Chapter 1 : SIGNALS AND COMMUNICATIONS

Electronic communications is exchanging signals. While these signals are symbolic in many communication schemes, they are almost exact electrical replicas of original information in analog wireless communications. Sound and vision are all such signals. Signals are converted into a form, by a *transmitter*, so that they can be transmitted in the air as part of *electromagnetic spectrum*, and are received by a *receiver*, where they are converted back to the original form. Two communicating parties can be quite far away from each other, and therefore the term *telecommunications* is used to describe this form of communications. What follows in this chapter is a descriptive theory of analog signal processing in communications.

Transceivers are wireless transmitters (TX) and receivers (RX) combined in a single instrument. This book is structured around building and testing a transceiver, TRC-10, operating in the 10-meter amateur band (28-29.7 MHz). The name is generic: *TRC* stands for transceiver and *10* indicate that it works in 10-meter band.

TRC-10 is an *amplitude modulation superheterodyne transceiver*. We have to make some definitions in order to understand what these terms mean.

1.1. Frequency

The two variables in any electrical circuit is voltage, V, and current, I. In electronics, all signals are in form of a voltage or a current, physically. Both of these variables can be time varying or constant. Voltages and currents that do not change with respect to time are called *d.c.* voltages or currents, respectively. The acronym *d.c.* is derived from *direct current*.

Voltages and currents that vary with respect to time can, of course, have arbitrary forms. A branch of applied mathematics called *Laplace analysis*, or its special form *Fourier analysis*, investigates the properties of such time variation, and shows that all time varying signals can be represented in terms of linear combination (or weighted sums) of sinusoidal waveforms.

A sinusoidal voltage and current can be written as

$v(t) = V_1\cos(\omega t+\theta_v)$, and

$i(t) = I_1\cos(\omega t+\theta_i)$.

V_1 and I_1 are called the *amplitude* of voltage and current, and have units of Volts (V) and Amperes (A), respectively. "ω" is the radial frequency with units of radians per second (rps) and $\omega = 2\pi f$, where "f" is the frequency of the sinusoid with units of Hertz (Hz). "θ" is the phase angle of the waveform. These waveforms are periodic, which means that it is a repetition of a fundamental form in every T seconds, where T=1/f seconds (sec).

Quite often, sinusoidal waveforms are referred to by their *peak amplitudes* or *peak-to-peak amplitudes*. Peak amplitude of v(t) is V_1 Volts peak (or V_p) and peak-to-peak amplitude is 2 V_1 Volts peak-to-peak (or Vpp).

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Now we can see that a d.c. voltage is in fact a sinusoid with f = 0 Hz. Sinusoidal voltages and currents with non-zero frequency are commonly referred to as *a.c.* voltages and currents. The acronym *a.c.* comes from *alternating current*.

If we know the voltage v(t) across any element and current i(t) through it, we can calculate the power delivered to it as

$$P(t) = v(t)i(t) = V_1 I_1 \cos(\omega t+\theta_v)\cos(\omega t+\theta_i)$$

or

$$P(t) = \frac{V_1 I_1}{2}\cos(\theta_v - \theta_i) + \frac{V_1 I_1}{2}\cos(2\omega t + \theta_v + \theta_i).$$

P(t) is measured in watts (W), i.e. (1V)×(1A)=1 W.

In case of a resistor, both current and voltage have the same phase and hence we can write the power delivered to a resistor as

$$P(t) = \frac{V_1 I_1}{2} + \frac{V_1 I_1}{2}\cos(2\omega t + 2\theta_v).$$

We shall see that the phase difference between voltage and current in an element or a branch of circuit is a critical matter and must be carefully controlled in many aspects of electronics.

P(t) is called the *instantaneous power* and is a function of time. We are usually interested in the *average power*, P_a, which is the constant part of P(t):

$$P_a = \frac{V_1 I_1}{2}\cos(\theta_v - \theta_i),$$

in general, and

$$P_a = \frac{V_1 I_1}{2}$$

in case of a resistor.

We note that if the element is such that the phase difference between the voltage across and current through it is 90^o, P_a is zero. Inductors and capacitors are such elements.

Radio waves travel at the speed of light, c. The speed of light in air is 3.0 E8 m/sec (through out this book we shall use the scientific notation, i.e. 3.0 E8 for 3×10^8), to a very good approximation. This speed can be written as

$$c = f\lambda$$

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where λ is the wavelength in meters. TRC-10 emits radio waves at approximately 30 MHz (actually between 28 and 29.7 MHz). The wavelength of these waves is approximately 10 meters. The amateur frequency band in which TRC-10 operates is therefore called *10-meter band*.

1.2. Oscillators

Electronic circuits that generate voltages of sinusoidal waveform are called "sinusoidal oscillators". There are also oscillators generating periodic signals of other waveforms, among which square wave generators are most popular. Square wave oscillators are predominantly used in digital circuits to produce time references, synchronization, etc. Oscillator symbol is shown in Figure 1.1.

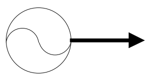

Figure 1.1 Oscillator symbol

We use oscillators in communication circuits for variety of reasons. There are two oscillators in TRC-10. The first one is an oscillator that generates a signal at 16 MHz fixed frequency. This oscillator is a square wave *crystal oscillator module*.

A square wave of 2 volts peak to peak amplitude is depicted in Figure 1.2.

