NAVSHIPS 0967-307-7010

U.S. NAVY

# SINGLE SIDEBAND COMMUNICATIONS

Prepared for

THE AFLOAT COMMUNICATIONS MANAGEMENT OFFICE

Of The

NAVAL SHIPS SYSTEMS COMMAND

November 1968

FOR OFFICIAL USE ONLY

TABLE OF CONTENTS

Page



FIGURES

## TABLES

# INTRODUCTION

Single sideband "SSB" communications equipments are common in the Navy today. However, what was once regarded as a "cure-all" has not eliminated all Naval communication problems. In some cases it appears that the older communications systems are getting the message through with fewer "garbles" than SSB systems. On the other hand the Navy's engineers and scientists tell us that SSB is theoretically capable of providing vastly improved communications. What's the problem? No single answer is possible. The problems vary greatly among ships and operators. Early estimates of the advantages of SSB over conventional AM systems were optimistic. Nevertheless, SSB is an improvement over AM, but the operational techniques are more complex.

The results of the BASE LINE communications exercises and actual combat operations in S. E. Asia have pointed out the superiority of SSB when properly used. It has also highlighted the fact that communications will be degraded if the proper care in operating is not observed. The key to maximizing the advantages of SSB in the Navy is a thorough understanding of its properties by all communicators.

This handbook is intended to highlight the important concepts of SSB and aid shipboard operators in getting the best communications with any of the SSB and associated equipments. It assumes only that the reader has a practical interest in SSB communications and desires to understand the subject well enough to operate his equipment more effectively.

This material was prepared and edited by men with real operational experience in communications. The handbook identifies and clarifies the areas where operators have had difficulty in developing an understanding of SSB.

You don't have to be an electronic or mathematical genius to understand the important concepts about radio equipment. You do have to study some basic terminology and apply some basic reasoning to see why an equipment is built a certain way and how it should be operated. This handbook presents the why and how of SSB.

# 1.0 HISTORY OF SSB COMMUNICATIONS

Single sideband (SSB) communications is not a new concept as many people believe it to be. SSB has received attention in theoretical work since 1914, and SSB equipments have been employed since 1923.

In 1923 the first trans-Atlantic radio-telephone (voice) demonstration was conducted. This project actually employed a SSB signal with a pilot carrier in order to conserve power. AT&T used SSB signals on a submarine cable established for trâns-Atlantic service in 1927. Radio amateur operators conducted SSB experiments in the early 1930's. It wasn't until 1957, however, that the Navy initially purchased and installed SSB equipment for general shipboard use. The reason for this long wait was that the equipment that was available was not practical as shipborne equipment, for SSB required extremely stable radio frequency oscillators to work well. If the receiver and transmitter oscillators drift apart in frequency more than a few tens of cycles, enough distortion is introduced to lose all intelligibility.

Today, modern equipment has more than met the requirements necessary for good SSB high frequency communications at sea. Even as the Navy looks to communication satellites and other means to fulfill its ever increasing demands for communications, SSB will continue to be a useful modulation device.

This handbook will explain in the sections that follow first how sideband signals are generated and then how they should be controlled for the best communications performance.

## 1.1 BASIC DEFINITIONS AND EXAMPLES

### A. Sine Waves, Frequency and Wavelength

In just about every book concerning radio equipment you read about sine waves, or the more formal term sinusoidal waves. These sine waves are used in many places in radio. For example, the 60 cycles, 115 volt power that goes into most equipments is a sine wave voltage. The sine wave is the simplest type of wave. More complicated wave forms can always be represented as a sum of sine waves of various wavelengths. Also amplitude modulation and single sideband modulation is easiest to explain in terms of sine wave.

The easiest way to picture the sine wave is to think of a mechanical wheel turned by a motor running at a constant number of RPM (see Figure 1). To the wheel is connected a shaft which is constrained to move one plane. Near the motor with the crank there is a moving roll chart like the the kind used in barometers. By placing a ball point pen at the end of one arm of the shaft and having the roll chart moving at a constant number of inches per second, the pen would draw sine waves as the crank turns. Half of the maximum swing from top to bottom is called the amplitude of the wave. The distance on the paper between any two corresponding points (two adjacent crests or any two adjacent valleys) is known as the <u>wavelength</u>. If you happened to have a stop watch and measure the time between passing the of two crests or valleys, this is the <u>period</u> of the wave. The number of wave crests or valleys drawn per second or minute would increase as you increased the motor RPM. The number of waves made per unit of time is the wave cycles per second or the frequency of the wave. If you generated two waves per second the period of wave would be 1/2 second. A <u>frequency</u> of three cycles per second would mean you had a period of 1/3 a second. We can generalize here and say that the <u>period is the reciprocal of the frequency</u>. Thus, if frequency is <u>f</u> cycles per second, the period is T seconds then:

$$T = \frac{1}{f} \text{ or } f = \frac{1}{T}$$

While you are experimenting with the sine wave machine someone might ask you "how much roll chart paper are you going to use?" If the roll chart is calibrated in inches per second it can be calculated readily. But suppose it is not. All that is known is that it is moving smoothly at a constant speed. With a ruler and stop watch you can figure it out in a few moments. Mark out a foot on the paper and time how long it takes the foot of paper to go by a mark on the deck. Actually this is a measure of the <u>velocity</u> of the wave too. That it, if you used a wavelength as the two marks on the chart and a period as the observation time, you would have measured the wave velocity. Thus, if <u>v</u> is wave velocity, <u>d</u> is wave distance or wavelength, then

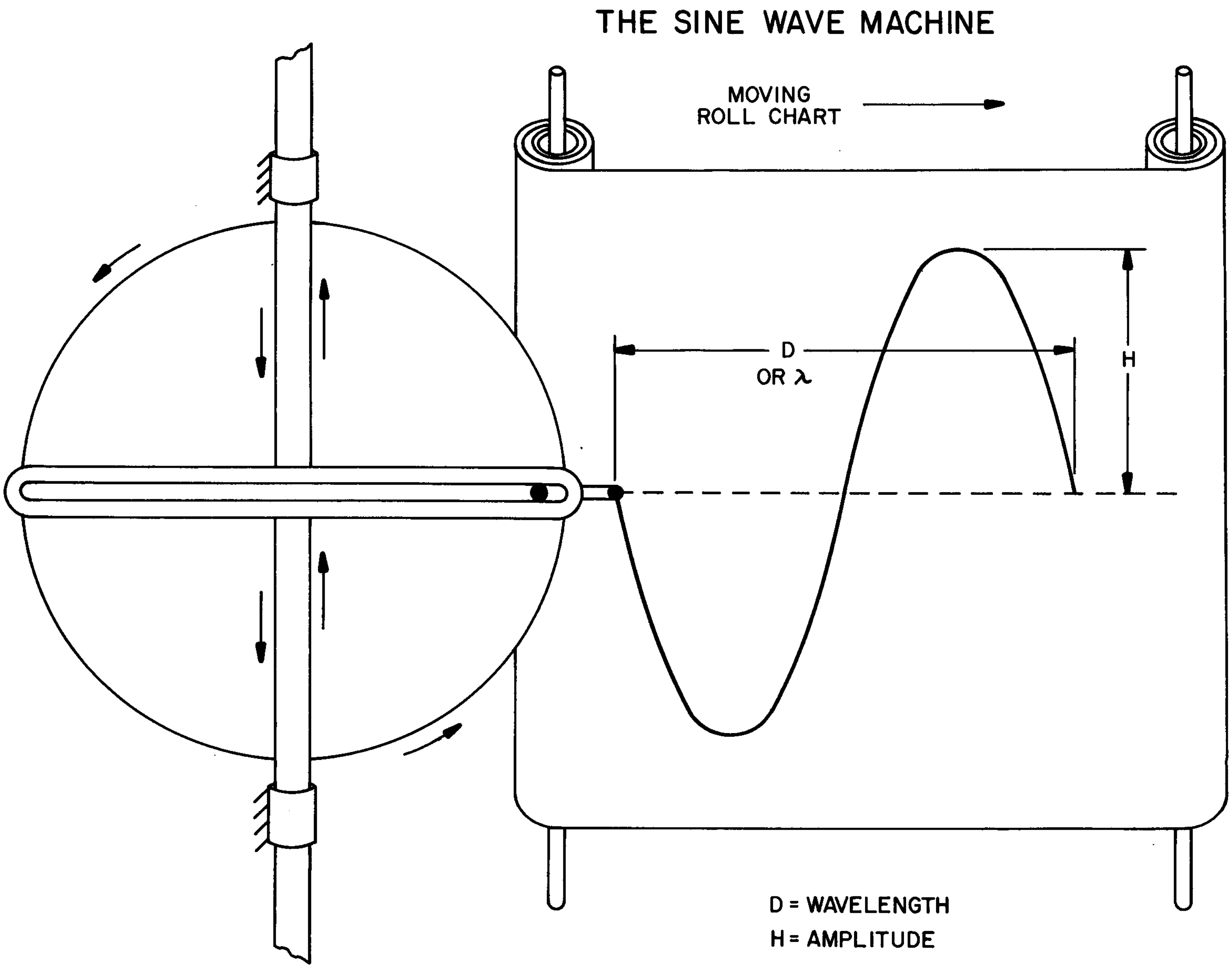



Figure 1. The Sine Wave Machine

$$v = \frac{d}{t}$$

and we know from the earlier discussion that

$$f = \frac{1}{T}$$

so that

$$v = df$$

and this is written in most texts as

$$v = \lambda f$$

since the Greek letter lambda ($\lambda$) is almost always used as the symbol for wavelength. Thus, by an experiment and some simple arithmetic you can determine the wavelength when you know the frequency and wave velocity. You can also determine the frequency when you know the wavelength and wave velocity.

Light waves and radio waves both travel at the same constant speed of 300,000 meters per second or about 186,000 miles per second in space. The relationship given above would apply and

$$\lambda = \frac{300{,}000{,}000}{f}, \quad f = \frac{300{,}000{,}000}{\lambda}$$

where

$\lambda$ = wavelength in meters
$f$ = frequency in cycles per second or "Hertz"

Of course, you have to move your decimal place if you convert frequency to kilohertz or megahertz.

In discussing actual radio equipment we will often refer to sine waves as just waves, and what you should picture in your mind is the sine wave. As stated earlier this is an over-simplification because many wave forms such as speech are a lot more complex than the familiar sine wave. Those more irregular waves could, however, be broken down into a series of sine waves of various amplitudes and frequencies. An example of a graphical buildup or combination of sine waves into square waves is shown in Figure 2.

## B. Carriers and Modulation

Although many people take radio for granted, if you want to know how it really works, you must master a few basic concepts. One of the hardest concepts to understand is modulation and its application. Modulation is any change (alternation, modification, variation, difference, deviation, transformation, mixing, diversion, shifting, translation, rotation, increase or decrease, widening or narrowing, chopping, adding or deleting) of a signal or wave for the purpose of communicating. All information is transmitted by means of changing something. If there is no change there is no message!

There are two terms that need to be carefully defined:

- A carrier is a steady source of energy that can be manipulated (modulated) to transmit or carry ingelligence.
- A modulating signal is a source of changing energy that is used to change the carrier to allow the transmission of intelligence.

Knowing these basic definitions we are now in a position to discuss in more detail how messages are actually sent and received.

A simple example of modulation would be the ordinary telephone (see Figure 3). The speaker's voice changes the resistance of carbon granules in the microphone by air compression. The carbon button in turn is connected in series with a battery. The DC current that flows in the circuit will vary with speech fluctuations. Thus the voice energy is converted into varying electrical current which can be transmitted over wires. Clearly, when no one is speaking, the battery current is steady and nothing can be heard at the receiver end. A more technical way of saying the same thing is that a voice signal modulates a direct current in the telephone transmitter to produce a single sideband of an amplitude-modulated carrier of zero frequency.

## C. Bandwidth

In the simple example of the voice and the carbon microphone, the latter behaved as a modulator to produce a modulated signal carrying speech information. The complex sounds in speech contain many different frequencies and intensities or amplitudes. The mouth and lips, could also be called modulators since they convert the stream of pulsating air produced by the vocal chords into complex frequency-amplitude patterns that we recognize as speech. Experimental analysis shows that intelligible speech requires a band of frequencies from about a hundred cycles per second to several thousand cycles per second.

A high-fidelity stereo amplifier must amplify the currents from a phonograph pickup equally well over a range of 20-20,000 Hz (cycles per second) in order to faithfully reproduce orchestra music.

A bandwidth of approximately 4.5 MHz is required to transmit standard U.S. television. This bandwidth is needed because a television signal contains a great deal more information than a voice channel.

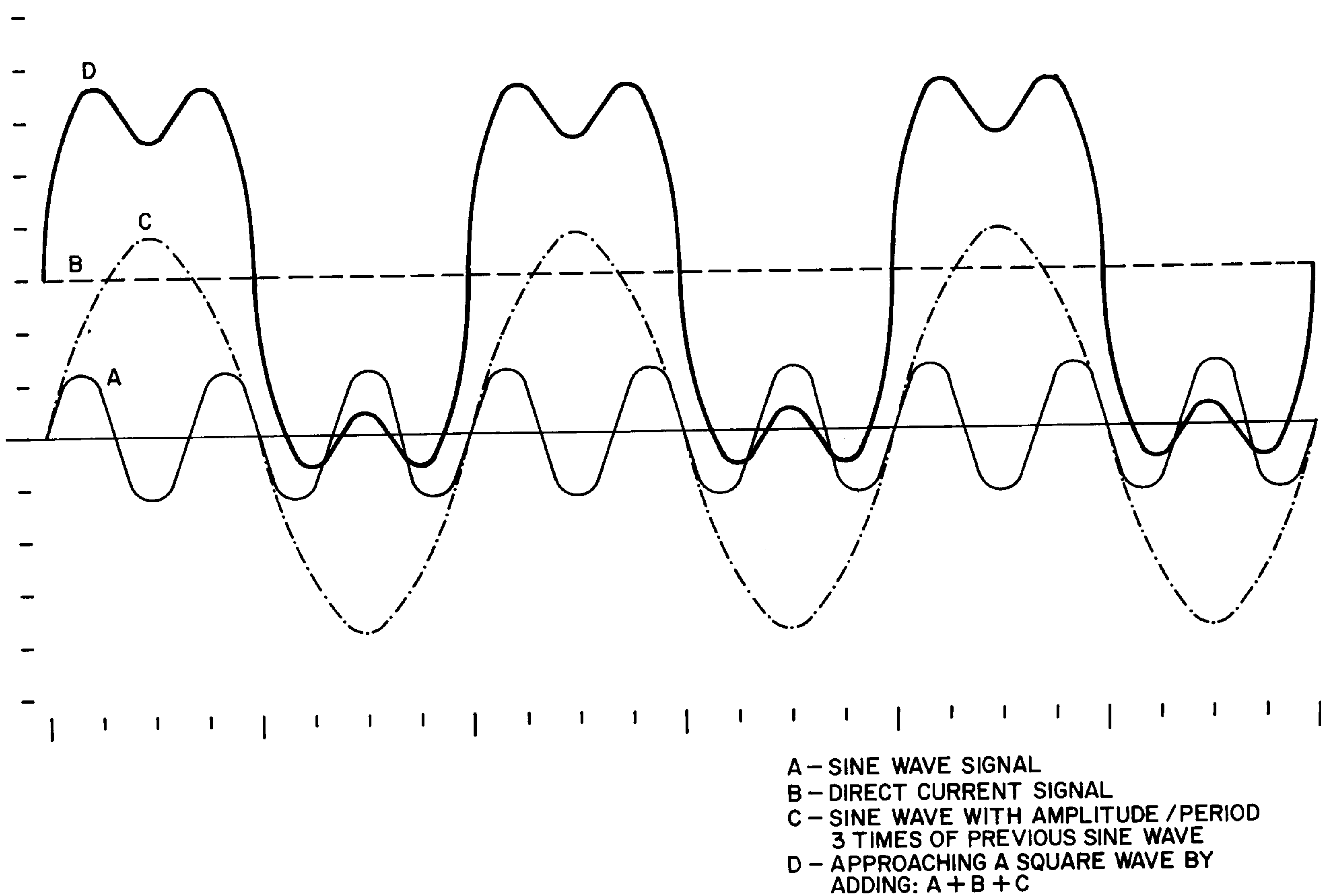
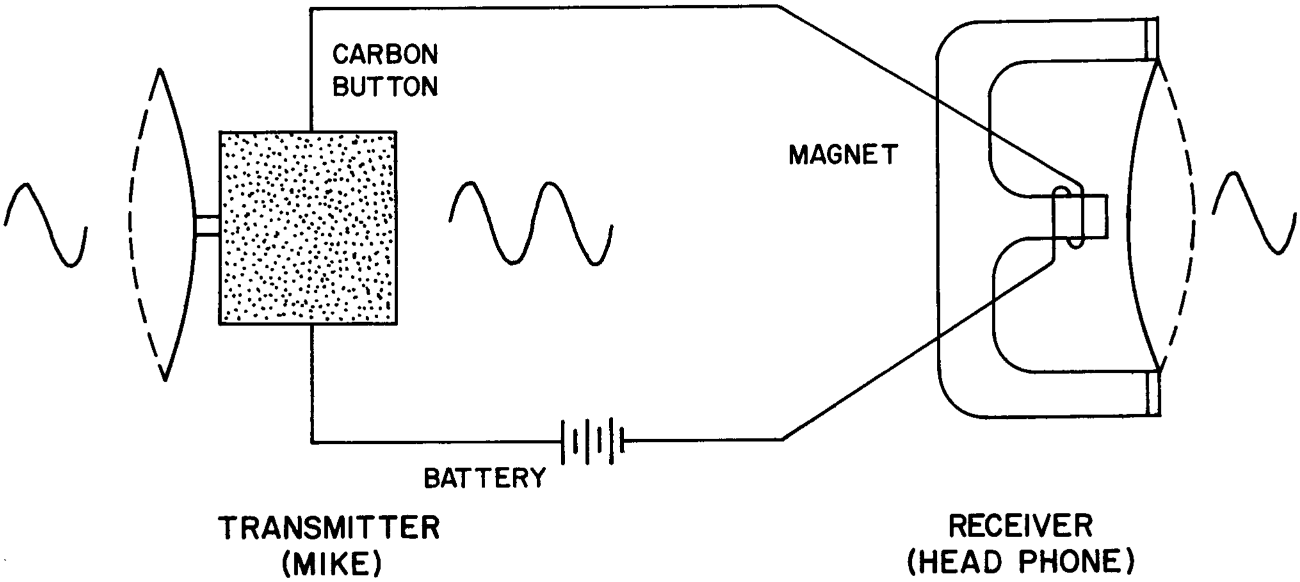



Figure 2. Square Wave Form Built from Sine Wave

## TELEPHONE SET



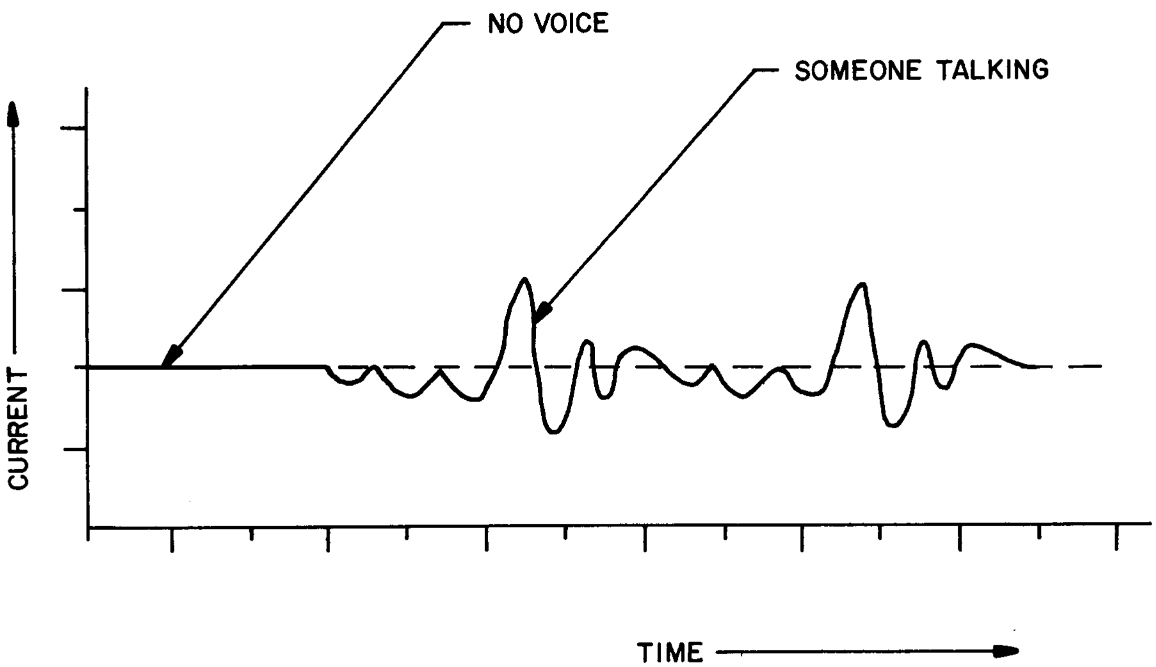




Figure 3. Telephone Set

A modulated signal capable of driving 16 different radio teletype machines at 100 words per minute would require approximately 3000 Hz or a 3 kHz band of audio tones in our present systems.

D. Octaves

An octave on the piano is an interval of eight whole notes. It also includes a frequency ratio of 2:1. That is, C, one octave above middle C, is a sound of twice as many cycles (vibrations) per second as middle C. A voice signal containing audio signals from 100-3200 Hz would contain five (5) octaves (i.e., 100-200 Hz, 200-400 Hz, 400-800 Hz, 800-1600 Hz, and 1600-3200 Hz). Generally speaking, the complexity and cost of amplifiers, speakers, antennas, modulators, transmitters and receivers relates to the number of octaves the equipment covers. Modulation systems can reduce the cost and difficulties of transmitting information by impressing the bands of modulating frequencies onto a higher carrier frequency. This is one of the most practical aspects of the modulating process.

To understand how this modulation works let's switch for a moment from radio to sonar: Consider the problem of communicating by voice signal underwater. A submariner wants to talk to a destroyer-man by means of an underwater loud speaker (transducer). He wants to actually transmit sound pressure waves from his voice directly through the water. This transmission is theoretically possible because sound waves go a long way in the water. Radio waves however do not penetrate sea water very well so radio is of less value here.

Why doesn't the submariner take a big audio amplifier and connect it up to a transducer to serve as a transmitter? For a detector or receiver he can use a waterproof microphone (also a transducer). Would that be the simplest and most efficient installation? The answer is no! Here is why. To transmit voice through the water with good intelligibility would require, a band 100-3200 Hz or 5 octaves. This means that we must have a 5 octave amplifier and a 5 octave transducer. A 5 octave transducer is hard to find. Also, each frequency component of the human voice in that 5 octave range will suffer a different degree of loss or attenuation as it goes through the water. All underwater noises created by the ship's screws, hull, and by fish will interfere with the signal in that 5 octave range.

If we decide to modulate a carrier frequency higher than any frequency that the voice contains, say a carrier of 8 kHz, then the modulated output would cover a range of frequencies from 4,800 Hz to 11,200 Hz (8K ± 3200). This range is a 6400 Hz bandwidth or twice the modulating wave bandwidth. Something else has happened, however, in terms of octaves. An octave at 8 kHz would be 8000 Hz to 16,000 Hz. To transmit the voice about the 8 kHz carrier uses 6400/8000 of an octave or about eight tenths (0.8) of an octave at 8 kHz. Now the transducer need only produce output over an 0.8 octave range. Also there will be less loss variations and noise signals over an 0.8 octave range. than over a 5 octave range. The complexity of transmitting the voice underwater has been reduced by amplitude modulating at 8 kHz carrier tone. If a single sideband system were used the transmission bandwidth requirement would have been reduced by a factor of two or a bandwidth of only 0.4 octave would be required. The modulation process that was just described has actually been built into an equipment sometimes called "Gertrude", the AN/UQC-1 Single Sideband Underwater Communications Set.

E. Adding and Mixing Sine Waves

Previously, it was pointed out that the frequency band in any amplitude modulated wave is equal to twice the highest frequency component in the modulating wave. This is because each frequency input gives a pair of sideband frequencies centered about the carrier frequency at the output. At this point, after seeing how a practical AM modulator connects into the transmitter one might ask: How are these sideband frequencies generated? Isn't this just varying the amplitude of the carrier wave? How can the transmitter radiate at anything but the carrier frequency, if the oscillator is adjusted to be precisely at the carrier frequency? These are valid questions to ask. Understanding sidebands requires a knowledge of what happens when two sine waves are superimposed in both linear systems and non-linear systems.

So far we haven't said what a wave sum would look like. What is the sum of two sine waves? How do you add up these waves? Well, if the waves are both on the same frequency and same amplitude then you can see that the two are rising and falling together. If you add their heights above the base line together at corresponding points from where the waves started together you would get one big sine wave of the same frequency but twice the amplitude of either wave. The effect is slightly different if the two waves are the same amplitude but are at slightly different frequencies. By referring to Figure 4, you can see when the two waves of different frequency start out together corresponding points on each wave gradually separate until one wave is out of phase with the other. They reach a point where one wave is at a crest and at the same point the other wave is at a valley. The

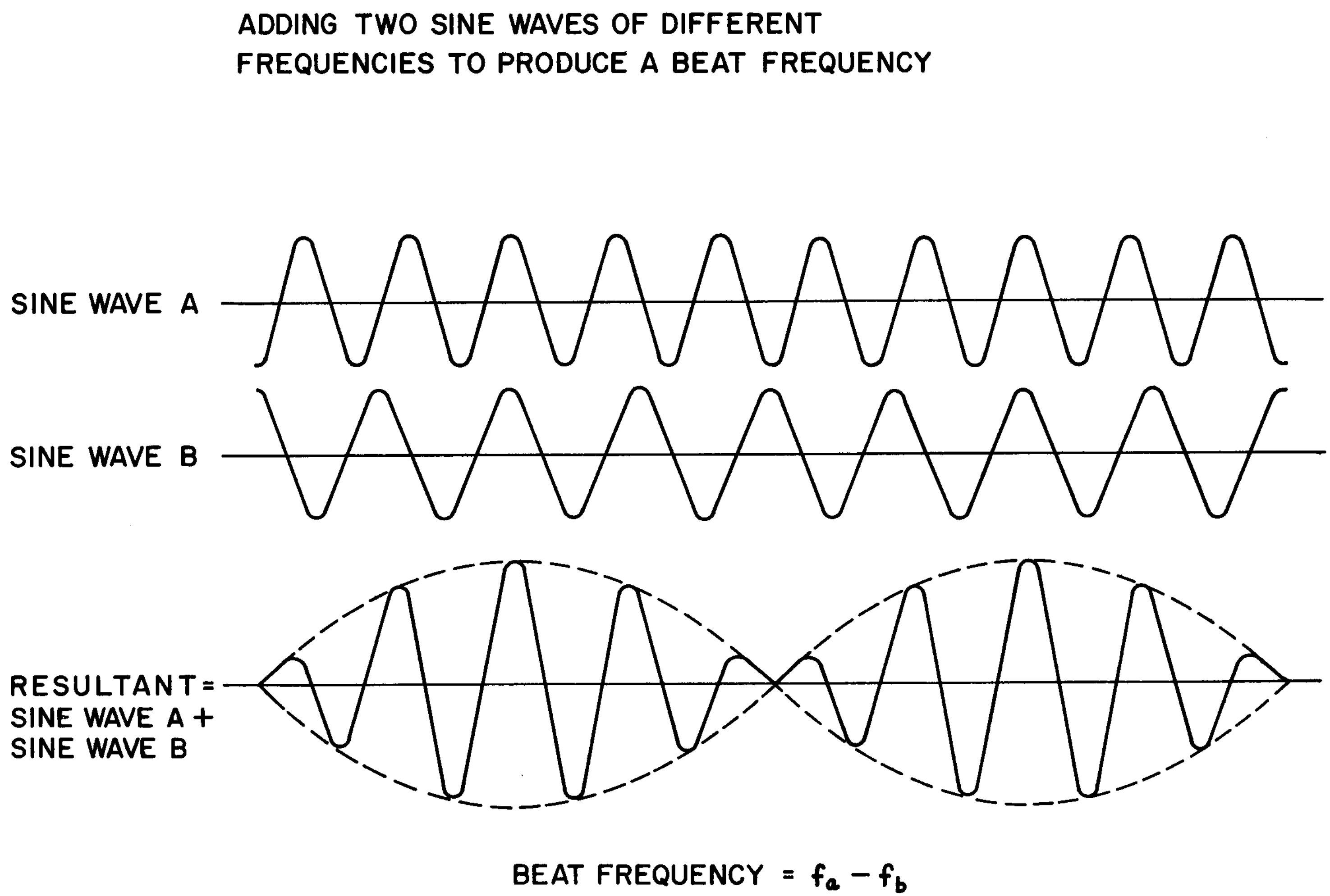



Figure 4. Adding Two Sine Waves of Different Frequencies to Produce a Beat Frequency

sum of the waves at this point is zero. Then they continue out of phase for a time only to come slowly back in phase again. At this in-phase point they are both crest at the same place on the graph. The sum of the waves at this point is twice the original amplitude. Thus, if the phase of the wave is uniformly changing relative to the other wave then adding the two waves results in a "sine wave" whose amplitude is pulsating from zero to twice the amplitude of either input wave. This produces what is called a beat frequency. Mathematically the beat frequency $f_{Beat}$ is:

$$f_{Beat} = f_1 - f_2$$

In Figure 4 wave A has a frequency of 10 Hz and wave B has a frequency of 8 Hz and they combine to form a wave which has $f_a - f_o = 10-8 = 2$ beats per unit of time, i.e. 2 Hz.

