MICROWAVE RECEIVERS

A microwave receiver is used to receive information transmitted at microwave frequencies (from 1 to 220 GHz). Most microwave receivers are integral parts of a system, that is, communication or radar, and they are designed together with the transmitter. In other words, the receiver is designed to receive a specific signal with maximum efficiency. For example, the video bandwidth of a pulsed radar receiver matches the bandwidth of the transmitted pulse so as to receive maximum energy in the signal while at the same time limit its noise bandwidth to a minimum value. The receiver of a frequency modulation (FM) radar has a dispersive delay line matching the transmitted signal and compresses it into a short pulse to increase the processing gain. In the global positioning system (GPS) the coarse/acquisition (C/A) signals are code division multiple access (CDMA). In a GPS receiver locally generated C/A codes are used to correlate with an input signal to perform acquisition and signal tracking. In an FM receiver, the frequency variation information is converted into amplitude information through a frequency discriminator. From these examples one can see that each receiver is uniquely designed.

Another type of microwave receiver detects uncooperative signals. This type of receiver is usually referred to as an intercept receiver. This type of receiver has some designs in common, because each subtype is designed to detect signals with very limited information. For example, there are police radar receivers and police radar detectors. The police radar receiver, designed to match the transmitted signal to achieve the highest sensitivity, is an integral part of a radar system. The police radar detector used in automobiles is an intercept receiver. The purpose of the radar detector is to detect whether police traffic radar is being used. An intercept receiver does not have the detailed information of the radar signal; instead, it uses coarse information, that is, the frequency range over which the radar operates. Although the intercept receiver has less sensitivity than the radar receiver, the intercepted signal is much stronger than the signal returned to the radar. The signal strength received by a radar receiver is proportional to $1/R^4$ where R is the distance between the radar and the target (or intercept receiver). The intercepted signal strength is proportional to $1/R^2$, which is stronger than the returned signal.

In this article, intercept microwave receivers will be the main subject, because of their similarities in design goals. Intercept receivers are very useful in military applications. Used in electronic warfare (EW), they are often referred to as EW receivers. These receivers are used to intercept hostile communication as well as radar signals. It is a more challenging task to design military intercept receivers, because it is a common military practice to design signals that cannot be detected by an intercept receiver and cannot be jammed.

Receivers can sometimes be used to detect unintentional radiation. For example, one can use a microwave receiver to check the microwave power leakage level of a microwave oven. This type of receiver is also useful in military applications. For example, the detection of spark plug radiation from automobiles can help locate enemy vehicles.

In the past, most microwave receivers were built using analog techniques. Because of recent advances in digital circuits, that is, with today's high-speed analog-to-digital converters (ADC) with their large number of bits and recent high-speed digital signal processing (DSP) techniques, it appears that the trend is to build digital receivers. Some narrowband receivers have already been built using digital techniques. Wideband digital microwave receivers are in the research stage. Digital receivers should be more reliable because they require less maintenance and adjustment.

This article includes these subjects: a generic receiver is described, followed by a discussion of the important terminology and definitions used in receiver design. A classification of receivers is then presented, with a discussion of analog and digital receivers to follow in subsequent sections. A comparison of different types of receivers concludes the article.

GENERIC RECEIVERS

In general, the signals received by an antenna are very weak. It is difficult to process the signals or even to detect their presence directly. A common approach is to amplify the signals to a higher power level before further processing or detection. This amplification is accomplished through radio frequency (RF) chain.

The RF chain usually contains the following components: RF amplifiers; filters; mixers; local oscillators; intermediate frequency (IF) amplifiers; IF filters; and attenuators. Amplifiers are used to raise the signal power level. Filters are used to limit out-of-band noise as well as spurious responses (undesired frequencies) generated from some components. The mixer and local oscillator are used in combination to shift the input frequency to another frequency, often referred to as the IF. At IF, additional filters and amplification can be provided. In many receivers the input signals are not converted to a different frequency and therefore, the mixer and local oscillator are not needed. Attenuators are used to adjust the overall gain of the RF chain. The overall gain in the RF chain must be of a specific value. Often commercial amplifiers with specific gain values are used in receiver design and it is difficult to implement the desired value. A common practice is to use amplifiers to provide more gain than the desired value and attenuators to lower the gain to the correct value.

In communication receivers, automatic gain control (AGC) is sometimes used. The AGC changes the gain of the RF chain according to the input signal strength: lower gain for strong signal and higher gain for weak signal. The AGC is seldom used in receivers intercepting pulsed signals, because it is difficult to build an AGC with very fast response time.

After the RF chain in an analog receiver, the signal is detected by a crystal video detector. The detector filters out the RF but retains the information of the signal, often referred to as the video signal. Further processing is needed to obtain the necessary information, which includes digitizing the video signal. A basic analog receiver is shown in Fig. 1(a). In some analog receivers, there are no RF amplifiers and a crystal video detector is used to detect signals directly. These receivers usually have low sensitivity and can detect only very strong signals.

After the RF chain in a digital microwave receiver, an ADC is used to convert the input into digital data as shown in Fig. 1(b). Because the output of the ADC is digital, digital signal processing (DSP) can be used to obtain the necessary information.

DEFINITIONS USED IN RECEIVERS

The two most important specifications to describe a receiver are sensitivity and dynamic range. These two parameters can be used to specify all kinds of receivers whether they are intercept receivers or receivers designed for a specific signal. Sensitivity can be briefly defined as the capability to receive the weakest possible signal. Dynamic range is the maximum signal amplitude range that a receiver can process without distortion. A strong signal above the upper limit of the dy-

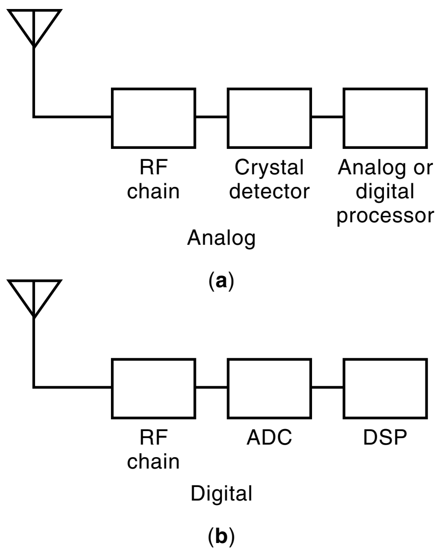

Figure 1. (a) A basic analog receiver with RF amplifier chain, crystal detector and analog/digital processor. The RF amplifier chain is used to amplify the input signal. The crystal detector changes the RF signal into a video signal. The processor takes the video signal and generates the desired digital information. (b) A basic digital receiver with RF amplifier chain, ADC and DSP. The ADC digitizes the analog input signals. The DSP processes the digitized data and generates the desired digital information.

namic range can produce distortion or generate spurious responses. It is desirable to have high sensitivity to receive weak signals and high dynamic range to receive a broad range of signals. However, high sensitivity often causes lower dynamic range and vice versa. Therefore, in designing a receiver, the trade-off between sensitivity and dynamic range becomes an important issue.

Sensitivity is closely related to noise floor, the noise figure of the receiver, and the gain in the RF chain. In an analog receiver, the sensitivity depends on the video bandwidth after the detector. In a digital receiver, the DSP algorithm used after the ADC affects the overall bandwidth of the receiver and determines receiver sensitivity. When a receiver can process only one signal, there is only one definition of dynamic range. If a receiver can process simultaneous signals, there are usually three definitions of dynamic range. They are denoted as the single signal, two signal spur free, and two signal instantaneous dynamic range. The lower limit of the dynamic range is always the sensitivity level. The upper limit depends on the definition of the dynamic range. These definitions, terminology and calculations to obtain some of the values will be discussed in the following paragraphs. Most equations can be found in Ref. 1.

Receiver Input Bandwidth and Instantaneous Bandwidth

The input bandwidth of a receiver refers to the frequency range in which the receiver can detect an input signal. This is also referred to as the operational bandwidth or the overall bandwidth of the receiver. The instantaneous bandwidth means that any signals with sufficient amplitude in the bandwidth will be detected immediately. Usually the input bandwidth is wider than the instantaneous bandwidth, but in some receivers they are the same. The instantaneous bandwidth can be assigned to different portions of the input bandwidth. For example, a receiver may have a 16 GHz (from 2–18 GHz) input bandwidth but a 1 GHz instantaneous bandwidth. This 1 GHz bandwidth can be placed in any one of the bands in the 2–18 GHz range to receive signals in that band. The input bandwidth and the instantaneous bandwidth are not used to determine the sensitivity. They are mentioned here only to distinguish them from the RF bandwidth.

RF Bandwidth

The RF bandwidth is used to determine the sensitivity of a receiver. In an analog receiver, the filter with the narrowest bandwidth in the RF chain is the RF bandwidth. In some receivers, the input, the instantaneous and the RF bandwidths may be the same. In a digital receiver, usually the inverse of the DSP length will be used as the RF bandwidth. For example, if the input is digitized at 1000 MHz, each sample is separated by 1 ns. If 256 samples will be processed through the fast Fourier transform (FFT), the RF bandwidth will be approximately 3.9 MHz ($1/256 \times 10^{-9}$ s).

Video Bandwidth

The video bandwidth of a receiver is determined by the video circuit following the crystal detector. The desired bandwidth is determined by the input signals. For an electronic warfare (EW) receiver, the video bandwidth is determined by the shortest pulse anticipated. In a digital receiver, it is determined by the processing scheme and is sometimes assumed to have the same as the RF bandwidth.