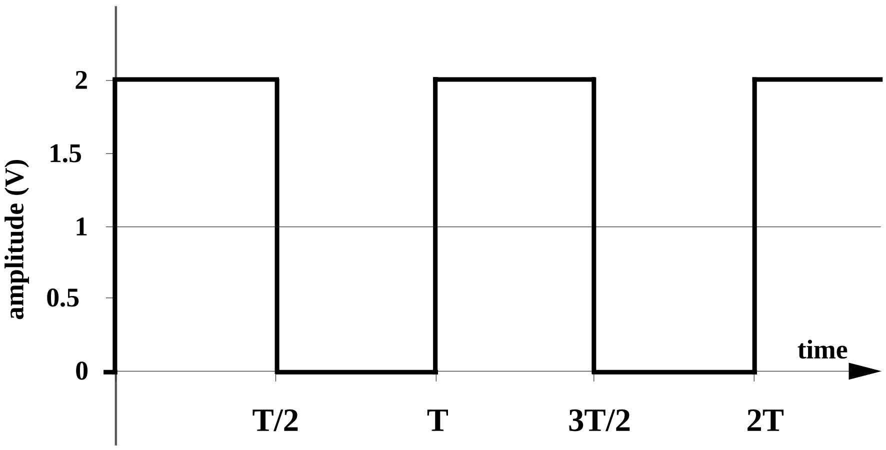

Figure 1.2 Square wave

In fact, such a square wave can be represented in terms of sinusoids as a linear combination:

$$s(t) = 1+(4/\pi)\sin(\omega t)+(4/3\pi)\sin(3\omega t)+(4/5\pi)\sin(5\omega t)+(4/7\pi)\sin(7\omega t)+\ldots$$

$$= a_o + \sum_{n=1}^{\infty} b_n \sin(n\omega t)$$

where a_o is the average value of s(t), 1 in this particular case, and

$b_n = (2/n\pi)[1-(-1)^n]$

Note here that

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(i) there are an infinite number of sinusoids in a square wave;
(ii) the frequencies of these sinusoids are only odd multiples of ω, which is a property of square waves with equal duration of 2's and 0's- we call such square waves as 50% *duty cycle* square waves;
(iii) the amplitude of sinusoids in the summation decreases as their frequency increases.

We refer to the sinusoids with frequencies 2ω, 3ω, 4ω,…, nω as *harmonics* of the *fundamental component*, sinωt.

We can obtain an approximation to a square wave by taking a_o, fundamental, and only few harmonics into the summation. As we increase the number of harmonics in the summation, the constructed waveform becomes a better representative of square wave. This successive construction of a square wave is shown in Figure 1.3.

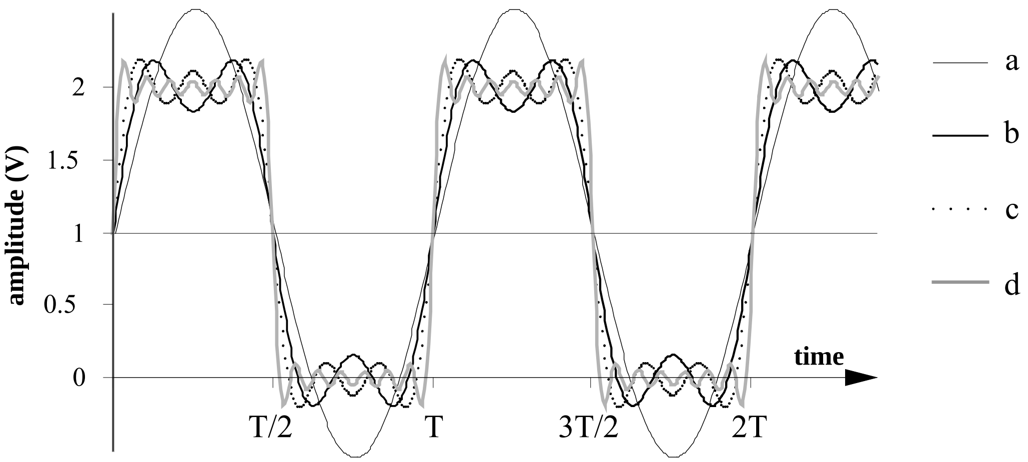

Figure 1.3 Constructing a square wave from harmonics, (a) only a_o+ fundamental, (b) waveform in (a) + 3rd harmonic, (c) waveform in (b) + 5th harmonic, (d) all terms up to 13th harmonic.

Even with only 3 terms the square wave is reasonably well delineated, although its shape looks rather corrugated.

A common graphical representation of a signal with many sinusoidal components is to plot the line graph of the amplitude of each component versus frequency (either f or ω). This is called the *spectrum* of the square wave or its *frequency domain representation*. Spectrum of this square wave is given in Figure 1.4, which clearly illustrates the frequency components of the square wave.

Figure 1.4 clearly shows that the square wave, being a periodic signal, has energy only at discrete frequencies.

We need sinusoidal voltages in TRC-10, not square waves. Indeed, we must avoid the harmonics of our signals to be emitted from our transceiver, because such harmonics will interfere with other communication systems operating at that frequency. We use this fixed frequency square wave oscillator module, because such modules provide a

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very accurate and stable frequency of oscillation and can be obtained at a low cost. We first filter out the harmonics of the waveform, when we use this module in our circuit.