You can physically observe this beat frequency effect. This principle is used to tune a guitar or a piano.

In an actual transmitter the two waves are really multiplied together. This product of audio and radio frequency waves is equivalent to adding three waves, none of which is the original audio signal. The three are the carrier, sum frequency and the difference frequency. Later, it will be shown that in single sideband transmitters the carrier and either the difference or the sum frequency will be eliminated.

## 1.2 AMPLIFIERS AND LINEAR AMPLIFIERS

An amplifier is used to control large amounts of power by a weaker input or control signal. When the term amplifier is used in technical publications visualize something like that shown in Figure 5. In Figure 5 you see an input or control signal going into an amplifier. Also power is going into the amplifier, furnished by a power supply. At the output is a signal that looks similar to the input signal but it has greater amplitude or greater power level. Note that the amplifier does not produce power on its own; it merely controls power supplied by some kind of power supply. If the ship's generators fail, all the amplifier circuits on board are useless because they have no power left to control.

The amplifier can be thought of as a type of modulator because the power supplied to the amplifier from the supply is being "modulated" or changed by the input signal. In all cases, the output power or signal is greater than the input signal power because supply power is being consumed. The output power could be 10 times the input or 100 or 1000,000 times greater than the input. The multiplication factor is called the power gain of the amplifier. If an amplifier is said to be linear its gain is constant over a range of frequencies and amplitudes. Amplifiers amplify voltage as well as power. If an amplifier has a voltage gain of 10, it will have a power gain of 100, because power is proportional to the square of the voltage. In this formula the power is expressed in watts (W) and the voltage in volts (V)

$$\text{Power gain} = \frac{\text{W output}}{\text{W input}} = \frac{\text{V}^2\text{ output}}{\text{V}^2\text{ input}}$$

It can be seen that $\frac{\text{V output}}{\text{V input}}$ is the voltage gain and that:

$$\text{Power gain} = (\text{voltage gain})^2$$

### A. Linearity

A way of describing a linear amplifier is to say that if two or more waves go into the amplifier what comes out should be (proportional to) the simple sum of what went into the amplifier regardless of the frequencies or amplitudes of the input signals. Each wave would be treated by the amplifier as if the other waves were not present. As an example: a linear amplifier had a voltage gain of 10 and two waves were applied to its input. Wave A has an amplitude of 3 volts and wave B of 7.5 volts. Wave A has a frequency of 256 cps and B 440 cps. If this is truly a linear amplifier then the output would be a wave which is the sum of the two component waves. We shall discuss a little later what two waves added together look like and sould like. Anyway the output energy has one component with a frequency of 256 cps and an amplitude of 30 volts. The other component is a wave with a frequency of 440 cps and amplitude of 75 volts. Notes these two waves were amplified together as if each came through by itself. No frequencies were shifted. The voltage amplitude of each wave got multiplied by the same factor (10).

Another thing to remember about linear amplifiers is that the output wave will rise to its crest at almost but not quite the same instant that the input wave does. The difference in time is referred to as the time delay of the amplifier. The time delay of a good high quality linear amplifier is the same for waves of all frequencies. If all these properties hold true, the output wave will have exactly the same shape as the input wave. The technical way of saying this is that the linear amplifier does not produce distortion, or it is distortionless. HiFi amplifiers are examples of linear amplifiers.

Naturally when a person talks over a communication system he hopes he sounds like himself at the other end. Since linear amplifiers are the

## AMPLIFIER

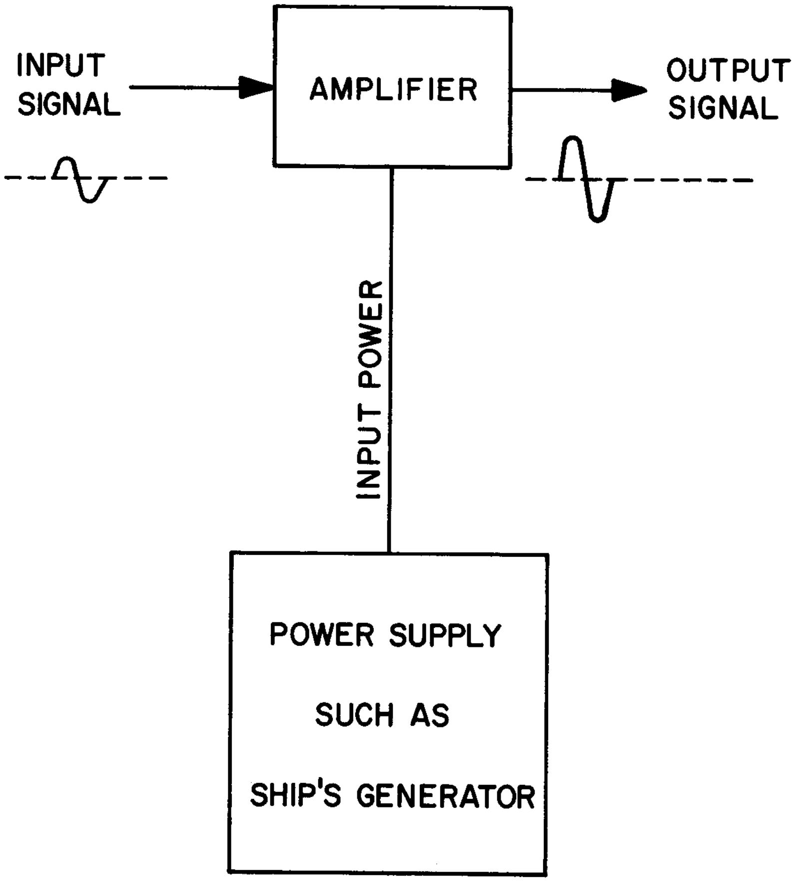



Figure 5. Amplifier

# LINEAR AMPLIFIER
# WITH DISTORTIONLESS OUTPUT

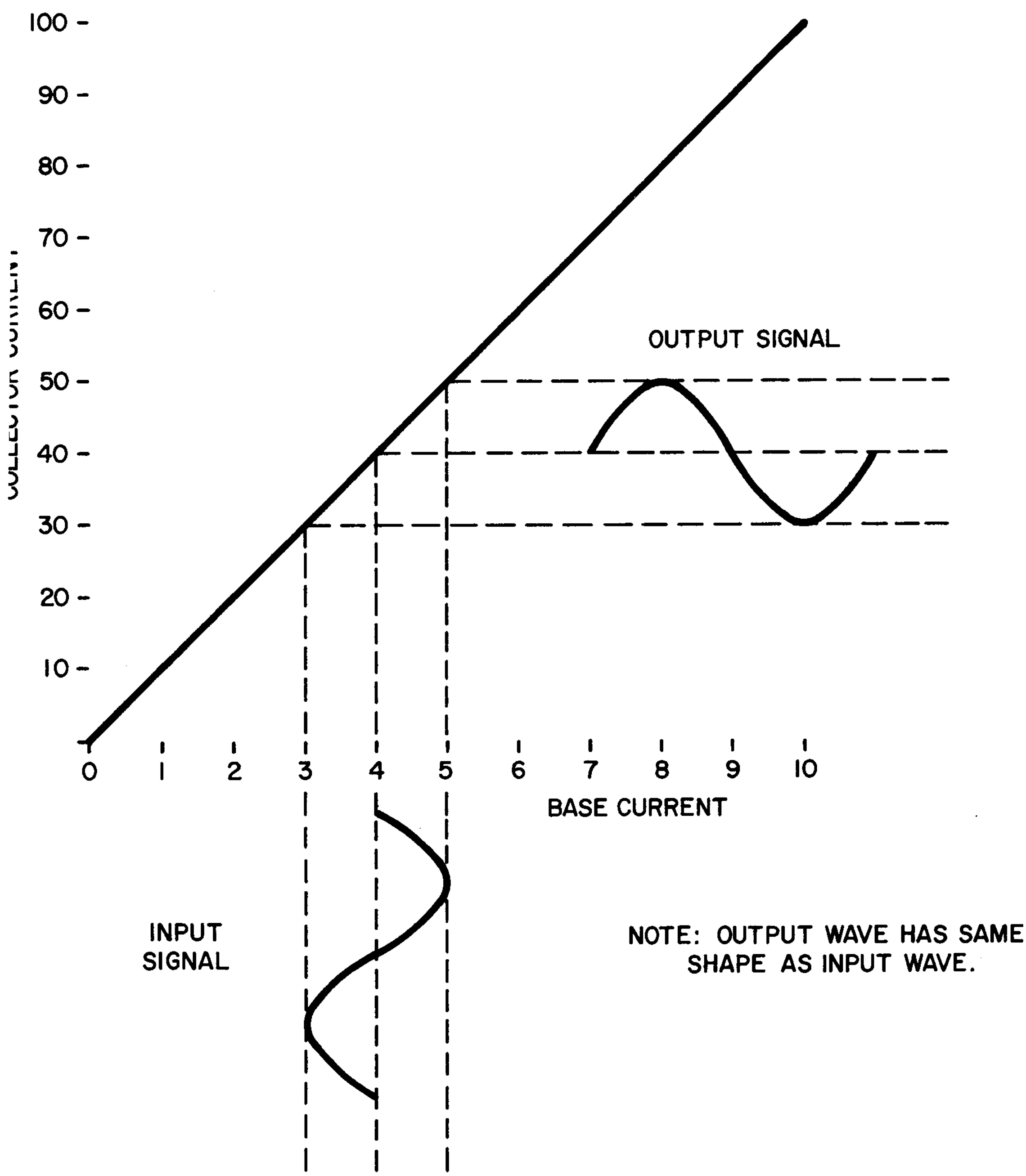



Figure 6. Linear Amplifier with Distortionless Output

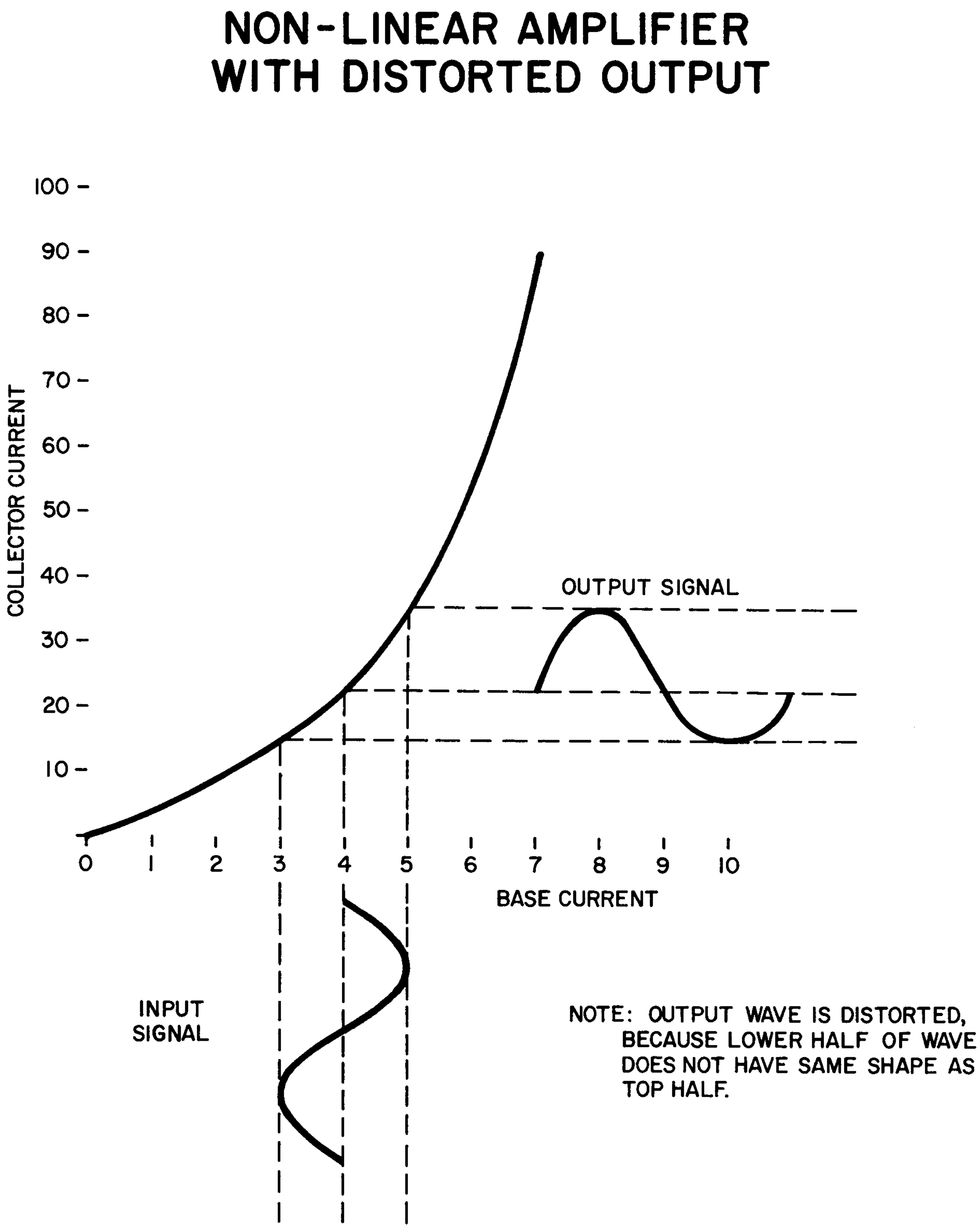



Figure 7. Non-linear Amplifier with Distorted Output

only type of amplifiers that can preserve individual voice qualities or modulating wave shapes, linearity is a most important property of amplifiers used for communications purposes.

The linear amplifier is one that produces an output wave identical to the input wave as shown in Figure 6. A non-linear amplifier will introduce distortion in the output as displayed in Figure 7. A distorted wave is rich in harmonics. Physically there is little difference between a linear amplifier and any other type of amplifier. Most amplifier circuits can be adapted to linear operation by the choice of proper operating voltages and load conditions. One of the least understood factors in SSB is that linear amplifiers are less efficient that non-linear amplifiers. The ratio of power output developed to supply power used in the linear amplifier is smaller. A technical point worth bearing in mind is that while a non-linear (Class "C") amplifier has a practical efficiency limit of about 80%, the linear (Class "B") amplifier efficiency will not exceed about 60% and is often 35-50%.

## 1.3 AMPLITUDE MODULATORS

Some basic terms and processes have been defined. We have also discussed amplifiers and their properties. We now want to talk in more detail about the process of Amplitude Modulators, and how they function.

In general, the frequency bandwidth containing an amplitude modulated signal is at least twice the highest frequency component in the modulating waves. Each frequency component that goes into an amplitude modulator produces at the output a pair of sideband frequencies centered about the carrier frequency. As we attempt to send more information over telephone wires or over the air, we have to transmit a wider band of frequencies or sidebands. In radio work we are primarily concerned with carrier frequencies from tens of kHz to several hundred MHz. These carrier frequencies are many times higher than the highest modulating frequency normally employed.

We still have to find a way to change or modulate the carrier wave in order to communicate or transmit information. The familiar telephone contains a carbon microphone which varies the amount of current flowing from the battery to the telephone line. We could call the carbon microphone an amplitude modulator because it causes changes in the amplitude or level or magnitude of the carrier signal, which in this case is a direct current.

When we wish to modulate a radio frequency carrier we need to use two amplifiers. One amplifier will be used to amplify the audio signal (voice) that is to be used to modulate the carrier. The other amplifier will be used to amplify a source of RF energy of the appropriate frequency. The power supplied to the RF amplifier will be modified and changed by the audio amplifier that is amplifying the voice signal. This can be done in several ways, but the most common method is to use a transformer whose primary is connected to the audio amplifier and whose secondary is placed in series with the power supply of the RF amplifier. When this is done, the power to the RF amplifier can be controlled or modulated in accordance with the audio signal.

When we amplitude modulate the radio frequency carrier with audio frequency sine waves, two new waves are generated by the process of modulation. If the carrier frequency is $f_c$ and the audio tone or modulating signal is $f_m$ then we get the difference-frequency, $f_c-f_m$, and the sum-frequency, $f_c+f_m$. These two are more commonly referred to in technical books as the lower sideband and upper sideband. These two sidebands are radiated by the AM transmitter right along with the carrier wave. At the receiver an AM wave is really three (3) waves added or superimposed together. For example, if a 1 kHz (1000 Hz) signal were inserted into an AM transmitter that had a carrier frequency at 3000 kHz, the signal going out over the air is a 3000 kHz carrier, a 3001 kHz upper sideband and a 2999 kHz lower sideband. To hear this signal you would need a receiver that was sensitive to a band of frequencies from 2999-3001 kHz, i.e. a 2 kHz bandwidth. If your receiver was so selective that it brought in only the 3000 kHz carrier, it would not reproduce the 1000 Hz signal.

If two tones are injected into an AM transmitter, two pair of side-band frequencies will be generated. With voice communications, of course, there are many complex waves going into the modulator-- this is equivalent to sine waves all added together. In this case, the transmitter will be emitting sidebands over a range of frequencies. All that need be remembered here is that the highest (or lowest) sideband frequency is the carrier frequency plus (or minus) the maximum frequency that the modulation signal contains. When voice is used to modulate a transmitter the highest frequency needs to be about 3 kHz for good intelligibility. Double sideband AM voice, therefore, can be delivered with a 6 kHz radio bandwidth frequency ($f_c \pm 3$ kHz) and should be detected with a receiver having about a 6 kHz bandwidth.

The two sidebands produced in an AM transmitter reach their maximum amplitude in relation to the carrier when the audio wave amplitude is equal to the carrier amplitude. You can generate stronger sidebands by increasing carrier or audio waves, but only up to the point where the audio is equal to carrier amplitude. The optimum situation then, produces two sidebands, each of which is one-half the carrier amplitude. If you go past this optimum condition i.e. increase the

# AMPLITUDE MODULATION

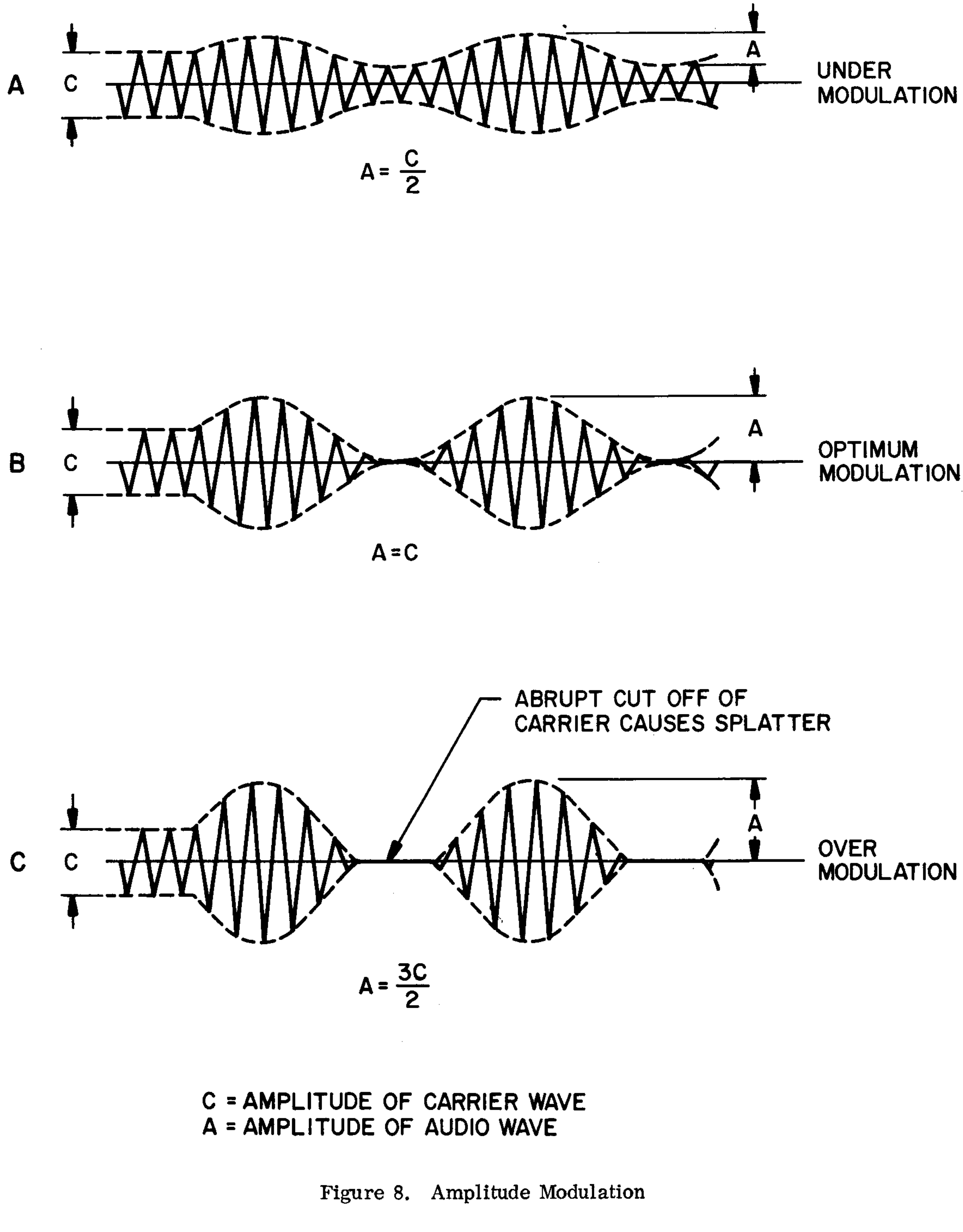



Figure 8. Amplitude Modulation

audio amplitude beyond the carrier amplitude you are, in engineering language, over modulating. Over modulation in the AM transmitter is harmful for communications because in addition to reducing the sideband power (which carries your voice) it generates spurious sideband frequencies called splatter and causes distortion of signals at the receiving end as shown in Figure 8.

In practical terms where we've used the term amplitude we could have substituted voltage or current. It is standard procedure to refer to a sine wave's amplitude as the peak voltage or current in AC volts or amps. A quick review of basic electricity gives us the relationship between the power dissipated and the current or voltage. Using Ohm's law we find that where the resistance stays constant the power follows the square of the applied voltage or current:

$$P = I^2 R \text{ or } P = \frac{E^2}{R}$$

The power relations can now be calculated from what has been already said about amplitudes (i.e. voltages). As an example, an AM transmitter which has a carrier output of 100 watts is modulated by a single audio tone at the optimum condition (the 100% modulation point). This modulation produces two sidebands which are the sum and difference frequencies. The voltage of each of the sidebands is half that of the carrier voltage. Therefore, the power in each sideband is $(1/2)^2$ times that of the carrier. Since it was assumed that the carrier output was 100 watts, the power in each sideband is 1/4 of that or 25 watts. The total sideband power, since there are two, is 50 watts. This sideband power comes from the audio amplifiers in the modulator. If the amplifiers are perfectly efficient a 50-watt modulator or 50-watt audio amplifier would be needed to get the full communications usefulness out of the 100-watt radio frequency amplifier. The total average transmitter power is 150 watts if you include the modulator power. Two-thirds of the available power is going into the carrier (100 watts) and one-third is going into the production of useful sidebands (50 watts). An AM transmitter could also be called a double sideband (DSBAM), full carrier, amplitude modulated transmitter. Amplitude modulators are one of the oldest and most common types of modulators used in radio. Single sideband (SSB) modulation is really just a special case of AM where the concept of changing the signal level or amplitude in accordance with the modulating wave is used to its best advantage.

### A. Single Sideband Modulation

There is nothing very special about the term "sideband" in radio telephone work. We have stated in the previous pages that there is no known way to modulate waves or transmit information that doesn't generate some kind of sideband frequencies. Frequency modulation (FM) and phase modulation (PM) systems also set up sidebands. Multiple sets of sidebands on either side of the carrier are generated when you attempt to modulate by changing frequency rather than changing amplitude. These modulation schemes are beyond the scope of this book.

Single sideband is just another form of amplitude modulation. The main difference is that available transmitter power is used to greater advantage. Having gotten over the hard part of understanding conventional AM, SSB is easy to comprehend.

The name "SSB" tells you that this is a system where only one sideband is transmitted. It normally implies (although the name doesn't tell you) that the carrier wave is not transmitted. The carrier wave does not change in amplitude or frequency in AM systems. Since nothing changes in the carrier, no information is sent by the carrier, and no information is lost if the carrier is suppressed or eliminated. Actually, this system of modulation is more properly called single sideband, suppressed carrier (SSSC) but it is usually shortened to just SSB.

In regular AM (DSBAM) we have seen that the total power contained in the two sidebands is one-half the carrier power. Since the carrier must be present in order to be modulated, although it conveys no intelligence in itself, it can be cancelled or balanced out. This is what a SSB transmitter does--it permits sideband frequencies to be generated but cancels out the carrier after it has served its purpose. When this is done at low power levels a lot of power is saved. In addition to the power saving feature of cancelling out the carrier, the sum and difference frequencies or upper and lower sidebands are identical and redundant as far as information carrying ability goes. If one sideband can be eliminated additional power will be saved. Also and most important, the bandwidth of a single sideband emission is half that of the double sideband emission--a fact which means a saving of frequency spectrum "space". Suppose that you inserted one kHz signal into the microphone of an SSB transmitter that had a carrier frequency of 3000 kHz. In the output of the SSB equipment the 3000 kHz carrier would be cancelled out as well as one sideband--for example, the lower sideband. What would go out over the air would be only a 3001 kHz sum frequency or upper sideband. Now a narrow band receiver sensitive only to a 1 kHz frequency band could conceivably reproduce your signal. Similarly, since a 3 kHz band of sine wave frequencies can convey your voice, a SSB receiver should have about a 3 kHz radio frequency bandwidth, as opposed to the 6 kHz required for an AM.

Now we can determine the power a SSB transmitter has to develop to be equal to an AM transmitter. Remember the AM transmitter with a 100 watt carrier output required 50 watts of audio power to get the maximum possible sideband power of 25 watts per sideband. Apparently, we can transmit the same amount of information by spending only 25 watts average power in the SSB setup instead of the total of 150 watts for AM. Thus the SSB system saves 125/150 or 5/6 of the average power because of suppression of the carrier and the one sideband. Naturally the 25 watt SSB transmitter which is equivalent in broadcasting ability to the 150 watt AM transmitter is going to be smaller and lighter. Weight and size are important factors on crowded ships.

There are other advantages for SSB which are not as obvious. On long distance hops at high frequencies (3-30 MHz) AM is subject to what is known as selective fading. The transmission route for long-haul radio communications is via the ionosphere. The ionosphere is a layer of charged particles floating about 60 miles above the earth and it reflects radio waves thus making possible long distance HF communications. Under poor propagation conditions (when the ionosphere is disturbed) an AM transmission will be distorted and fade in and out more than a SSB signal. The upper sideband, lower sideband, and carrier must be received exactly as transmitted to get a clear, full power AM signal. If one or both sidebands are received weaker (attenuated) than the carrier signal, or if the carrier is received weaker than the sideband frequencies (as happens during fading due to the ionosphere), the received signal will not be detected properly by the receiver. Since SSB is the "sum" of only one band of waves, a single sideband, there is a much better chance of getting through with a good signal. Then too, the SSB receiver requires only half the bandwidth which means only half as much interference and only half as much noise due to receiver hiss. Another important factor is that on crowded AM frequencies you hear a lot of "squealing" or heterodyning, caused by all the carriers "beating" together. This effect is not present if all users of a particular radio telephone band transmit SSB, since no carriers will be present.

The main disadvantages of a SSB system are the need for extremely stable oscillators in both the transmitter and receiver, and the somewhat greater cost and complexity of circuits which cancel out the carrier and the unwanted sideband.

B. Single Sideband Generation

The question now arises as to how the SSB transmitter actually eliminates the carrier and one of the two sidebands. There are several methods for generating SSB signals. All methods employ a special type of AM modulator called a balanced modulator. A balanced modulator takes a carrier wave and modulating wave input, multiplies them together (mixing) to produce the sum and difference sidebands, and then cancels out the carrier energy. The easiest balanced modulator circuit to understand employs two amplifiers as shown in Figure 9.