Noise

Noise generated by a resistor R can be represented by a noise generator in series with the resistor. Maximum power transfer from a generator to a load occurs when the load impedance is matched to the generator impedance. Available power refers to the power that would be delivered to a matched load. The available thermal noise power N_i in watts at the input of a receiver can be expressed as

$$N_i = kTB \tag{1}$$

where k is Boltzmann's constant ($= 1.38 \times 10^{-23}$ J/°K) T is the temperature of resistor R, and B is the bandwidth of the receiver in hertz. The power level in a typical receiver system is very low and is usually expressed in milliwatts or in dBm, which is defined as

$$P(\text{dBm}) = 10\log(P) \tag{2}$$

where the P on the right-hand side is power in milliwatts and the base of the log is 10. The thermal noise at room temperature where $T = 290$ °K can be expressed in dBm as

$$P(\text{dBm}) = -174 \text{ dBm/Hz}$$

or (3)

$$P(\text{dBm}) = -114 \text{ dBm/MHz}$$

These two values are commonly used in receiver designs. It should be noted that with an antenna aimed skyward, the noise temperature can be very low.

Gain

The gain of an amplifier is defined as

$$G = \frac{S_o}{S_i} \tag{4}$$

where S_o and S_i are the available output and input signal powers, respectively. The gain is often defined in decibels as

$$G(\text{dB}) = 10\log(G) \tag{5}$$

When N amplifiers are connected in cascade, the overall gain can be expressed as

$$G = G_1 G_2 \cdots G_N$$

or (6)

$$G(\text{dB}) = G_1(\text{dB}) + G_2(\text{dB}) + \cdots + G_N(\text{dB})$$

where G_1, G_2, . . . G_N are the gain of each individual amplifier.

Noise Figure

The noise figure is defined as

$$F = \frac{N_o}{GN_i} = \frac{\text{noise output of practical receiver}}{\text{noise output of an ideal receiver at temperature } T} \tag{7}$$

where N_o is the noise at the output of the receiver, G is the gain of the RF chain in ratio and N_i is the input thermal noise ($= kTB$). Substituting Eq. (4) into Eq. (7), the result is

$$F = \frac{S_i/N_i}{S_o/S_o} = \frac{\text{signal-to-noise ratio at input of receiver}}{\text{signal-to-noise ratio at output of receiver}} \tag{8}$$

The noise figure is often defined in decibels as

$$F(\text{dB}) = 10\log(F) \tag{9}$$

If there are N amplifiers connected in cascade, the noise figure in ratio can be expressed as

$$F = F_1 + \frac{F_2 - 1}{G_1} + \frac{F_3 - 1}{G_1 G_2} + \cdots + \frac{F_N - 1}{G_1 G_2 \cdots G_{N-1}} \tag{10}$$

where G_1, G_2, . . . and F_1, F_2, . . . are the gain and noise figure of the first, second . . . amplifiers respectively, and are expressed in power ratio rather than in dB. From Eq. (10) it can be shown that if the first component in the RF chain is a high gain amplifier, the overall noise figure can be approximately equal to the noise figure of the first amplifier.

Sensitivity

The sensitivity of a receiver depends on the noise power at the input of the receiver, which is related to the bandwidth of the RF chain, gain, and video bandwidth for an analog receiver and the DSP algorithm used for a digital receiver. It is often specified along with false alarm rate and probability of detection. The false alarm rate is defined as the number of false measurements when there is no input signal. In some cases more restrictions can be added to the definition of sensitivity, that is, the parameters measured by the receiver must be within certain limits. In designing a receiver, the sensitivity can be determined from curves generated based on the RF bandwidth, the video bandwidth, the probability of detection, and the probability of false alarm from this author's work (1). Once a receiver is built, one can apply specific requirements to evaluate its sensitivity. The sensitivity is usually frequency dependent, which means its value varies across the frequency range of the receiver.

Tangential Sensitivity

As the sensitivity of a receiver with given false alarm and probability of detection is tedious to calculate and measure, an easily calculable and measurable sensitivity is defined. For an analog receiver, the tangential sensitivity is measured through visual display on an oscilloscope that monitors the output of a video amplifier following the detector. The input must be a pulse signal. On the scope display, when the bottom of the noise trace in the pulse region is roughly tangential to the top of the noise trace between pulse as shown in Fig. 2(a) the receiver is at tangential sensitivity. At tangential sensitivity, the signal-to-noise ratio is 8 dB at the output of the detector with a standard deviation of 0.4 dB. Based on these values, the tangential sensitivity of a digital receiver can be illustrated from the signal-to-noise ratio of 8 dB as shown in Fig. 2(b). When there is sufficient gain in the RF chain, the tangential sensitivity is independent of the characteristics of the video detector. This case is referred to as the noise-limited case. If there is insufficient gain, the tangential sensitivity depends on the characteristics of the detector and is referred to as the gain-limited case. The minimum gain required for the noise-limited case is to raise the noise floor to the tangential sensitivity of the crystal detector (approximately −35 to −45 dBm). Above this gain value, the sensitivity is gain independent. Most modern microwave receivers fulfull this condition. The tangential sensitivity (TSS) for the noise limited case is

$$\begin{aligned} TSS &= -114 + 10\log(F) + 10\log\left(3.15B_v + 2.5\sqrt{2B_R B_v - B_v^2}\right) \text{ dBm for } B_v \le B_R < 2B_v \\ TSS &= -114 + 10\log(F) + 10\log\left(6.31B_v + 2.5\sqrt{2B_R B_v - B_v^2}\right) \text{ dBm for } B_R \ge B_v \end{aligned} \tag{11}$$

where −114 is the noise floor of 1 MHz bandwidth, F is the overall noise figure of the receiver and B_v and B_R are the video and RF bandwidths, respectively.

The tangential sensitivity is usually too low to be used as the operating sensitivity level, because at this level the false alarm rate is very high. The rule of thumb is that the operating sensitivity is approximately 6 dB higher than the tangential sensitivity level.

Single Signal Dynamic Range

This dynamic range is applicable to all receivers. The lower limit is the sensitivity level. When an input signal is very

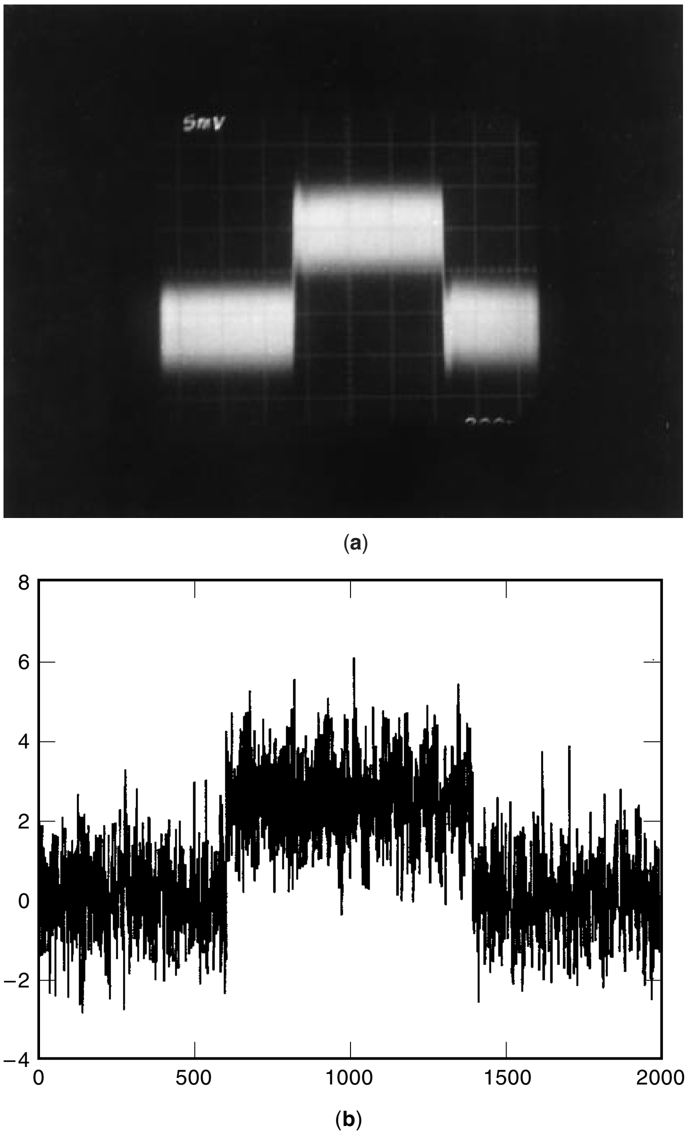

Figure 2. (a) Tangential sensitivity of an analog signal from an oscilloscope display. The output signal-to-noise ratio = 8dB. (b) Tangential sensitivity of a digital signal where the signal-to-noise ratio = 8dB.

strong, it may cause some components in the RF chain or the ADC in a digital receiver to become saturated. Under this condition, the output signal will be distorted or spurious signals will occasionally be generated. Depending on the specifications of a certain receiver, the upper limit is determined accordingly.

Two-Tone Third-Order Intermodulation Products and Third-Order Intercept Point

If the passband of a receiver is less than an octave, the third-order intermodulation products are the lowest-order intermodulation products that can fall within the passband. That is why they are used as the upper limit of the two-tone spur-free dynamic range. The third-order intermodulation is measured with two input signals of equal amplitude. When the input signals f_1, and f_2, are strong enough to drive the RF chain into saturation, spurs will be produced at frequencies $2f_1, - f_2$ and $2f_2 - f_1$ as shown in Fig. 3. The outputs at these two frequencies are referred to as third-order intermodulation. The amplitude of the third-order intermodulation can be experimentally measured.

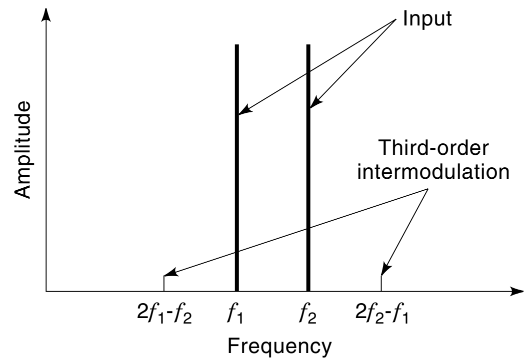

Figure 3. Third-order intermodulation products. The input signals are at f_1 and f_2 with equal amplitude and the third-order intermodulations are at $2f_1 - f_2$ and $2f_2 - f_1$.

