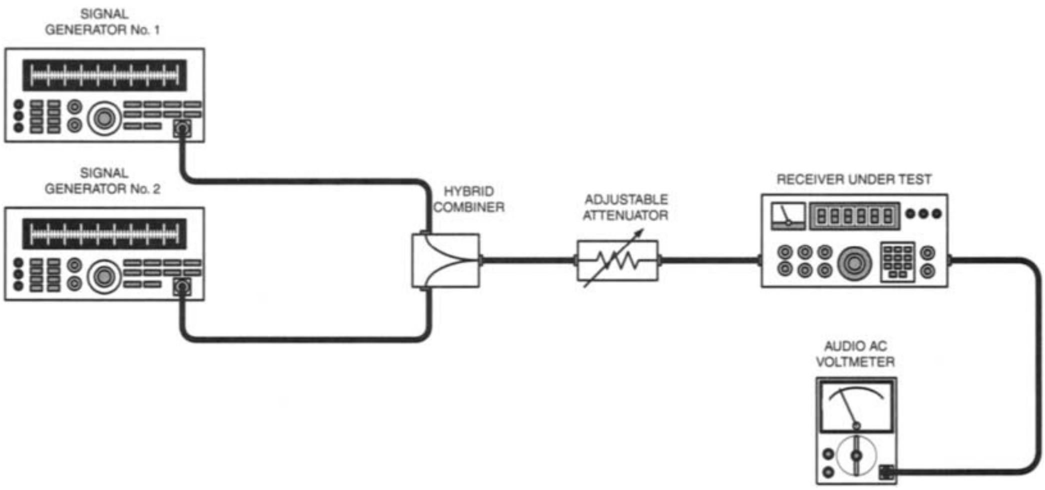

Figure 8.29 IM distortion test set-up.

ing should be used on different receivers when comparisons are made, because it seriously affects the results.

At the start of the measurement the F1 is turned on, F2 is turned off, and the receiver is adjusted for maximum output of F1. Turn on F2 at some level such as −100 dBm. Increase the level of F2 in 1-dB steps until the output level of the receiver drops 1-dB. The level at which this occurs represents the blocking dynamic range.

Perform this test on automatic test equipment, then the output levels will be automatically stepped. Also, in some cases the frequencies will also be stepped up and down the band, although maintaining the frequency separation ΔF. Adequate *settling time* between measurements must be programmed into the protocol. The signal-generator setting time rarely dominates this case, but be rather wary of the AGC settling time. By their nature, AGC circuits have a time constant, and it might be a relatively long period (e.g., 5 seconds). Understand the AGC time constant before programming a frequency or signal-level stepping protocol.

Third-Order IMD Dynamic Range (TOIMDDR)

A test set-up similar to that for BDR is used. Set the signal generators for a convenient output level such as −20 dBm, and the frequencies in-band with a specified ΔF spacing. The attenuator is set for a high degree of attenuation (low-signal level). Increase the signal by decreasing attenuation in 1-dB steps until a third-order response appears in the output. The actual signal level applied to the receiver can, as usual, be calculated from the signal generator and attenuator settings. The difference between the MDS and the level that is determined by this procedure is the TOIMDDR.

A Cautionary Note

In both this measurement and the IMD measurement (see below) it is important to ensure that only the receiver IMD products are being measured. Anytime a receiver or amplifier is overdriven into a nonlinear operating region, the possibility exists for the creation of unwanted distortion products. Even a single signal can produce first-order effects. If, for example, frequency F is applied to the input in sufficient power to overload the front-end, then harmonics at 2F, 3F, and so forth might be generated. This is also the case with normal mF1 ±nF2 IMD products higher than first-order.

Several mechanisms exist, and must be addressed. First, it is possible that the hybrid coupler will be nonlinear. This can occur when broadband ferrite or powdered iron-core transformers are used in the hybrid coupler. The coupler must produce very low levels of IMD, or it will affect the results of the test. Typically, a third-order intercept point (also known as TOIP or IP3) of > 50 dBm is required of the coupler. If the hybrid coupler being used has a lower TOIP, or if the receiver has a particularly high TOIP (> 45 dBm), then do not use any test configuration that places attenuation between the receiver input and the output of the coupler. That configuration forces the coupler to operate at too high a level.

Another scenario is a signal from one signal generator entering the output circuits of the other. Most amplifiers produce IMD products when signals arrive from the outside. This problem is reduced to negligible levels if the signal generator outputs have a high degree of inherent isolation, if attenuation is used between the coupler and the signal generators, and if the input-to-input port isolation of the hybrid is high. Try to get as much isolation as possible between signal sources, and between the sources and the receiver (> 90 dB). It is also prudent to use a signal generator with a high-level output (i.e., $\approx +15$ to $+22$ dBm).