Transformers are used to couple or insert signals in and out of the circuit. The two amplifiers are connected together in what is known as a push-pull circuit. In this arrangement both the input and output leads from the two amplifiers are connected to the opposite ends of the transformer windings in a balanced circuit. For the moment ignore the RF oscillator in the diagram and regard the input transformer, secondary winding center tap as being grounded. At any instant the ends of the input transformer secondary will be at opposite polarity with respect to the center tap, if a sine wave signal is applied to the transformer primary winding. This means that while one amplifier is getting the positive crest of the input wave the other amplifier is getting the negative excursion of the same wave. Hence the output voltage or current of one amplifier is completely (180°) out of phase with that of the other amplifier.

To make a balanced modulator out of the basic push-pull amplifier, the RF oscillator is added to the circuit as is shown in Figure 9. Note that the oscillator is injecting carrier at the transformer center tap. A sine wave applied to the secondary at this point will appear at both ends of the transformer winding in phase. The push-pull configuration has the unusual property of amplifying only those signals which are of opposite phase at the inputs of the two amplifiers. The carrier wave is balanced out and does not appear in the output transformer, since it is applied in phase to the push-pull amplifier. The audio waves are applied out of phase by means of the input transformer; therefore, the sidebands they generate by mixing with the carrier do not cancel in the output circuit.

In any balanced modulator there will be no output until the audio is applied because of the carrier cancellation process. When the modulating audio is applied the balance is upset and the sidebands or sum and difference frequencies appear in the output. The carrier never appears at the output. What you have at this point could be called a double sideband, suppressed-carrier (DSBSC) transmitter. It would require the same bandwidth as regular AM because both sidebands are being generated, but would save two-thirds of the power because of suppression of the carrier. (Recall example of AM 100-watt carrier and 50-watt sidebands).

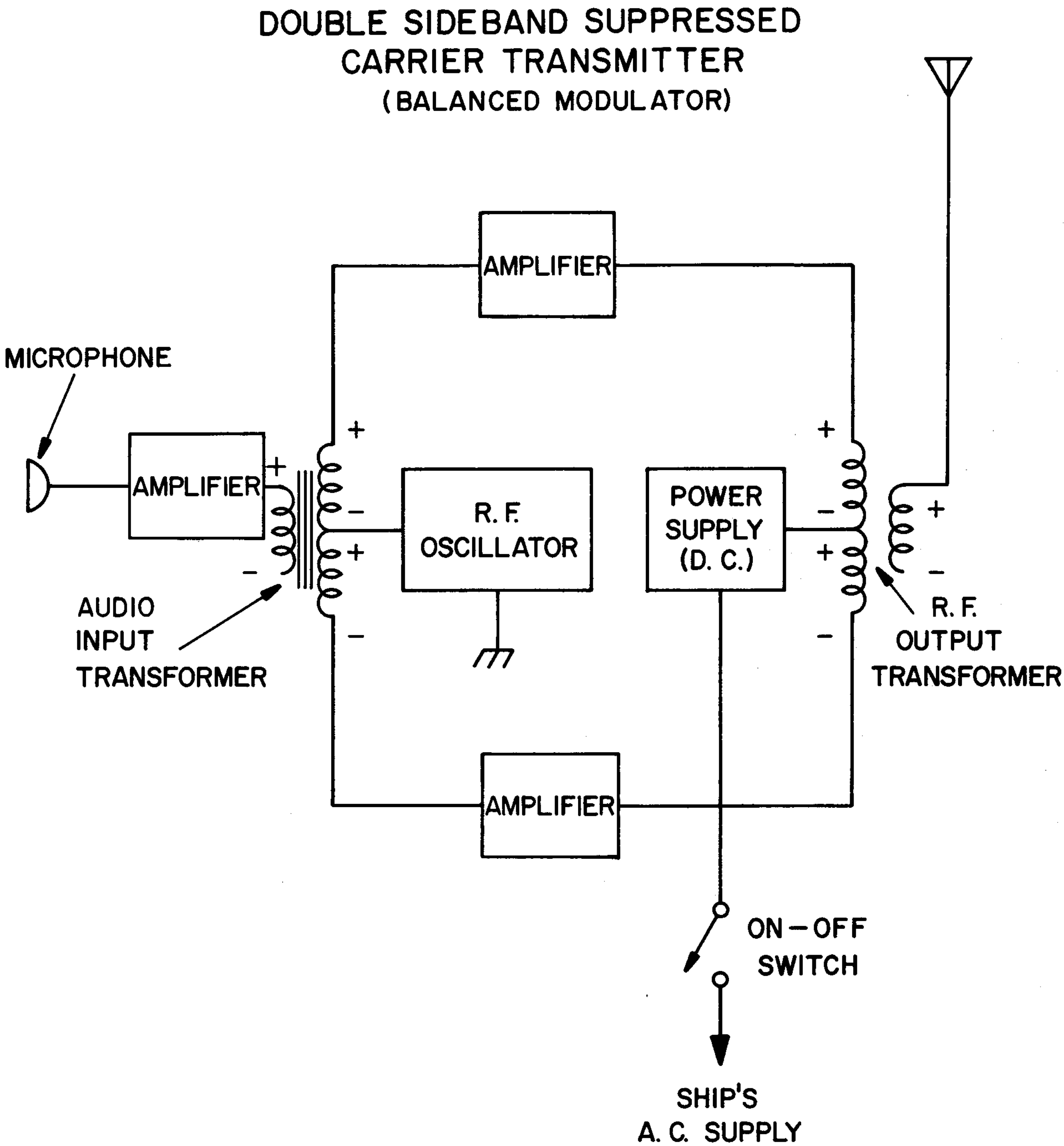



Figure 9. Double Sideband Suppressed Carrier Transmitter (Balanced Modulator)

## 1.4 TRANSMITTERS

We have discussed the concepts of frequency and wavelength. We have talked about carriers and modulation. We have discussed the theory and practice of adding versus mixing several sinewaves together. We have spoken about amplifiers and amplitude modulators, the electronic devices that do the adding, the mixing, the amplifying, and the modulating. We now wish to discuss the uses that these devices are put to, in the design and construction of a series of simple transmitters: a CW (Morse Code) transmitter, an Amplitude Modulated (AM) transmitter and a Single Sideband transmitter.

First, to generate the carrier frequency you will need an oscillator. An oscillator is an amplifier whose output is connected back to the amplifier input through a crystal or resonant circuit tuned to a particular frequency. As the output signal is fed back to the input it picks up more power from the amplifier power supply. This process repeats until a sine wave signal is produced that can be used in the radio transmitter.

### A. The Elementary CW Transmitter

At this point we can define the necessary parts of a continuous wave (CW) transmitter. To do this, connect the oscillator to an amplifier capable of handling the oscillator radio frequency waves. In addition, some sort of switch or key is needed in the power supply circuit to send Morse code. Finally a suitable antenna is necessary to radiate the signal. Such a transmitter is shown in Figure 10.

### B. The Elementary Amplitude Modulated (AM) Transmitter

Transmitting voice or teletype tones with an AM transmitter is not quite so simple. It is necessary to add a modulator to the CW transmitter to make it into an AM transmitter. To build a modulator, a microphone, an audio amplifier and a transformer are needed. This amplifier should be capable of handling all the sine wave component ponents in your voice. The transformer couples or combines the audio output from this amplifier into the power line going to the radio frequency (RF) amplifier from its power supply. The block diagram is shown in Figure 11 and can be compared to Figure 10. As you speak, the audio amplifier is adding and deleting power in series with the RF amplifier's power supply. This changes the amplitude of the radio signal at an audio or voice frequency rate. That is what "AM" means. If no one is talking into the microphone no voice power is being injected into the RF amplifier

### C. The Elementary Single Sideband Transmitter

In the previous section of the handbook concerned with Single Sideband Generation, we described the balanced modulator and the way it operates to suppress the carrier while allowing the generation of both the upper and the lower sideband. We mentioned that if this signal were radiated, a 2/3 saving in power over the conventional Double Sideband AM transmitter would result. In experimenting with emissions of this kind, scientists and engineers have discovered the DSB Suppressed Carrier Transmission is very very critical to frequency drifts at the receiver. Just a few hertz drift in the receiver local oscillator or BFO and the transmission is unintelligible. The simple addition of a filter converts the balanced modulator into an entirely acceptable single sideband transmitter of low power. All that remains to make a useful SSB transmitter out of the filtered modulator, is the addition of a suitable linear amplifier. Such an elementary transmitter is shown in Figure 12.

Of course the filter that follows the balanced modulator in this circuit must have sufficient selectivity so that it can pass one sideband and reject the other. The filter also helps reject any residual carrier, because no balanced modulator is completely perfect. In conventional SSB transmitters, a switching arrangement is often included with a pair of filters so that either the upper or the lower sideband can be selected or rejected at the operator's choice.

### D. Emissions of the Elementary Transmitters

We have described how very simple forms of the typical CW, AM, and SSB transmitters can be made. We would like to summarize in pictorial form the emissions of these transmitters when they are operating properly. In Figures 13A and 13B these emissions are portrayed for a carrier frequency of 3000 kHz (3.0 MHz). In Figure 13A1 we see the emission of the CW transmitter with its key "down". The emission is a single spectral line at the frequency of the carrier (3000 kHz). In Figure 13A2 we see the emission of the AM transmitter being modulated with a 2 kHz audio tone. The carrier is present, and the upper and lower sidebands are present, spaced as single spectral lines 2 kHz above and below the carrier.

Figure 13A3 portrays the transmitter emission of the balanced modulator when no single sideband filter is used. The emission consists of two spectral lines 2 kHz above and below the position of the suppressed carrier. In practice it is impossible to completely suppress the carrier, because no balanced modulator is perfect. The state-of-art at present allows suppressions of the order of 60 decibels (a power ratio of 1,000,000 to 1). Finally, Figure 13A4 shows the emission of the SSB transmitter that has been set to pass the upper sideband (USB) and to suppress both the carrier and the lower sideband (LSB). The

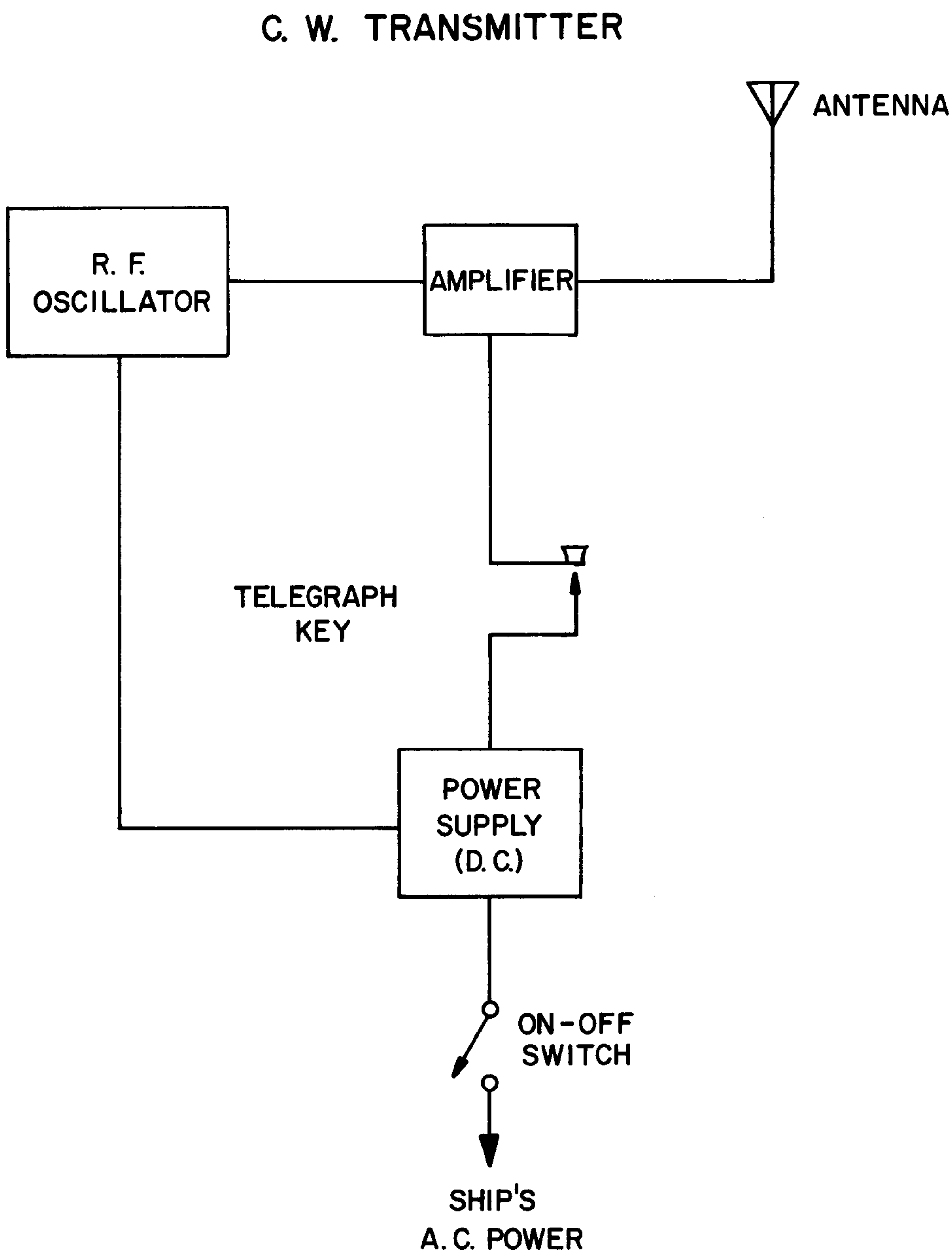



Figure 10. CW Transmitter

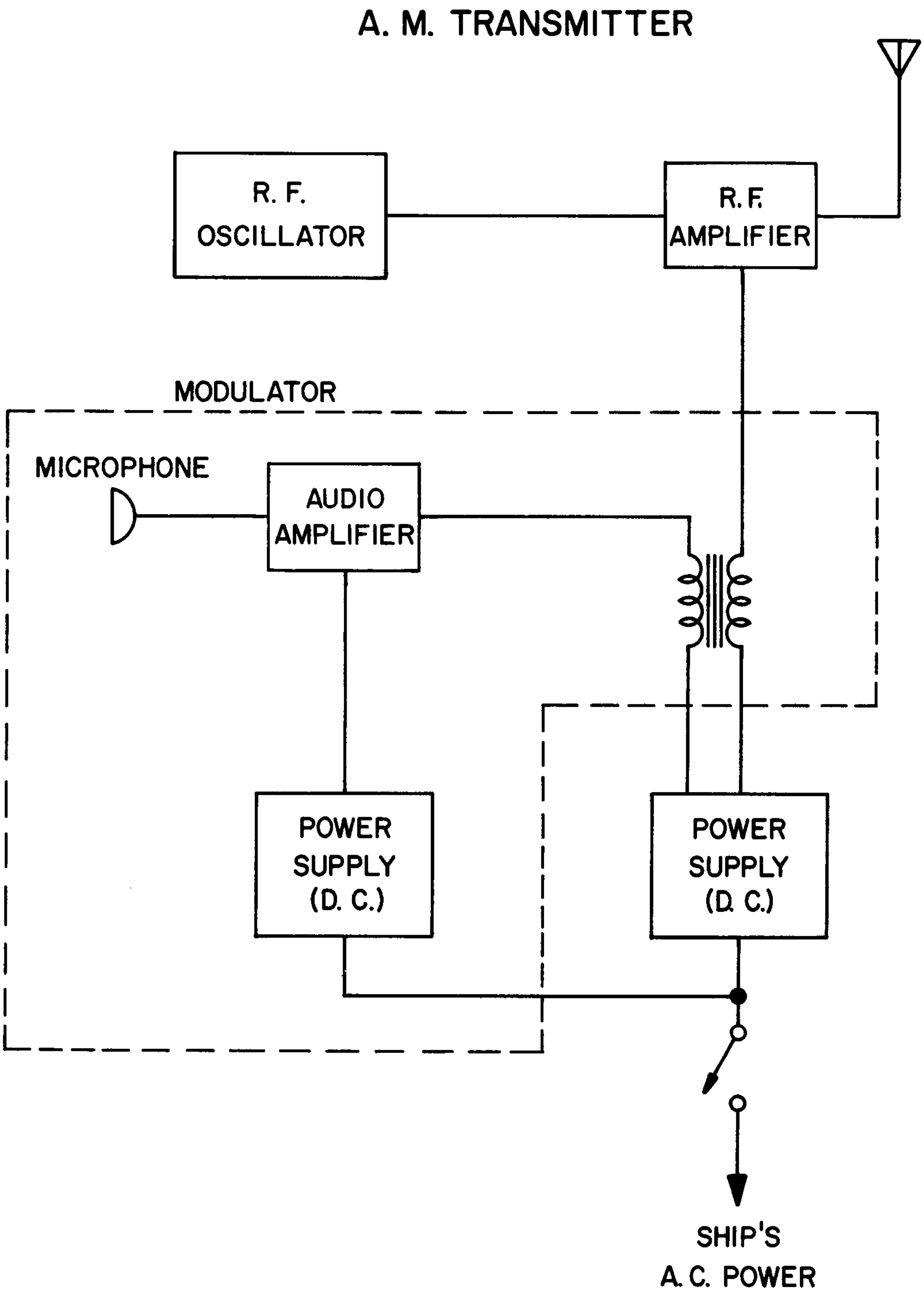



Figure 11. AM Transmitter

# SINGLE SIDE BAND TRANSMITTER

(FILTER TYPE)

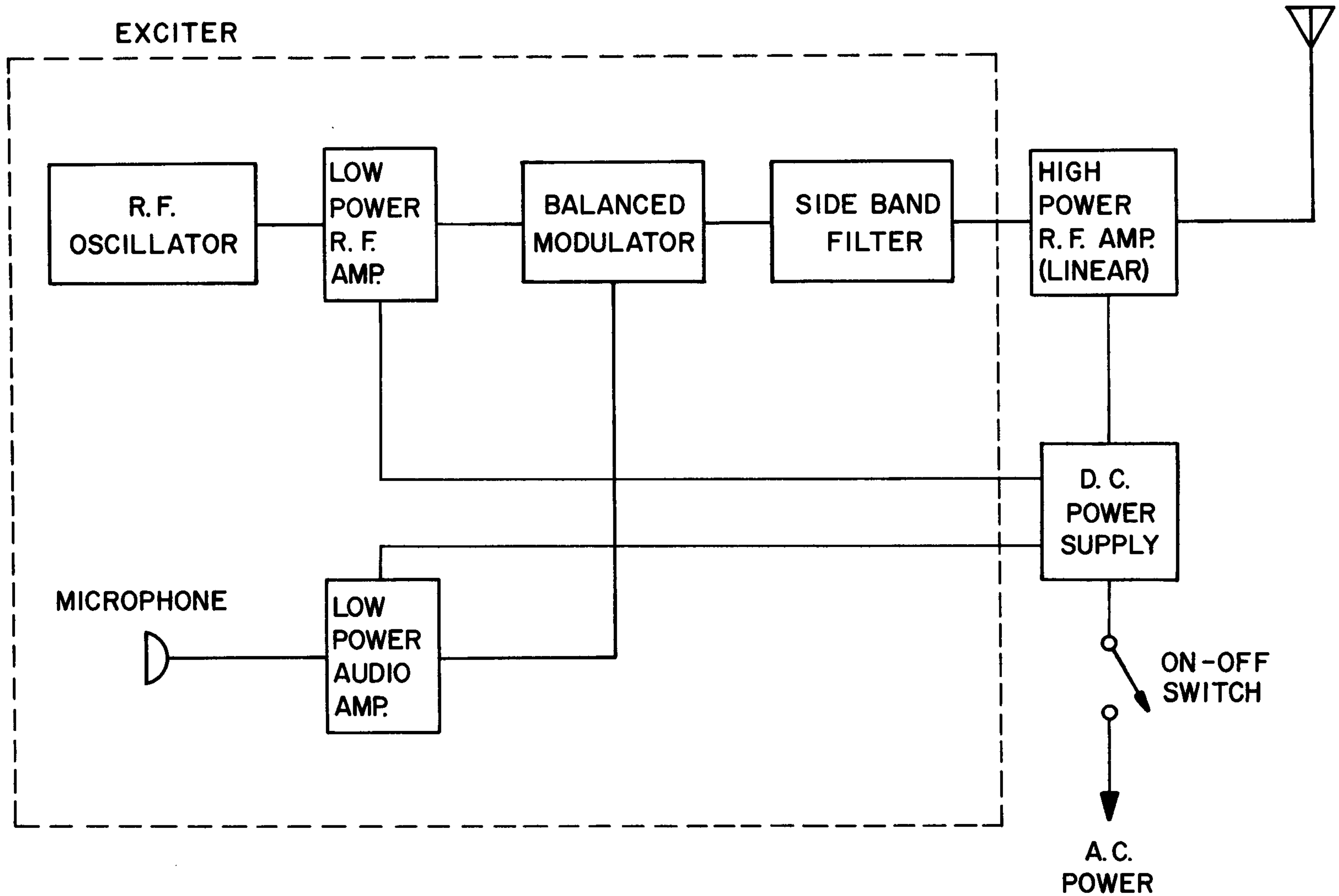



Figure 12. Single Sideband Transmitter (Filter Type)

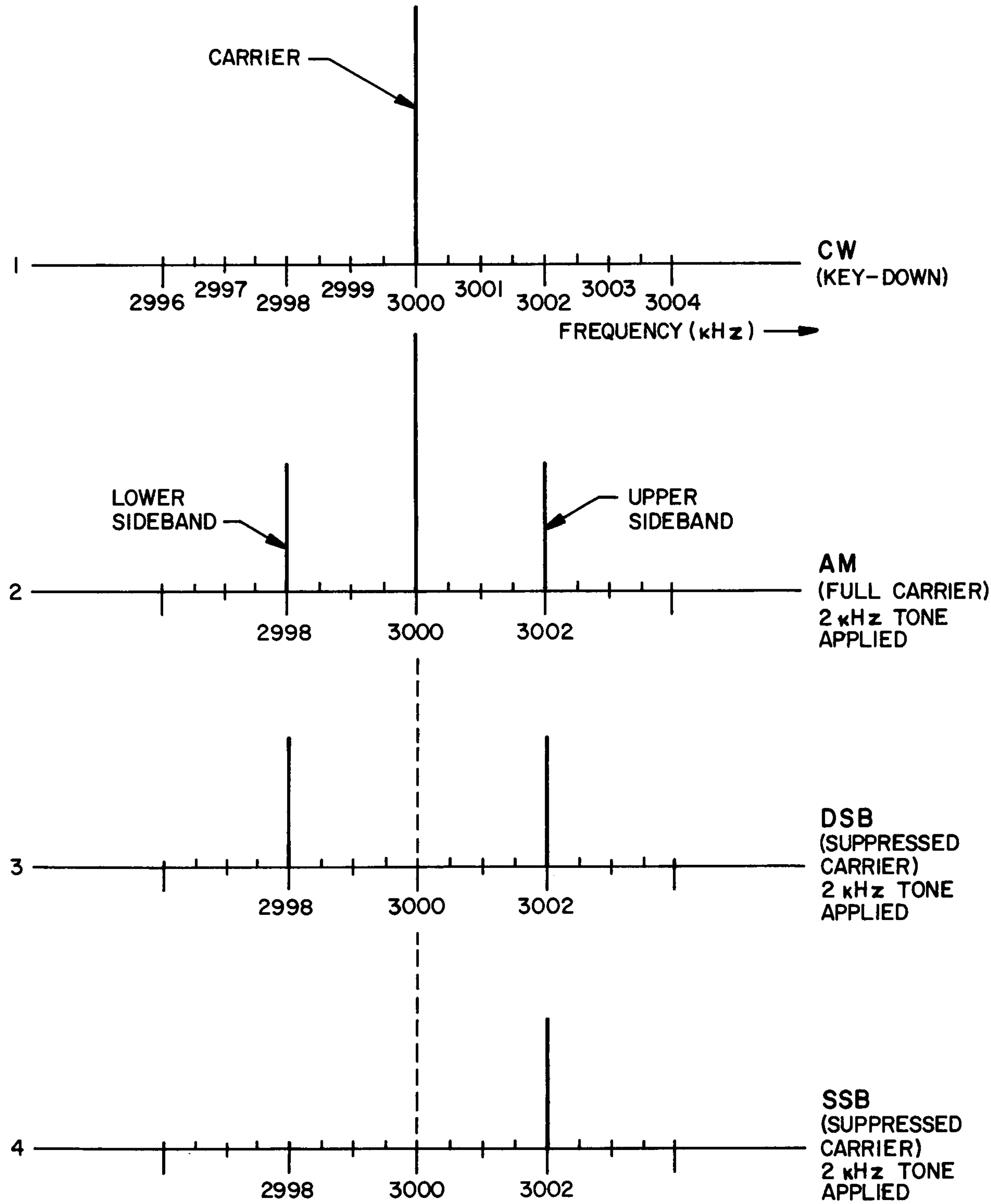



Figure 13A. Transmitter Emissions (Carrier Frequency at 3000 kHz)

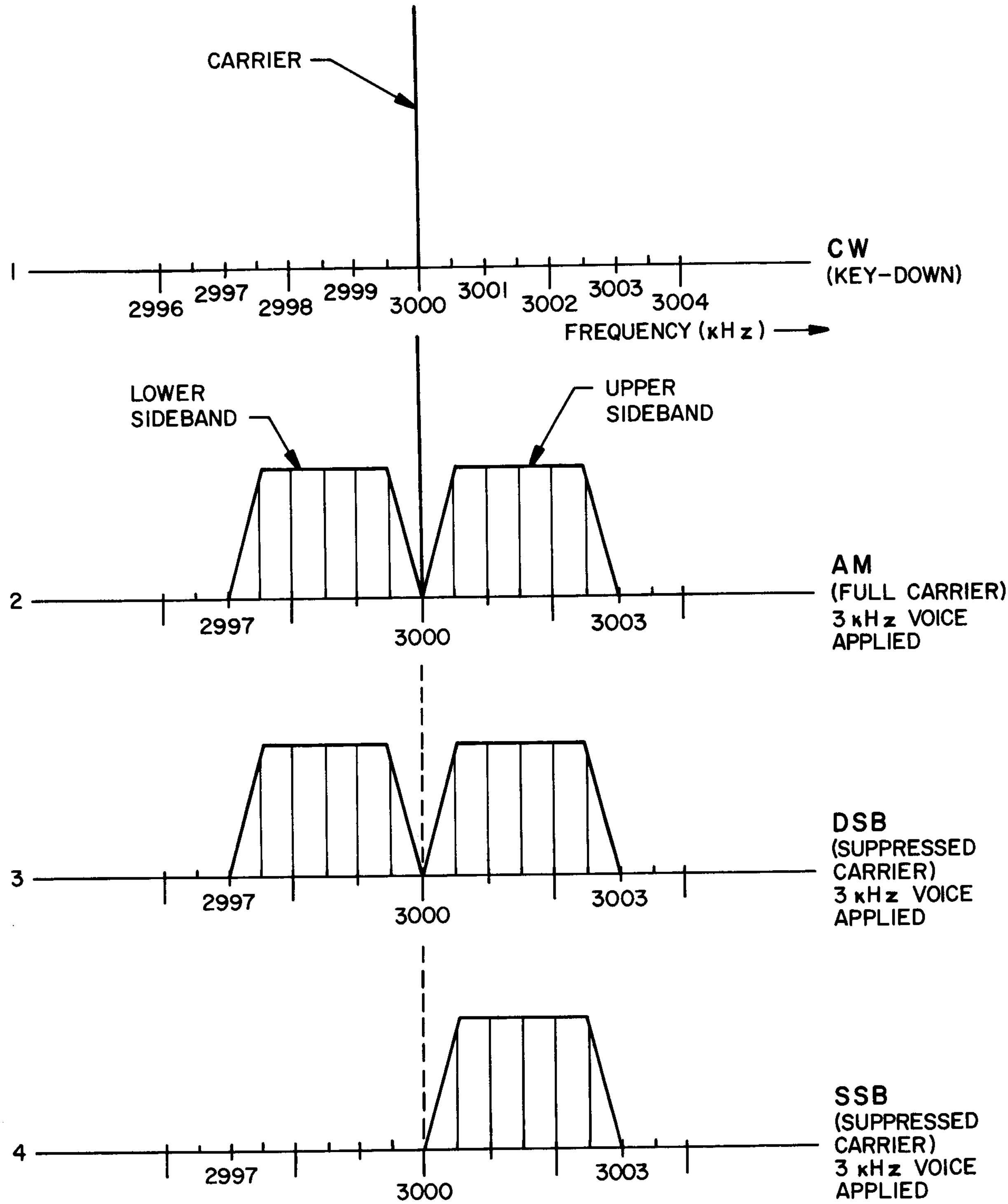



Figure 13B. Transmitter Emissions (Carrier Frequency at 3000 kHz)

The emission is a single spectral line spaced 2 kHz above the frequency of the missing carrier.