Another quantity related to third-order intermodulation is referred to as the third-order intercept point. The third-order intercept point can be obtained from the third-order intermodulation in Fig. 4. The straight line with a 1 : 1 slope represents the gain of an amplifier. Another straight line is drawn that passes the measured third-order intermodulation point with a 3 : 1 slope that represents the anticipated amplitude change of the third-order intermodulation as a function of input signal. The intercept point between these two lines is the third-order intercept point and the value is often read from the output axis. It should be kept in mind that these two straight lines are projected results. They can not actually be measured because, when a component is approaching saturation, the actual measured lines will bend downward. This quantity is used to determine the spur-free dynamic range.

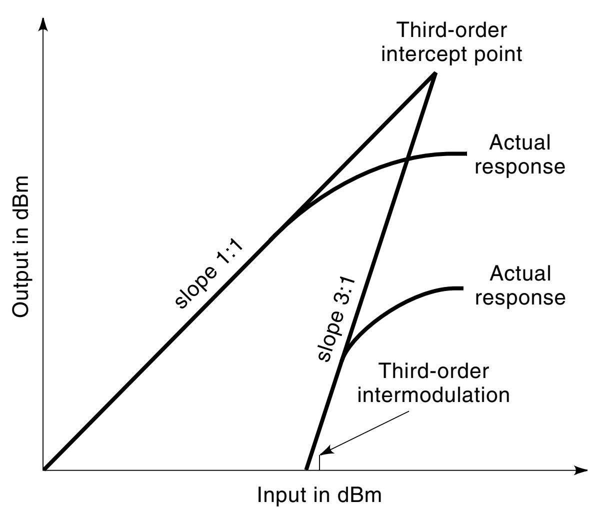

Figure 4. Third-order intercept point. The fundamental output has a slope of 1 : 1 and the third-order output has a slope of 3 : 1. The third-order intercept point is where the extrapolated fundamental and third-order outputs intercept. The third-order intermodulation is also shown.

The third-order intercept point of many microwave components is provided by the manufacturers.

Third-Order Intercept Point of Cascade Components

The overall third-order intercept Q_i of N components connected in cascade is given by:

$$Q = \frac{G_1 G_2 \cdots G_N}{\dfrac{G_1}{Q_1} + \dfrac{G_1 G_2}{Q_2} + \cdots + \dfrac{G_1 G_2 \cdots G_N}{Q_N}} \tag{12}$$

where Q_1, Q_2, . . . are the third-order intercept point of the first, second . . . components. If a component is passive, for example, as with an attenuator, a high third-order intercept value can be assigned to the device and its effect will be negligible from that seen in Eq. (12).

Two-Signal Spur-Free Dynamic Range

This dynamic range is usually applied to receivers that can process simultaneous signals. The lower limit is the sensitivity level. The upper limit is reached when two strong signals of equal amplitude begin to generate a detectable third-order intermodulation. For convenience, the detectable level is often chosen to be the noise floor. With this assumption, the two-signal spur-free dynamic range can be written as

$$DR = \frac{2}{3}(Q - G - N_o)\ \text{dB} \tag{13}$$

where Q is the third-order intercept point, G is the gain of the RF chain and N_o is the noise power at the output of the RF chain and can be expressed as

$$N_o = -114 + F + 10\log(B_R)\ \text{dBm} \tag{14}$$

where -114 is the noise floor of 1 MHz bandwidth, F the overall noise figure of the receiver, and B_R is the RF bandwidth. To achieve a certain sensitivity and dynamic range in a receiver, the gain of the RF chain must be of a specific value. Although higher gain may improve sensitivity, it degrades the dynamic range. However, a higher third-order intercept point than the designed value will not produce any adverse effect.

Two-Signal Instantaneous Dynamic Range

This dynamic range is applicable to receivers that can process simultaneous signals. It represents the capability of the receiver to receive a strong as well as a weak signal simultaneously. The instantaneous dynamic range is defined as the maximum amplitude separation between two simultaneous signals such that the receiver can measure both correctly. Generally speaking, this dynamic range depends on the frequency separation of the two signals. When two signals are close in frequency, the instantaneous dynamic range is low. When the frequencies of two signals are widely separated, the instantaneous dynamic range is high.

If a microwave receiver is used to intercept radar signals, two additional parameters are important: the throughput time (throughput rate) and delay time.

Throughput Time (Throughput Rate)

If a receiver can process only one signal at a time, the throughput time is the shortest time between two pulses the receiver can process. If the receiver can process N simultaneous signals the throughput time is the shortest time between two groups of N simultaneous pulses the receiver can process. The information of a pulsed signal, that is, RF and pulse amplitude, can be obtained from the front of the pulse, but the pulsewidth must be measured at the end of the pulse. Therefore, the throughput time is pulsewidth dependent. To keep the throughput time an intrinsic characteristic of the receiver, it should be measured with signals of minimum pulsewidth. The inverse of the throughput time is called the throughput rate.

Delay Time

The delay time is measured from the time a pulse reaches the input of a receiver to the time it is completely encoded. If delay lines are used in the receiver to store information temporarily, the delay time can be only a few microseconds. Delay time is important for intercept receivers used to generate information to respond on the same signal. If delay time is too long, the information can not be used by a jammer to respond on the same signal.

CLASSIFICATION OF RECEIVERS

There are many ways to classify microwave receivers. One popular classification uses operating frequency range, that is, Ku band, X band or extremely high frequency (EHF) receivers. Another way of classifying them is by application, for example, satellite receiver, global positioning system (GPS) receiver, etc. Even the intercept receivers can be subdivided by application, for example, warning, electronic intelligent (ELINT) receivers, etc. Although these classifications can reveal some specific information about the receiver, they do not reveal the technology upon which the receiver is based.

In this article receivers are classified by their structure and only intercept receivers will be discussed. These receivers are designed to receive different kinds of signals and all receivers have similar input. Different techniques can be used to build intercept receivers. These techniques, fundamental to receiver designs, are adopted in many other types of receivers. For example, the superheterodyne technique is used in most receiver designs. Almost all communication receivers irrespective of their operating frequency and applications often use superheterodyne techniques.

A very important factor in a receiver is whether the receiver can process multiple simultaneous signals. Since it is usually easy to isolate one desired signal, it is relatively easy to build a receiver that can process only one signal at a time. Most commercial communication receivers belong to this category. A scanning receiver can listen to many stations in a sequential manner, but it can process only one signal at a time. It is relatively difficult to build a receiver that processes more than one signal, especially when high instantaneous dynamic range is required. High instantaneous dynamic range requires the receiver to distinguish a weak signal from a spurious response, which is difficult to achieve. For low instantaneous dynamic range requirement, the problem is not as severe. For example, a GPS receiver must receive simultaneous signals from many satellites. As the signals from different satellites have about the same amplitude, the required instantaneous dynamic range is low and it is relatively easy to build such a receiver. If a large number of simultaneous signals need to be processed, considerable hardware will be required and the receiver can become rather complicated.

The receivers are divided by structure into two major groups: analog and digital. Comparatively speaking, analog receivers are technologically more mature. In fact, most commercial and military receivers are analog. Therefore, there are several techniques used in designing analog intercept receivers that will be discussed here. Although the digital receiver is in its infancy, it is anticipated that this kind of receiver will become popular because advances in digital hardware and software can be applied to receiver design and processing.

ANALOG INTERCEPT RECEIVERS

This discussion is limited to radar intercept receivers. The receivers will be divided into six types depending on their structures. The first three types, which process only one signal at a time, are the crystal video, superheterodyne and instantaneous frequency measurement (IFM) receivers. The next three types, which can process simultaneous signals, are the channelized, Bragg cell and compressive receivers.

In a radar intercept receiver five parameters will be measured. These parameters are RF, angle of arrival (AOA), pulse amplitude, pulsewidth, and time of arrival (TOA). Pulse amplitude is measured from the amplitude of the detected video pulse. Pulsewidth is measured from the width of the video pulse. The TOA is measured from the leading edge of the video pulse. These three parameters are measured similarly for different receivers. The AOA is measured from several antennas and receivers combined together and will not be included in this article. A digital EW receiver must also be able to generate these five parameters as output.

Crystal Video Receiver

This is the simplest analog receiver and has existed for many years. The receiver consists of an RF chain and a crystal video detector. In the RF chain there is a wide bandpass filter and RF amplifiers. Along with the detector there is a video filter, a video amplifier and a comparator as shown in Fig. 5. Sometimes, the RF amplifiers are not available to cover the desired bandwidth or they do not have sufficient gain. Under this condition, the sensitivity of the receiver depends on the sensitivity of the detector and the tangential sensitivity can not be calculated from Eq. 11. As a result, sensitivity will be poor. To improve sensitivity, the detector is occasionally biased in the forward direction as shown in Fig. 5. However, improving the sensitivity of the detector may decrease bandwidth. At the output of the detector a video filter will be used to limit the output bandwidth. The video filter bandwidth should match the shortest pulsed signal anticipated. For example, if the shortest pulse anticipated is 100 ns, the video filter will be approximately 10 MHz ($1/100 \cdot 10^{-9}$). The video amplifier is used to amplify the video level to a level that can be properly processed. Sometimes a logarithmic (log) video amplifier is used instead of a video amplifier. A log video amplifier takes

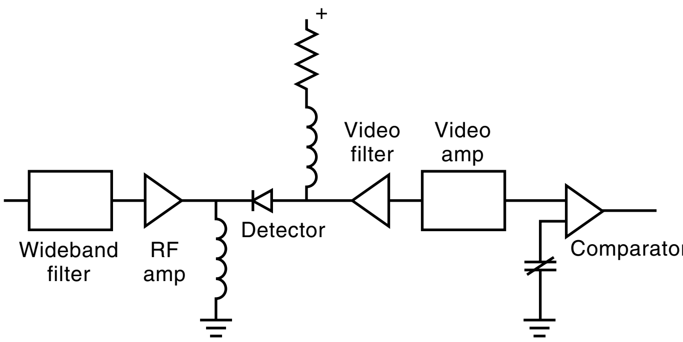

Figure 5. A basic crystal video receiver. This receiver consists of a wideband filter, an RF amplifier chain, a biased detector, a video amplifier and a comparator. The wideband filter is used to limit out-of-band signals. The RF amplifier is used to amplify the input signal. The detector is used to change the RF signal to a video signal. The bias applied to the detector is to increase the detector sensitivity. Video filter is designed to match the anticipated minimum pulse width to maximize the detection efficiency. The video amplifier is used to amplify the video signal. The comparator is used to detect signals crossing a certain threshold.

a video signal as input and generates a video output that is proportional to the logarithm of the input signal. Finally, a comparator is used to determine whether a signal is crossing a certain threshold.