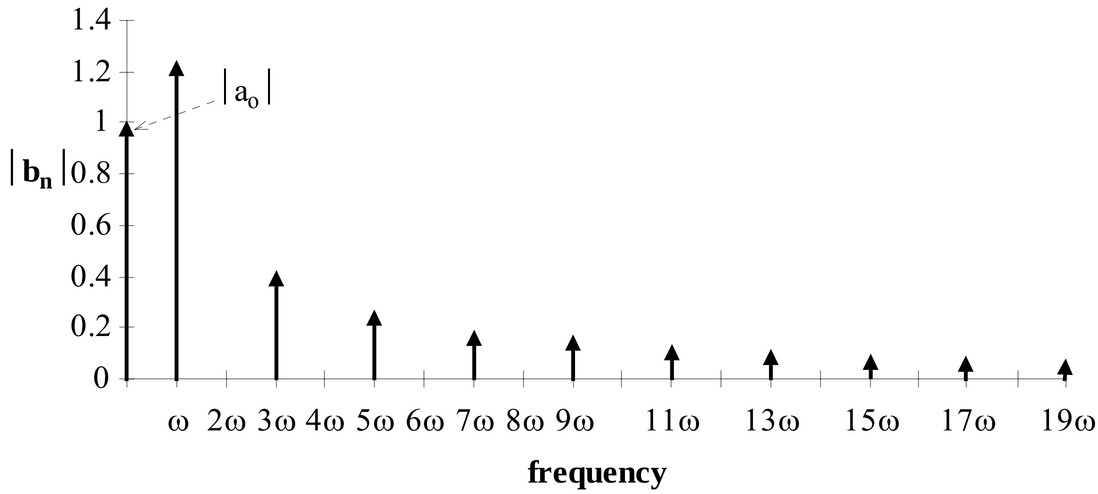

Figure 1.4 Spectrum of the square wave

The other oscillator is a *Variable Frequency Oscillator* (VFO). This oscillator produces a sinusoid of frequency that varies between 12 MHz and 13.7 MHz. This frequency is controlled by a d.c. voltage. We discuss the VFO in Chapter 7.

1.3. Modulation

A sinusoidal waveform does not carry any information on its own. In order to transmit any information, in our case voice, we need to make one parameter of a sinusoid dependent on this information.

In electronics, the information must of course be converted into an electrical signal, a voltage or a current, first. For example, a microphone converts sound into a voltage, which we first amplify and then use as information signal, $v_m(t)$. This signal is again a time varying signal, which can be represented as a linear combination of sinusoids. The frequency band of this signal, however, is not suitable for transmission in air, as it is. This frequency band is rather low for transmission and it is called *base-band*. The information signal occupying this band is referred to as *base-band signal*. Converting the information-carrying signal to a form suitable for (electromagnetic) transmission is called *modulation*.

There are three parameters that we can play with in a sinusoid: amplitude, frequency and phase. We must mount the information signal $v_m(t)$ on a sinusoid of appropriate high frequency, so that it can be *carried* on air at that radio frequency. We call this operation, modulation. In TRC-10 we use *amplitude modulation (AM)*, which means that we make the amplitude of a sinusoid dependent on $v_m(t)$. Note that the frequencies of sinusoids that are present in a voice signal is within few kHz, while we wish to transmit this voice signal at 30 MHz. Let us assume that $v_m(t)$ is a simple

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signal, $V_m\cos(\omega_m t)$. In order to modulate the amplitude of a *carrier signal*, $V_c\cos(\omega_c t)$, we construct the signal,

$$v(t) = V_c\cos(\omega_c t) + v_m(t)\cos(\omega_c t) = V_c\left(1 + \frac{V_m}{V_c}\cos(\omega_m t)\right)\cos(\omega_c t)$$

$v_m(t)$ is called the *modulating signal*. In AM, the maximum peak variation of $|v_m(t)|$ must always be less than V_c, otherwise it cannot be *demodulated* by simple *envelope detector* circuits which can be used with this modulation scheme, and $v_m(t)$ cannot be recovered. $V_c[1 + (V_m/V_c)\cos(\omega_m t)]$ part in AM signal is called the *envelope*. An AM signal is depicted in Figure 1.5(b).