Figure 13B portrays much the same information except that a voice signal containing a band of audio frequencies extending from a few hundred Hz up to about 3 kHz was used to modulate the transmitter. Figure 13B2 shows the emission of the AM transmitter and the carrier is seen as the single spectral line at 3000 kHz, in the middle of the upper and lower sideband energy. Figure 13B3 shows the Double Sideband Suppressed Carrier emission from an unfiltered balanced modulator. Both sidebands are present, but the carrier has been cancelled out. Finally, Figure 13B4 shows the emission of the voice modulated SSB transmitter. The emission is in the upper sideband, ranging up to 3 kHz higher in frequency than the suppressed carrier.

## 1.5 STABLE OSCILLATORS/FREQUENCY SYNTHESIZERS

Frequency control in SSB communication is much more critical than for CW or AM modes. Formerly, the only type of transmitter that could meet the unusually high degree of frequency stability required by SSB was one that was crystal controlled. Today, frequency synthesizers are the standard means of frequency control. A synthesizer uses an internal or external crystal oscillator as a reference. In a crystal oscillator the usual frequency determining resonant circuit is replaced by a mechanically vibrating piezoelectric crystal. Generally, due to crowded frequency assignments and variations in propagation conditions it is desirable to be able to tune the 2-30 MHz. HF military band in increments of 0.1 kHz. Without a synthesizer, this tuning range would require a set of 280,000 crystals. Needless to say, no one could even find room aboard ship for such a set of crystals. The only problem of using a box of crystals concept for frequency control, in fact, was their availability. If you have ever had experience with the old TED/URR-35 UHF equipment, you are surely familiar with this problem.

The name "Frequency Synthesizer" sounds very technical and complicated, but actually the general idea is much easier to understand than anything we have discussed. A frequency synthesizer is a device that takes a frequency from a master crystal oscillator, generates other frequencies by using multipliers and dividers, and then combines these multiplier and divider outputs into the desired "synthesized" output frequency.

The main idea in a frequency synthesizer is to employ the beat frequency concept discussed earlier to reduce the number of crystals. This is done by using harmonic generators and frequency dividers.

A harmonic generator is a non-linear amplifier with the output tuned to one of the distortion harmonics. Harmonics are the additional sine waves that are multiplications of the basic or fundamental frequency. Thus, a non-linear amplifier rich in second harmonic distortion produces a substantial output wave which is twice the input or fundamental frequency. In this way any desired frequency multiplication such as the generation of second, third, fourth, and higher-order harmonics can be obtained. Filters must be employed to select the desired harmonic and reject undesired harmonic distortion. There also exist classes of amplifier circuits which behave as frequency dividers.

By mixing combinations of various frequency multipliers and dividers practically any number of "synthesized" output frequencies can be generated from one master crystal oscillator. The block diagram in Figure 14 is a frequency synthesizer composed of a 1000 kHz master oscillator. Using this synthesizer one can tune to, say, 1300 kHz. In one branch the 1000 kHz signal is divided by 10, then multiplied by 7 for a mixer input of 700 kHz. Another circuit takes some of the master oscillator 1000 kHz waves and multiplies them by 2 for a second mixer input of 2000 kHz. In the mixer-amplifier the 2000 kHz and 700 kHz waves are combined to obtain a beat frequency of 1300 kHz and a 2700 kHz sum frequency. The 2700 kHz wave is filtered out leaving only 1300 kHz signal. In a like manner any 100 kHz step from 1000 kHz to 2000 kHz could be obtained only by changing the multiplier.

Extensive filtering and extremely judicious selection of operating frequencies is required for even the simplest circuits. Spurious frequency problems increase as the output frequency range desired increases. Even so, frequency synthesizers utilizing a single master oscillator are nearly standard today in military SSB equipment.

To further enhance the excellent stability characteristics of a frequency synthesizer a servo mechanism may be used with an internal or external precision (oven-controlled) reference oscillator or frequency standard, e.g. the AN/URQ-9 or the AN/URQ-10. A block diagram of a stabilized master oscillator is shown in Figure 15. The system principle is that a sample of the master oscillator output is compared with the frequency standard output. The comparator or error detector is similar to a FM receiver. It develops an amplitude variation in response to a frequency shift. This error signal is amplified and used to drive a small DC motor. The motor, in turn, operates a variable resistor or other feedback device which modifies the oscillator circuit. If properly designed, such a servoloop exactly compensates the oscillator's frequency for any drift. If the master oscillator drifted up 2 Hz because

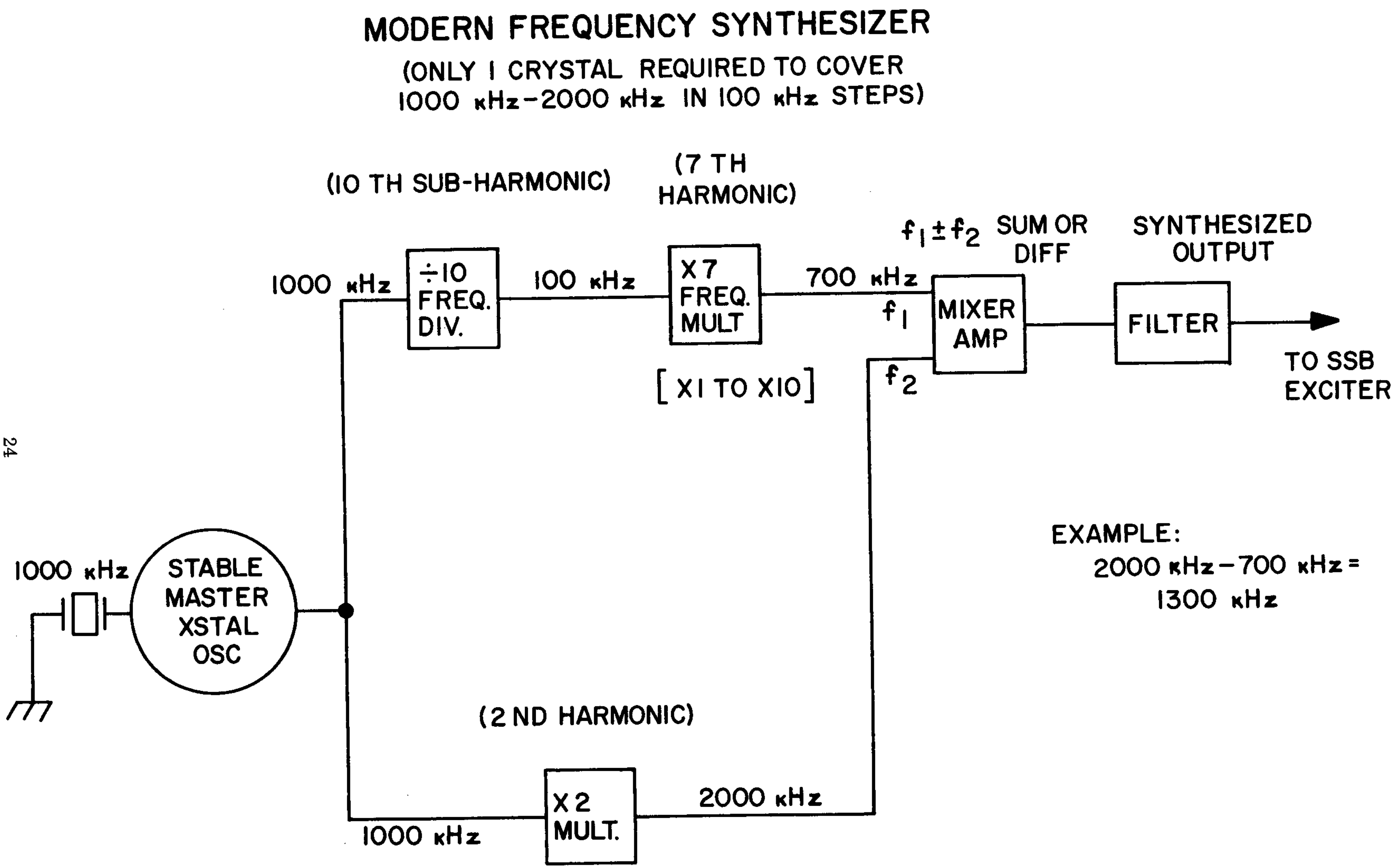



Figure 14. Modern Frequency Synthesizer

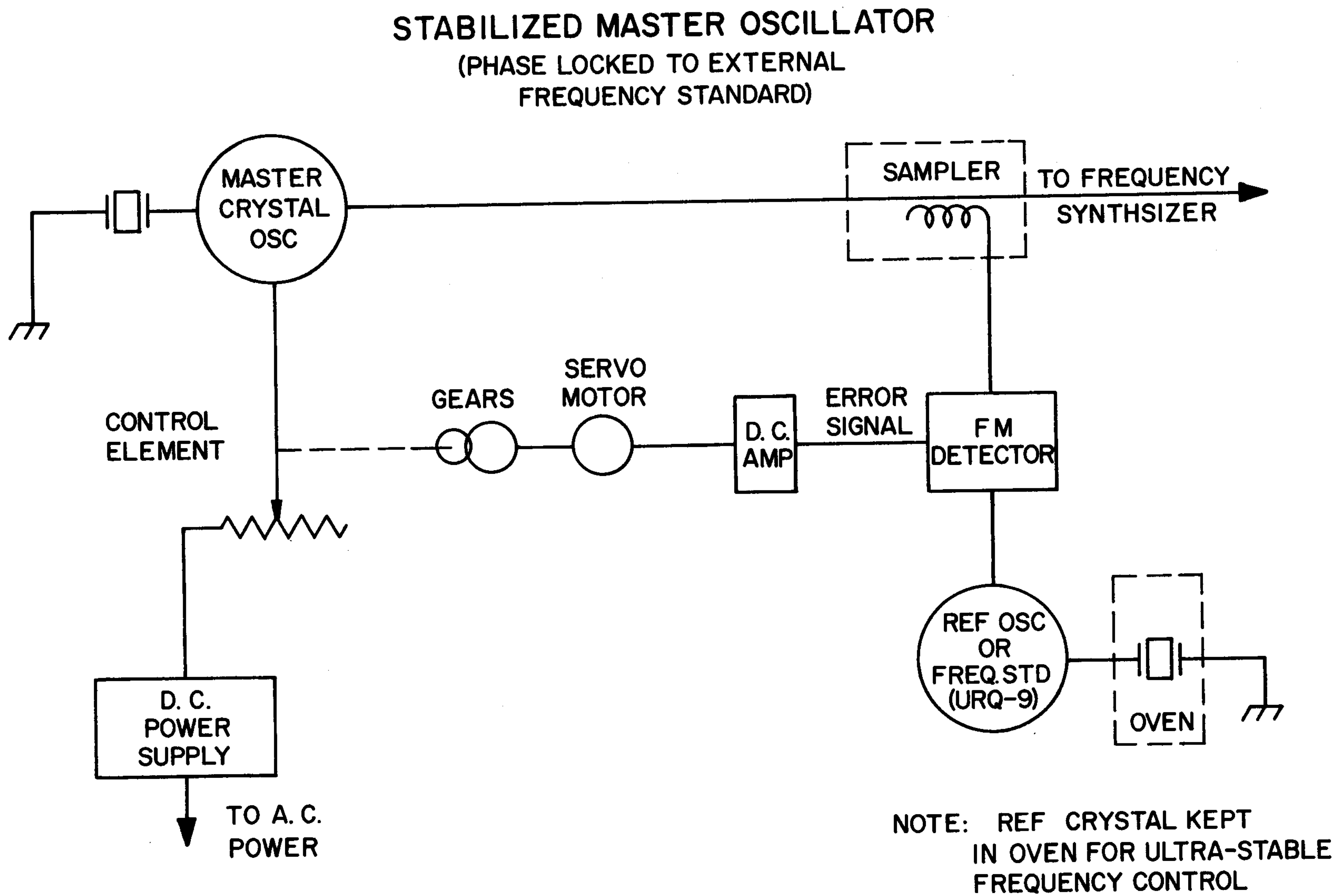



Figure 15. Stabilized Master Oscillator

of a slight change in room temperature, then the servo-mechanism would develop a feedback signal that would pull down the master oscillator's frequency by 2 Hz. In this manner all the synthesized output frequencies will be exactly right because the master oscillator is constantly comparing and correcting itself to be in complete agreement with the frequency standard.

## A. The Elementary AM Receiver

The simplest type of AM receiver is the "crystal set" which uses a germanium crystal as a diode detector. Perhaps you made one in grade school. Besides the crystal, a coil and capacitor are needed to form a resonant circuit. This provides a means of selecting a particular station. Capacitors and inductors (coils) are circuit elements whose A.C. resistance or reactance to an alternating current varies with frequency. A series inductor-capacitor circuit in which the inductive and capacitive reactance are equal in magnitude (for some frequency) is said to be resonant. The principle of resonance allows you to select one frequency desired from many signals. By moving the slider on the crystal set coil you are tuning a circuit to resonance for some particular frequency. When an AM signal (that is, a carrier plus two sidebands) reaches the detector, it is rectified. Usually rectification implies a process of changing an alternating current into a direct current. Actually rectification is a two-step process. First, on positive half-cycles a diode passes a wave while rejecting the wave on the negative half-cycles. This raw, rectified AC current is really an intermittant fluctuating DC current. Second, a smoothing filter is used with the rectifiers to convert the fluctuating DC into pure direct current.

In the case of the AM detector, the rectifier filter is designed to smooth out or eliminate the radio frequency variations, but retain the audio frequency variations brought to the detector by the varying sideband. The diode detection process is illustrated in Figure 16. This detected signal will be the same type of signal as the one produced in the microphone at the transmitter input. It is fluctuating direct current--fluctuating at an audio rate. This signal can now be directly connected to a set of headphones to reproduce the speech or other information contained in the AM wave. All AM or SSB receiving systems contain the basic components just discussed. As shown in Figure 17 the basic functions performed by any receiving systems are: Interception, selection, amplification, detection, and reproduction.

## B. Receiver Refinements

The communications type receivers found in the Fleet are more complicated than indicated in the basic receiver set example. The most important factors which distinguish communication grade receivers are sensitivity, selectivity, and stability. Sensitivity is the ability of a receiver to bring in very weak radio signals and amplify them enough so that the detector can function. Selectivity is the ability to distinguish between radio signals separated by only a very small frequency difference. This property is of paramount importance in HF work because Navy communications services in that band are shared with foreign broadcast stations, long-range navigation systems, amateur radio stations, and other radio services. Stability is the ability of a receiver to tune to and retain a signal under varying conditions of volume control settings, temperature variations, AC supply voltage changes, ionospheric changes, and mechanical shock.

## C. Detecting SSB Signals

An SSB receiver is in many ways similar to standard AM, superheterodyne receivers. It generally has two or more stages of frequency conversion or mixing. As in a SSB transmitter, frequency stability in a SSB receiver is very critical, much more so than when AM is used. In some cases frequency synthesizers are used to generate the local oscillator injection signal which is fed to the mixer amplifier. This is generally the case for a SSB transceiver. A transceiver combines the function of transmitter and receiver in a single package. Transceivers use many of the same active and passive circuit elements for both transmitting and receiving. They are well suited for SSB because once the receiver is properly tuned, the transmitter is set up to transmit on exactly the same frequency. Transceivers, e.g. the AN/URC-32, AN/URC-35 and AN/URC-58 are by their nature more compact than separate transmitter and receiver combination and thus lend themselves well to shipboard use.

In what ways is a SSB receiver uniquely different from the standard AM receiver? The primary difference (other than the order of frequency stability required) lies in the type of detector. When suppressed-carrier SSB signals are being received on an ordinary AM receiver, the diode detector will still produce a varying DC output; but it would sound garbled at the speaker because the carrier wave has been left out or suppressed. A SSB receiver must "simulate" a carrier wave by injecting a sine wave signal into the detector. Such a detector is sometimes called a product detector. It is called a product detector because its output is similar to the mathematical product of the two separate inputs. A product detector is similar to a balanced modulator in reverse, because in it the carrier is added in instead of being

# HOW A DIODE DETECTS AN A.M. WAVE

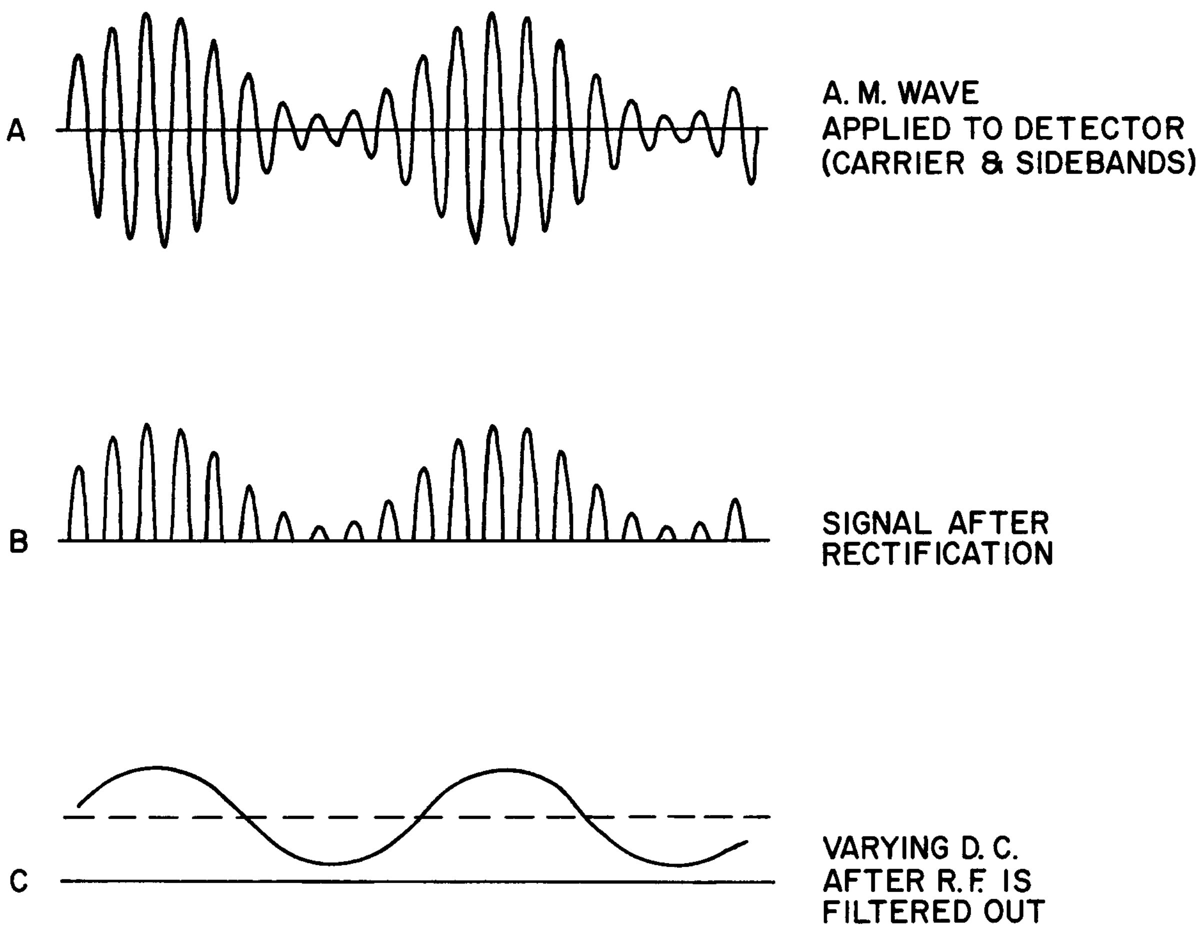



Figure 16. How a Diode Detects an A. M. Wave

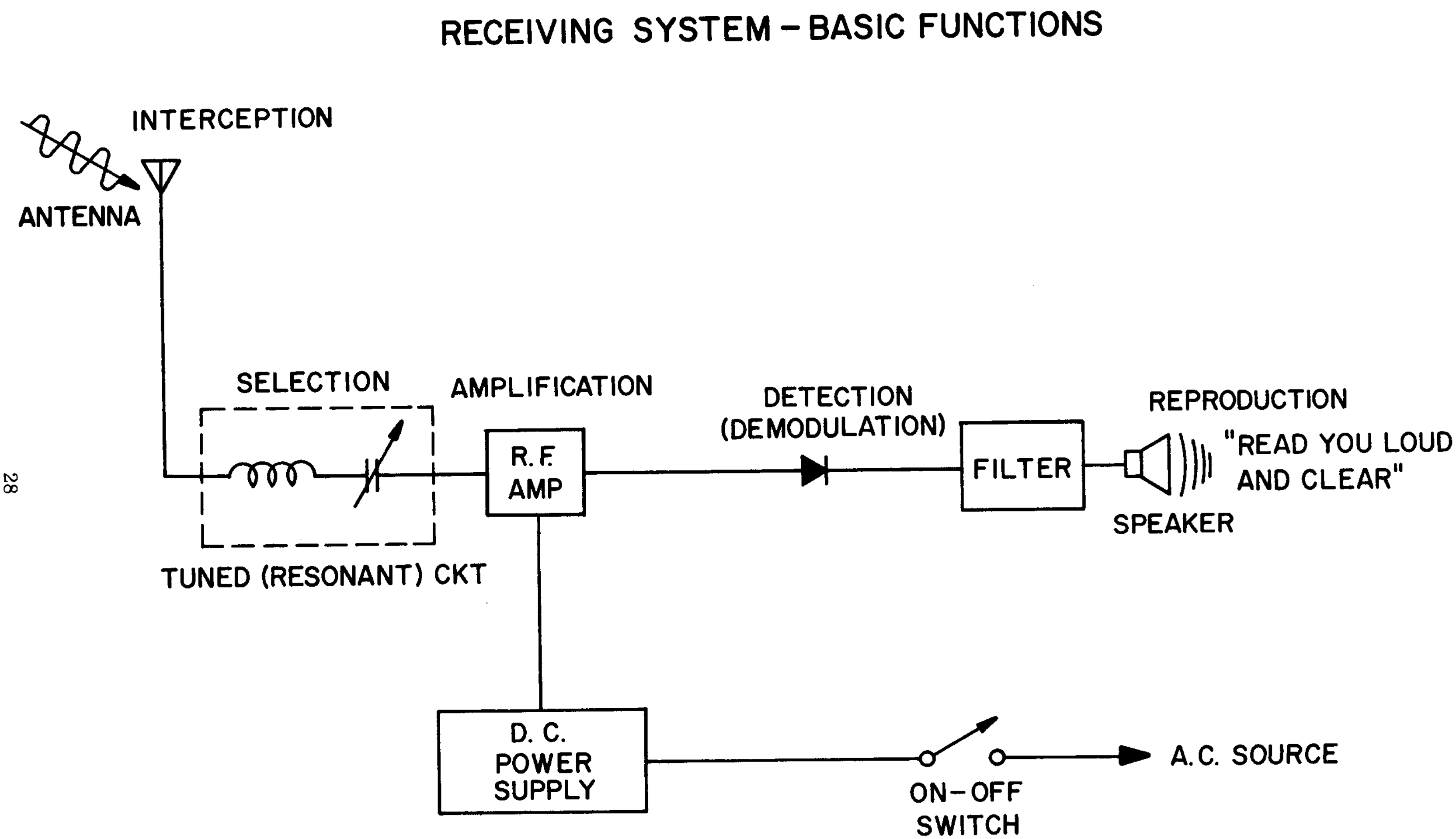



Figure 17. Receiving System – Basic Functions

cancelled out. As a balanced modulator produces zero sideband output when no audio is applied to its input; a product detector will give no audio output when the receiver's carrier oscillator input is removed. The concept is also similar to the beat frequency oscillator (b. f. o.), usually provided on AM communications grade receivers for the purpose of copying CW. The product detector is better for SSB signals and it is also an ideal detector for DSBAM and CW. It provides a more linear type of detection for all three signals and, therefore, generates less distortion at the output. The simplest circuit form of product detector is two diodes or rectifiers "back to back" with the receiver carrier oscillator injecting the carrier signal at their common connection point as shown in Figure 18.

Tuning a SSB receiver is a little more difficult than an AM receiver. This is particularly true if the SSB receiver is an older AM receiver that uses SSB adaptors. Discussion of tuning SSB equipment will be covering in Operating Procedures.

## 1.6 DIVERSITY OPERATION

One of the terms that appears in discussions of modern communications is "Diversity Operation." The word itself implies a difference of operations. This topic needs to be explained before getting into specifics about Single Sideband Communications.

There are several "diversity" techniques. Such techniques are invariably used to improve communications quality and reliability. They directly combat the vagaries of the ionosphere. When ionospheric conditions are poor, when solar flares have occurred, or when the day/night or night/day shifts of the ionosphere are disturbing communications, something has to be done to preserve communications quality.

One of the first diversity techniques that was tried and used was called "Space Diversity". In this system two or more antennas are spaced some distance from one another and are each connected to a separate receiver. The theory behind this system was that a signal that was fading badly at one antenna might not be fading at the same instant at another. A diversity combiner circuit was used that would automatically extract the strongest or cleanest signal from one of the receivers and send it on to the terminal equipment.

Another diversity technique is called "Frequency Diversity". In this system two or more frequencies are used to send the same information. The ionospheric disturbance that was affecting one of the frequencies severely might not be affecting another frequency. In this system it is not necessary to have antennas separated so widely, although this is sometimes done.

A third technique is known as "Polarization Diversity". The basis of this technique is that radio waves that reflect off the ionosphere come back to earth in an elliptically polarized state. Sometimes the radio wave arrives at the receiver horizontally polarized and sometimes vertically polarized. If a single antenna is used at the receiving station there is often very deep fades of the signal due to changes in its polarization. When two antennas, one horizontally polarized and one vertically polarized, are used with special RF combining circuits or with two receivers, a considerable advantage is apparent.

When the rise of the Audio Frequency Tone Shift method of transmitting radioteletype signals, a fourth diversity technique has come into being. This technique is known as "Tone Pair Diversity." In this system different tone pairs, located in the same audio channel of the same SSB transmitter, are used to carry the same teletype information. This is basically frequency diversity, with very close (inter-channel) frequency spacing. Considerable improvement to communications quality occurs when using this form of diversity, and only one radio receiver is needed. Some of the more modern terminal equipments such as the AN/UCC-1 have switches on the front panel that allow this type of diversity to be selected. The AN/UCC-1 equipment allows "Quadruple Tone Pair Diversity" to be set up. In this mode of operation, four tone pairs, spaced through the 3kHz sideband, may be used to transmit and receive the same teletype signal. Various interconnections of the AN/UCC-1 equipment will allow "Tone Pair Diversity" operation as well as one or more of the various RF diversity techniques described above. More will be said about this in the section of the handbook concerned with multichannel operation.

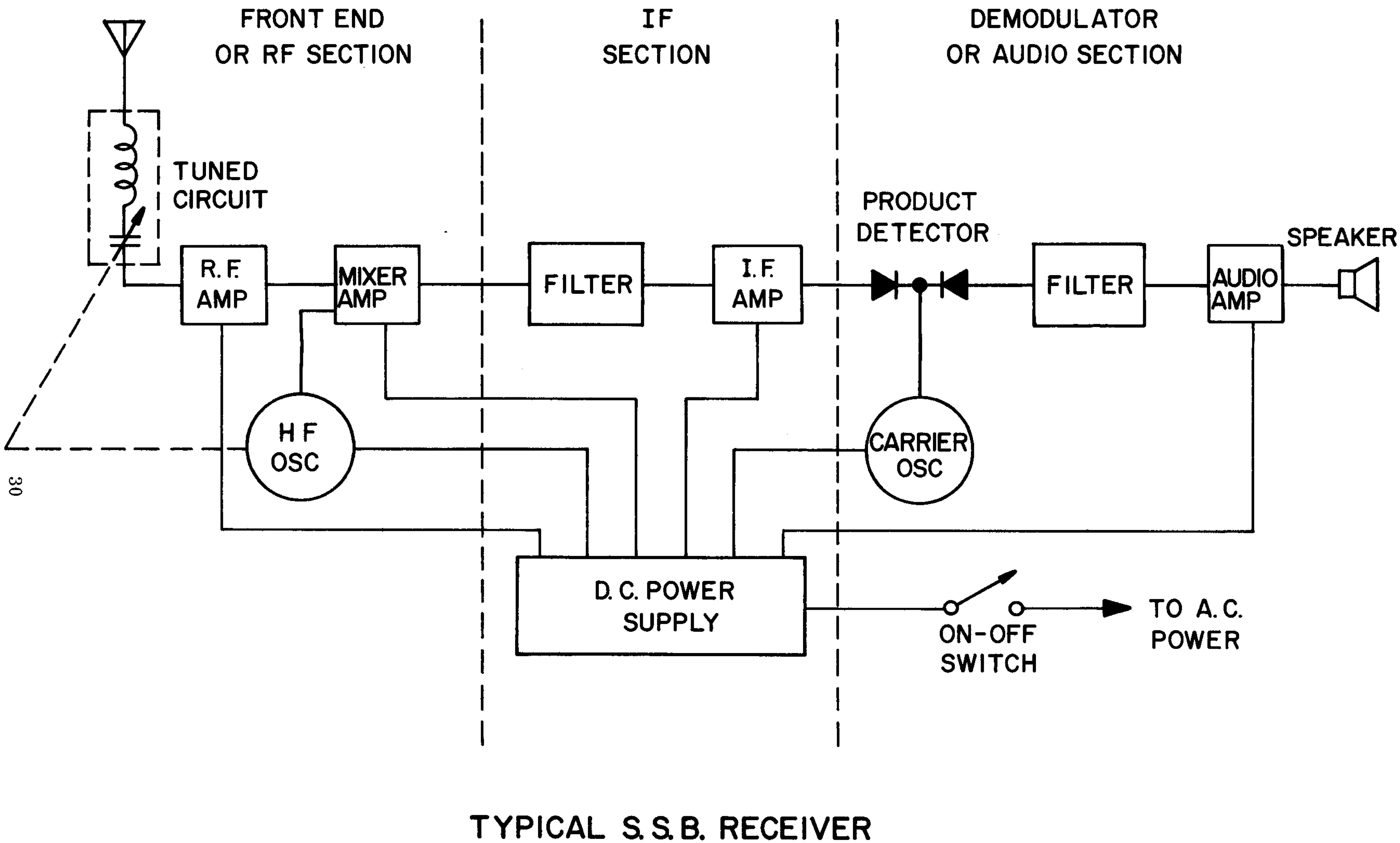



Figure 18. Typical S. S. B. Receiver

In the first section of this handbook we took a look into the background, basic theory, and terminology of SSB communications. We have tried to draw a good picture of what's going into and coming out of SSB transmitters and receivers, and how this equipment differs from the older, AM/CW communication sets.