A crystal video receiver usually has a very wide input bandwidth and often covers a bandwidth of an octave or more. A crystal video receiver does not provide frequency information. The only frequency information is the detected signal within the input band of the receiver. The receiver can measure pulse amplitude, pulsewidth and TOA. When simultaneous signals arrive at the input of the receiver, the receiver can receive and process all of them. However, the receiver does not have the capability to indicate the existence of simultaneous signals. The pulse amplitude measured will be the sum of the amplitudes of the simultaneous signals if the frequency separation of the signals is much greater than the video bandwidth. However, when this condition is not met, the pulse amplitude measured will be the vector sum of the signals. The pulsewidth measured will be from the first leading edge to the last trailing edge of the pulses in the group. The TOA measured will be the first leading edge of the group.

Because of its relatively low sensitivity and poor frequency accuracy, this type of receiver is no longer widely used. However, due to simplicity of the receiver, sometimes it is used to provide AOA information. The AOA information is often obtained through amplitude comparison from four directional antennas and four crystal receivers. In this application, log video amplifiers are used after the detectors. The difference between two adjacent channels is equal to the amplitude ratio due to the logarithmic relationship. The amplitude ratio can be converted into AOA information.

Superheterodyne Receiver

In the superheterodyne concept, the input signal is changed from one frequency to a different frequency while also maintaining all of the signal information. This is very important to the technology, with most of today's communication receivers using superheterodyne techniques. Almost all receiver sys-

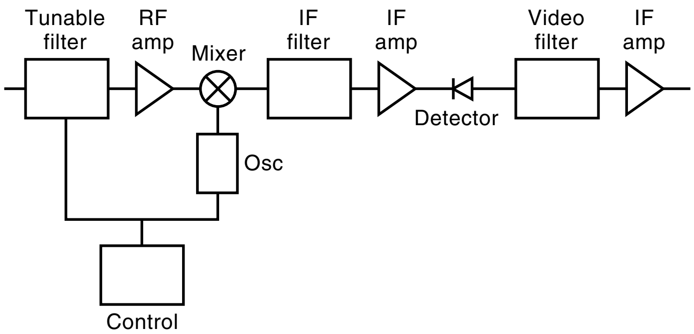

Figure 6. A basic superheterodyne receiver. This receiver consists of a tunable filter, an RF amplifier chain, a mixer and local oscillator, an IF filter, IF amplifiers, a detector, a video amplifier and a video amplifier. The tunable filter and the local oscillator are controlled by the same control circuit and their frequencies are tracked to filter out unwanted input signals. The RF amplifier is used to amplify the input signal. The mixer and local oscillator combined changes the input frequency to the IF. The IF filter is used to limit spurious signals generated by the mixer. The IF amplifiers are used to amplify the IF signal. The detector changes the RF signal into a video signal and the video amplifier is used to amplify the video signal.

tems use this methodology at one stage or another. For example, an intercept receiver may have an instantaneous bandwidth of 1 GHz and an input band of 18 GHz. One common approach is to divide the input band into eighteen 1 GHz bands and convert each band to a common frequency range through the superheterodyne technique. The intercept receiver is time-shared among these bands to measure the input signals.

A generic superheterodyne receiver is shown in Fig. 6. In this design, the first element is a tunable filter and an RF amplifier. If higher sensitivity is desirable, an RF amplifier can be placed in front of the filter. A combination mixer/oscillator is used to change the input frequency to a different frequency, usually referred to as the IF. The IF can be either higher or lower than the input signal. If the IF is higher than the input frequency, it is called *up conversion.* If the IF is lower than the input, it is referred to as *down conversion.* The IF signal is further amplified and filtered before reaching the crystal detector. Sometimes an IF logarithmic (log) video amplifier is used. An IF log video amplifier takes IF as input and generates a video signal proportional to the logarithm of the input. In an IF log video amplifier, many IF amplifiers and detectors are used. Each detector covers approximately 10–15 dB of dynamic range. Therefore, the amplifier can cover a wide dynamic range and provide high accuracy. Along with the detector there are video amplifiers and video filters to shape the video signal.

A mixer is a nonlinear device that converts a signal from one frequency to another. In order to change frequency the device must be nonlinear. However, in receiver design, a mixer is considered to be a linear component with a negative gain (loss) and a third-order intercept point. The desired output IF frequency f_{IF} of a mixer is

$$f_{IF} = f_i \pm f_o \quad \text{or} \quad f_{IF} = f_o \pm f_i \tag{15}$$

where f_i is the input frequency and f_o is the oscillatory frequency. The plus sign is used for up conversion, the minus sign for down conversion. If $f_i > f_o$, the first part of the equation is used when $f_o > f_i$ the second part is used. The input frequency is usually down converted to a lower IF, because amplifiers and narrowband filters are more available at lower frequencies. The IF is fixed at a certain value and the frequency of the oscillator is tuned across the input bandwidth of the receiver to find signals. Once the input signal is converted to the IF, the signal will be detected and the video signal will be processed. The frequency of the input signal can be measured from the preceding equation, because f_{IF} and f_o are known.

When the input frequency is higher or lower by f_{IF} than the local oscillator frequency, they will be detected by the receiver; these are images of each other. If the input bandwidth is not properly limited, both the signal and its image can be received by the receiver. An image rejection mixer can be used to separate signals above the local oscillator frequency from signals below it. However, it is often desirable to limit the input band of the receiver either above or below the local oscillator frequency. Limiting the bandwidth to one side of the local oscillator frequency also reduces the noise by 3 dB.

As a mixer is a nonlinear device, many other frequencies will be generated in addition to the desired frequency. These extraneous frequencies are referred to as spurs. The frequencies f_I of the spurs, including the desired IF, can be determined by:

$$f_I = Mf_1 + Nf_2 \tag{16}$$

where f_1 and f_2 are used to represent the input and oscillator frequencies; M and N are integers (either positive or negative) and the output frequency f_I must be a positive value. In order to limit the spurious outputs, two filters are often used in a superheterodyne receiver. An IF filter with a fixed center frequency f_{IF} is used to reject spurs at the output of the mixer. Because the input bandwidth is much wider than the IF filter, several signals present at the input of a mixer will cause it to generate spurs. A tunable filter with bandwidth comparable to the IF filter can be placed at the input of the receiver to limit the instantaneous bandwidth and reduce spur generation. The tunable filter and the local oscillator must be synchronized and the difference frequency between them must equal f_{IF}. Thus, one control unit is often used to tune the tunable filter as well as the oscillator. The frequency range of the input filter and the oscillator must be wide enough to cover the input bandwidth of the receiver.

Because the filter bandwidth of a superheterodyne receiver is very narrow, the sensitivity is high. The spurious responses generated by the mixer are carefully filtered, so the dynamic range is usually high. It is relatively easy to build superheterodyne receivers with matched performance, that is, amplitude and phase, because of the narrow bandwidth. The probability of intercept of a superheterodyne receiver is low and it can not process simultaneous signals. Therefore, a superheterodyne receiver alone is seldom used as an intercept receiver. It is often used as part of an intercept system such as one that measures AOA information through multiple antenna and receiver combinations.

IFM Receiver

An IFM receiver uses the autocorrelation function to measure the input frequency. A signal is correlated with its delayed

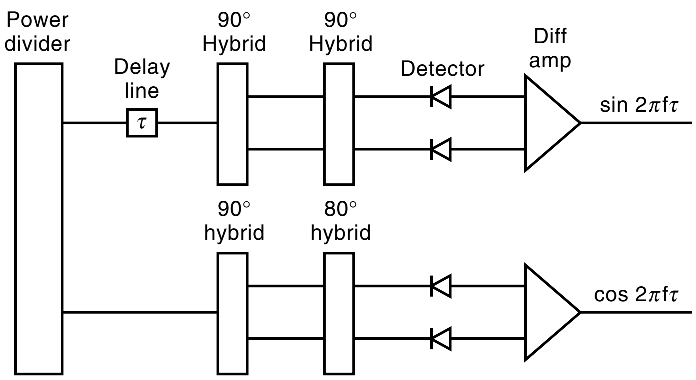

Figure 7. A correlator for the IFM receiver. The correlator consists of a power divider, a delay line of known delay time, three 90° and one 180° hybrids, four detectors, and two differential amplifiers. The power divider divides the input signal into two parallel paths and the signal in one path is delayed by a known time. The four hybrids are used to provide the necessary phase shifts. The detectors are used to convert RF signals into video signals as well as to perform multiplication of two signals. The two differential amplifiers are used to cancel a dc bias term. The outputs from the correlator are $\sin(2\pi ft)$ and $\cos(2\pi ft)$.

version. The outputs are lowpass filtered to generate the desired video signals, which in turn can be used to produce the frequency information. There are many different ways to build a correlator by using different RF components such as 90° hybrids and in-phase power dividers. One common approach is shown in Fig. 7.