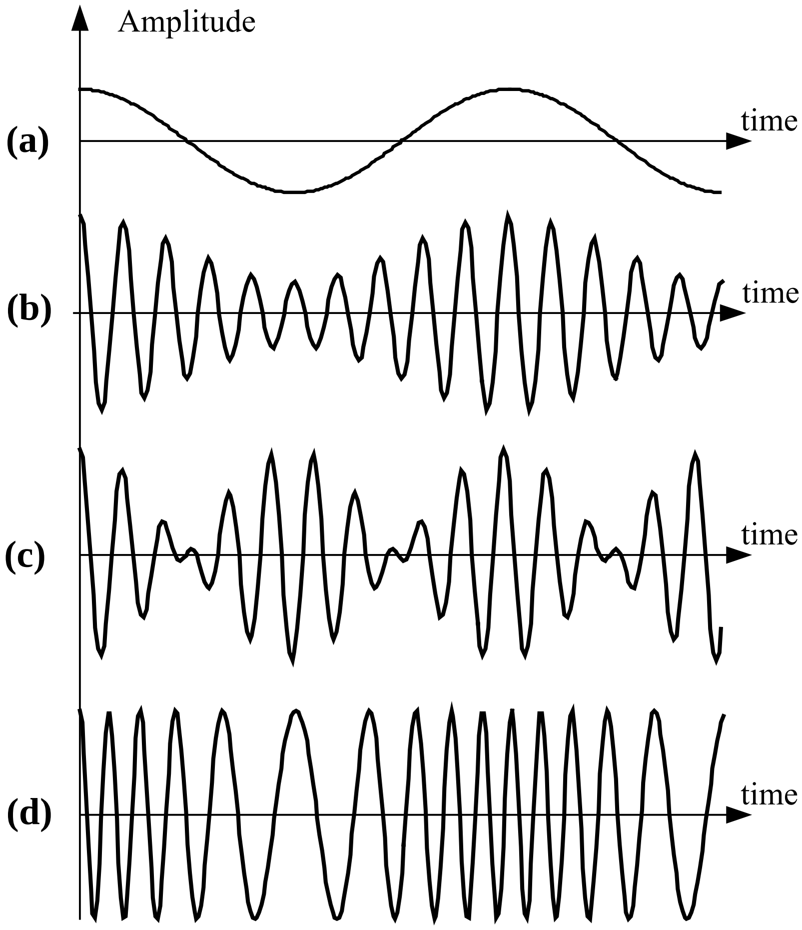

Figure 1.5 Modulated signals: (a) modulating signal, (b) AM signal, (c) DSBSC AM signal and (d) FM signal

The *depth of modulation* is determined by the maximum value of the normalized modulation signal $|v_m(t)/V_c|$. If

$$\left|\frac{v_m(t)}{V_c}\right|_{max} = 1,$$

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AM signal is said to have *100% modulation*, or *modulation index is one*. Similarly, if this maximum is, for example, 0.5, then the modulation index is 0.5 and the AM signal has 50% modulation.

A special case of AM is *double side-band suppressed carrier AM* (*DSBSC AM*),

$v(t) = v_m(t)\cos(\omega_c t) = V_m\cos(\omega_m t)\cos(\omega_c t)$.

In this scheme a different demodulation must be used. A DSBSC AM waveform is shown in Figure 1.5(c).

Other parameters that can be modulated in a sinusoid are frequency and phase. In analog communication systems different forms of amplitude modulation and frequency modulation (FM) are used. In FM, we construct the following signal:

$v(t) = V_c \cos\{\omega_c t + k_f \int v_m(t)dt\} = V_c \cos\{\omega_c t + \beta V_m \sin(\omega_m t)\}$,

such that $\beta = k_f/\omega_m$, and $\omega(t) = d\{\theta(t)\}/dt = \omega_c + k_f v_m(t)$.

Here we change the frequency of the carrier signal around the carrier frequency, ω_c, according to the variation of modulating (information) signal. An FM modulated signal is shown in Figure 1.5(d).

Long wave and middle wave radio broadcasting is done by AM, and radio broadcasting in 88-108 MHz band is done by FM. Analog terrestrial television broadcasting employs a version of AM (called *vestigial side-band AM*) for image and FM for sound.

1.4. Amplifiers

The most frequently done operation on signals is amplification. The signal received at an antenna is often very weak, may be at power levels of few tens of *fW* (1 *f*emto*W*att is 1 E-15 W). This power level corresponds to a few μV (*micro volt*, micro is 1 E-6) into a 50 Ω, which is a typical value of input resistance for a receiver. This signal level must be increased so that it can be demodulated, and further increased so that it can be heard. The device that performs this function is called *amplifier*. We use operational amplifiers (OPAMP) for amplification in TRC-10. Amplifiers relate the signal at their input and at their output by a gain. We are usually interested in two types of gain, the voltage gain and power gain. We shall denote voltage gain by A and power gain by G:

$A = V_o/ V_i$, and

$G = P_o/ P_i$.

Gain is a quantity, which does not have any units. We use *decibels* (dB) to describe the amount of gain. We can express the same gain in decibels as,

$A = 20 \log (V_o/ V_i)$ dB, and

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$G = 10 \log(P_o / P_i)$ dB.

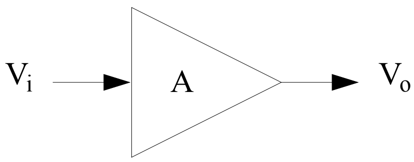
The coefficient of 10-base logarithm is 10 for power gain and 20 for voltage or current gain. With this definition, both a voltage gain and the corresponding power gain yield the same value in dB. For example, if a peak voltage of V_1 appears across a resistor R, then the peak current through R is V_1/R, and average power delivered to R is $V_1^2/2R$. Now, if this voltage is amplified two folds and applied across the same resistor, then there is a voltage gain of A = 2 and power gain of G = 4. In decibels, the value of both A and G is 6 dB. Also note that 3-dB corresponds to a power gain of 2 and a voltage gain of $\sqrt{2}$.

Decibel is also used to define absolute levels. For example 0.5 miliwatt of power is expressed in decibels as "-3 dBm". Here, "m" denotes that this value is relative to 1 miliwatt. Similarly, 20 Watts is expressed as "13 dBW". Another way of writing absolute levels in decibels is to directly write what it is relative to. For example we can write 32 microvolts as "30 dB re μV".

Amplifier block diagram and some easy to remember approximate dB values are given in Figure 1.6.

dB	$A=V_o/V_i$	$G=P_o/P_i$
0	1	1
3	1.42	2
6	2	4
7	2.24	5
9	2.83	8
10	3.16	10
20	10	100

(a)



(b)

Figure 1.6 (a) gain conversion table, and (b) amplifier block diagram

1.5. Mixers

We want to transmit and receive voice signals, which are limited to a few kHz frequency band. We are allowed to do that at frequencies orders of magnitude higher, because all frequency bands are shared between different services and carefully regulated both nationally and internationally. We must, therefore, have the capability of shifting the frequency of information signal, up for transmission, and down for reception. Mixers do this by multiplying two signals.