Another problem with coupling between signal generators is that many high-quality signal generators employ a feedback-controlled automatic level control (ALC) that samples and rectifies the RF output signal, and then feeds back the derived DC control voltage to an amplitude modulator. When external signals arrive, they can affect the ALC in two ways. First, the extra signal level may influence feedback levels. Second, and more likely, the IMD products will get into the feedback system, causing beat notes that modulate the regular output signal.

> Tidbit: How do you tell if the IMD is due to the receiver or the test fixture? In most cases, reducing the signal levels will tell the tale. If the ratio of the desired to IMD responses changes when the input signal level to the receiver is changed, then it is a reasonable assumption that the IMD is generated inside the receiver. If the ratio does not change, then it is probably due to the test fixture.

Intermodulation Distortion

Intermodulation distortion (IMD) occurs when two frequencies, F1 and F2, mix together to produce heterodyne products that were not in the original set of signals. Figure 8.30 shows this behavior. When F1 and F2 are sufficiently strong, the receiver becomes nonlinear, so mixing will occur. When this happens IM products rise up out of the noise.

There are several different methods for measuring the IMD performance of a receiver. Figure 8.31 shows one standard set-up. Two signal generators are used to provide the two different signals required for the IMD test. Each signal generator

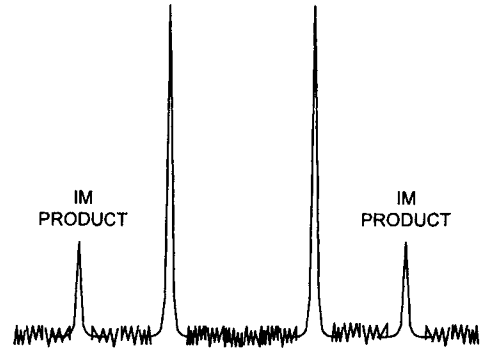

Figure 8.30
Third-order IM products are usually most important because they fall close to the test signals and may be within the receiver passband. Second-order IM products (F1 ± F2) tend to fall outside the band, so they are easily filtered out. Third-order IM products (2F1 ± F2 and 2F2 ± F1) are usually outside the band.

is equipped with an adjustable attenuator, which may or may not be external to the generator. In some cases, both internal and external attenuators may be used.

Optional bandpass filters are sometimes used to clean up the signal generator output spectrum. These filters are used to suppress harmonics of the output frequency. If the signal generator has sufficiently low harmonic output, then these filters can be eliminated. Keep in mind that some filters use ferrite or powdered iron cores, so may saturate and cause IMD products of their own. The two signals are combined in a two-port hybrid. Following the hybrid is another attenuator. This attenuator supplies signal to the receiver input.

The output signal is monitored by any of several means. Some procedures use the audio AC output level, as measured by an AC voltmeter. In other cases, the spectrum of the audio output signal is measured using a spectrum analyzer. Alternatively, one might also use a frequency selective voltmeter (also called a wavemeter). The latter method is out of favor because spectrum analyzer prices have fallen significantly. Some people will use the receiver S-meter (if it has one) to

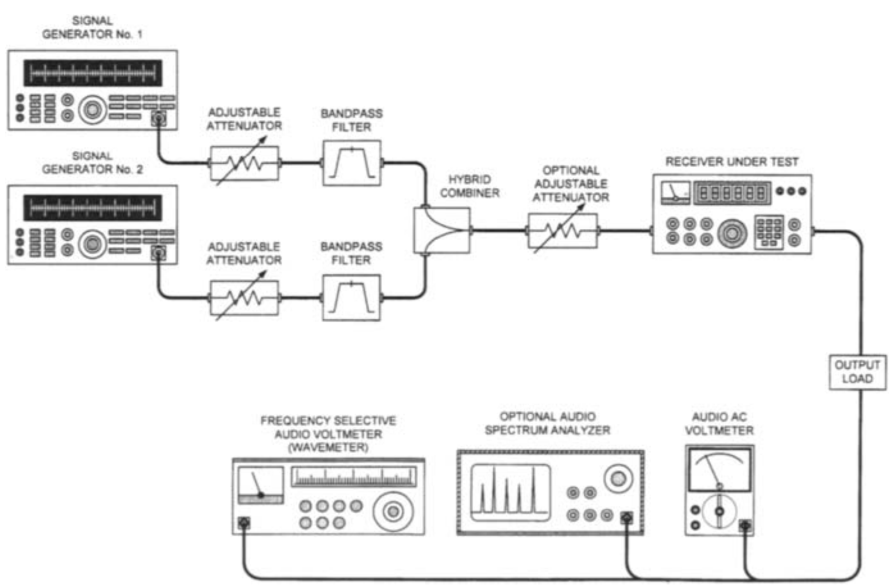

Figure 8.31 Test set-up for making complete measurement of IMD performance of the receiver.