The input signal is divided into two paths and in one of the paths a known delay time τ is added. Four hybrids are used to obtain the desired phase relations. The detectors are used to perform multiplication. If the input frequency is f, the outputs of the detectors consist basically of two terms—a double frequency term and a dc term. The outputs of the detectors are lowpass filtered to stop the high frequency and pass the dc components. They represent the autocorrelation functions of the input signal. The differential amplifiers are used to remove a constant term in the dc components and the outputs are $\sin(2\pi f\tau)$ and $\cos(2\pi f\tau)$. As the delay time is known, the frequency of the input signal can be found as

$$f = \frac{\theta}{2\pi\tau}$$

where (17)

$$\theta = \tan^{-1}\left(\frac{\sin(2\pi f\tau)}{\cos(2\pi f\tau)}\right) = 2\pi f\tau$$

By measuring the $\sin(2\pi f\tau)$ and $\cos(2\pi f\tau)$, the angle θ can be found and the frequency can be calculated.

As the angle θ calculated from the sine and cosine is less than 2π, this relation limits $f\tau < 1$. If the desired input bandwidth is 2 GHz, the delay time must be less than 0.5 ns. Using this τ value to measure frequency, the accuracy is rather poor because of the poor angle resolution. In order to cover a wide instantaneous bandwidth and at the same time produce fine frequency accuracy, several correlators are needed. Correlators with short delay times are used to resolve ambiguity; correlators with the longest delay line provide frequency accuracy. The longer the delay line, the better the frequency accuracy a correlator can provide. The longest delay line must be shorter than the shortest pulse anticipated and allows the pulse and its delayed version to have sufficient overlap. The delay line lengths are commonly selected to be multiples of each other. The two common delay line ratios are 1 : 2 and 1 : 4. Two examples will be used to illustrate these two design ideas.

To cover a 2 GHz bandwidth, it is common practice to select the unambiguous bandwidth wider than the desired value, that is, 2.56 GHz, and the corresponding shortest delay time is 0.390625 ns (1/2.56 GHz). In the 1 : 2 ratio case, the delay line lengths are 1, 2, 4, 8, 16, 32 and 64 and the shortest delay line is considered as unit length. There are seven correlators in this design. In the 1 : 4 ratio case, the delay line lengths are 1, 4, 16 and 64 and there are only four correlators. The correlators with the longest delay line have the same design in both cases and they provide the frequency accuracy (usually about 1 MHz). Decoding schemes for the other correlators are slightly different. The decoding scheme of the correlators with a 1 : 2 ratio is simpler than the ones with a 1 : 4 ratio because the former one is only required to generate 1 bit and the latter one must generate 2 bits of information. Therefore, both approaches are popularly adopted in IFM receiver designs.

An IFM receiver uses a very unique RF front end design. The RF chain of most receivers uses linear components to avoid generating spurs and retain the amplitude information of the input signals. In an IFM receiver it is common practice to use a limiting amplifier in the RF chain. This kind of amplifier raises all signals above a certain threshold to a constant level. Therefore, amplitude information is lost with the limiting amplifier. A separate circuit must be used to measure pulse amplitude. As the limiting amplifier is a nonlinear device, it generates strong spurious responses. When two signals are present in the amplifier, the strong one will suppress the weak one. This is referred to as the *capture effect*. Because an IFM receiver can process only one signal at a time, the capture effect will enhance the receiver performance to measure the strong signal under simultaneous signal conditions.

An IFM receiver has many advantages over other types of intercept receivers. The receiver can cover a very wide instantaneous bandwidth [possibly 16 GHz (2 to 18 GHz)] and provide fine frequency accuracy on short pulses, for example, 1 MHz accuracy on 100 ns pulse. As the structure of the receiver is very simple, the receiver can be very compact, low cost, and reliable. The only deficiency is that the receiver can not process simultaneous signals. Not only is it unable to process simultaneous signals, but simultaneous signals with amplitude within about 5 dB may cause the receiver to produce an erroneous frequency report without reporting the mistake. This is considered a major problem with the IFM receiver and limits its usage. However, the concept of the IFM receiver is very important and can be adopted in many receiver designs, including digital receivers.

Channelized Receiver

A channelized receiver can intercept simultaneous signals. The basic concept is very simple. It uses a bank of filters with adjacent frequencies to separate signals. Signals with different frequencies will exit from different filters. By measuring

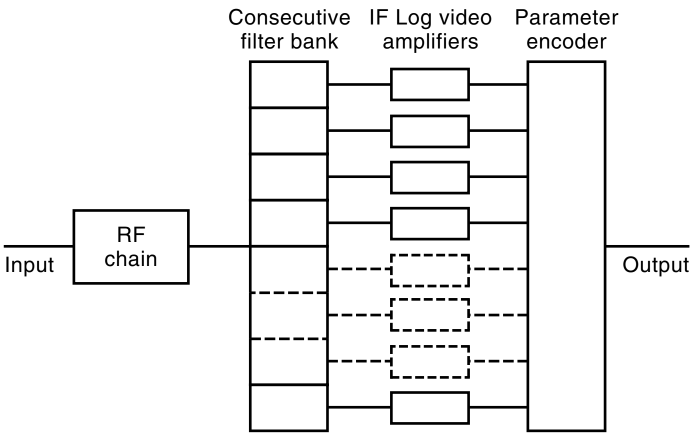

Figure 8. A basic channelized receiver. The receiver consists of an RF amplifier chain, a filter bank with contiguous filters, IF log video amplifiers and a parameter encoder. The RF amplifier is used to amplify the input signal. The filter bank is used to separate signals with different frequencies into different output ports. The IF log video amplifiers amplify the IF signals and covert them into video signals. The parameter encoder generates the desired digital information from the input video signals.

the outputs of the filters one can determine the frequencies of the input signals. A basic channelized receiver is shown in Fig. 8 and it consists of four major components. The first one is the RF chain. The second component is a filter bank with consecutive center frequencies. In order to retain the amplitude information, IF log video amplifiers are used after the filters. The last component is a parameter encoder, which takes the outputs from the filters and converts them into the desired information. These components are discussed here.

The RF chain usually consists of amplifiers, filters and mixers to shift the input frequency. The RF chain must operate in the linear region to avoid generation of spurs. As the bandwidth of a channelized receiver is wide, some spurs generated from mixers will not be filtered out but be present in the output.

The output of the RF chain is fed to the input of the filter bank. A common way to feed the filter bank is through a power divider. Using a power divider can improve impedance matching to the filters, but they cause insertion loss. Every time a signal is divided into two paths there is a 3 dB loss. These losses must be recovered by placing additional amplifiers in the receiver. Although a frequency multiplexer is a better approach to feed the input of the filter, it is usually difficult to achieve the desired filter requirements. In each filter usually only one signal can be processed. If more than one signal is in one filter, they will be processed as one signal and may produce erroneous results. Depending on the center frequency of the filters and their bandwidth, different techniques can be used to build the filters, for example, surface acoustic wave (SAW) technology and lumped *LC* elements. The two general requirements of these filters are low insertion loss and small size. The filter must also have low sidelobes in the frequency and time domains. The required shape of the filter is usually based on the encoding circuit design.

The outputs of the filters are further amplified by IF log video amplifiers. If the amplitude information after a filter is not of interest, a limiting amplifier followed by a crystal detector can be used to convert the IF into a video signal. With only one signal processed within a filter, intermodulation and spurs are not of concern. Therefore, in designing a receiver, the third-order intercept point in the IF channel is not of concern. If amplitude is not retained after the IF amplifiers, pulse amplitude information must be obtained from another part of the receiver.

A parameter encoder takes the video signals from all of the filters as input and produces the desired information. Most of the effort in encoder design is directed toward obtaining frequency information. There are usually two ways to obtain frequency information. One is to compare amplitudes from adjacent filter outputs. In this approach, IF log video amplifiers must be used to generate amplitude information at the filter outputs. One problem with this approach is that it is difficult to balance the gain in all of the channels. If the gain from one channel changes slightly, that is, due to temperature drift, the encoder must be adjusted accordingly. Sometimes it can be a major problem in a receiver with many channels. Another approach is to detect the transient response of a filter output. If a pulsed signal passes the center of a filter the transient effect is not significant, which means the pulse shape is slightly distorted. On the other hand, if a pulsed signal passes the skirt of a filter, the transient effect is very significant, which means the pulse shape is drastically distorted. By measuring the transient on the output pulse one can determine whether a signal is in the middle or on the skirt of a filter. In this design, both limiting amplifiers and IF log amplifiers can be used after the filters. The problem with this approach is that the variation on the leading edge of the pulse can change the transient response and the performance of the encoding circuits. In some receiver designs both the amplitude comparison and transient phenomenon are used to obtain better results. The filter shape, which controls the amplitude and the transient of the video signal, is often determined from the encoding circuit design. The encoder design is the most critical element in building a channelized receiver.

In a receiver that can process simultaneous signals, there are two quantities related to frequency measurement. One is frequency resolution, which tells the receiver to separate two simultaneous signals that are close in frequency. The other one is frequency accuracy, which is related to the error in the frequency measurement. The minimum pulsewidth determines the minimum filter bandwidth, which determines the frequency resolution of the receiver. Frequency measurement must be carried out after the transient dies off. The transient is approximately equal to the inverse of the filter bandwidth. During the transient period both the shape and the RF of the pulse change. If the desired minimum pulsewidth is 100 ns, the minimum bandwidth of the filter is approximately 10 MHz and the frequency resolution is close to 10 MHz. However, the receiver can be designed with a filter bandwidth that is much wider than the minimum filter bandwidth. A wideband filter has a shorter transient time. If additional processing, that is, the concept of an IFM receiver is used, better frequency accuracy can be obtained. It is difficult to design a receiver with both high-frequency resolution and frequency accuracy on short pulses because it takes a longer time to generate a fine frequency reading. In general, the anticipated minimum pulsewidth deters the minimum filter bandwidth and thus the minimum frequency resolution. This design rule applies to all receivers with simultaneous signal capability.