Consider the amplitude modulated signal,

$$v(t) = V_c\cos(\omega_c t) + v_m(t)\cos(\omega_c t) = V_c[1 + (V_m/V_c)\cos(\omega_m t)]\cos(\omega_c t).$$

Let us define

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$$m(t) = \left(\frac{V_m}{V_c}\right)\cos(\omega_m t)$$

so that

$v(t) = V_c[1+m(t)]\cos(\omega_c t)$.

The multiplication of 1+m(t) with carrier $\cos(\omega_c t)$ can be done by a mixer. In Figure 1.7 a mixer performing this operation is shown.

The signal received by the antenna must be *mixed* down to a frequency where it is properly *filtered* and converted back to *audio* signal. This is called *heterodyne* reception.

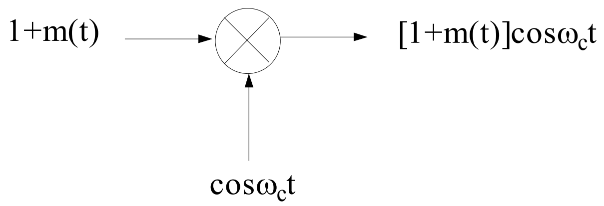

Figure 1.7 Mixer

1.6. Filters

Analog electronics is all about signals. Every signal occupies a frequency band in the spectrum. This band can be quite large, 8 MHz as in the case of a television signal, or very small such as for a single sine wave. The most important matter is that all signals are in the same frequency spectrum, they share the spectrum. Also apart from the signals of well-defined nature like the ones that we are interested in, there exists *natural* and *man-made noise*, which further complicates the task of electronic engineers. Whenever a particular signal is to be processed, we must make sure that we use a good and clean sample of that signal without any irrelevant and unwanted components in it. We employ *filters* for this purpose.

We use *low-pass* filters, *high-pass* filters and *band-pass* filters in TRC-10. As the names imply, low-pass filters (LPF) allow the signals below a certain frequency to pass through the filter, and *attenuate* (decrease their amplitude) the signals of higher frequency. This threshold frequency is called *cut-off* frequency. Filtering effect is not abrupt, but it is gradual. The signal components and noise beyond cut-off frequency are not completely eliminated, but attenuated more and more as their frequencies are further away from cut-off. This behavior of a LPF is shown in Figure 1.8(a). The filtering property of filters is described by their transfer function, $H(\omega)$. $H(\omega)$ is a complex function (in algebraic sense) of radial frequency, ω. At those frequencies where $|H(\omega)|$ is equal to or close to 1, then the signals at the input of the filters pass through the filter unaffected. On the other hand, if $|H(\omega)|$ is significantly lower than 1, the signals at those frequencies are attenuated. The cut-off frequency $\omega_{-3dB} = 2\pi f_{-3dB}$ is defined as the frequency where $|H(\omega)|$ is $1/\sqrt{2}$.

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The transfer function of a filter is also described in decibels. We can plot |H(ω)| as 20log|H(ω)| versus ω. In that case cut-off frequency is the frequency where 20log|H(ω)| is –3dB. 20log|H(ω)| of the same LPF is also depicted in Figure 1.8(b).

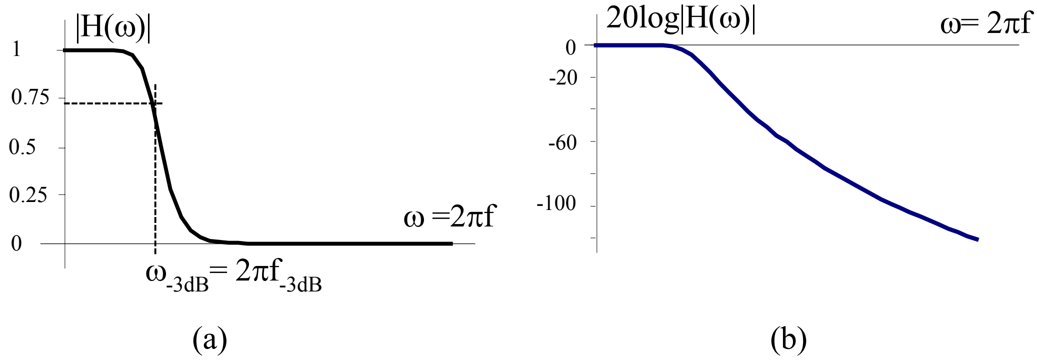

Figure 1.8 Low-pass filter (a) |H(ω)|, and (b) 20log|H(ω)|

High-pass filters (HPF) eliminate the signals below the cut-off frequency. Band-pass filters (BPF) allow the signals that fall in a certain frequency band to pass through the filter and attenuate other signals. HPF and BPF are shown in Figure 1.9.

Again, the *bandwidth* (BW) of a BPF is defined as $\Delta\omega=\omega_2-\omega_1$ in rps, where ω_2 and ω_1 are the -3dB *upper* and *lower cut-off frequencies*. Often these are expressed in terms of Hz, where $\Delta f=f_2-f_1$, and f_2 and f_1 are $\omega_2/2\pi$ and $\omega_1/2\pi$ respectively. The *center frequency* of a BPF, f_o, is the geometric mean of f_2 and f_1. BPF usually have the lowest attenuation at f_o.