make this measurement. Still others couple the IF signal to an RF/IF spectrum analyzer. The latter method may show more information, but it has the disadvantage of requiring entry inside the receiver. The other methods treat the receiver as a "black box," and thus require no modification of, or entry into, the receiver. The IMD test is best run in one of the linear reception modes (SSB or CW), but that is not always possible (e.g., when the receiver is FM-only or AM-only).

Audio Signal Level Method

The audio output level is monitored on an audio spectrum analyzer (or measured on a wavemeter). The signal levels are turned up until the IMD product being investigated rises up out of the noise level.

The spectrum analyzer method can be particularly useful for measuring products below the noise floor of the receiver. Recall that the noise floor is proportional to bandwidth. Typical bandwidths vary from 500 Hz for CW receivers to 200 kHz for FM broadcast receivers (and more for microwave radar receivers). If the bandwidth filter on the audio spectrum analyzer is set to some narrow value, such as 5 or 10 Hz, then the noise is much lower, so low-level IMD problems show up better.

Signal-to-Noise Ratio Method

This approach to measuring IMD uses either an audio signal-to-noise ratio meter or a SINAD meter. The audio output is set to produce a 1-kHz signal for this method. Care must be exercised to prevent excess noise contribution from the signal generator output noise. This noise is indistinguishable from receiver noise, so it makes the IMD look worse. It is also possible that AGC action will interfere with this test.

S-Meter Method

In this method the level of the IMD product is noted on the receiver's S-meter. A reference signal is then provided that matches the S-meter reading. This yields the level of the IMD product. Problems with this method include the fact that some receivers compress gain when the signal level gets to a level above S9 or S9 + 10 dB. This method is better than some, however, for measuring the IMD performance of receivers with very high IMD performance.

Standard Method

The normal method for measuring the IMD performance is to set the signal generators to some convenient high-level output (e.g., −20 dBm). Select test frequencies (F1 and F2) and calculate the third-order products (2F1 + F2, 2F1 − F2, 2F2 + F1, and 2F2 − F1).

Set the receiver to a channel frequency, F1. If possible, turn off the AGC or clamp it to a low value (highest receiver gain), if possible. If the receiver uses a front-end RF or IF attenuator, then set it to 0 dB. If there is an RF preamplifier being used, turn if off. Adjust both the receiver tuning and F1 to the same frequency, and maximize the receiver output. Set the second signal generator to F2 at a specified spacing (e.g., 20 kHz) away from F1. Set both signal generators to a convenient output level such as −10 dBm. Set the in-line attenuator to the highest setting (most attenuation).

Once the set-up is completed, turn off the signal generators and measure the receiver output noise level on an AC audio voltmeter. Turn on the signal generator

F1 and decrease the attenuator setting in 1-dB steps until the output noise level of the receiver increases 3 dB. This is the minimum discernible signal (MDS) reference level. Return the attenuator settings to maximum. Record this signal level as $P_{IM} = -10$ dBm − (Attenuator setting).

Tune the receiver to either of the close-in third-order product frequencies (either 2F1 − F2 or 2F2 − F1), while leaving the signal generators at F1 and F2 (both −10 dBm output). Reduce the attenuator setting until the receiver output response at this frequency increases 3 dB (the same as the reference MDS). Record this level as $P_A = -10$ dBm − (Attenuator setting).

$$IP_N = \frac{N\,P_A - P_{IM_N}}{N - 1} \qquad [8.14]$$

Where:

IP_N is the intermodulation product of order N
N is the order of the intermodulation product
P_A and P_{IM} are signal power levels in dBm

EXAMPLE

A 162.55 MHz receiver was tested using frequencies F1 = 162.55 MHz and F2 = 162.57 MHz. The close-in third-order products would be 2F1 − F2 = 162.53 MHz and 2F2 − F2 = 162.59 MHz.

The minimum discernible signal at 162.55 MHz required an attenuator setting of −89 dB, so P_{IM} = [−10 dBM − 89 dB] = −99 dBm.

The response at 162.53 MHz required 19 dB of attenuation for the third-order response to equal the MDS level. So P_A = [−10 dBm − 19 dB] = −29 dBm.