The problem at this point is to build up a knowledge of operating techniques by becoming familiar with the types, functions, and placement of controls that will be encountered on the various equipments around the Fleet. This handbook cannot present the optimum procedures to use with each operational equipment nor can it be a substitute for actual experience. The actual technical manual for each equipment must be used. This book is designed to present basic information that will allow an individual to effectively develop his own experience and to take advantage of the experience of other communicators, and more detailed manuals. Presented here are some rules-of-thumb for a "typical" or "average" piece of voice sideband gear plus a description of how various nomenclatured receivers, and transceivers are used in practical communication systems, e.g. radio teletype and independent sideband systems.

Table I gives a summary of the high frequency SSB equipments most commonly used in the Navy at the time of printing of this handbook. This part of the book will point out unique differences between those equipments and their relationships of auxilliary terminal equipments used in SSB radio systems. Interjected between these two main discourses are comments on differences between peak and average power, and the various forms of radio frequency interference (RFI) that SSB is capable of generating under improper operating conditions.

## 2.1 TRANSMITTING SSB PHONE (VOICE OPERATION)

The front panel of a "typical" modern SSB transmitter is shown in Figure 19. This typical front panel will be used to discuss the general operation of a SSB transmitter--specific equipments will be discussed later. On many types of equipment the controls are located on separate racks, drawers, or modules. Figure 19 shows all controls together for simplicity. This hypothetical transmitter may have too many or too few controls, depending upon the viewpoint of the operator.

The first thing to notice is that a SSB transmitter naturally divides itself into an exciter and radio frequency power amplifier. This is understandable when you recall what the functions of these two components are (reference page 41, Figure 12). Actually the great bulk and weight of the SSB transmitter is due to the power supplies for the exciter and final power amplifier. The amplifier section usually has several meters and large tuning knobs or dials. The exciter section controls the radio frequency for the rest of the transmitter, and is most easily identified by looking for digital windows, dials or knobs that are associated with the frequency synthesizers. Generally, there are five or six digital controls. On older equipments there may be only one frequency setting knob with a counter dial with five or six digits--like the mileage dial in an auto speedometer--for frequency readout. In any case, finding the exciter and power amplifier (P.A.) sections will orient you in the understanding of the equipment. Strange as it may seem, the hardest control to find is the on-off switch. This vital control can appear just about anywhere.

### A. Setting the Carrier Frequency

Assume that you want a SSB transmitter to "net" or tune to an upper sideband voice channel on an assigned frequency of 12,371.5 kHz (USB). There has been considerable confusion in the shift from AM to SSB as to what the published frequency assignment means. In this assignment the carrier frequency, as in AM, or is it the center of the transmission? SSB and other new forms of modulation have forced the revision of frequency assignment definitions. Now, by international agreement, a frequency assignment refers to the center of the emission bandwidth. This type of assignment requires you to do little arithmetic before you can set the frequency dial of the exciter. Most military SSB transmitters follow the practice of the older AM equipments and have the dial calibrated in terms of carrier frequency. Referring to Figure 20, you see that the carrier frequency is also the center of the emission bandwidth for AM or DSB emissions. In SSB, however, the carrier frequency (which isn't radiated) is at either the high or low

## Table 1

## HIGH FREQUENCY SSB EQUIPMENTS MOST COMMONLY USED IN THE FLEET

| | NOMENCLA-TURE | FUNCTION | FREQ. RANGE (MHz) | MODES OF OPERATION | P.E.P. Watts | AVG. PWR. Watts | TYPE TUNING INCREMENTS (kHZ) | SYN. ? | EXT. FREQ. STD. (MHz) | AFTS TONE (Hz)* | COMMENTS |
|---|---|---|---|---|---|---|---|---|---|---|---|
| TRANSMITTERS | AN/WRT-2 | XMIT | 2 - 30 | CW, SSB, ISB, AFTS, AM, MULTI-CHAN. RATT | 1000 | 500 | 1.0 | YES | 3.0 | 0 | 425 Hz tone in USB & LSB for AFTS. |
| | AN/URT-23 Series | XMIT | 2 - 30 | CW, SSB, ISB, AFTS, AM, MULTI-CHAN. RATT | 1000 | 1000 | 0.5/or 0.1 with latest model | YES | 5.0 | 2000/ 2550* | Companion Antenna Coupler is AN/URA-38. |
| | AN/FRT-39B | XMIT | 2 - 28 | CW*, SSB, ISB, AFTS AM, MULTI-CHAN. RATT | 10,000 | 5000 | 0.1 | YES | 1.0 | 2000/ 2550* | |
| | AN/URT-24 | XMIT | 2 - 30 | CW, SSB, ISB, AFTS, AM | 100 | 50 | 0.1 | YES | 5.0 | 2000/ 2550* | |
| RECEIVERS | R-390A & CV-591A | RCVE | 0.5-32 | CW, SSB, AM, AFTS | | | CONTINUOUS | NO | NONE | | General Purpose Rcvr. Not used for MULTI-CHANNEL RATT |
| | AN/WRR-2 (FRR-59) Series | RCVE | 2 - 32 | CW, SSB, ISB, AM, AFTS MULTI-CHAN. RATT | | | 1.0/ or 0.5 with FLD CHANGE + CONTINUOUS | YES | 1.0 | | |
| | R-1051/ URR Series | RCVE | 2 - 30 | CW, SSB, ISB, AM, AFTS MULTI-CHAN. RATT | | | 0.5/ or 0.1 with latest model + CONTINUOUS | YES | 5.0 | | |
| TRANSCEIVERS | AN/URC-32 Series | TRANS-CVE | 2 - 30 | CW**, SSB, ISB, AM, AFTS, MULTI-CHAN RATT. | 500 | 500 | 1.0/ or 0.1 with FLD CHANGE | YES | 0.1 | 2000 | |
| | AN/URC-35 | TRANS-CVE | 2 - 30 | CW, SSB, AM | 100 | 50 | 0.1 + CONTIN-UOUS | YES | 5.0 | NONE | |
| | AN/URC-58 | TRANS-CVE | 2 - 15 | CW*, AM, SSB | 100 | 100 | 1.0 + CONTIN-UOUS | YES | NONE | NONE | |
| TRANSMITTER/ RECEIVER | AN/WRC-1 Series | XMIT & RCVE | 2 - 30 | CW, SSB, ISB, AM, AFTS | 100 | 50 | 0.5/ or 0.1 with latest model + CONTINUOUS ON RECEIVE | YES | 5.0 | 2000/ 2550 | |

* CW tone is 1000 Hz in the upper sideband
** CW tone is 1000 Hz or 1500 cps selectable

* = Center Freq. - Tones are ± 425 Hz.

SSB TRANSMITTER
TYPICAL ARRANGEMENT OF CONTROLS

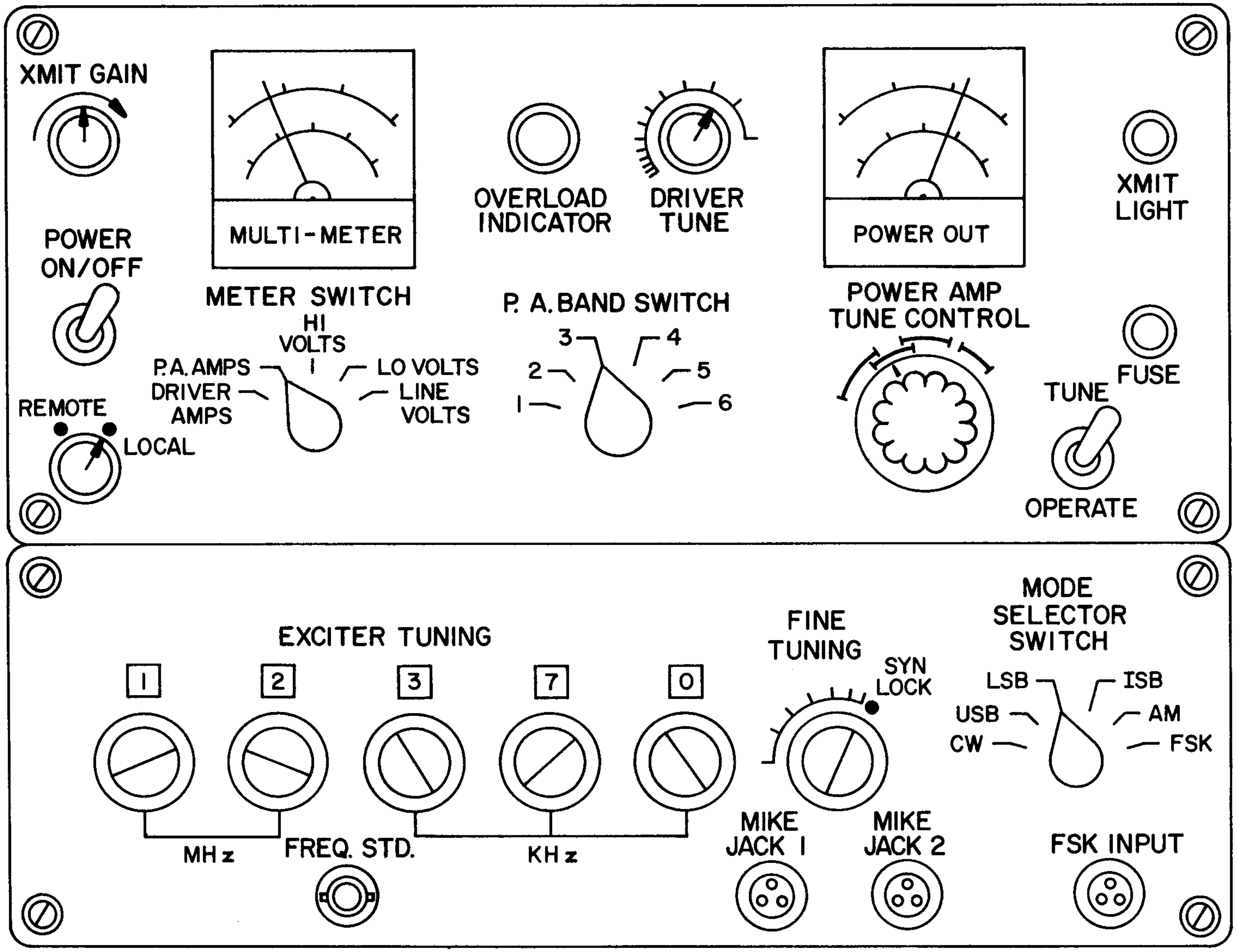



Figure 19. SSB Transmitter, Typical Arrangement of Controls

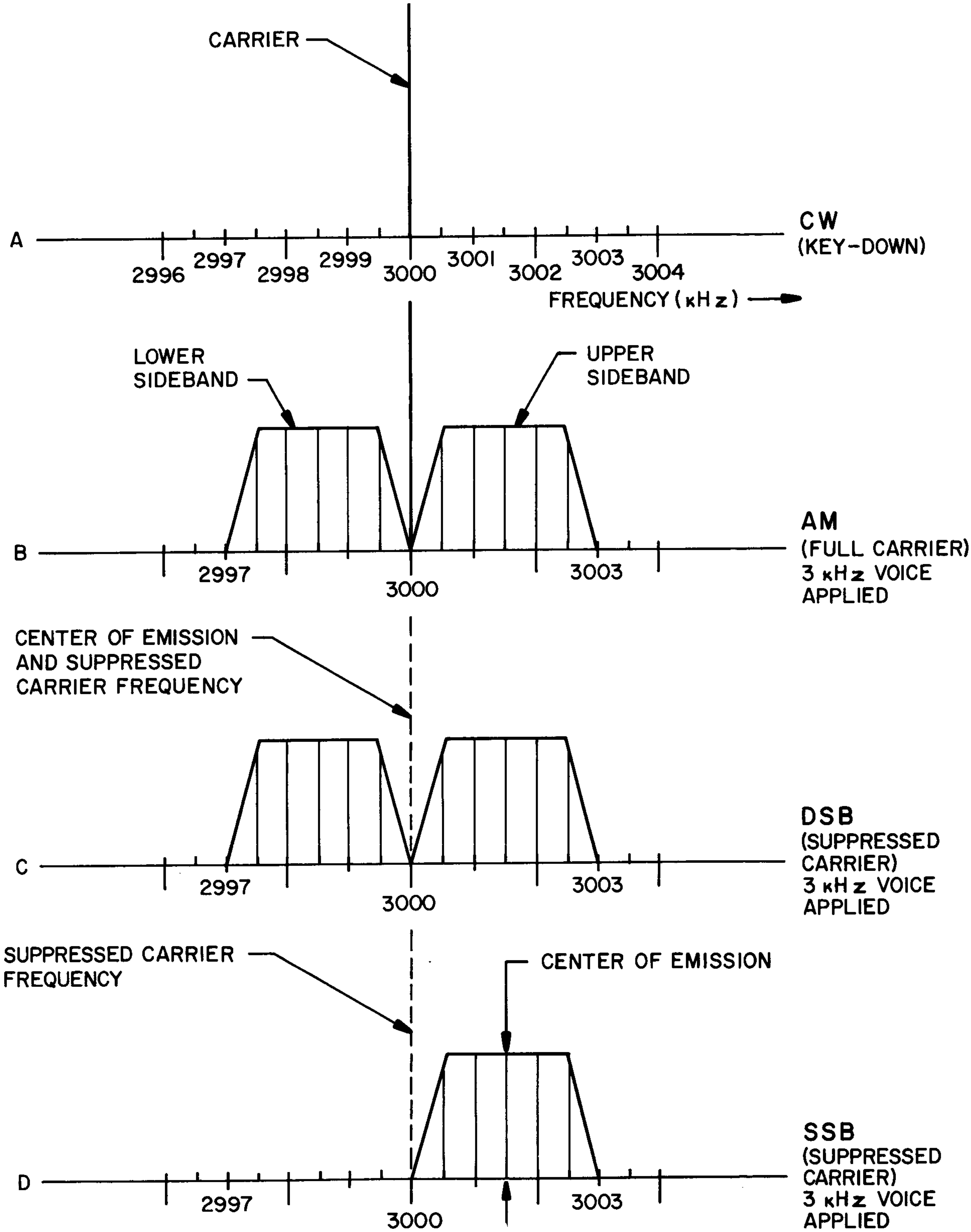



Figure 20. Transmitter Emissions (Carrier Frequency at 3000 kHz)

end of the emitted bandwidth, depending on whether the emission is LSB or USB. Therefore, the carrier is not the center of the SSB voice-modulated emission. But, since the dial refers to the carrier frequency you have to offset the dial reading for the carrier one-half the total amount of bandwidth contained in the emission. A voice SSB channel requires a standardized 3.0 kHz bandwidth. With this bandwidth, the carrier frequency should be set 1.5 kHz below (or above) the assigned frequency. If the frequency assignment of this "net" were 12,371.5 kHz (USB), then you would set the frequency control for 12,370.0 kHz--i.e. the carrier is set 1.5 kHz below the center of the USB voice channel assignment. If the "net" assignment were given as 12,371.5 (LSB), then you would set the carrier frequency at 12,373.0 kHz--i.e. the carrier is 1.5 kHz above the emission bandwidth. Needless to say, it pays to take a moment and double check the figures because a simple error could mean missing the "net" assignment by 1.5 kHz or possibly even 3.0 kHz.

It has become general practice in the Navy to use the USB for all SSB RATT and Voice Transmissions. When SSB RATT transmissions are considered, the dial setting of either transmitter or receiver is set 2.0 kHz below the center of emission (except for AN/WRT-2 transmitter - see Table 1).

Some transmitters or transceivers have a fine tune or frequency vernier dial after the last digit. This control varies the transmitter frequency in increments less than the last digital dial--usually less than 1 kHz. It is useful for "zero beating" in the CW mode. It should not be used on multichannel afts however, because using it may "unlock" the stabilized master oscillator and the synthesizer frequency may drift. When operating multichannel afts, always use the fully-synthesized mode of operation if possible. If a frequency standard is available, such as the AN/URQ-9 or AN/URQ-10, check the manufacturer's technical manual to see if the transmitter will accept external frequency standard signals. If it does have provision for injection of an external standard frequency, use it because then the master oscillator frequency can then be continuously compared with the standard frequency. Using the frequency standard will result in the optimum long term frequency stability. Synthesized equipments should be left on 24 hours a day (i.e. continuously) for maximum stability and reliability. Most new equipments used in the fleet today have internal frequency standards. These equipments should have their internal standards calibrated against an AN/URQ-9 or 10 periodically for maximum accuracy. These internal frequency standards are primarily intended for installations not having an external ships frequency standard system (i.e. small craft). The internal frequency standards do not have the long term accuracy and stability as provided by a ships frequency standard system.

B. Adjusting the Power Amplifier

The power amplifier consists of two or more high-vacuum amplifier tubes, designed to operate under linear conditions. These tubes are to bring the SSB signal (that the exciter has developed) up to the desired power level. Adjustment of the power amplifier is a two-step process. The first step is to tune the amplifier to a CW signal at the carrier frequency. Second step is to adjust the output from the SSB generator or exciter to the proper level for voice operation. The power amplifier has tuned circuits in its input and output stages like those in the front end of the receiver--these transmitter circuits are able to handle more power. These tuned circuits must be resonated to the operating frequency in order to filter out all signals other than the desired frequency. A CW signal is used while tuning because no sideband energy appears at the output of the balanced modulator, until audio is applied. The AN/URT-23 equipments tune in the AM mode at reduced power and this protects the associated antenna coupler from being tuned at full power. The AN/WRC-1, AN/URC-35 and the AN/URT-23, 24 do not require manual tuning for they are automatic. In some transmitters, a "CW Switch" allows signals from the exciter to bypass the balanced modulator and go directly into the amplifier section. After the power amplifier stage is correctly tuned, then the output from the exciter may be adjusted. This is very critical. If the exciter output level (which is feeding the power amplifier stage) is too high in amplitude, then the power amplifier will operate in a non-linear manner due to over-driving and over-load. A power amplifier operating non-linearly will distort voices and create splatter or spurious products that will interfere with other stations operating on nearby frequencies. Greater detail on these two steps are given below:

Step No. 1 - Tuning the Power Amplifier

Tuning the linear power amplifier stage looks complicated because it involves using more controls than any other transmitter adjustment. It is really not difficult and should be familiar to operators who have previously tuned CW or AM transmitters. The tuning should be done using a "dummy load". A dummy load is a large non-inductive resistor which matches the transmitter output impedance and therefore absorbs the output energy. The dummy load is used to avoid causing radio interference to other stations on the air while tuning. After the transmitter tuneup is accomplished with the dummy load, one can then proceed to adjust the antenna coupler for the most efficient transfer of radio energy to the transmitter antenna.

Having identified the power amplifier section, you will find several meters, one or more P. A. tuning controls, a driver tuning control and (usually) a band switch. First check to see if the transmitter has a local-remote switch. This local-remote switch must be in the local position so that the transmitter may be keyed with either the mike push-to-talk (PTT) button or tune/operate switch on the front panel. No one else can key the transmitter from a remote location while it is being tuned with the switch in the local position. Locate the tune-operate switch and place it in the tune position. The AN/WRC-1, AN/URT-23, AN/URC-35, and AN/URT-24 are in the AM mode (carrier only) when the selector switch is in the tune position.

Now the transmitter can be keyed by pressing the mike button. Start the tuning procedure by placing the band switch in the position corresponding to the operating frequency. There may be four or five positions, or as many as nineteen (19) positions on this band switch, depending upon the equipment. Double check to make sure that the exciter is set to the right frequency (because a mistake can mean tuning to a harmonic instead of the desired fundamental frequency). No one wants to be heard on "Orestes Net" while attempting to tune up to "Harbor Common"!

Next, set the power amplifier and driver tuning controls. There are approximate frequency calibration points provided on a dial behind the tuning knob. The final adjustment shouldn't end up too far from "what the dial says". Click the mike PTT* button very briefly, to see if everything is connected up properly. If all settings are correct up to this point, a transmit light will come on or one or more of the meters will kick up scale as the transmitter is keyed. Upon getting such an indication, you are then ready to tune up.

To tune the unit, you must use some sort of meter (unless you're lucky enough to rate one of the latest auto-tune equipments, e.g. the AN/URT-23 or AN/WRC-1). The most common types of meters are power output meters, P. A. plate current meters, or multi-meters that have volt/amp readouts at various points in the transmitter. If the equipment has a multi-meter, then select (by using meter selector switch) the P. A. amps or P. A. Milliamps setting. Check over the "short-form" tuning procedure found posted on the transmitter or in the Operator's section of the manufacturer's technical manual. These instructions will insure that you observe the right meter and use the proper controls.

* Push to Talk

Generally, the tuning procedure is: peak a power-out meter, or dip a current meter that belongs to the particular stage being tuned. Whichever you use, take your time and do it right--but don't under any circumstances hold the key or button down for more than 10 or 15 seconds without allowing the transmitter to cool off. Release the key for a period of time at least equal to the "on-time". A SSB transmitter is designed to put out full power only on voice peaks, or on a long dash (dah) in CW. You may damage the amplifier if you hold the key down for long periods of time during tuning. Remember, once again, tune the P. A. tuning control for maximum power out or for minimum P. A. current. You tune a stage for the "dip" on its current meter, because parallel-type resonant or tuned circuits are employed. These tuned circuits give a high resistance at resonance instead of low resistance which occurs in the series-type tuned circuit. When properly tuned, however, the circuit is still peaking radio energy on the desired frequency. Thus the higher reading on the output power meter occurs nearly simultaneously with minimum P. A. current. When adjusting the tuning controls of the driver stages, they may be set for peak P. A. power output or for peak driver output. It will be noticed that when peak driver output is obtained, the P. A. plate current will also be a maximum. After initial adjustment of the driver and power amplifier tuning controls, one can go back and carefully re-peak or re-dip to get the highest power out permissible within the power limitation of the particular transmitter. Be careful, however, not to exceed any "red lines" on the meters during the power amplifier tuning procedures. Note the meter readings for they will be needed later in connection with audio level adjustment.

After the power amplifier is properly tuned, turn the mode selector switch to USB or LSB--depending upon which mode is desired. Push the PTT button while observing the plate current or power output meter. Do not speak into the mike. You should see only a very slight kick on the P. A. plate current or power out meter. Now you are ready to perform the next step, setting the modulation level.

Equipments such as the AN/WRC-1, the AN/URT-23 and 24 have automatic modulation level control circuits. Internal circuits are preset and do not need to be adjusted during tune-up.

### Step No. 2 - Setting the Modulation Level

Setting the modulation is under-emphasized in many instruction manuals. If done correctly, it can create problems by driving the power amplifier into the non-linear range. The engineers who design military sideband equipment have taken great care to select power tubes and supply voltages to insure operation on the linear part of the power amplifier's characteristic curve. They have also designed and developed speech processing equipment which filters, clips, and compresses the voice. This reduces the possibility that the SSB transmitter will distort on voice peaks. Not withstanding all this built-in protection, provision is still made for operator control of the amount of voice sideband power coming out of the exciter into the power amplifier. This control goes by various names--probably one of the following: mod level, transmit gain, transmit audio, mic gain, increase power; or words similar to these. This "mod level control" is the main "accelerator" for the SSB transmitter, and it is left to the operator's judgement to set this control. Like the accelerator in your car, if you hit it too hard you're eventually going to wind up in trouble. It is tempting to advance this control in an effort to "squeeze" a little more voice power out of the transmitter when propagation conditions are poor and signals are weak. Don't do it! Overdriving the power amplifier creates more interference (especially in your part of the ocean) and poorer intelligibility at the receiving end. If the transmitter has an audio input level-meter, use it and follow the instructions set forth in equipment technical manuals.

Generally, the manuals will tell you to hit the red area of the Exciter Power output or the P. A. Power output meter only occasionally on voice peaks. If you don't have an audio input or modulation level meter on your transmitter, here is a rule of thumb: While talking into the mike (hold it close to your lips) at your usual voice level adjust the control so that on the average the power amplifier plate current meter or power output meter is peaking to about one-third (1/3) the level that had been obtained during the (CW) power amplifier tuning procedure. That is, if the final amplifier plate current reading during tuning had been 300 milliamps, you should be hitting 100 milliamps on the average during voice modulation peaks. If the power output meter reads 500 watts in the CW mode, it should be peaking at about 170 watts, on the average in USB or LSB voice mode. If the meter is hitting average peaks at about 1/4 scale, the transmitter is being undermodulated. In order to set the modulation level correctly, turn up the audio-mic or transmit gain until the meter reads up to the 1/3 point on peaks. If, on the average, the meter is peaking 1/2 scale, then the transmitter is definitely being overmodulated. Reduce gain or turn down the "mod-level" control. The 1/3 point, typically, gives the SSB equivalent of optimum or 100% AM modulation. If no control settings can be adjusted to yield the 1/3 point, then call a technician and have him check the transmitter. Something is wrong. The above described rule has been derived from practical operating experience and has been confirmed by sensitive test equipment. It can be explained by saying that the meters of the transmitter cannot follow voice inflections perfectly. Such meters give a type of average reading. A voice signal is much more irregular than a sine wave, having an occasional very high amplitude peak or valley. This causes the meters to give readings than can be misleading on SSB voice if not properly interpreted. Observe this 1/3 scale rule-of-thumb and no one will ever accuse the transmitter that you have tuned of either undermodulation or causing splatter or other forms of overmodulation distortion-products.

## 2.2 RECEIVING SSB PHONE (VOICE OPERATION)

### A. General

If you are using one of the new SSB transceivers for transmitting, then no change in adjustments are necessary for receiving. Specifically, the AN/URC-32, AN/URC-35, and AN/URC-58 transceivers fall into this group. Frequency setting remains exactly the same for transmitting and receiving. The correct sideband mode (LSB or USB) for transmitting is the identical mode used for receiving that sideband. The only other thing to worry about at this point (assuming the transmitter is connected to a good antenna with a properly adjusted antenna coupler) is turning the receive audio or A. F. gain to a level such that all other stations on the "net" can be heard.

On the other hand you may be using a separate receiver and transmitter. In this case, the receiver is usually located in a different space from the transmitter. Thus it is necessary to move to the new location to set up a receiver on the same frequency as the transmitter, assuming duplex operation on this net.

Receivers usually fall into one of two classes with respect to sideband operation. Older equipments form one class. The R-390A/URR is an example of those receivers designed originally for AM/CW only. The second class consists of the newer receivers like the R-1051/URR that are designed primarily for SSB operation. The latter class can also be used with CW and AM. The older receivers such as the R-390A/URR, if stable enough, can be used on SSB voice channels with fair to good performance. To use the older receivers a SSB adapter such as the CV-591A/URR is required. This adapter contains detectors and filters adequate for sideband reception. The newer receivers, naturally, will give superior SSB copy and generally work better on AM and CW, too.