Because the concept of channelized receivers is very simple, a general misunderstanding is that it is very easy to build. As a result, the design goals might be set too high to achieve. In general, a receiver has fewer problems when the filter bandwidth is wide because the receiver has fewer parallel channels. However, wider bandwidth means poor frequency resolution and lower sensitivity. When a high instantaneous dynamic range is desired, the receiver must detect a weak signal in the presence of strong signals. Sometimes it is difficult to distinguish a spurious response from a weak signal. Therefore, a high dynamic range receiver may produce a spurious signal report. When the instantaneous dynamic range requirement is low, the receiver can be designed to produce fewer spurious responses.

The input bandwidth of a channelized receiver is proportional to the number of channels and their bandwidth. When a large number of channels are built, a channelized receiver can be bulky and expensive but have better frequency resolution. If the channel bandwidth is wide, a small number of channels can cover a wide frequency range. Because the encoding circuit also has fewer inputs, the receiver can be relatively small.

Bragg Cell Receiver

A Bragg cell receiver can be considered another type of channelized receiver where channelization is accomplished by optical means. The name Bragg cell receiver derives from the concept of the Bragg angle of optical diffraction. A basic Bragg cell is shown in Fig. 9. The optical arrangement will be discussed first. A laser is used as a coherent light source. A diode laser is often preferred to a gas laser because of its small size. The beam expander and the collimator are used to form the light beam into the desired shape to shine on the Bragg cell. A Bragg cell is used to diffract input light. A Bragg cell is made of a crystal transparent to the laser light. At one end of the crystal is a transducer to change the input electric signals into acoustic signals. At the other end of the Bragg cell is absorption material used to eliminate acoustic wave reflection to avoid the generation of standing waves. The acoustic wave produces variations in the refractive index of the crystal. This refractive index modulation causes the laser beam to deflect. The time bandwidth product of a Bragg cell is defined as the time of the acoustic wave traveling through the window of the Bragg cell multiplied by the bandwidth. The laser beam is incident on the Bragg cell at the Bragg angle where the desired diffracted light is at maximum intensity. The light output from the Bragg cell passes through a Fourier transform lens and focuses on a photodetector array. The photodetector changes the optical signal back into a video signal.

The input signal to the receiver is applied to the transducer of the Bragg cell through an RF chain containing amplifiers and filters. There is an impedance matching network to match the output of the RF chain to the input impedance of the Bragg cell. The output power from the RF chain should keep the Bragg cell operating in its linear region. High input power can drive the Bragg cell into the nonlinear region and generate spurious responses. The input signal is converted into a traveling acoustic wave in the Bragg cell. The position of the diffracted light beam on the detector array is related to the frequency of the input signal. If more than one signal is present in the Bragg cell, there will be multiple outputs on the detector array. The maximum number of outputs is equal to the time bandwidth product of the Bragg cell. The amplitude of the output is related to the power of the input signal.

The video outputs from the photo detector array can be considered as outputs from a filter bank with adjacent frequencies. The parameter encoder takes these outputs and generates the desired information. In a channelized receiver, IF log video amplifiers or limiting amplifiers are used after the filters to provide more RF gain. To provide the same gain in a Bragg cell receiver, a light amplifier must be used in each output channel, however, such a device is not available with today's technology. Thus, the sensitivity of the receiver depends on the laser power and the characteristics of the photodetectors. A laser with higher power can be used to improve sensitivity. As the outputs from the photodetectors are video signals, some of the encoding schemes applicable to a channelized receiver may not be adopted for Bragg cell receivers.

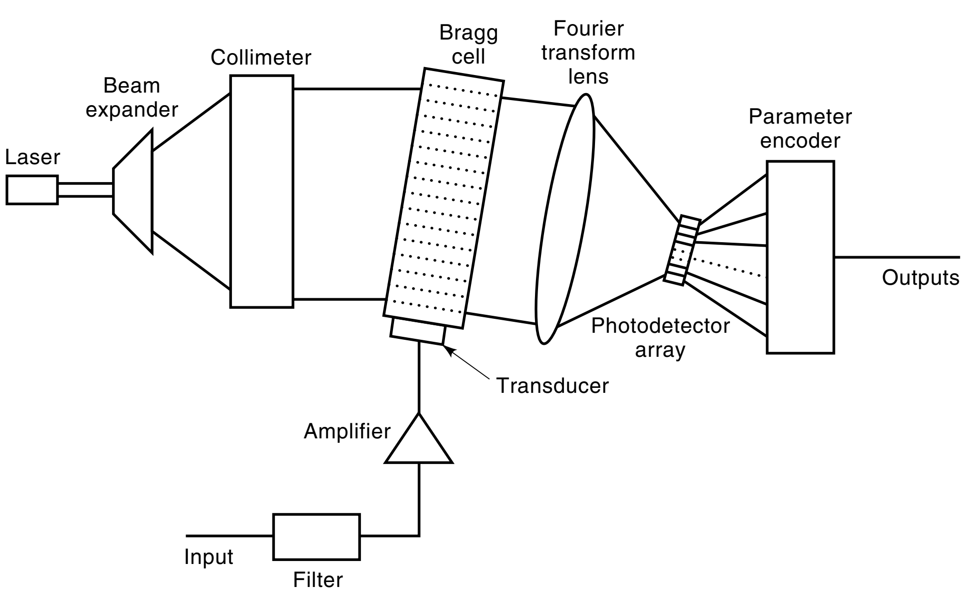

Figure 9. A basic Bragg cell receiver. The receiver consists of an RF chain, a laser, a beam expander, a collimator, a Bragg cell, a Fourier transform lens, a photodetector array and a parameter encoder. The RF amplifier is used to amplify the input signal. The beam expander and the collimator shape the laser light into desired shape. The input signal is converted into an acoustic wave in the Bragg cell to form a grating and diffract the light from the laser. The Fourier transform lens focuses the light on the photodetector array which coverts the light into video signals. The light position on the photodetector array determines the frequency of the input signal. The parameter encoder generates the desired digital information from the input video signals.

Another difference between the photodetector and the crystal detector is that the output from a photodetector is proportional to the input light power and, therefore, proportional to the input power. The output from a crystal detector is proportional to the voltage of the input signal, which is related to the square root of the input power. Therefore, a crystal detector covers more dynamic range.

Compared to a channelized receiver, a Bragg cell receiver has less dynamic range. The major advantage of a Bragg cell receiver is its simplicity and compactness. A Bragg cell receiver including the optical bench (consisting of beam forming lenses and Bragg cell), laser and detector array, but excluding the RF chain, is only a few cubic inches in size. Such a small receiver can provide over one hundred parallel outputs. The size of the optical bench can be further reduced using integrated optics. In the integrated optical approach, the entire optical bench can be fabricated on a single chip. The light is transmitted through a light waveguide and the Bragg cell uses SAW technology. All optical components can be made on the chip, and the laser and detector array can be attached from the ends of the chip. Thus, the size can be a few cubic centimeters.

The Bragg cell receiver already discussed here is often referred to as the *power Bragg cell receiver,* because the detector output is proportional to the power of the input signal. In order to improve the dynamic range of a Bragg cell receiver, an interferometric approach can be used. The main goal of this approach is to make the photodetector proportional to the voltage of the input signal rather than the power. In this arrangement, the laser beam is split into two paths through a beam splitter with a Bragg cell in each path. The input to one Bragg cell is a locally generated spread spectrum signal covering the entire bandwidth of the Bragg cell. The input signal is applied to the other Bragg cell. The outputs of the two Bragg cells are focused on a photodetector array and the location represents the input frequency. Each photodetector is used as an optical mixer and the output is an IF signal rather than a video signal. Thus, IF amplifiers and filters can be used to improve the performance of the receiver. After the IF chain, crystal detectors are used to convert the IF signals into video signals. Although this approach has the potential to improve the dynamic range of the receiver, it sacrifices the simplicity of the Bragg cell receiver. Individual IF channels have to be built separately and the optical bench is equivalent to the filter bank in a channelized receiver.

As in designing channelized receivers, the parameter encoder of a Bragg cell receiver is a major portion of the effort.

Compressive Receiver

A compressive receiver can also process simultaneous signals. As opposed to a channelized receiver where all outputs are in parallel, the outputs from a compressive receiver are in series. A basic compressive receiver is shown in Fig. 10. In this figure the RF chain is not included. The two major components in a compressive receiver are the local oscillator and the dispersive delay line. The output from the local oscillator is a repetitive FM signal. The frequency range of the FM signal can be very wide and the period very short, such as 1 GHz/200 ns. This receiver is also called a *microscan receiver* because the scan time is short. The input signal after mixing with the local oscillator output is converted into an FM signal. This FM signal passes through the dispersive delay line (often referred to as compressive line) and is compressed into a pulse. The position of the pulse relative to the beginning of the scan represents the frequency of the input signal. The amplitude of the compressed pulse represents the amplitude of the input signal.