Because this is a third-order response, N = 3, so

$$IP_3 = \frac{(3 \times (-29\,dBm)) - (-99\,dBm)}{3 - 1}$$

$$IP_3 = \frac{-87\,dBm + 99\,dBm}{2} = \frac{+12\,dBm}{2} = +6\,dBm$$

Once the P_A and P_{IM} points are found, any IP can be calculated using Equation 8.14.

Selectivity Testing

Selectivity is a measure of the receiver's ability to reject off-channel signals. It is measured in terms of bandpass, so it has the units of frequency. The operational definition of selectivity is that it is the bandpass (Figure 8.32A) between the two points (F1 and F2) where the frequency response drops −3 dB from its mid-band point (F_o).

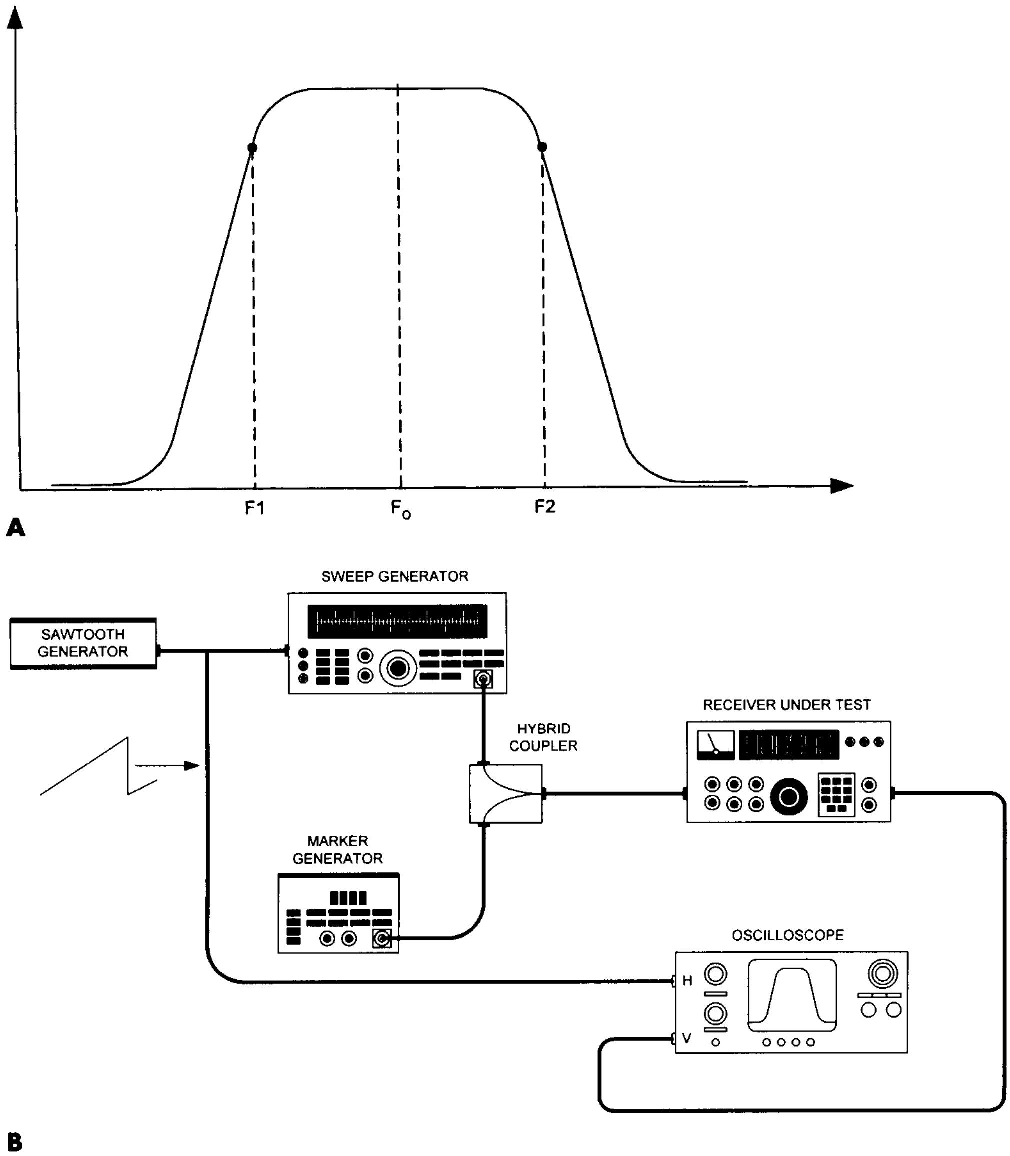

Figure 8.32 (A) Selectivity bandpass; (B) Sweep-generator measurement of selectivity.