B. Receiver Tuning

The main adjustment problem in a SSB receiver is tuning the high frequency and carrier oscillators correctly. In addition to selecting the right frequency the correct bandwidth/gain adjustments have to be made. The frequency setting part is the same, in most cases, as described for SSB transmitters. Figure 21 shows the front panel of a "typical" SSB communications receiver. Many of the details and the location of controls will be different for various receiver models. The general layout and types of controls found would, however, be very similar to the "typical" model. If transmitter emissions are thought of as "hills" or "humps", picture the receiver passband or tuning curve as a hole or slot. The slot "accepts" the emission bandwidth hump, when the receiver is tuned on frequency. Figure 22 is a picture of how the typical receiver's selectivity curve appears when adjusted for different types of signals. The receiver main tuning dial, digital readout, or incremental digit window (as shown) will normally indicate the carrier frequency of the emission being received. The main tuning must be adjusted 1.5 kHz above or below the frequency assigned for SSB operation just like the transmitter adjustment. This adjustment places the center of emitted bandwidth from the desired SSB station in the center of the receiver passband slot. To take the earlier example: a "Net" frequency assignment listed as 12,371.5 kHz (USB) would mean a dial setting of 12,370.0 kHz--i.e. carrier set 1.5 kHz below the center of the USB voice channel. If a signal is on LSB, tuning the receiver to a lower frequency will lower the voice pitch. On USB tuning to a lower frequency will raise the voice pitch. Some receivers have a calibrate provision for checking dial accuracy against an internal frequency standard. An external frequency standard, such as the AN/URQ-9 or AN/URQ-10 should be used with the newer receivers (R-1051/URR). Check the manufacturer's technical manual to see if the receiver will accept external frequency standard signals. The AN/URQ-9 or 10 can supply frequency standard signals at three frequencies simultaneously: 100 kHz, 1 MHz, and 5 MHz. Using such standards is very desirable, as it insures the highest frequency stability and accuracy.

C. Setting Bandwidth

The curves of Figure 22 indicate how much signal is lost when the receiver is tuned off-frequency. Figure 22 also shows the minimum bandwidth requirement adequate for reception in CW/SSB/AM modes. It is clear that the bandwidth suitable for SSB lies between that used for CW and AM. Another way of saying this is that: SSB emission is narrower than AM but wider than CW. In the Figure, the center of emission being received is at the 0.0 kHz line. As you tune off frequency you get less receiver output voltage for a given receiver input voltage. These voltages are plotted in terms of ratios on the left side of the graph. Figure 22 also shows the minimum bandwidth requirement adequate for reception of CW/SSB/AM modes. Ideally an AM emission occupies a 6 kHz channel and a SSB station occupies a 3 kHz channel. Most recently built receivers, however, have many tuned circuits or a crystal filter which tends to cut off the higher sideband frequencies. The receiving bandwidth is usually specified at the 2:1 voltage ratio point. The typical receiving bandwidths specified are 8 kHz for AM, 3 kHz for SSB and 1 kHz for CW and this can be verified in Figure 22. Older receivers would have somewhat wider bandwidths. Some receivers give automatic selection of the optimum bandwidth through the mode selector switch. Others give the operator a choice of bandwidth by means of the I. F. bandwidth switch. The choices given will vary, but for voice SSB you should select a bandwidth from 2.0 - 3.0 kHz.

D. Setting the Gain Controls

Generally speaking, you should manipulate the receiver controls as if to copy a CW signal. The RF gain control is set near maximum and headset or speaker level volume adjusted to a comfortable listening level with the AGC gain or audio level control. The operator must be careful to avoid receiver overload on very strong SSB signals. It is helpful here to use the AGC position. AGC (automatic gain control) helps combat fading and overloading by regulating the gain of the receiver automatically. If an I. F. gain control (intermediate frequency gain) is provided, adjust it to the minimum level for a good signal-to-noise ratio.

E. Noise Limiter Notes

A noise limiter switch is often included on mobile receiving equipment (AN/URC-35, 38). Its function is to activate circuits which clip out noise signals that are of a higher level (on the average) than the desired received signal. This device works best on impulsive interference such as that generated by internal combustion engine ignition systems and welding arcs. Since SSB involves a modulation which has a high peak to average power characteristic, you may find the noise-limiter action reducing the signal level as much as the interference. The noise limiter is more useful in receiving the AM signals. The signal level meter (S-meter) comes in handy for comparing relative strength of two stations or for

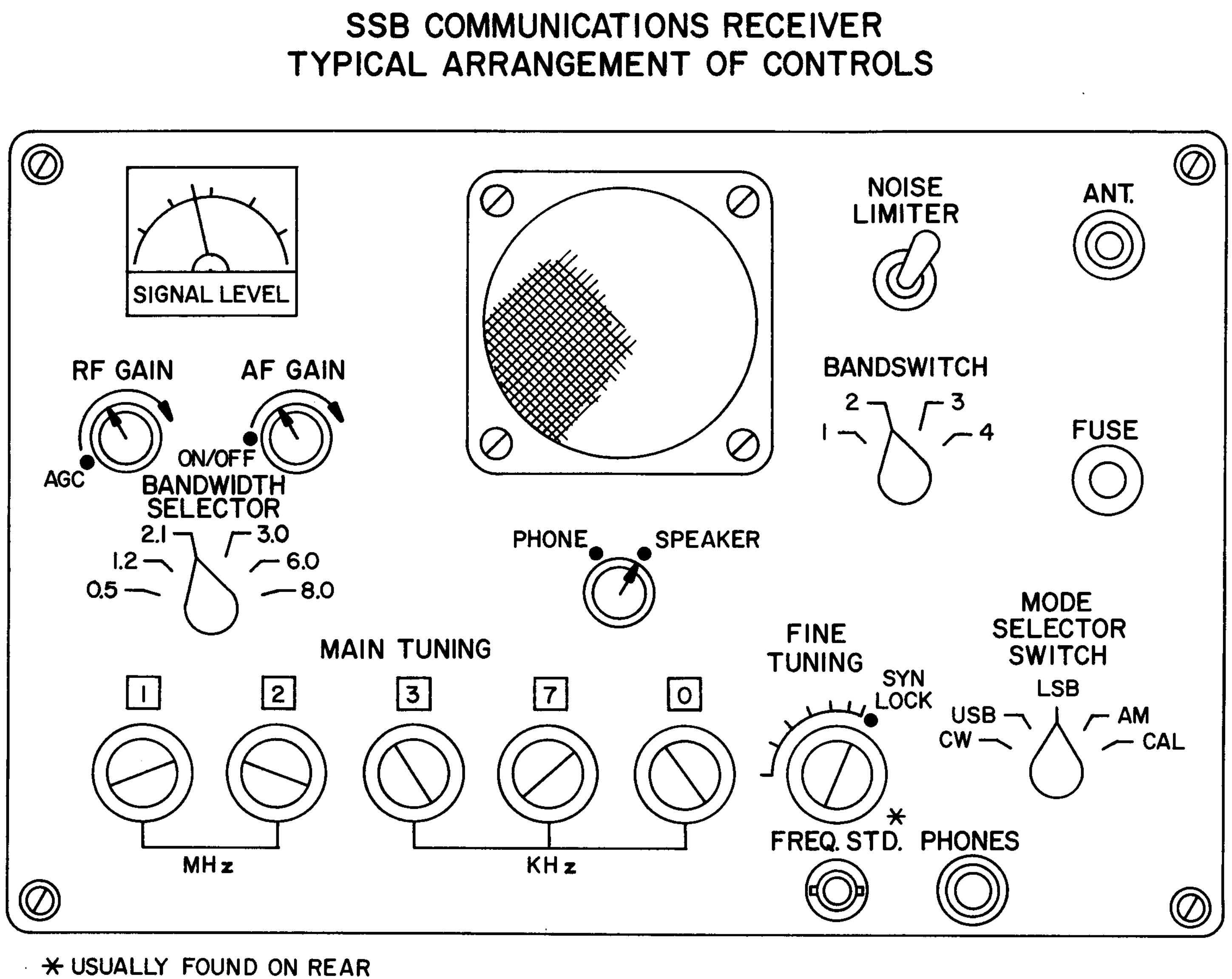



Figure 21. SSB Communications Receiver, Typical Arrangement of Controls

# TYPICAL RECEIVER TUNING CURVES

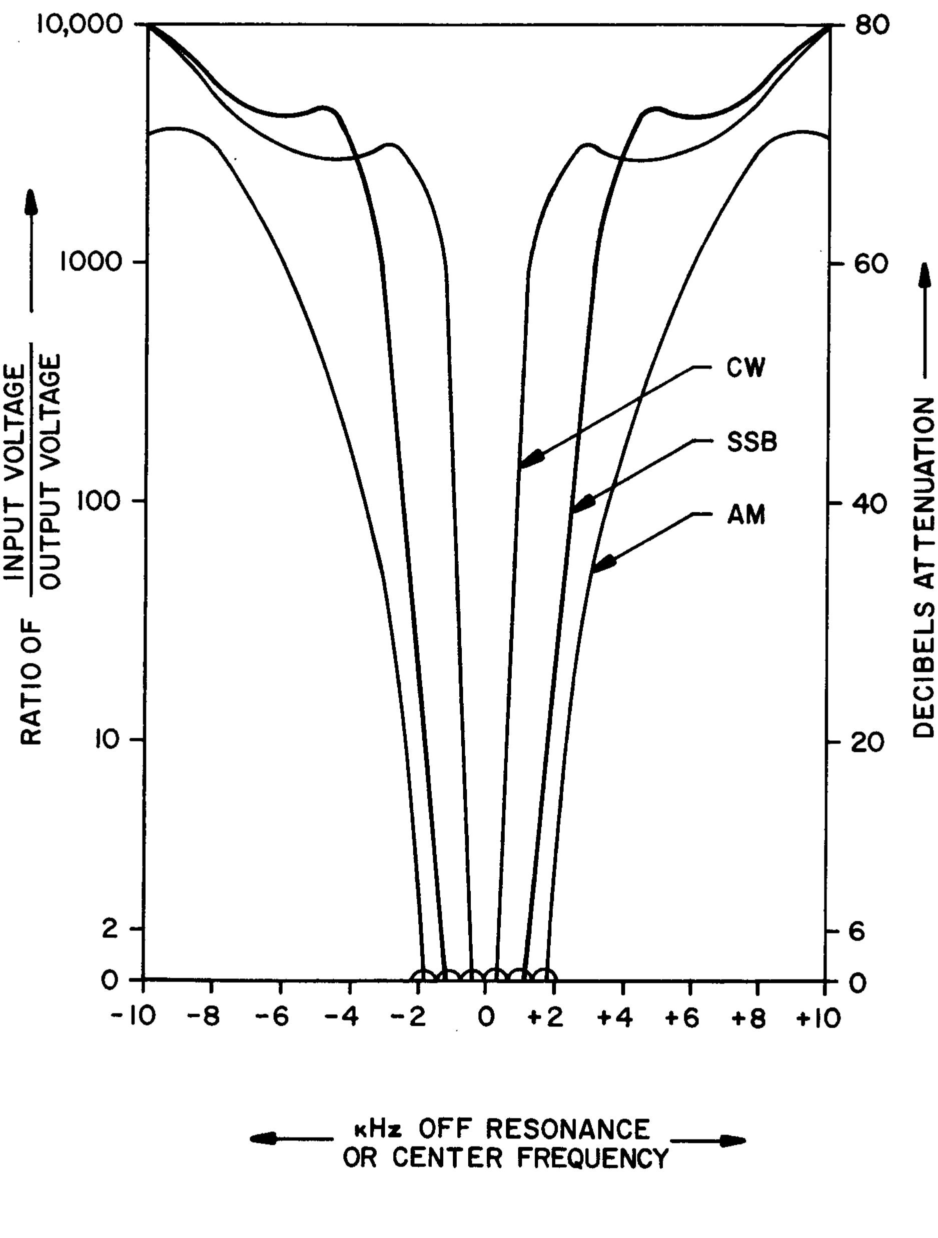



Figure 22. Typical Receiver Tuning Curves

tuning stations whose exact frequency assignment is not known. This meter however gives a true indication of signal strength only when the AGC is in use. With the AGC off, the meter reading will be largely dependent on the setting of the RF gain control.

## 2.3 RECEIVING SSB ON AM OR CW EQUIPMENT

SSB voice is compatible with the older AM and CW receivers that have adequate frequency stability. The Model R-390A/URR is a typical example. This receiver is used both aboard ship and at shore stations throughout the Navy. Considered a general purpose receiver, the R-390A nevertheless has good frequency stability (although not synthesized) and the generally high performance characteristics necessary for voice SSB communications. It also provides reception of CW, AM, and single channel radio teletype signals. On SSB the R-390A is used with CV-591A/URR, most common SSB adapter. In fact, SSB adaptor CV-591A/URR is usually installed adjacent to the R-390/URR receiver. The adaptor contains filters which give the selectivity response needed for SSB reception. Tuning of SSB signals is simplified because final tuning is done at the adapter, not the receiver. A band-spread control tunes over the I. F. bandpass of the R-390A. This enables the operator to separate adjacent stations with a "slower" tuning rate than the basic receiver gives.

Check over the adapter technical manual to make sure the receiver-adapter interconnections, intermediate frequency, and power supply voltages are correct. Review of the Operators Section in the adapter manuals plus some actual practice, is the quickest way to get used to interaction between receiver and adapter controls.

In some cases it is possible to receive SSB even without the use of an adapter. This procedure is not recommended for normal use, because frequent retuning is usually necessary. The trick is to use the beat frequency oscillator (BFO) the older receivers employ for CW reception. The BFO can supply the missing local carrier. The AM/CW receiver will not have a product detector so if this procedure is used some distortion will occur. The best adjustment is to turn the audio gain control as high as possible and turn i.f. or r.f. gain control as low as possible. With the BFO switched "on" this adjustment will provide the maximum oscillator injection and, therefore, minimum distortion. This trick may fail, however, because turning up the audio gain control may raise the hum and noise high enough to make the SSB signal unreadable. It all depends on the design of the receiver. Use of either the BFO pitch control or the main tuning/fine tuning controls gives satisfactory tuning with this procedure. In either case, tune for the most natural sounding understandable voice.

## 2.4 INDEPENDENT SIDEBAND OPERATION (ISB)

Some sideband transmitters and transceivers are designed to handle two different or independent information channels, one in each sideband. In other words, if you wished to transmit RATT on USB and voice LSB to another ship, this dual or twin channel information could be sent out at the same time by an ISB transmitter. These transmitters (or transceivers) include principally the AN/URC-32, AN/URT-23, 24, AN/WRC-1, AN/WRT-2, and AN/FRT-39/40 series. The station at the other end can employ separate receivers, one for the RATT tones on USB and another for voice detection on LSB, or one ISB receiver. Certain receivers such as the R-1051/URR and AN/FRR-60 can simultaneously demodulate ISB dual signals.

At this point in time it is not possible to transmit and receive on the same (carrier) frequency from one ship or station simultaneously. The "spillover" from the filtered sideband would still be of such a strength that it would block the receiver even though tuned to the opposite sideband. On the other band, shore stations, whose transmitting and receiving sites are separated by tens of miles, can often receive on the opposite sideband while an associated SSB transmitter is on the air. Therefore, Independent Sideband operations has come to mean its transmission or its reception of two different sets of information, one set in USB and one set in LSB.

An ISB transmitter block diagram is given in Figure 23. Note that the master oscillator or (carrier) frequency synthesizer supplies the same signal to two, separate balanced modulators. The balanced modulators, of course, produce a DSB signal from the carrier and respective audio input channels. The carrier is cancelled in the output of each balanced modulator. This cancellation leaves two distinctive double sideband signals bracketing the same carrier frequency. Sideband filters reject the upper sideband on one channel and the lower sideband on the other channel. Thus the two SSB emissions can be mixed together without creating mutual sideband interference or cross-talk. Finally, this double channel "package" is amplified and radiated as a single emission--see Figure 24.

### A. Setting Power Level in ISB

In ISB operation the transmitter shares its power between the two channels. The operator

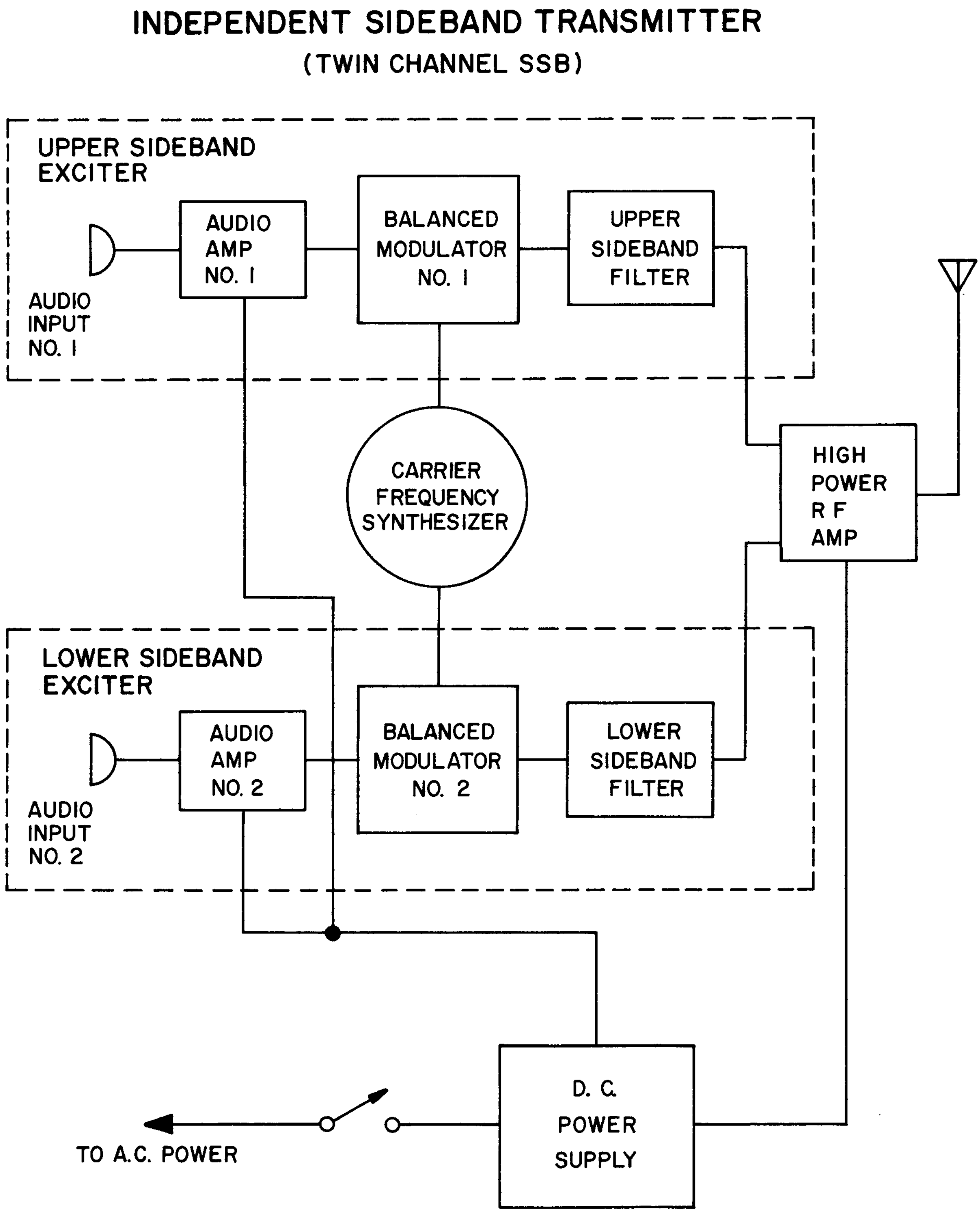



Figure 23. Independent Sideband Transmitter (Twin Channel SSB)

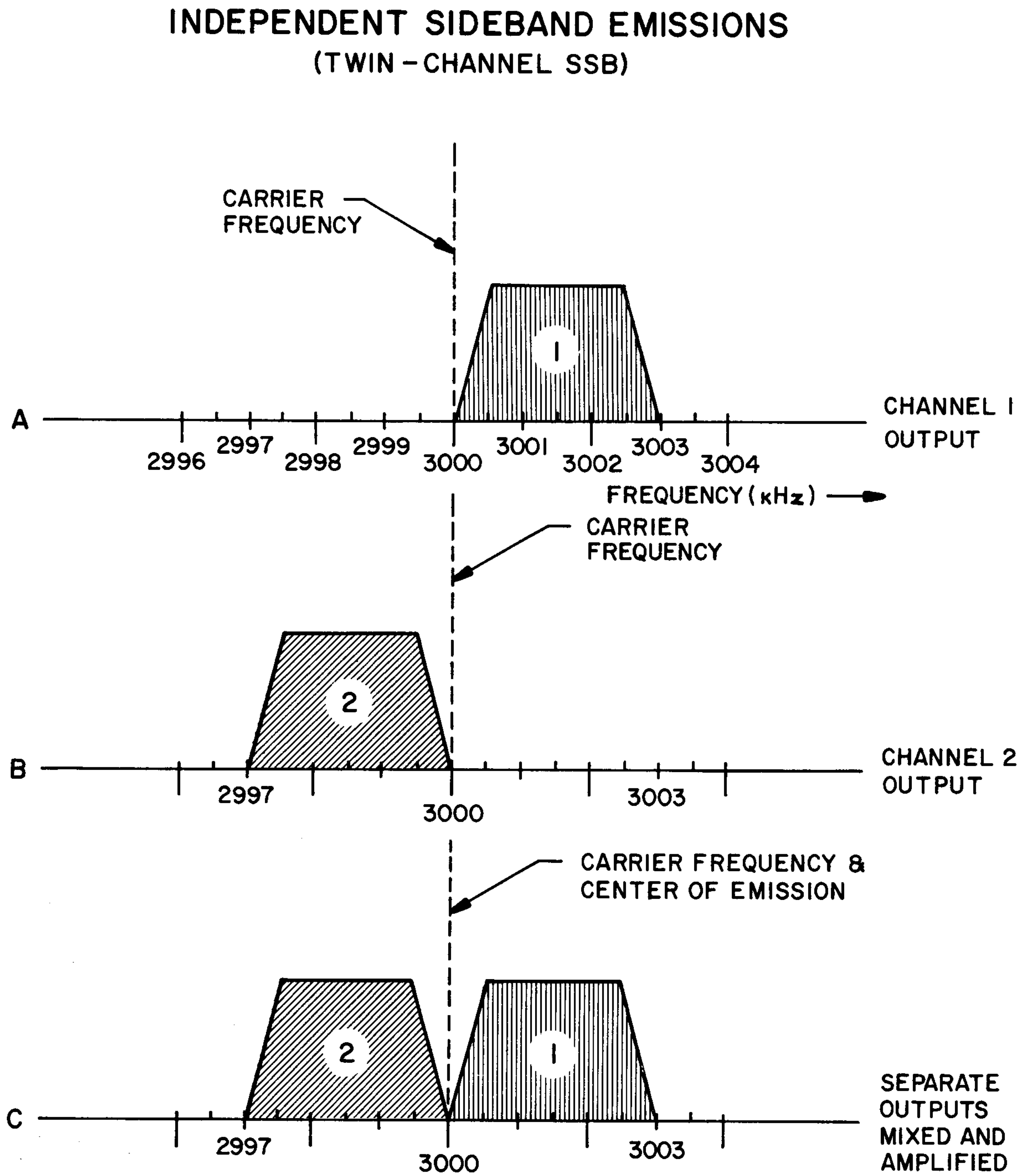



Figure 24. Independent Sideband Emissions (Twin-Channel SSB)

must be careful to avoid overloading the transmitter. The best procedure here is to tune the transmitter like the normal SSB transmitter, using only one active audio input. Then, back down on the transmit gain or audio level for that channel by 3 db (on the input level meter, or until the output power meter is peaking 50% of the 1/3 scale reading or 1/6 scale). Now channel 1 is developing half the transmitter peak power. Disconnect the audio source from channel 1 and connect source number 2 to audio input 2. Adjust the transmit gain or audio level on channel 2 so that the power output meter is peaking 50% of 1/3 scale or reading about 1/6 scale, the same as for channel 1. Now connect both audio inputs and the transmitter will run at full peak power. This procedure has assumed a voice-type signal in each channel. If multi-channel teletype tones are used on one or both channels the settings will again be different, because peak to average power levels in multitone teletype are different than that for voice or single channel teletype. When transmitting multitone teletype ISB with a modern transmitter adequate metering in the exciter circuits is usually provided. Some stations also use spectrum analyzers to check up on the clarity of the emission.

## 2.5 SINGLE CHANNEL RADIOTELETYPE WITH SSB

### A. General Notes and Comments

Some confusion exists today on the relationship between radioteletype (RATT) and SSB. The reason is that the Navy now uses two types of RATT emissions. Both variations require the use of two discrete or separate radio frequencies or audio tones to produce one channel of RATT: one frequency for the Mark signal and the other for the Space signal. At any instant of time only one frequency or tone is being emitted by the transmitter. For the moment we shall call the two methods the Old RATT and New RATT. The same teletypewriter machines are used in transmitting and receiving RATT with single sideband--the new system-- as with frequency shift keying (FSK)--the old system. The transmitters and keyers have been changed in order to use the new RATT. The teletypewriter sets, tape cutters, and perforator/reperforators remain the same.

The teletypewriter operates by mechanical linkages, direct current switches, and electromagnets. In the send (or encoding) mode the operator's pressing of the teletypewriter keys enables the machine to generate a sequence of current pulses. These pulses are the teletypewriter's code signals. A space pulse is the absence of current; whereas, a Mark is a 60 milliampere d.c. pulse. In some machines it is the difference between current polarity that determines whether a Mark (+) or a space (-) is being transmitted. Some machines merely provide contact closures and assume that the power source is in the equipment being keyed by the teleprinter. In recent years a low-level teleprinter signal has come into use. This standard signal requires the teleprinter receiver to function correctly with a maximum current of .0001 amperes. This low level keying signal has become important where radio frequency interference (RFI) suppression of the 60 mA. signal is a problem. Each character (letter or figure) has a unique mark-space, sequence associated with it, generated by the action of various cams, levers, and contactors. In addition an initial start element (space) and a terminal stop element (mark) is established to maintain synchronism between sender and receiver. The most common system used in Navy adds a start pulse the same length as the five character pulses, and a stop pulse of somewhat longer duration. This is known as the five level Baudot code. Further detailed information about codes and teletypewriter machines can be obtained by reference to: PRINCIPLES OF TELEGRAPHY NAVSHIPS 0967-255-0010 and MIL-STD-188( ).

In the receive (or decoding) mode the incoming current pulse sequences drive the selector magnets and mechanical parts which print the correct letters.

A list of most common teleprinter equipments and ancillary units used aboard ships are given in Table 2.

### B. RFCS RATT

This old system is properly referred to as RFCS RATT (Radio Frequency Carrier Shift Radio Teletype). The system concept is that shown in Figure 25. Figure 25, for simplicity, does not include the typical patch panels and cryptographic equipment found in every communications facility. Here a send-teletypewriter operates a keyer which shifts an unmodulated radio-frequency carrier or CW signal back and forth between two discrete or separate frequencies. General practice is to use a total shift of 850 Hz (± 425 Hz) between the mark and space elements. The mark frequency is the higher of the two frequencies.* The center of the emission is halfway between the mark/space frequencies. This center frequency would be the same as the carrier frequency prior to hooking up the RATT keyer. In this FSK or RFCS RATT system you transmit only one single channel of teletype unless time division multiplexing (TDM)

*Confusion exists about a suitable standard for this criteria. Existing Navy equipment is set up so that Mark is high and Space is low. Many foreign governments do exactly the opposite.