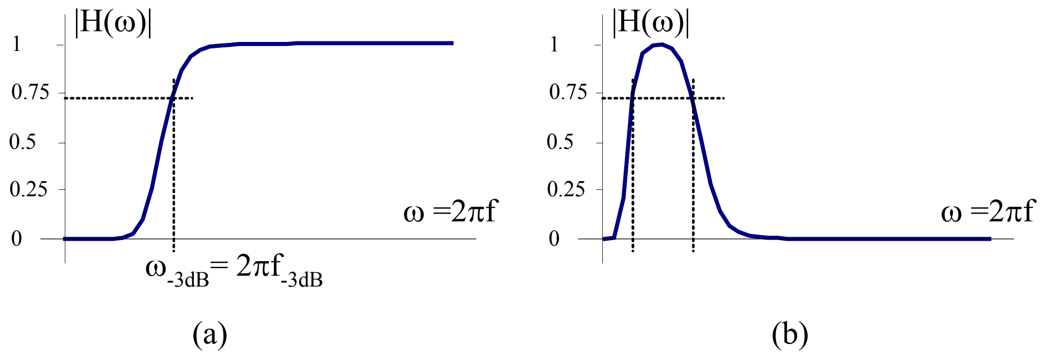

Figure 1.9 (a) High-pass filter, (b) Band-pass filter

Now consider that we wish to filter out the fundamental component of the square wave at 16 MHz, using the BPF in Figure 1.9(b). We must tune the filter so that center frequency is in the vicinity of the fundamental, and the bandwidth is sufficiently small to attenuate particularly the nearest harmonic sufficiently (which is third harmonic in this case). This situation is depicted in Figure 1.10.

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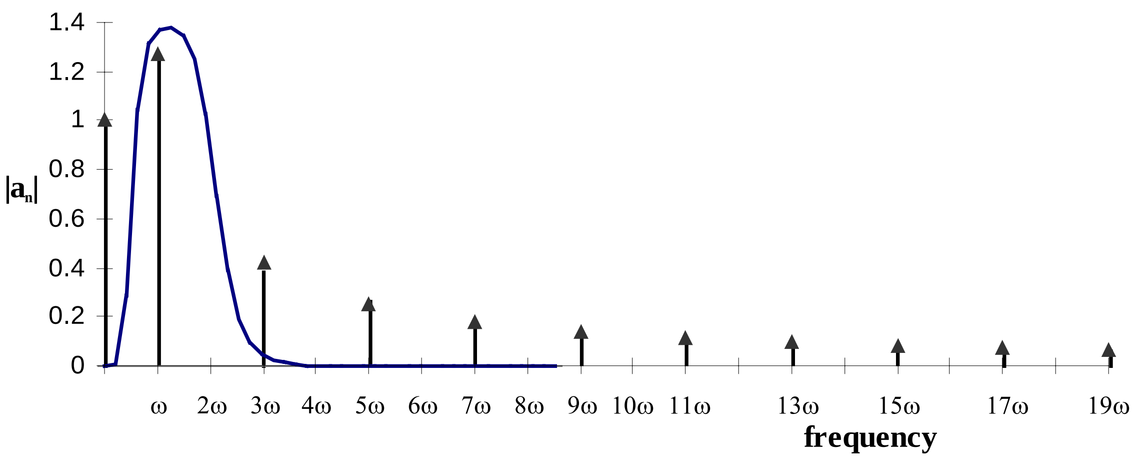

Figure 1.10 A BPF superimposed on the spectrum of the square wave

1.7. Receivers

We modulate a carrier to transmit the information signal over a radio frequency (*RF*). When received, we must demodulate this RF signal, and *detect* the information. We use *envelope* or *diode detectors* to do this. Envelope detectors separate the envelope from the carrier. This is the simplest way of detecting AM signals. We use envelope detector in the receiver of TRC-10.

Another way of detecting AM signals, is to reverse the modulation process. We can use mixers to demodulate AM signals, simply to multiply the received signal by an *exact* replica of the carrier signals. Indeed this is the only way to recover the information signal in DSBSC AM, and can of course be used in AM.

Frequency deviation carries the information in FM. The frequency of the received signal must be estimated in order to get the information signal, in this modulation scheme. There are different ways of doing this, among which, *FM discriminators* are the ones, which are most widely used.

When we modulate the amplitude of a carrier with a voice signal, the transmitted signal becomes:

$$v(t) = V_c\left(1 + \frac{V_m}{V_c}\cos(\omega_m t)\right)\cos(\omega_c t)$$

where $V_m\cos(\omega_m t)$ represents the voice signal (this would sound like a whistle) and $\cos(\omega_c t)$ is the carrier. If we carry out the algebra, we can see that v(t) contains three sinusoids,

$$v(t) = V_c\cos(\omega_c t) + (V_m/2)\cos[(\omega_c-\omega_m)t] + (V_m/2)\cos[(\omega_c+\omega_m)t]$$

Now, assume that ω_m is 2π x 3E3, i.e. a 3kHz signal (a *very* high pitch whistle sounds like this). On the other hand, ω_c is anything between 2π x 28E6 and 2π x 29.7E6 in

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TRC-10, let us take it as $2\pi \times 29E6$. Then the frequencies of the three components become 29MHz, 28.997MHz, and 29.003MHz, respectively. The information content in the modulated signal is confined to only a 6kHz bandwidth around 29MHz. This bandwidth is only 0.2% of the carrier frequency. All the signals and noise outside this bandwidth is irrelevant to this communication setup, and hence they must be eliminated as much as possible. We must use a filter with such a narrow bandwidth. Of course, if we had chosen another carrier frequency in 10-meter band, then the center frequency of the filter should be adjusted to that frequency.