CW Method

It is possible to use a standard CW generator to make this measurement, although it is tedious and may not yield the best information. Set up a signal generator and output indicator per Figure 8.24. Center the signal-generator frequency inside the receiver passband, and adjust the receiver output for a convenient level indicated on the AC voltmeter. Adjust the signal generator below F_O to a point where the output drops the specified amount. This frequency is F1. Repeat the procedure above F_O until F2 is determined. The receiver bandwidth is F2-F1.

Sweep Method

The CW method produces a raw indication, but lacks certain information, at least in its simplest form. We are not simply interested in the F2-F1 value, but also in the shape of the passband. Ideally, a receiver passband is flat inside the passband, rolling off gently on the upper and lower ends. That may not be the actual case. By using the sweep method, we can determine the shape of the passband, as well as its bandwidth. There are sometimes some surprises lurking there, so the information is needed. To get shape data by the CW method requires collecting a lot of data points and then manually graphing them. The sweep method produces an oscilloscope trace that can be photographed.

Figure 8.32B shows the test equipment set-up for a simple sweep test. A sweep generator will sweep through a specified band of frequencies. In the simplest form, it consists of a *voltage controlled oscillator* (VCO) with its tuning voltage in the form of a sawtooth waveform (modern sweepers are a bit more sophisticated, but the idea is the same). The same sawtooth can be used to control the horizontal deflection of an oscilloscope. The sweep signal is applied to the receiver's antenna input, while the receiver output is applied to the vertical deflection input of the oscilloscope. The result is a trace representing amplitude-versus-frequency.

A *marker generator* can be used to indicate specific frequencies within the passband (typically, F1, F2, and F_0). In some cases, discrete frequencies are used, while in others the harmonics of a standard frequency may be used. For example, a 1-kHz marker generator that is sufficiently rich in harmonics will provide markers every 1 kHz throughout the passband. There is, however, a limit to the use of harmonic markers, as their upper harmonics may not be strong enough to produce the desired display.

The sweep rate setting deserves some comment. The sweep rate is the repetition rate at which the signal passes through the swept band. This attribute is controlled by the frequency of the sawtooth generator. If the sawtooth rate is too slow, then the display will flicker and be hard to read. If it is too high, then there may be ringing in the receiver's IF filters, causing a distorted reading. Flicker fusion tends to occur in the 8- to 10-Hz range for most human operators, so the minimum sweep frequency will have to be at least these values. When you get up to above 40 Hz or so, the danger of causing a ringing response in high-Q IF filters becomes more likely. You should, therefore, select a sweep frequency between roughly 8 and 40 Hz for manually operated measurement systems.

Squelch Tests

Communications receivers are often equipped with a *squelch* circuit. These circuits turn off the receiver's audio output when no signal is being received. Noise occupies the bandwidth when no signal is received, and it is uncomfortable to hear. When the squelch circuit detects noise, rather than signal, it turns off the receiver output. The operator sets the squelch from the front panel.

There are two levels of squelch: *critical squelch* and *tight squelch*. The critical squelch level exists when the control is set so that it just barely quiets the receiver when no signal is being received. Indeed, when a particularly high-level noise burst is heard, it might degrade the squelch enough to pass through momentarily.

Tight squelch requires a much larger signal to cause break through, and it occurs when the squelch control is set to maximum.

Critical Squelch

Connect a signal generator to the antenna input of the receiver. Adjust the signal generator frequency on-channel.

Modulation. For AM receivers set the signal-generator modulation depth to 30% with a modulating frequency of 1,000 Hz. For FM receivers, use a deviation that is approximately 60% of the normal deviation for that receiver, with a 1,000-Hz modulating frequency.

1. Set the signal-generator output level to the lowest possible level so that no signal is heard in the receiver output.
2. Set the squelch control to the point where the output noise just disappears when no signal is present.
3. Bring the signal generator's output level up very slowly until the squelch breaks. Record the output level as the critical squelch point (dBm or μV).

Tight Squelch

The tight squelch test is performed identically to the critical squelch, but with the squelch control on the receiver set to the maximum (most squelched) position. The signal level required to break tight squelch will be much larger.

Squelch Range

This characteristic is the range between critical squelch and tight squelch. It may be expressed in signal level units (dBm or μV), or in dB. For example, suppose the signal levels are:

Critical Squelch: 0.4 μV (−115 dBm)
Tight Squelch: 58 μV (−71 dBm)
Squelch Range: 57.6 μV (44 dBm)