Table 2

CURRENT TELETYPEWRITER EQUIPMENT ABOARD SHIPS

| | |
|---|---|
| **TELETYPEWRITER SETS:** | |
| AN/UGC-6 or AN/UGC-16 | Deck mounted consisting of page printer, keyboard, tape perforator, transmitter distributor, and reperforator. |
| TT-187/UG | Transmitter distributor. |
| **PAGE PRINTERS:** | |
| AN/UGC-25A | Desk model machine without keyboard. |
| AN/UGC-20A | Desk model machine with keyboard. |
| **PERFORATORS/REPERFORATORS:** | |
| TT-192/UG | Desk model reperforator |
| TT-253/UG | Desk model perforator/reperforator |
| **RECEIVING CONVERTERS (COMPARATORS):** | |
| AN/URA-8 Series | Vacuum tube |
| AN/URA-17 Series | Solid State |
| **MULTICHANNEL (FREQUENCY DIVISION MULTIPLEX) TONE CONVERTER-KEYER TERMINAL:** | |
| AN/UCC-1 Series | Fully duplex, 1 to 16 channels i.e. 16 send/16 receive |

# OLD RATT SYSTEM
## RADIO FREQUENCY CARRIER SHIFT (RFCS)

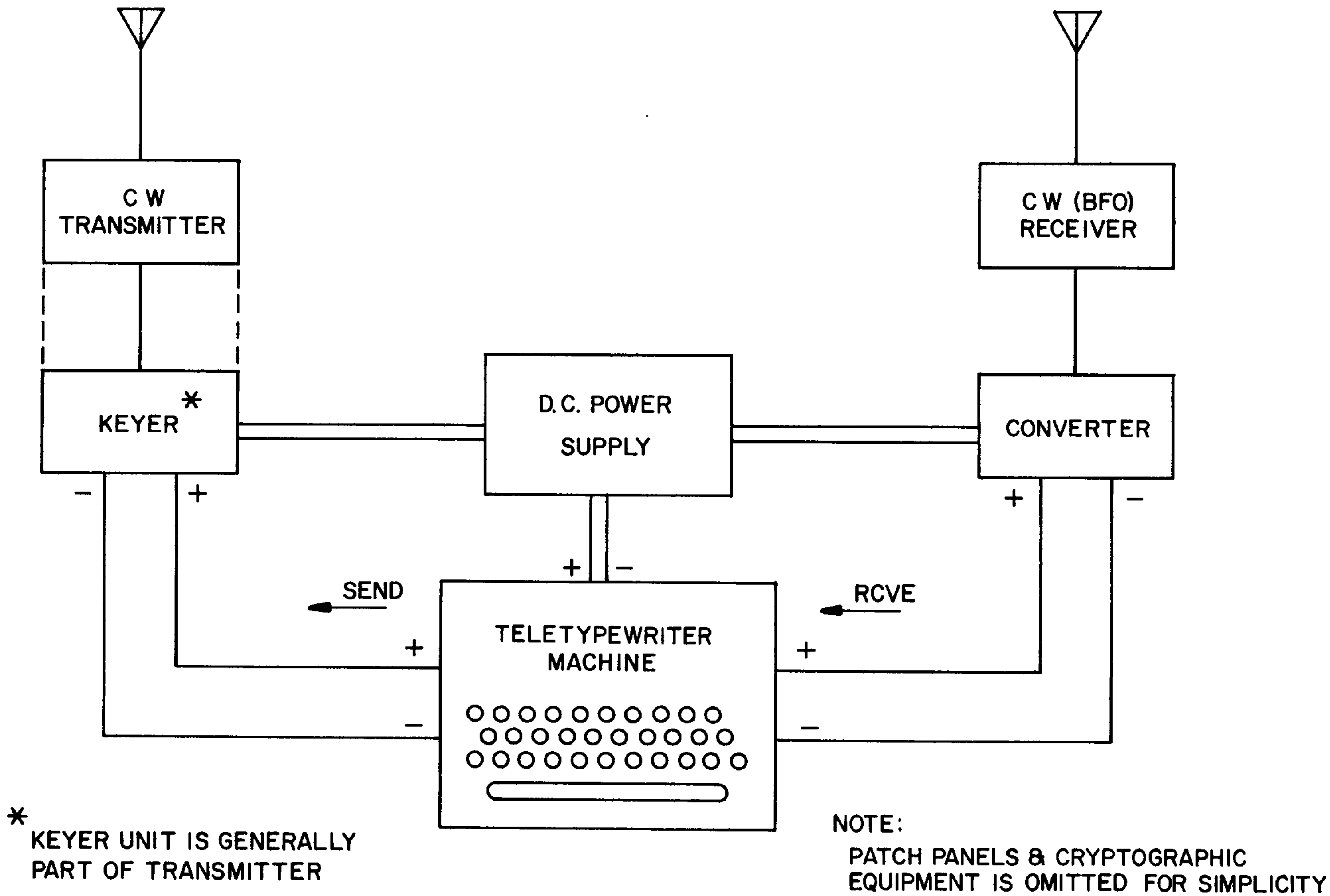



Figure 25. Old RATT System, Radio Frequency Carrier Shift (RFCS)

equipment such as AN/UGC-1 is employed to enable up to four teletypewriter circuits to "share" the same pair of radio frequencies. Nowadays this is seldom done because a better multichannel capability has evolved with New RATT systems.

Figure 26 shows the principle of the type of circuit that is used with FSK keyers. Here the (plus and minus) current pulses originating from the teletypewriter machine alternately forward and reverse bias a semiconductor/solid state diode. The diode acts as a switch. When the diode is reverse biased (negative space pulse) it is effectively open and no frequency shift occurs. A position mark pulse, however, causes the diode to conduct, which permits an additional capacitor to be inserted in parallel with the capacitor nominally employed in an LC tuned circuit. The addition of this extra capacitance shifts the frequency of carrier oscillator in the transmitter.

Other variations are possible, of course, but the general idea is the same for any FSK unit. Remember, that in the RFCS RATT or FSK the marks and spaces are sent by simply shifting the carrier radio frequency of a CW transmitter. RFCS has nothing whatsoever to do with AM or SSB. In fact RFCS is more similar to FM than any other modulation discussed, because the carrier amplitude is kept constant while carrier frequency is changed. A separate FSK unit, the KY-75/SRT, is needed to transmit RFCS RATT signals when using some of the older HF Transmitters, such as (AM/CW only) models TBK and TBL. All the newer SSB equipments, however, have keyer circuitry built into the transmitter cabinet.

If told to set up a transmitter in RFCS RATT on an assigned frequency of 3000 kHz, you would simply dial 3000 kHz on the front panel of any transmitter capable of CW/FSK. The dial always reads the carrier frequency. Activation of the FSK unit causes the two CW signals to be emitted alternately; neither of which are exactly on 3000 kHz. Thus, the teletypewriter mark will be transmitted 425 cycles above the center frequency, and the space will be transmitted 425 cycles below the center frequency. This transmitter then develops a mark-shifted signal at 3000.425 kHz and a space-shifted signal of 2999.575 kHz. Thus, for a case of old or RFCS RATT the carrier and center of emission (bandwidth) are one and the same.

In order to receive RATT, the communications system requires a CW receiver with a bandwidth great enough to accept the 850 Hz shift between mark and space frequencies. A suitable RATT converter completes the receiver-side loop. The two most common RATT converters found aboard ship are the AN/URA-8 and AN/URA-17 series. They both perform the same function: the units convert the audio frequency (tone) output of the receiver into d.c. pulses. The pulses operate the teletypewriter machine. The AN/URA-17 is the later, entirely solid state model. This converter is about half the size of the AN/URA-8.

Converters usually employ a discriminator circuit to transform the tones into pulses. This circuit is the audio analogy of an FM radio. An AM detector with special filters can accomplish the same purpose. Figure 27 shows the concept of the latter, simpler type. Basically it uses two diode type detectors, each mated to a special filter. The "forward" (upper one in the figure) diode has a filter in front tuned to accept only the mark tone. The "backward" diode (lower one in the figure) has a filter tuned to pass only the space tone. This arrangement of diodes gives positive rectification of mark tones and negative rectification of space tones. These pulses of positive/negative direct current drive a pair of vacuum tubes or transistor amplifiers. The amplifiers act as high-current switches capable of driving the page printers or typing reperforators. Until recently FSK converters have centered their + and - 425 Hz shift around an audio center frequency of 2550 Hz. This is being changed, however, and most new equipments will use a center frequency of 2000 Hz. The frequencies of mark and space tones are 2425 Hz and 1575 Hz, respectively.

## C. AFTS RATT

New RATT system has been referred to in the past by a variety of terms including Tone Modulated RATT, VFTG RATT, SSB RATT, and Audio Frequency Shift Keying. It is now most often called (in the Navy) AFTS RATT, for Audio Frequency Tone Shift RATT. In this new RATT system the mark and space signals are developed by shifting back and forth between two discrete or separate audio frequency tones. The system concept is shown in Figure 28. As seen from the diagram the FSK unit has now been replaced by a tone generator. SSB transmitters and receivers are employed instead of CW equipment. But a converter is still required between the receiver and teletype machine. In this AFTS system the d.c. mark and space pulses shift the frequency of a tone oscillator in the tone generator equipment. The alternating tones are then fed into the audio input of the SSB transmitter. The "tone keyer" circuit used in AFTS RATT is very similar to the FSK unit shown earlier in Figure 26. The only difference is that an audio oscillator or tone generator replaces the carrier oscillator. Refer to Figure 29 for comparison of RFCS vs AFTS emission. In single channel RATT, an 850 Hz (± 425) shift is the standard employed. The frequency of the emission center will vary in various equipments, but 2000 Hz is now standard. For equipments in the field today, the mark audio tone is at 2425 Hz, and the space tone is 1575 Hz. If the

# RADIO TELETYPEWRITER KEYER

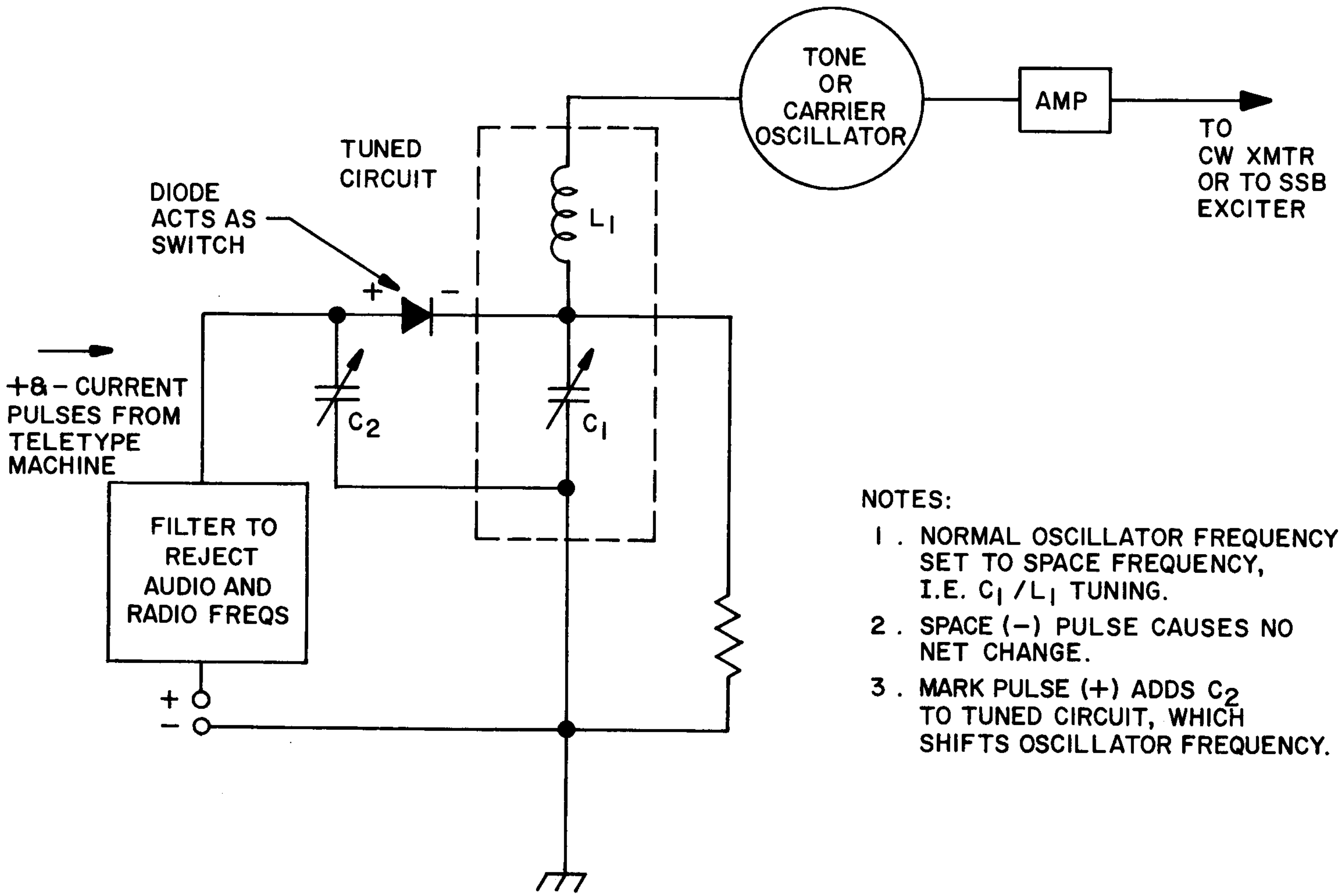
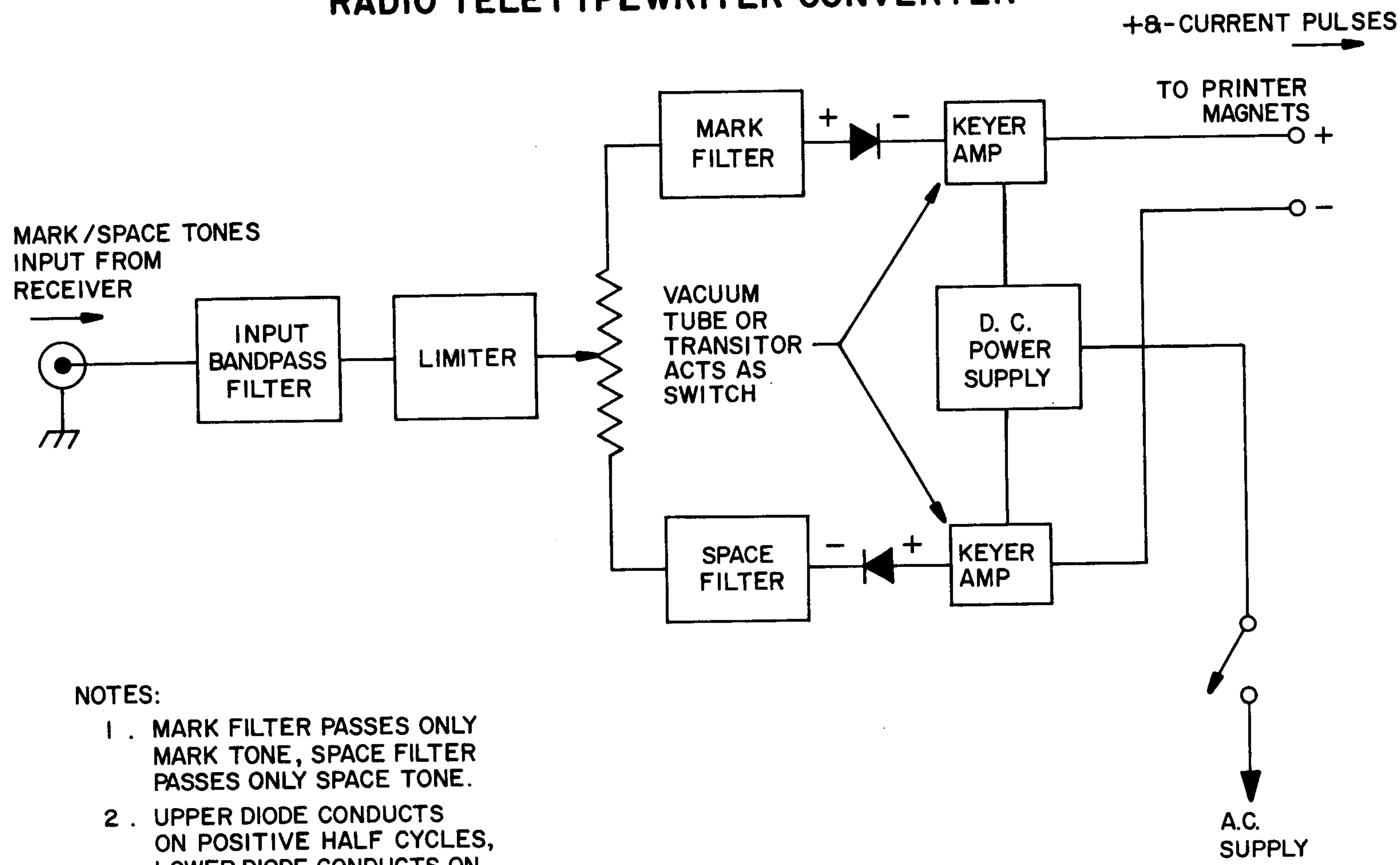



Figure 26. Radio Teletypewriter Keyer

## RADIO TELETYPEWRITER CONVERTER




NOTES:

1. MARK FILTER PASSES ONLY MARK TONE, SPACE FILTER PASSES ONLY SPACE TONE.
2. UPPER DIODE CONDUCTS ON POSITIVE HALF CYCLES, LOWER DIODE CONDUCTS ON NEGATIVE HALF CYCLES.
3. KEYER AMP SUPPLIES POSITIVE / NEGATIVE CURRENT PULSES TO DRIVE TELETYPEWRITER MACHINE.

Figure 27. Radio Teletypewriter Converter

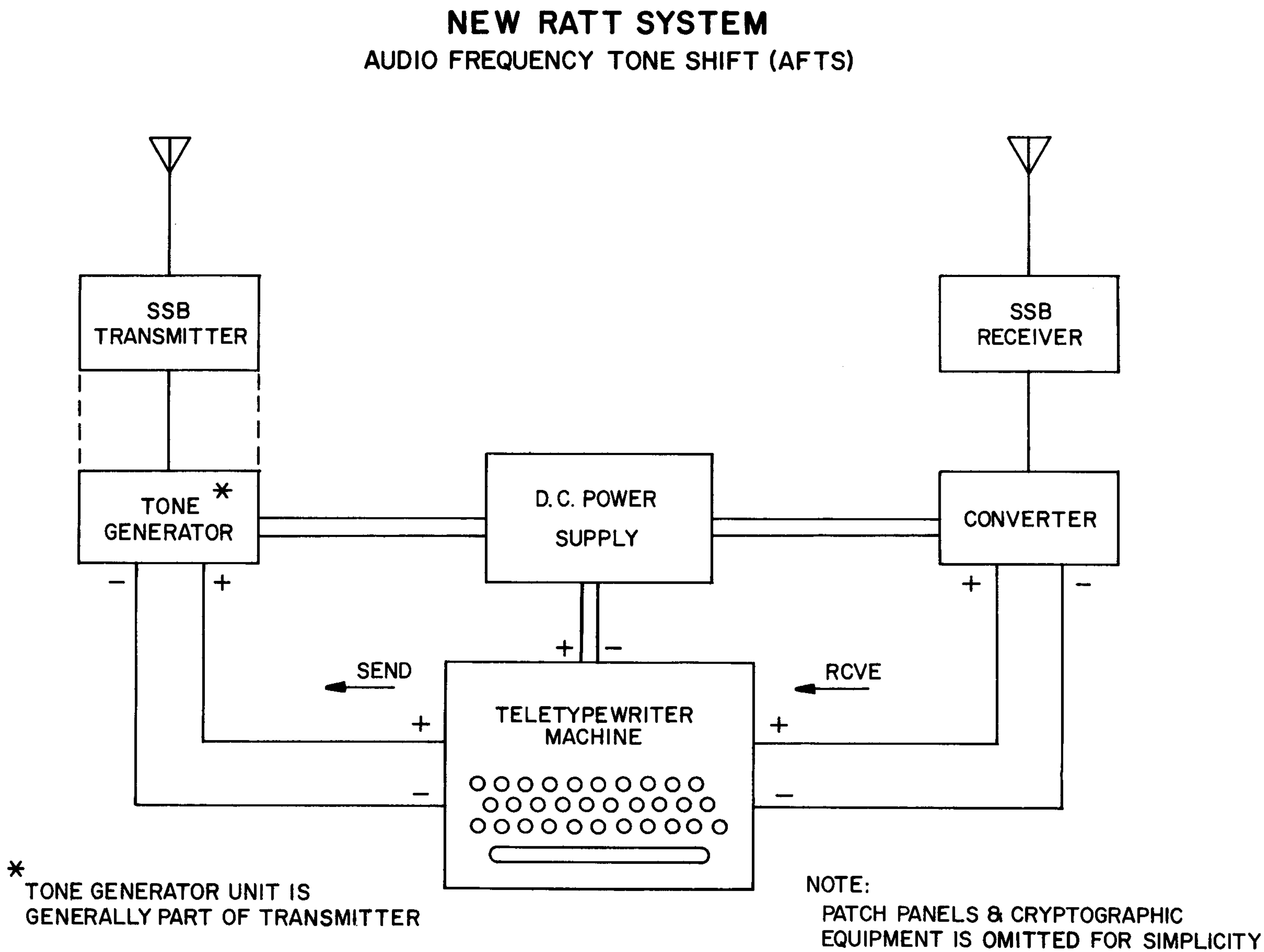



Figure 28. New RATT System, Audio Frequency Tone Shift (AFTS)

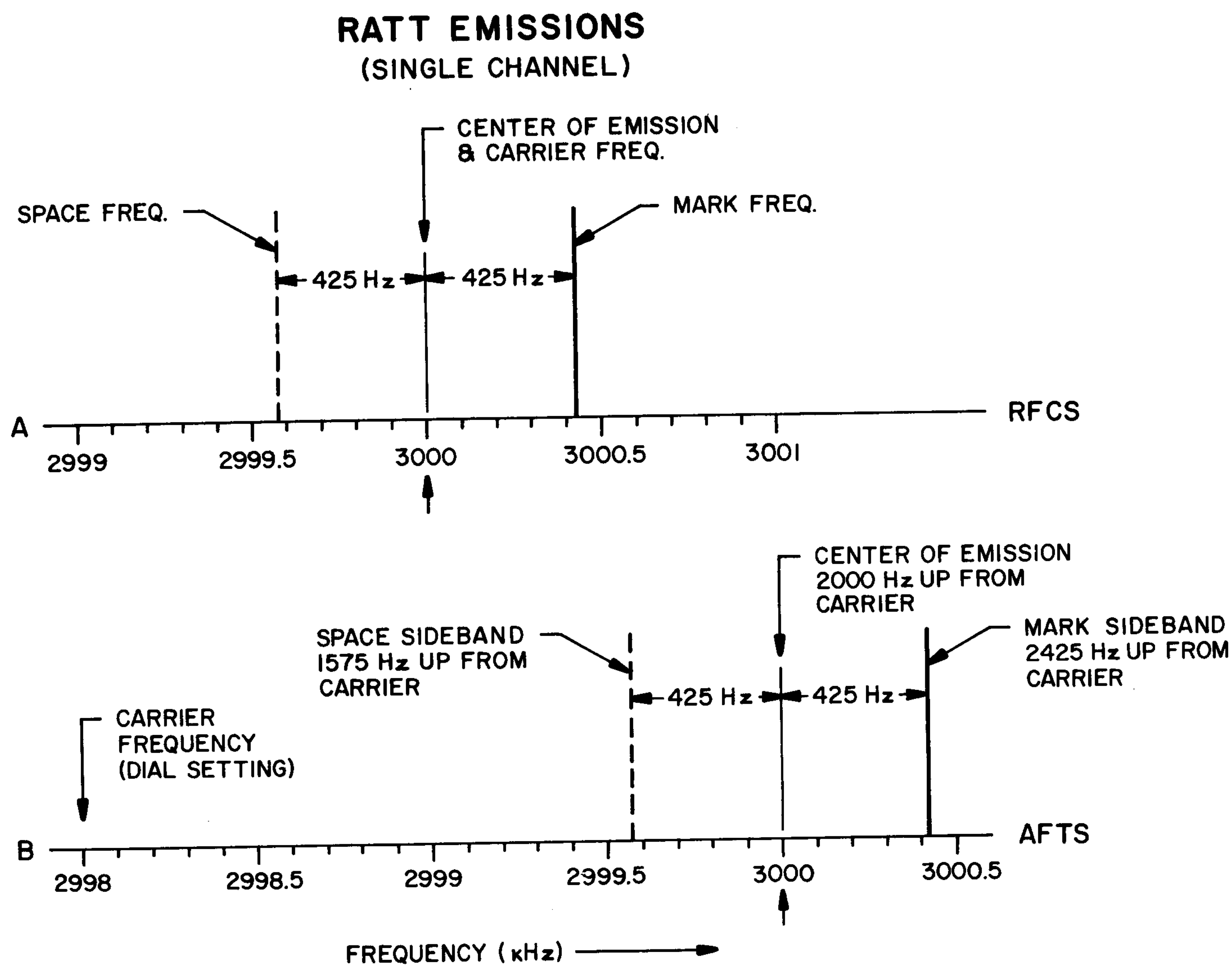



Figure 29. RATT Emissions (Single Channel)

emission center was 2550 Hz (the old standard), the mark tone would be 2975 Hz and the space tone would be 2125 Hz. The occasional difference in tone generator center frequencies sometimes causes confusion as to what carrier or dial setting should be utilized when transmitting AFTS RATT, SSB emission. The universal rule: the frequency assignment is always the center of the emission bandwidth, applies again. You must offset your carrier setting by an amount equal to the center frequency of the tone package. This is the same thing as saying that the sidebands emitted by mark and space modulation are equally displaced (± 425 Hz in frequency) from where the emission center frequency would be if it were ever generated. Therefore the carrier must be offset by the same amount in kHz. Consider the earlier example using a tone generator which has an unshifted or center tone frequency of 2 kHz, you are told to set up for AFTS RATT on a net frequency of 3000 kHz (USB). In this situation, set the transmitter/exciter frequency dial 2 kHz down from the published frequency assignment so the carrier frequency is 2998 kHz for USB. In Naval communications RATT tones are most usually transmitted using upper sideband.

In USB voice emission, the sidebands start near the suppressed carrier frequency and extend up to 3 kHz, the highest frequency voice sideband. It is not the same in SSB RATT. The mark/space tones originate away from the carrier frequency at a spacing equal to the frequency of the individual tone, i.e. 1575 Hz for a space and 2425 Hz for a mark. This discussion of single channel AFTS RATT should prepare the reader for the last RATT topic: Multi-channel radioteletype, or frequency division multiplexing.

## 2.6 MULTICHANNEL (FREQUENCY DIVISION MULTIPLEX) RADIOTELETYPE WITH SSB

One of the most practical advantages of using the new AFTS RATT over the old RFCS RATT is that the new system easily permits a number of channels information to be transmitted and received over one SSB (equivalent) voice channel (3 kHz bandwidth). Up to 16 channels of 100 wpm RATT can be accommodated with just one SSB transmitter and one SSB receiver at each end of the circuit. This is made possible by means of using the technique of frequency division multiplexing.

The system now being installed on most ships is the AN/UCC-1. This versatile piece of terminal equipment is almost entirely solid state design, with plug-in modules. Three (3) types of modules are employed in the AN/UCC-1 system: Keyer units, converters, and control-attenuators. Various types of shipboard installations are found, depending on whether the AN/UCC-1 functions are being employed in receive only, transmit only, or receive-transmit capacity. Cabinets are added as greater number of channels or functions are desired.

Each cabinet contains one control attenuator and a bank of either eight converters or eight keyers. The individual converter or keyer modules are wired through a plug on the back to the control-attenuator unit. The control-attenuator unit combines all the tones into a composite tone package when transmitting. On receive it accepts the composite tones for distribution to individual converters.

Each keyer module generates a tone pair (mark-space with an 85 Hz shift). A complete cabinet rack of eight keyers and one control attenuator will generate eight audio tone pairs spaced throughout a 3000 Hz audio passband. Such a keyer cabinet will accommodate data from eight teletype machines when each keyer module is operating in the "independent" mode. When "Dual Diversity" or "Twinned" operation is desired, each cabinet of eight keyers and one control attenuator will accept data from four teletype machines. When "Quad Diversity" is employed, each similar cabinet will accommodate two teletypewriter machines.

When receiving RATT signals, multiplexed in a voice bandwidth, a similar breakdown is possible with a single cabinet rack of eight converters and one control attenuator will provide the following:

| Mode of Diversity | Number of teletype machines operated |
|---|---|
| Individual | 8 |
| Dual | 4 |
| Quadruple | 2 |

In each case, the control attenuator gives precedence to the strongest signal that it receives.

A block diagram of a communications system employing an eight channel version of the AN/UCC-1 is shown in Figure 30.

Figure 31 portrays a four channel version of the AN/UCC-1, operating in a dual diversity mode or "Twinned." This is often done to improve communications quality and reliability when ionospheric conditions are poor.

The AN/UCC-1 can be expanded to allow the multiplexing and demultiplexing of 16 channels of teletypewriter data into a single 3 kHz audio passband. Auxiliary equipment is available to double this number of channels, but 6 kHz of bandwidth is required.