It is very difficult to design a filter that both has a very narrow bandwidth and can be tuned to arbitrary center frequencies. However it is possible to have a very precise narrow bandwidth filter at a fixed frequency, easily. In this case we must shift the frequency of the received signal to the center frequency of the filter using a mixer, since we cannot let the filter to follow the signal frequency. The receivers that use this concept are called *superheterodyne receivers*. The frequency to which the carrier is shifted, is called the *intermediate frequency*, or simply *IF*, and the filter is called the *IF filter*.

TRC-10 is a superheterodyne receiver with 16MHz IF.

1.8. TRC-10

The block diagram of TRC-10 is given in Figure 1.11. The adventure of a voice signal through a transceiver can be observed on this figure.

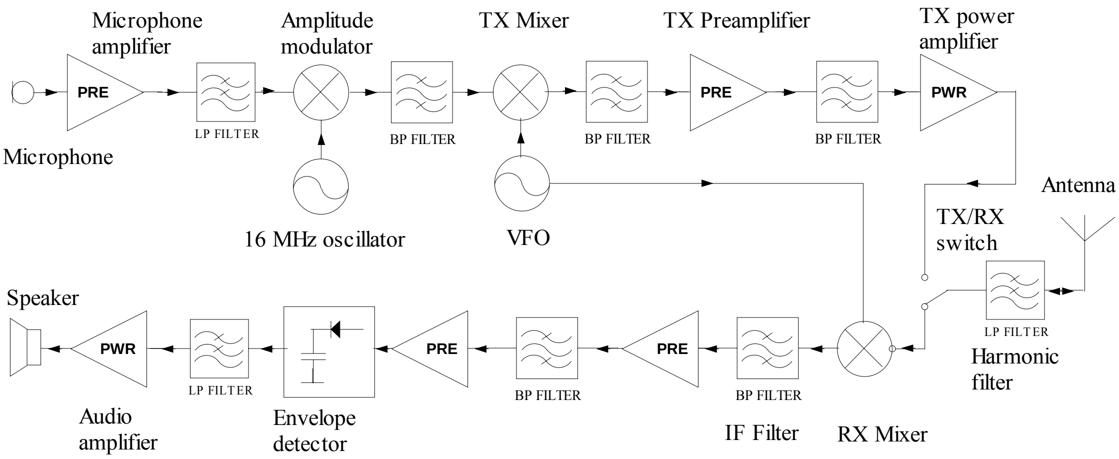

Figure 1.11 TRC-10 block diagram

Acoustic voice signal is first converted into an electrical signal at the microphone. The signal power is very low at this point, at pW level. Microphone amplifier increases the signal power level to about a mW. Voice or speech signal is not a periodic signal, and therefore its frequency spectrum is not a *line spectrum* like square wave spectrum. Speech signal has a continuous energy distribution over the frequency domain. Most of the energy contained in this signal is confined to a band between 100Hz and 3 kHz, approximately. Orchestral music, for example, is also an audible sound, but has a richer frequency spectrum. Orchestral music has a bandwidth of

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approximately 10 kHz. The LPF after the microphone amplifier limits the bandwidth of the audio signal to 3 kHz, since we only speak to the transceiver microphone, and any possible noise like signals with higher frequency content are avoided.

The filtered audio signal is then used to modulate the amplitude of a 16 MHz carrier. The signal spectra at the input and at the output of the amplitude modulator are shown in Figure 1.12.

Fourier analysis tells us that any signal with a *real algebraic model* must have a spectrum defined for *negative* frequencies as well as for positive frequencies (with a lot of symmetry properties), in order to be mathematically correct. Audio signal is such a signal and can be represented by a real function of time, m(t). Hence, we show the spectrum of m(t) by a two-sided spectrum in Figure 1.12, although it is customary to show a single side for positive frequencies only.

Once modulated, the voice spectrum is shifted such that it is now centered on IF frequency instead of zero. Also there is a sinusoidal carrier.

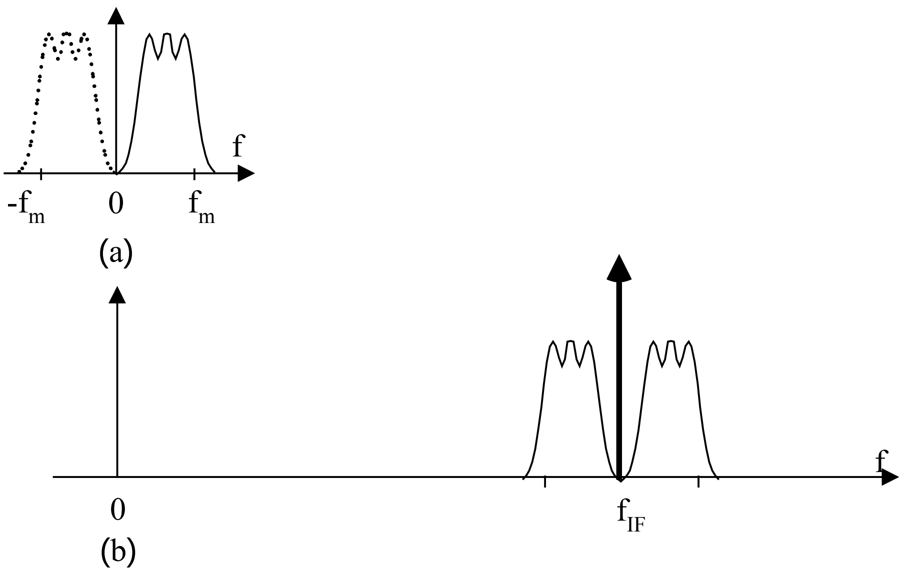

Figure 1.12 Modulator (a) input and (b) output signal spectra (not scaled)

TX mixer shifts that further to the transmission frequency, $f_c = f_{IF} + f_{VFO}$. This second mixing produces two spectral components. First one is the required one, and second one is at $f_{IF} - f_{VFO}$, which falls into the frequency band 2.3 MHz to 4 MHz. This component is an unwanted component and must be eliminated. We filter this component out by two filters, before and after the TX preamplifier. This process is depicted in Figure 1.13.