The instantaneous bandwidth of the compressive line is generally equal to the IF bandwidth of the receiver. The input bandwidth can be either wider or narrower than the IF bandwidth. In one common design, the local oscillator scans the

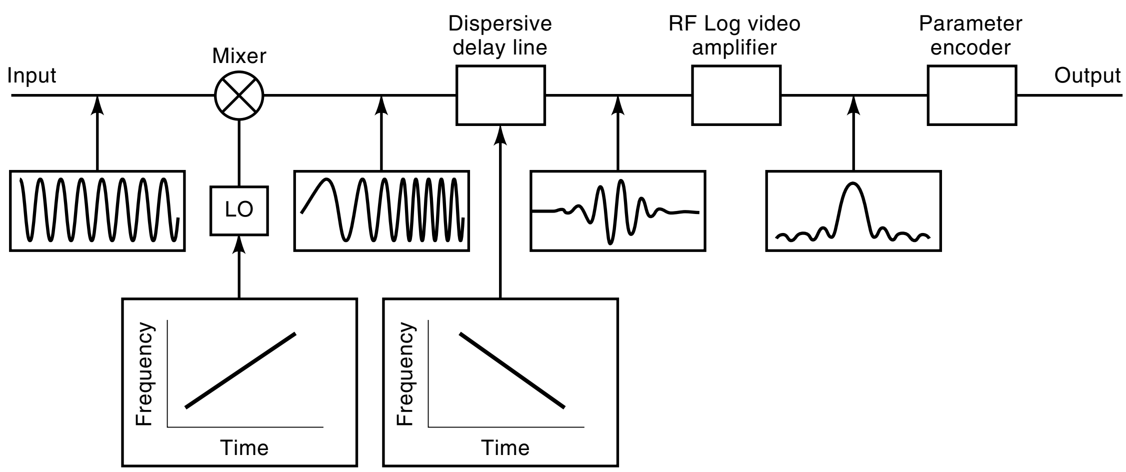

Figure 10. A basic compressive receiver. The receiver consists of a local oscillator, a mixer, a dispersive delay line, an RF log video amplifier and a parameter encoder. The local oscillator generates a linear FM signal which changes the input to a linear FM signal through the mixer. The dispersive delay line compresses the FM signal into a short pulse. The RF log video amplifier is used to generate the logarithm from the compressed pulses. By measuring the output time of the compressed pulse with respect to the beginning of the scan, the signal frequency can be determined. The parameter encoder converts the scan to scan information to a pulse by pulse information.

sum of the IF bandwidth and the receiver input bandwidth. In order to compress the FM signal from the output of the mixer into a short pulse, the frequency vs time slope of the local oscillator and the slope of the compressive line must be properly matched. In most designs, the compressive line is made from SAW technology with bandwidths of up to 1 GHz and dispersive delays of a few hundred nanoseconds. The local oscillator is often implemented from a dispersive delay line because a short pulse applied to the input of a dispersive delay line generates an FM signal at the output. This signal can be amplified and used as the local oscillator output. Because the frequency of the local oscillator is wider than the compressive line bandwidth, the requirements on the dispersive delay line to generate the local oscillator signal are more stringent. An oscillator built from a dispersive delay line is easier to match the frequency vs time slope of the compressive line. The time bandwidth product of the compressive line is defined as the dispersive delay time multiplied by the bandwidth. The maximum number of compressed pulses generated per scan is approximately equal to the time bandwidth product of the compressive line. The maximum number of compressed pulses can be considered equivalent to the total number of parallel channels in a channelized receiver.

The compressed pulse is very short and it is inversely proportional to the bandwidth of the compressive line. The pulse output rate equals approximately the bandwidth of the compressive line. If the bandwidth of the compressive line is 1 GHz, the pulsewidth is close to 1 ns (1/1 GHz) and the output rate is about 1 GHz. The center frequency of the pulse equals the center frequency of the compressive line. The compressed pulse has a main peak and many sidelobes. The sidelobes should be reduced to simplify the frequency encoding circuit design. The sidelobes can be reduced by adding a weighting (window) function to the compressive line. An IF log video amplifier is used to convert the compressed pulse into video pulses for further processing. The video bandwidth of the IF log video amplifier must be wide enough to accommodate the compressed pulsewidth.

A compressive receiver usually intercepts a pulsed signal in many consecutive scans. It is desirable to report the information on the intercepted signal on a pulse-by-pulse basis rather than a scan-by-scan basis. The parameter encoder will sort and combine the scan-by-scan information into pulse-by-pulse information. The outputs from each scan are the number of simultaneous signals intercepted during the scan time. From this operation, it is easily seen that the time resolution in generating pulsewidth and TOA equals the scan time. Thus, it is usually coarser than the time resolution in other types of receivers.

The main function of the parameter encoder is to find the frequency of the input signal. Because the compressed pulse output rate is usually rather high, the encoder must be able to process the signals at the same rate. Although the parameter encoder is quite different from a channelized receiver, they face the same basic challenges: detecting the signal and avoiding the sidelobes and spurious responses. The bandwidth of a compressive receiver is limited by the technology used in the compressive line as well as the speed of the logic circuit in the parameter encoder.

There are several different ways to design a compressive receiver. One of the most common designs is to make the receiver bandwidth equal to the IF bandwidth. Under this condition, the outputs of the compressive line are present only 50% of the time. This affects the probability of intercept. If a short pulse (less than half the scan time) falls in the silent half, the receiver will miss the pulse. To improve the probability of intercept, another mixer and local oscillator can be added in parallel. The beginning of the FM signal from this local oscillator is shifted to the middle of the original one. For example, if the scan time is 200 ns, the original scan is from 0 to 200 ns and the additional scan is from 100 to 300 ns. Outputs from both channels are combined and fed into the compressive line. With this arrangement, the compressive line can have output 100 percent of the time. This technique is called the interlace scan. The outputs from the IF log video amplifier are in series, so less hardware is required in the parameter encoder as compared with a channelized receiver; however, operational speed is very high and matches the bandwidth of the compressive line.

The minimum pulsewidth a compressive receiver can process is approximately equal to the scan time or half the scan time if the interlace scan is used. The frequency resolution is close to the inverse of the minimum pulsewidth. Compared with a channelized receiver, it shows that the same laws of physics govern the performance of both receivers, that is, the frequency resolution is inversely proportional to the pulsewidth.

DIGITAL RECEIVERS

A digital receiver consists of three building blocks: the RF chain; the ADCs; and the DSP. Once a signal is digitized the data are less affected by ambient conditions such as temperature changes. In an analog receiver, the performance of the components may change due to temperature variation and aging, while digital circuits do not have these problems. The digitized data can be processed with many different DSP approaches. Although most of the DSP are still hard wired for receiver applications, increasing the processing speed may permit changing receiver functions through software switching. Thus, the software receiver concept is becoming popular. With this concept many different receivers can be implemented using the same hardware. The software receiver idea is particularly popular for military communication receivers because of its potential versatility.

Strictly speaking, an ADC is a nonlinear device. It can be considered as a linear device for a large number of quantization levels. The discussion in the following sections is based on the linear model of an ADC. Nonlinear operation still exists if strong signals drive the ADC into saturation. Under this condition, the output of the receiver may produce many spurious responses. Special procedures must be considered to deal with this phenomenon.

To cover a wide bandwidth, the ADC must operate at high speed. The Nyquist sampling theorem requires that the minimum sampling speed be twice the information bandwidth. To cover a wide dynamic range, the ADC must have a large number of bits. The number of quantization levels is related to the number of bits b as 2^b. If the dynamic range is defined from a signal at the highest level to a signal at the lowest level, it can be readily expressed as

$$\mathrm{DR} = 20\log(2^b) \approx 6b \tag{18}$$

In this equation, the ADC is assumed ideal, that is, the quantization levels are uniform and there is no jitter in the sampling window. In a typical ADC, there are noise, nonuniform quantization level and sampling window jitter. An effective bit is often used to characterize a nonideal ADC. The number of effective bits is less than the actual number of bits and is defined as[2]

$$b_{\text{eff}} = b - \log_2 \left\{ \frac{\text{rms error (actual)}}{\text{rms error (ideal)}} \right\} \tag{19}$$

The effective bits change with frequency and have fewer bits at higher frequency. The number of effective bits should be used in Eq. (18) to determine the dynamic range. If the highest spurious response is used as the lower limit of a receiver, the dynamic range varies. However, Eq. (18) provides a simple estimation.

Digital receivers can be divided into two generic groups: narrowband and wideband receivers. Usually, one can consider a narrowband receiver to receive only one kind of signal and the receiver is not required to process simultaneous signals. For each type of signal a narrowband receiver will be designed individually. There are exceptions to this definition. For example, a GPS C/A (coarse/acquisition) code receiver has a narrow bandwidth, because the input signals have the same frequency. However, this receiver can receive multiple signals with different code. A wideband receiver can receive simultaneous signals over a wide frequency range. The input signals can be either known or unknown. For a wideband communication receiver the signals are known. For an intercept receiver the signals are unknown.

Another way of differentiating narrowband and wideband digital receivers is by the frequency tuning schemes. In a narrowband receiver, the frequency tuning is accomplished through analog means by changing the frequency of the local oscillator and selecting the proper filters. For this type of tuning, input to the ADC is usually an isolated signal rather than the full input bandwidth of the receiver. Thus, the ADC and the DSP following it can operate at a lower frequency.

In a wideband digital receiver the frequency tuning method is implemented digitally. The sampling frequency of the ADC and the following DSP must be high enough to accommodate the input bandwidth of the receiver. For example, one can build an FM radio through either a narrowband or a wideband approach. The input bandwidth of an FM radio is 20 MHz (88 to 108 MHz). If a single FM station is selected by analog tuning, the signal bandwidth is 200 kHz. If the sampling rate is 2.5 the signal bandwidth, an ADC with a sampling speed of 500 (2.5 × 200) kHz is required. If the frequency will be tuned digitally, the sampling speed must be 50 (2.5 × 20) MHz. Although the signal information bandwidth is only 200 kHz, the following DSP must match this operational speed of 50 MHz. The potential advantages of digital tuning are superior filter shape and flexible tuning ability.

In the following sections, a narrowband receiver with a band folding concept and a wideband channelized receiver will be presented.