The RF signal is once more amplified by TX power amplifier to increase the power level to 40 mW. It is then filtered once more, to suppress the possible harmonics, and fed to the antenna.

The voice is now "on air".

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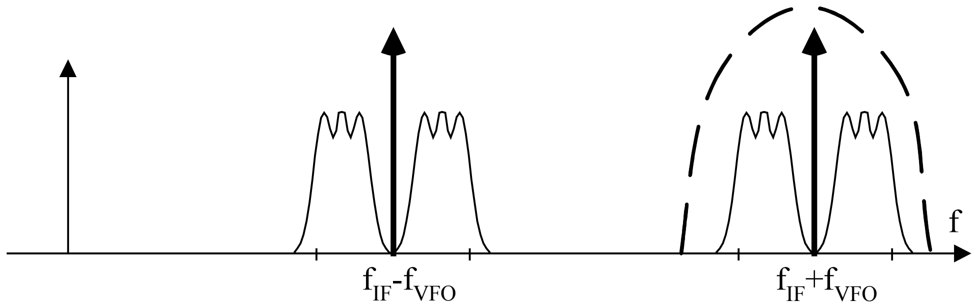

Figure 1.13 Signal at the TX Mixer output (not scaled)

When the transceiver is switched to reception mode, the diode switch isolates the antenna from TX and connects it to the RX input. Antenna receives all the signals and noise that reach to it, in air. Therefore what comes from antenna is first band pass filtered to 28-29.7 MHz, and then mixed with VFO signal down to 16 MHz. At this stage, the signal is passed through a very narrow band filter of just 6 kHz bandwidth. This ensures that we filter out all signals other than the one we are interested in.

The IF signal is then amplified by two amplifiers and then fed into the diode detector. Detector produces a replica of the envelope, which is the received voice signal. This signal is now at base band, and hence amplified in the audio amplifier before we hear it from the speaker.

1.9. Bibliography

a. D.B. Rutledge's book "The Electronics of radio", published by Cambridge University Press, 1999.
b. "The ARRL Handbook", published by ARRL, new edition every year, any recent edition.
c. P. Nahin's book " The Science of Radio", 2nd Edition, published by Springer-Verlag.

1.10. Laboratory Exercises

Soldering exercise

1. You will learn good soldering in this exercise. You will build a house on copper clad board using resistors and capacitors. Soldering is not gluing. Soldering is a chemical process to form an alloy of solder and the soldered metal pieces. Soldering iron must be hot and its tip must be shiny in order to make good solder joint. Put some water on the soldering sponge and keep it wet through out the soldering session. Turn the soldering iron on and wait until it is hot. Solder must immediately melt on the tip when it is hot enough. Put some solder on the tip and wipe the tip with wet sponge. The tip will shine. This process is called *tinning*. Now the iron is ready to make a solder joint. If the tip is not shiny, the heat transfer from the tip to the component is poor.

   The joint to be soldered must be mechanically sturdy enough before solder is applied, so that when the solder is hot and in fluid form, the joint must not

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move. Place the tip in contact with the joint, touching all parts to be soldered. Place the solder in contact with the parts (*not the tip*) opposite to the tip. Solder must melt within a second. Remove the tip and the solder.

1.11. Problems

1. Signal s(t) = 4 cos(ω_st) is multiplied by a carrier c(t) = cos(ω_ot) in a mixer. Calculate the signal at the output of the mixer as a sum of sinusoids. Plot the magnitude of the individual sine wave components wrt frequency if ω_s=1000 rps and ω_o=5000 rps, as in Figure 1.4.

2. Let s(t) = cos(ω_st)+ 2 cos(2ω_st)+ 2 cos(3ω_st)+ cos(4ω_st), where $\omega_s = 2\pi f_s$ and f_s = 300 Hz. s(t) is mixed with cos(ω_ot) where f_o = 5000 Hz. Calculate the signal at the output of the mixer as a sum of sinusoids (*no powers, no products*). Plot the magnitude of the individual sinusoidal wave components in this output signal and in s(t) wrt frequency.

3. Let s(t) = A(t)cos(ω_st). s(t) is mixed and filtered to obtain A(t)cos(ω_ot). What is the signal that s(t) must be mixed with and what kind of filter is needed?

4. Show that
$$\tan(A+B) = \frac{\tan A + \tan B}{1 - \tan A \tan B}.$$

5. Construct a square wave with 3 and 5 components and calculate the mean square error, using anything, a spreadsheet, etc.

6. Do we need a filter at the output of amplitude modulator in TRC-10?

7. What must be the BW of the filter after detector?

8. A particular filter is experimentally measured and the following data is obtained:

|H(ω)|	1	0.986	0.95	0.832	0.707	0.6	0.447	0.351	0.287	0.148	0.01
f -Hz	0	500	1000	2000	3000	4000	6000	8000	10000	20000	30000

Draw $|H(\omega)|$ with respect to ω. Calculate $|H(\omega)|$ in decibels and plot against ω.

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