Narrowband Digital Receiver

The Nyquist sampling theory requires that the minimum sampling speed be twice the signal bandwidth, not twice the highest frequency of signals. For example, the C/A code of the GPS signal is at L_1 band (1575.42 MHz) with a bandwidth of 2 MHz from null to null. The minimum sampling frequency to acquire this signal is 4 MHz, although the signal is at 1575.42 MHz. If the input bandwidth of the ADC can accommodate the 1575.42 MHz frequency, the signal can be sampled directly at slightly higher than 4 MHz. Sampling will alias the input signal to a baseband as shown in Fig. 11. In this figure the sampling frequency is f_s, thus, the maximum unambiguous bandwidth is $f_s/2$. All of the input bandwidth from 0 to $(n + 1)f_s$ will fold into the bandwidth $f_s/2$. From this frequency folding property, the RF chain of the receiver should be designed with at least two bandpass filters. The first one should be placed near the front of the RF chain to reject out-of-band signals as in an analog receiver. The second filter should be placed in front of the ADC to limit the out-of-band noise generated from the amplifiers in the RF chain.

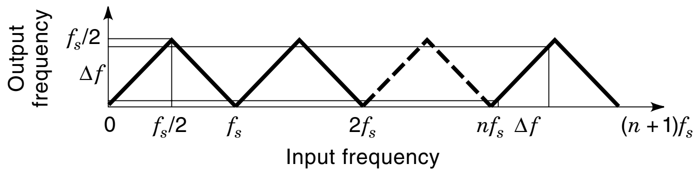

Figure 11. A sampling scheme transferring the input to baseband. The sampling frequency is f_s, the maximum unambiguous bandwidth is $f_s/2$. A signal with bandwidth of $\Delta f < f_s/2$ in the frequency range of nf_s to $(n + 1/2)f_s$ can be transferred to the baseband of 0 to $f_s/2$ through sampling.

One can design a receiver to receive several narrowband signals separated in frequency through sampling at a proper frequency. An example will be used to illustrate this approach. The Y code of the GPS signal at L_2 band has a bandwidth of 20 MHz centered at 1227.6 MHz. If one desires to receive the C/A code at L_1 and the Y code at L_2 of the GPS signals, the total bandwidth is 22 MHz (2 + 20). The minimum required sampling frequency should be 44 MHz (2 × 22). Some specific sampling frequencies can be used to fold the two signals into the baseband. One sampling frequency is 51.6 MHz. Under this condition, the Y code is aliased to 5 to 25 MHz and the C/A code is aliased to 0.62 to 2.62 MHz. These two frequency bands are not overlapped in the baseband. In general, it is desirable to fold the two signals into separate regions of the baseband to avoid interference. If interference is not a problem, it is possible to overlap the two signals to save bandwidth. However, overlapping two signals into one frequency range also folds the noise together. The noise floor will increase at the overlap region. This concept can be extended to more than two signal bands.

Wideband Digital Receiver

The most popular wideband digital receiver is a digital channelized receiver. The basic idea is the same as it is for an analog channelized receiver—to separate the input into many parallel consecutive frequency channels. In a digital channelized receiver, the filter bank is implemented digitally. To cover a wide bandwidth and high dynamic range, the ADC must operate at very high speed and have many output bits.

The simplest way to build a channelized digital receiver is to use discrete Fourier transform (DFT) implemented using

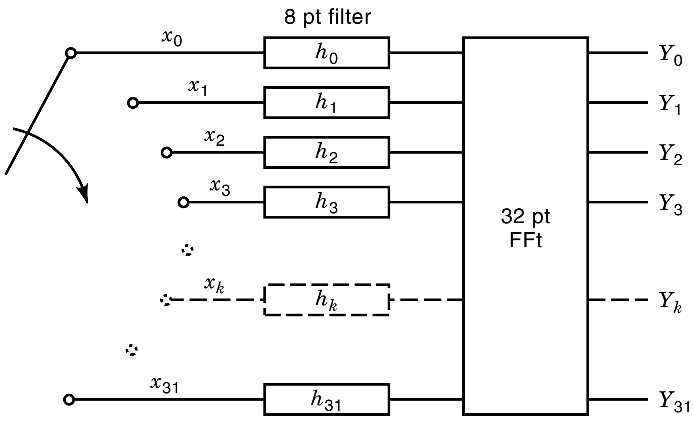

Figure 12. A 32-output filter bank using decimation scheme. The main goal is to perform a 32-point FFT with 256 input data points. The input is decimated into 32 parallel channels. In each channel, there is an 8-point filter. The output from each filter contains 8 input data points. These outputs are used as the input to the 32-point FFT.

the fast Fourier transform (FFT). The FFT should operate on the input data on a continuous manner. The FFT outputs from different time intervals can be considered as the outputs of each individual filter. This technique requires the FFT to operate at high speed. If a receiver covers a 1 GHz bandwidth, the sampling speed must be above 2 GHz. Suppose that the sampling speed is 2560 MHz and a 256 point FFT can produce 256 parallel channels. However, only 128 channels provide useful information, and the other 128 outputs are the complex conjugate of the first ones. It takes 100 ns to accumulate 256 data points and the output channel bandwidth is 10 MHz ($1/100 \times 10^{-9}$ s). As the bandwidth of the receiver is 1 GHz, only the center 100 outputs will be used and the remaining 28 end channels will not be monitored. To prevent missing data, the FFT must perform 256 point FFT at least every 100 ns. If any overlap of the FFT is desired, the processor must operate at a faster speed.

In general, it is difficult to match the digital processing speed to the speed of the ADCs. One common way to solve this problem is to reduce the total number of output channels, but keep the input at 256 data points. This approach reduces the frequency resolution of the receiver. For example, one can perform a 32-point FFT to reduce the output to sixteen 80 MHz channels. Under this condition, a bandpass filter will be used to limit the input bandwidth to 1 GHz, which will partially block the input of the two end channels. The input data are decimated 32 times, then each 8-data points will convolve with an 8-point filter as shown in Fig. 12. The coefficients of the 32 filters are obtained from decimating a 256-point filter, which is designed to shape the frequency response of the 16 filters. The 32 outputs from the filters become the input to the 32-point FFT. In order not to miss data, a 32-point FFT must be performed every 100 ns, which is a less stringent requirement than the previous case. If the FFT can operate at a higher speed, the input data can be processed in an overlap mode, that is, performing FFT every 50, 25, or 12.5 ns. This overlap mode is a common practice in channelized receiver design. Finally, the outputs from the FFT must be correctly decoded to generate the desired information. If the receiver is used to intercept radar signals, the encoder will provide the five parameters: frequency, AOA, TOA, pulse amplitude, and pulse width.

Hybrid Receivers

One common hybrid receiver design is to build a wideband receiver such as channelized or compressive receiver with a coarse frequency resolution. For example, one can divide the input bandwidth of 1 GHz into 10 to 20 uniform bands with a frequency resolution of 100 to 50 MHz. Once a signal is detected and its coarse frequency is measured, a narrowband receiver or receivers can be used to obtain fine grain information on the signal. One approach is to rapidly tune a narrowband IFM receiver to measure the frequency of the input signal, because the IFM receiver can provide improved frequency resolution. Another design is to rapidly tune several narrowband receivers which are connected to different antennas to measure the AOA of the input signal. In this design, it is easier to match the amplitude and phase of narrowband receivers. It is a common practice to combine these two approaches in one hybrid receiver design. These approaches can be implemented in both analog and digital receivers. In an analog hybrid receiver design, RF delay lines must be used to temporarily store the input signal while the narrowband receiver or receivers can be tuned to the desired frequency. Wideband low loss RF delay lines are not available and this is one of the major problems in analog hybrid receiver design. In digital design, since the digitized data can be stored easily,

Table 1

	Crystal Video	Superheterodyne	IFM	Channelized	Bragg[a] Cell	Compressive	Digital[b] Channel
Instant. BW	very wide	narrow	very wide	wide	wide	wide	wide
Simult. signal capability	none	none	none	good	good	good	good
Frequency accuracy	Poor	excellent	excellent	good	good	good	good
Sensitivity	poor-fair	excellent	good	good	fair-good	good	good
Single-signal DR	fair	excellent	excellent	good	fair-good	good	good
Two-signal instant DR	N/A	N/A	N/A	good	fair	good	good
Two-signal spur-free DR	N/A	N/A	N/A	good	fair	good	good
Structure	simple	moderate	simple	complex	moderate complex	complex	complex
Size	small	small/moderate	small	bulky	small-moderate	moderate	bulky

[a] This represents the power Bragg cell receiver. The interferometric Bragg cell receiver performance should be comparable to channelized receiver but the structure is more complicated.
[b] The digital receiver is in the development stage and performance should improve in the near future.

delay lines are no longer required. Once the basic properties of a receiver are understood, they can be combined in various ways to solve specific problems.

COMPARISON OF DIFFERENT TYPES OF RECEIVERS

It is difficult to compare the performance of different types of receivers. For example, one can consider a channelized receiver to be complicated if there is a large number of channels. If there are only a few channels the receiver can be rather simple. The same argument holds for Bragg cell and compressive receivers. To make the assessment meaningful, it is assumed that the four types of receivers—channelized, Bragg cell, compressive, and digital channelized—all have the same number of output channels. The other three types of receivers—crystal video, superheterodyne, and IFM—can not process simultaneous signals. It is also difficult to put quantitative measures on the performance. For one specific receiver this should be the correct way to present its performance. However, for one type of receiver it may be difficult to do so. For example, one channelized receiver is designed for higher sensitivity and lower dynamic range and another one is designed for higher dynamic range and lower sensitivity. If the best performance of each receiver is reported, the results can be misleading because they can not be achieved in one receiver design. Therefore, the performance is listed in a qualitative manner as in Table 1.

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