The AN/UCC-1 multiplex equipment is naturally more complex than a single channel tone shifted RATT equipment, but the theory of operation is exactly the same in both cases. The multichannel equipment keyers are separate tone pair oscillators and the control attenuator is a linear

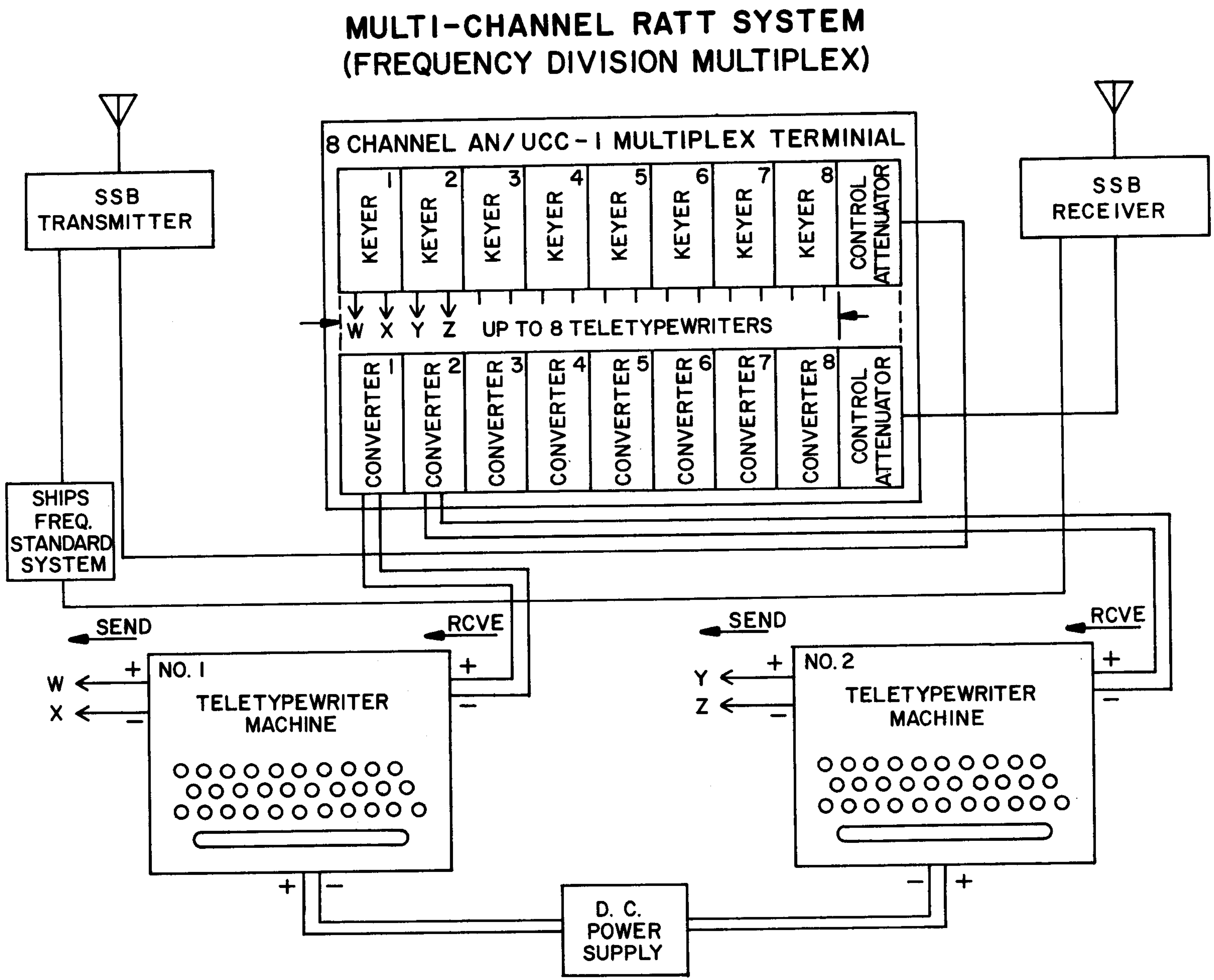



Figure 30. Multi-channel RATT System (Frequency Division Multiplex)

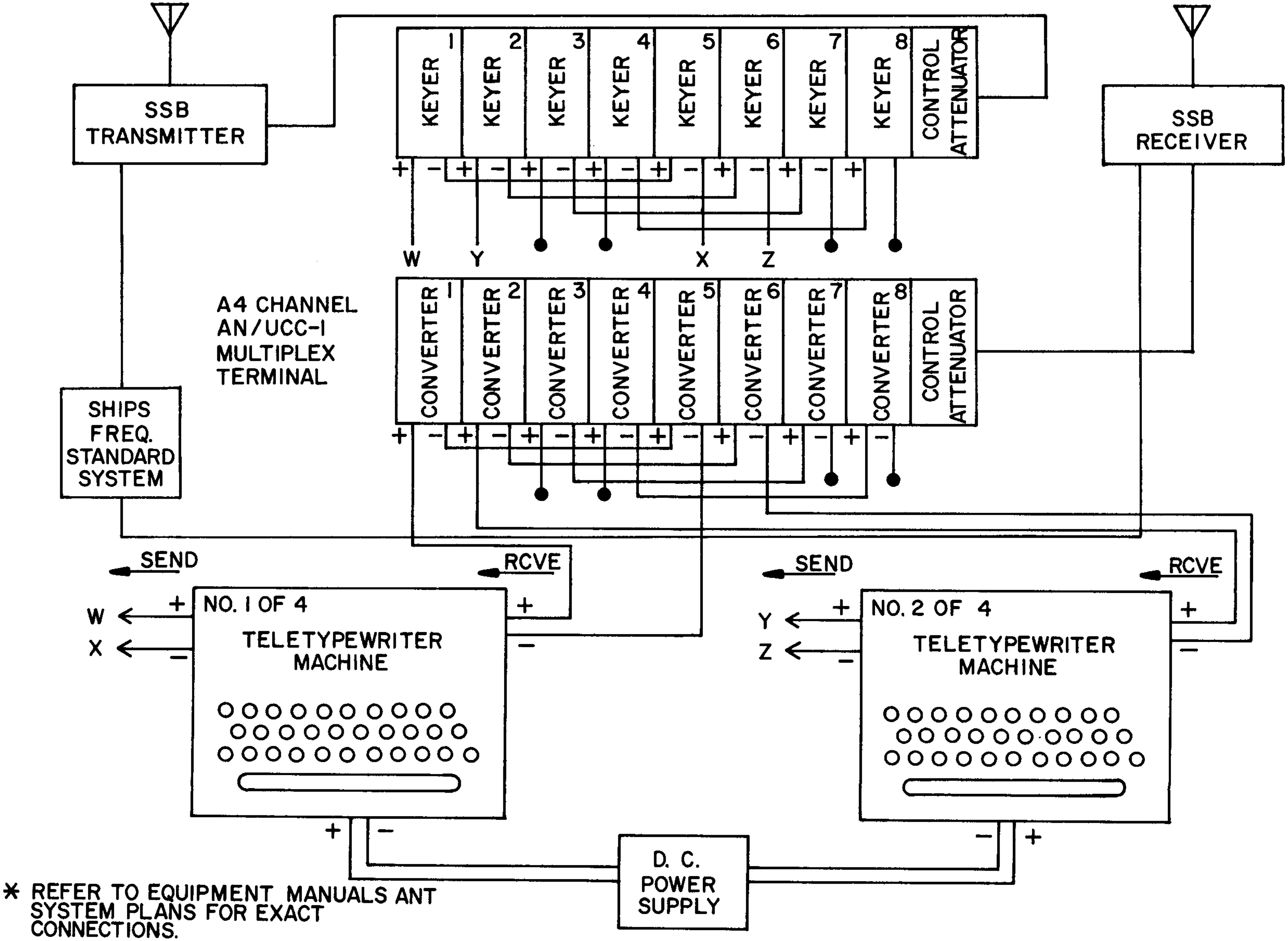



Figure 31. 4 Channel Version of a Multi-channel RATT System Operating in the "Twinned" Mode

mixer that combines all the tone pairs together. By the same token, it requires a complex array of filters and detectors in the converter modules for sorting out the individual component tone pairs from the received tone package. This added complexity affects only the D. C. teletypewriter loop circuits. As shown in Figure 30 there is still only one audio line between the output of the keyer control attenuator unit and the input of the transmitter. In a similar manner just one audio line goes from the single sideband receiver to the converter control attenuator. The emission from the AN/UCC-1 multichannel terminal is shown in Figure 32.

The AN/UCC-1 multi-channel terminal equipment employs a tone shift that is one-tenth as wide as single channel (AFTS) system; i. e., 85 Hz (±42.5 Hz) instead of the 850 Hz (±425 Hz) shift. Channel designations are normally standardized to correspond with particular tones or audio frequencies. These designations are tabulated in Table 3.

Table 3A lists the audio tone frequency assignments for a full 16 channel communications system. Since the tones roughly cover 3 kHz (382.5 Hz through 3017.5 Hz), this entire tone package occupies the same spectrum width as a SSB voice channel.

Table 3B lists the audio tone frequency assignments for an eight channel communications station. This plan uses the full 16 tone pairs (two cabinets of eight keyers each, plus their control attenuators) operating in the dual diversity mode. It can be seen that channel 1 and 9, channel 2 and 10, etc., are combined or "twinned" when dual diversity operation is desired.

When only eight-channel operation is required, and no diversity operation is desired, a single cabinet of eight keyers and one control attenuator plus a single cabinet of eight converts plus one control attenuator is all that is required. When such a configuration is set up, the alternate channel tone assignments would be used; i. e. channels 1, 3, 5, 7 etc. In this manner the eight-channel operation would still use the entire 3 kHz voice bandwidth in the transmitter.

There is a multichannel emission, however, used in the LF spectrum, where only the first eight tone frequency assignments are used. In this mode of operation about half of the full 3.0 kHz bandwidth is required.

## A. Telegraphic Signal Distortions

No treatment of telegraphic SSB communications is complete without a discussion of Signal Distortion. Distortion is considered to be of two types, systematic and fortuitous. Fortuitous distortion is caused by random variations in the signals by propagation phenomena, hum, noise spikes, momentary battery fluctuations, etc. On the other hand, systematic distortions can be traced to regular periodic displacement of the mark or space pulses due to a variety of causes. The telegraph signal, as received on a SSB receiver, consists of mark and space pulses (Mark = 1, +, "on," or high tone) (Space = 0, -, "off,", or low tone). The actual changing of the signal from mark to space or vice versa is called a transition. It is in the precise timing of these transitions that the criteria for distortion definitions and measurement are established.

The systematic distortion types that RATT is subject to are the following:

- Bias Distortion
- Characteristic Distortion
- Cyclic Distortion

Bias Distortion is the uniform lengthening or shortening of the Mark or Space elements, one at the expense of the other. The degree of bias distortion is measured as a percentage of the unit interval. Common causes of bias distortion include frequency error of the SSB receiver, improperly adjusted relays, earth potential differences, leakage currents, transmission line unbalance, and unequal battery tap resistance or potentials. If the bias distortion is severe enough, the signal will not lie in the range of the teletypewriter and errors will occur in the teletypewriter copy.

Characteristic Distortion is the displacement of signal transitions resulting from the persistence of voltage or current transients caused by or related to previous transitions. This type of distortion is related to the characteristics of the transmission lines (inductance, resistance, and capacitance) interconnecting receiver/converter and teletypewriter. If the line is too inductive, the current pulse of the Mark will not die out soon enough and will affect the timing of each subsequent transition. When a signal consists of text or random characters, it is nearly impossible to distinguish characteristic distortion from fortuitous distortion.

Cyclic Distortion has a periodic nature. This type of signal distortion usually originates from some defective or improperly adjusted device in the circuit. Induction (noise and hum) from power circuits is a typical example.

Two other phenomena are defined as telegraphic signal distortion of a special nature or measurement. These are "End Distortion" and "Peak Distortion".

End Distortion is not caused by equipment malfunction. This is an artificial distortion produced by the test signal generator for test and maintenance purposes.

Peak Distortion, too, is not a type of distortion but rather a measurement of the maximum

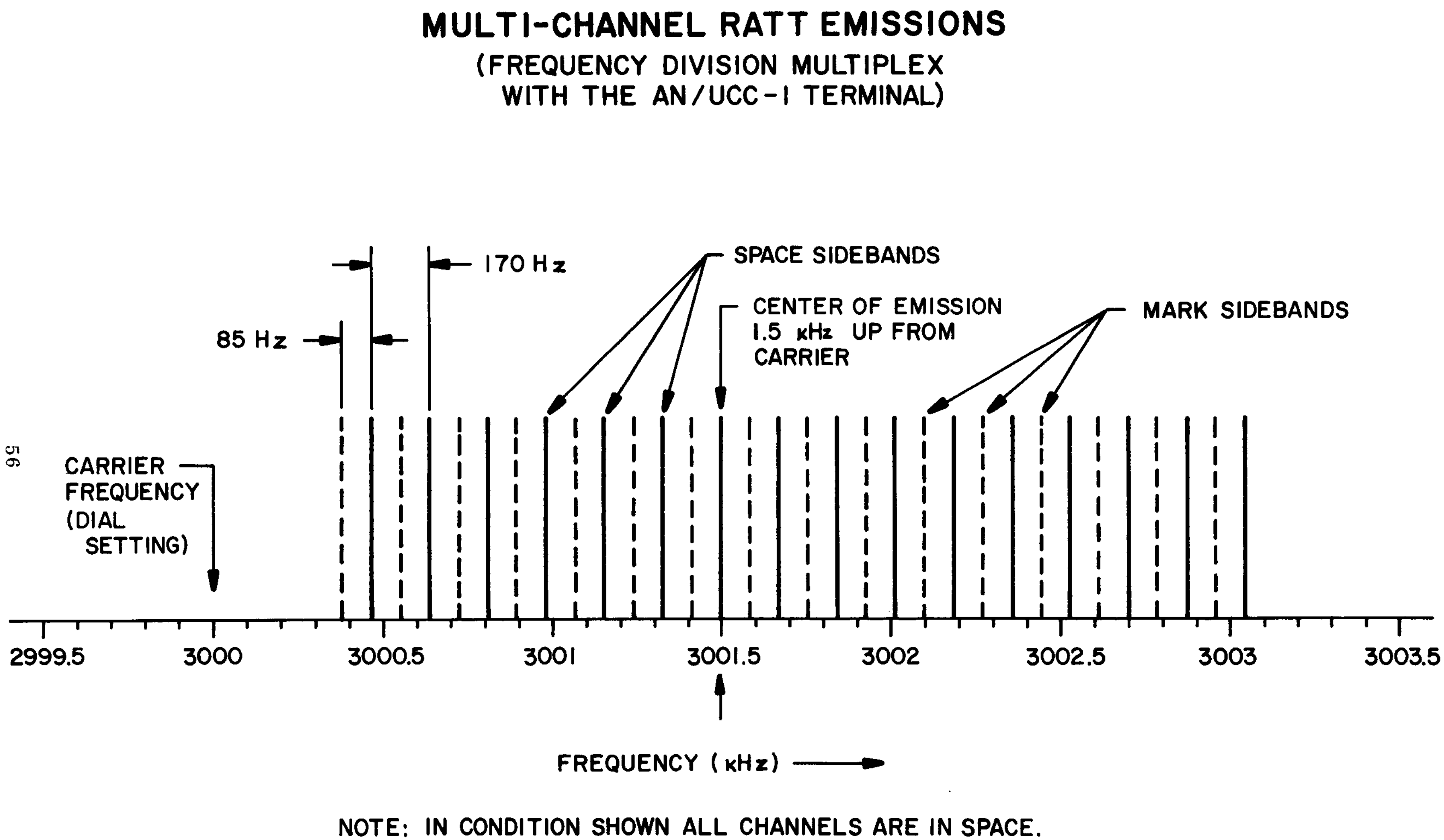



Figure 32. Multi-channel RATT Emissions (Frequency Division Multiplex with the AN/UCC-1 Terminal)

Table 3

MULTI-CHANNEL

MULTIPLEXED TONES

TABLE A

Non-diversity* mode

| Channel Designation | Mark Frequency (Hz) | Center Frequency (Hz) | Space Frequency (Hz) |
|---|---|---|---|
| 1 | 382.5 | 425 | 467.5 |
| 2 | 552.5 | 595 | 637.5 |
| 3 | 722.5 | 765 | 807.5 |
| 4 | 892.5 | 935 | 977.5 |
| 5 | 1062.5 | 1105 | 1147.5 |
| 6 | 1232.5 | 1275 | 1317.5 |
| 7 | 1402.5 | 1445 | 1487.5 |
| 8 | 1572.5 | 1615 | 1657.5 |
| 9 | 1742.5 | 1785 | 1827.5 |
| 10 | 1912.5 | 1955 | 1997.5 |
| 11 | 2082.5 | 2125 | 2167.5 |
| 12 | 2252.5 | 2295 | 2337.5 |
| 13 | 2422.5 | 2465 | 2507.5 |
| 14 | 2592.5 | 2635 | 2677.5 |
| 15 | 2762.5 | 2805 | 2847.5 |
| 16 | 2932.5 | 2975 | 3017.5 |

TABLE B

Dual diversity Mode

| Channel Designation | Mark Frequency (Hz) | Center Frequency (Hz) | Space Frequency (Hz) |
|---|---|---|---|
| 1 | 1827.5 | 1785 | 1742.5 |
| | 382.5 | 425 | 467.5 |
| 2 | 1997.5 | 1955 | 1912.5 |
| | 552.5 | 595 | 637.5 |
| 3 | 2167.5 | 2125 | 2082.5 |
| | 722.5 | 765 | 807.5 |
| 4 | 2337.5 | 2295 | 2252.5 |
| | 892.5 | 935 | 977.5 |
| 5 | 2507.5 | 2465 | 2422.5 |
| | 1062.5 | 1105 | 1147.5 |
| 6 | 2677.5 | 2635 | 2592.5 |
| | 1232.5 | 1275 | 1317.5 |
| 7 | 2847.5 | 2805 | 2762.5 |
| | 1402.5 | 1445 | 1487.5 |
| 8 | 3017.5 | 2975 | 2932.5 |
| | 1572.5 | 1615 | 1657.5 |

*Signal sense in Normal position on AN/UCC-1 multiplex terminal; with signal sense switch in reverse position interchange Mark & Space.

displacement of signal train transitions occurring during a measurement period.

The net result of all types of distortion is to cause false signal transitions, garbles in the copy--character errors.

### B. Multichannel Frequency Stability Requirements

When receiving multichannel RATT communications over an HF SSB link, the extreme importance of frequency accuracy and stability cannot be minimized. It is important enough to warrant continuous checking and calibration of the stable local oscillators of both receivers and transmitters. The continuous use of the AN/URQ-9 and 10 frequency standards for this purpose is strongly recommended.

Frequency stability is very critical in RATT work, particularly when using the narrow shift, multi-channel equipment with SSB. In receiving multi-channel AFTS RATT, use of anything but a synthesized SSB receiver is definitely not recommended. In HF range the best receiver for this purpose is the R-1051/URR receiver. The reason for the extreme importance of frequency accuracy and stability is that gross frequency error could be interpreted as a mix up in the spaces and marks by the multi-channel converter. When the tones shift in multi-channel RATT, the individual sidebands corresponding to these tones must shift also. Lesser frequency errors produce bias distortion in the converter and control attenuator or D. C. loops. The result of a frequency error on multi-channel RATT is ten times more critical than the same error would be to a single RFCS or AFTS RATT channel. The frequency shift of the multichannel emission is one-tenth of the shift of single channel. To stress the seriousness of frequency stability problems, consider the following example. A frequency error of 50 Hz on a single channel RATT circuit will introduce approximately 5% bias distortion to the copied signal. A frequency error of only 5 Hz on a multi-channel RATT circuit will produce the same 5% bias distortion. A 50 Hz frequency error might result in excellent copy on the single channel circuits; but the same 50 Hz error would be sufficient to render the multi-channel completely garbled and totally unreadable.

### C. Frequency Assignments in Multichannel RATT

As previously discussed, the 16 channel or 16 tone pair package results in approximately 3 kHz emitted bandwidth on SSB. At the risk of being redundant we should mention again the rule:

Frequency assignment is the center of emission bandwidth.

The tones generated by an AN/UCC-1 "send" terminal, or the AN/FGC-60 equivalent shore "send" terminal, cover the audio frequency range of 382.5 Hz to 3017.5 Hz. Normally, two tone pairs are utilized for each intelligence channel (twinning) to give close frequency diversity and decrease the effects of selective fading. The AN/UCC-1, normally, has an eight-channel intelligence capability with 16 tone pairs. The sets of audio tone pairs used on a given intelligence channel are separated by 1360 Hz.

Due to limitations on system equipments, traffic requirements, teletypewriter machines, and personnel, some ships utilize less than eight (some as few as two) intelligence channels. The remaining tone pairs for the unused channels of the AN/UCC-1 serve no useful purpose.

In the multichannel (MUX) the emission center frequency of the 16 tone pair package would be hard to compute. It is standard practice, therefore, to assume that the tones extend to exactly 3000 Hz; therefore, the center of this band is 1500 Hz. Therefore, the same offset rule learned for SSB voice work applies to multi-channel RATT with SSB. Thus, if the assignment for Fleet broadcast multi-channel RATT is 3000 kHz (USB), the 3 kHz multi-channel tone package will exactly straddle 3000 kHz. Use a carrier or dial setting 1.5 kHz below this, i.e. 2998.5 kHz. Generally, shore stations will employ upper sideband. If a Fleet broadcast multi-channel RATT assignment of 3000 kHz (LSB) is used, simply add the 1.5 kHz instead of subtracting it.

An interesting aspect of frequency assignment is the emission 1.7 A7J. This emission is used in the LF spectrum and contains eight channels of teletypewriter information. The bandwidth of the emission is 1700 Hz and the center of the Band is 850 Hz from the carrier. This emission requires some careful arithmetic in order to set the carrier frequency of an LF transmitter that is required to radiate it.

### D. Setting Audio Levels in Multi-channel RATT

A problem occurs with multi-channel RATT. The audio levels transmitted with SSB are quite different from single channel RATT or voice operation. This fact should be apparent from earlier discussions of Independent Sideband operation. Injecting two tone pairs into the exciter-modulator is like operating ISB with two different audio voice inputs. Ideally a transmitter should split its power equally if the two audio inputs are at the same level of amplitude. Similarly, if two tone pairs (each pair of which is actually alternating ±42.5 Hz between mark and space) are applied to the input of a SSB exciter/transmitter a splitting of power is also required. A transmitter will be overmodulating if an attempt is made to insert two tone pairs at the same level as one tone pair. In theory the procedure to follow, then, is to halve the input audio power per tone pair every time you double the number of tone pairs. This can be done either with the aid of the transmit gain/mod level controls in the exciter or with the composite tone/attenuator individual tone level (screw driver adjustment) controls in the UCC-1. Whichever controls are used, the net effect of a halving of tone power every time the number of tones in use is doubled. This prevents the transmitter from overmodulating when the audio levels are too high.

In addition, the overall power output level of the typical SSB voice transmitter may have to be reduced somewhat when using any form of RATT emission, including both AFTS or RFCS keying. This is necessary to prevent the transmitter final amplifier tubes from overheating. Overheating may occur because the average power level or duty cycle that RATT presents to a transmitter is much higher than either voice SSB or CW modes. A general rule of thumb is to start with about one-half (1/2) the maximum power output transmitter CW rating. (Most CW power figures are arrived at by assuming a key down condition approximately 50% of the time. Since, in any form of RATT, either a mark or space tone is being transmitted continually, the duty cycle is practically 100%. You should use only half as much power to begin with. This will prevent "burning up" the amplifier tubes.) Adjust the tone level for single channel AFTS RATT to this point, and use the guide lines that the instruction book gives in terms of 3 db setting on the audio level input meter. When setting up the gain, modulation level and/or tone level controls using the AN/UCC-1 equipment,* it is proper to turn on one tone pair at a time. Advance control for a single tone pair until the output power meter reads 50% of the full power (single channel) reading, i.e. 3 db down. At this setting the first tone pair is developing one-half the transmitter power. Next, disconnect or "block" the channel #1 and connect tone pair channel #2. Leave xmit gain setting alone and proceed to adjust unit tone level for same meter indications on the transmitter. Now both tones can be activated together. The composite tone level will be correct. The transmitter will "run" at the proper output. If you want to run a four-channel transmission, then you have to back down the power of each tone pair another 3 db in order to have the same composite tone level at the transmitter. If full eight-channel diversity or 16 channel non-diversity is desired, the tone levels do not have to cut back exactly in steps of 3 db due to the fact that there is less chance of all the tones being in phase. Figure 33 shows average power settings for 1 kw PEP (500 watt Avg) transmitter, e.g. the AN/WRT-2. Note that transmitter meters indicate average power, not peak envelope power.

It is worth remembering, when setting audio levels, that you would have to increase your transmitter power 2-3 times before anyone would notice the difference in listening level or improvement in RATT copy at the other end. Generally sideband design engineers do not increase the power capability of a transmitter unless they can get at least 4-10 times greater yield in power. Therefore, it's better to use conservative power settings than to try and squeeze a few more watts of power out of any given transmitter. Others will notice the increase in interference level if you overmodulate even slightly and produce spurious sidebands.

The blocking mode of operation (sometimes called "idling") can often be employed in multi-channel operations over AFTS circuits to overcome slow fading, weak signals, or noise which may not permit operations in the twinning mode. Blocking is accomplished by pulling the individual modules of the tone channels desired blocked for the AN/UCC-1, 1A and 1B type units. The tone switch of the associated channels must be positioned to OFF for the AN/UCC-1C and 1D type units.

By halting transmission on unused channels, the transmitted power normally used in these channels is concentrated in the channels actually used. The number of channels blocked depends upon the conditions of the frequency or frequencies in use at the time.

CAUTION

> When blocking or idling is employed be sure to readjust the input levels to the transmitter to maintain proper peak output power. The transmitter output is directly related to the audio input power.

During periods of marginal communications, unused tones should be blocked to obtain the following advantages:

(1) More power per tone channel used can be transmitted without overloading the transmitter. This is only a slight increase in

*Adjustment of the tone levels of the AN/UCC-1 is complex enough to warrant careful attention to the equipment manuals.

## MULTI-CHANNEL RATT POWER LEVELS
(SSB TRANSMITTER POWER LEVELS)

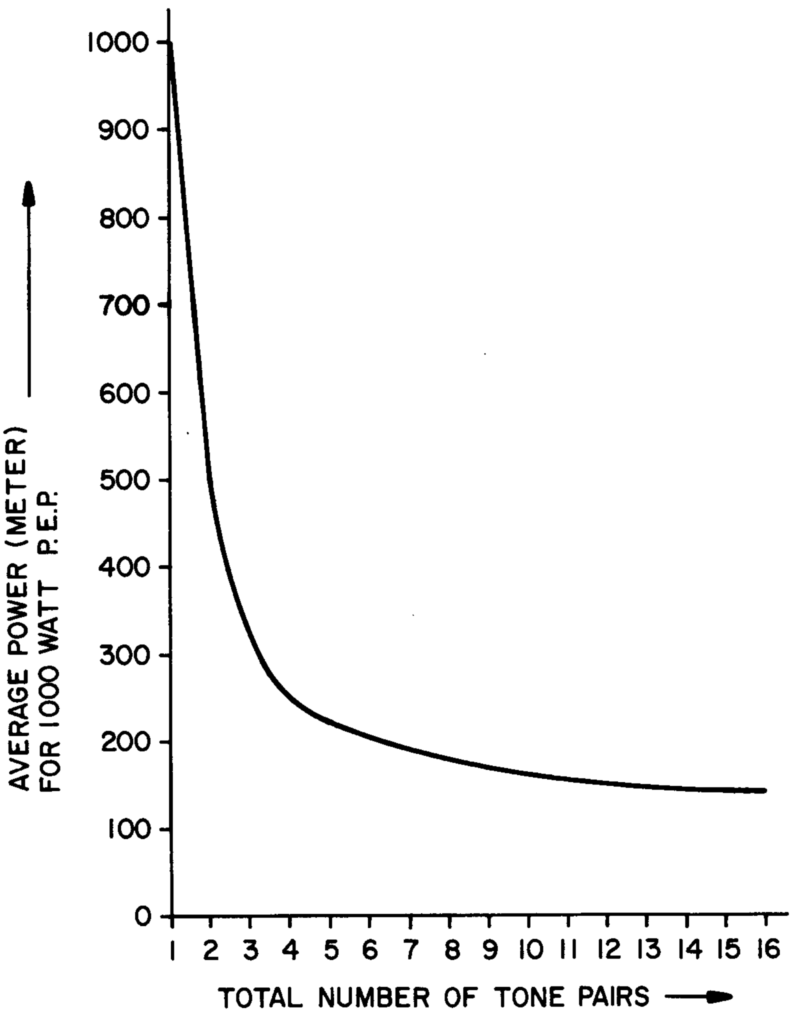



Figure 33. Multi-channel RATT Power Levels (SSB Transmitter Power Levels)

power, however, and care must be given to ensure that the transmitter is not overdriven to the point of distortion. If 16 tone channels were used and 8 were blocked, the average power would increase by 3 db. Readjust levels at the AN/UCC-1 and check the transmitter to ensure that it is not being overmodulated.

(2) With fewer tones being transmitted, the probability of shipboard-generated radio frequency interference is decreased. With fewer tones, the probability of intermodulation distortion significantly decreases the number of byproducts mathematically possible.

(3) Decreased bandwidth also lessens the chances of EMI on other ships using adjacent-channel operation. Blocking the higher audio frequencies from the tone package, in effect establishes a guard band between this tone package and the next higher channel frequency. This reduces interference throughout the Fleet.

(4) Should the front end of the associated receiver be driven into the nonlinear area of conduction, fewer tones in the tone package will result in the generation of fewer byproducts within the receiver. Therefore, tone blocking should be employed as a standard practice. Only the tones that are necessary to convey intelligence should be transmitted, and all other should be blocked.

## 2.7 UNCLASSIFIED CRYPTOGRAPHIC CONSIDERATIONS

Descriptions of individual crypto equipments are classified and are not included in this handbook. Suffice it to say that the automatic or "on-line" systems all depend on a random scrambling of the teletypewriter code pulses, initiated at certain time intervals. Timing and synchronization is the essence of the operation of these systems. Cryptographic devices are D. C. pulse operated, and must be inserted between teletypewriter equipment and FSK/MUX keyers; or between teletypewriters and receiving FSK/MUX converters. See Figure 34. The statements concerning telegraph signal distortion are of particular importance in the proper operation of cryptographic equipment.

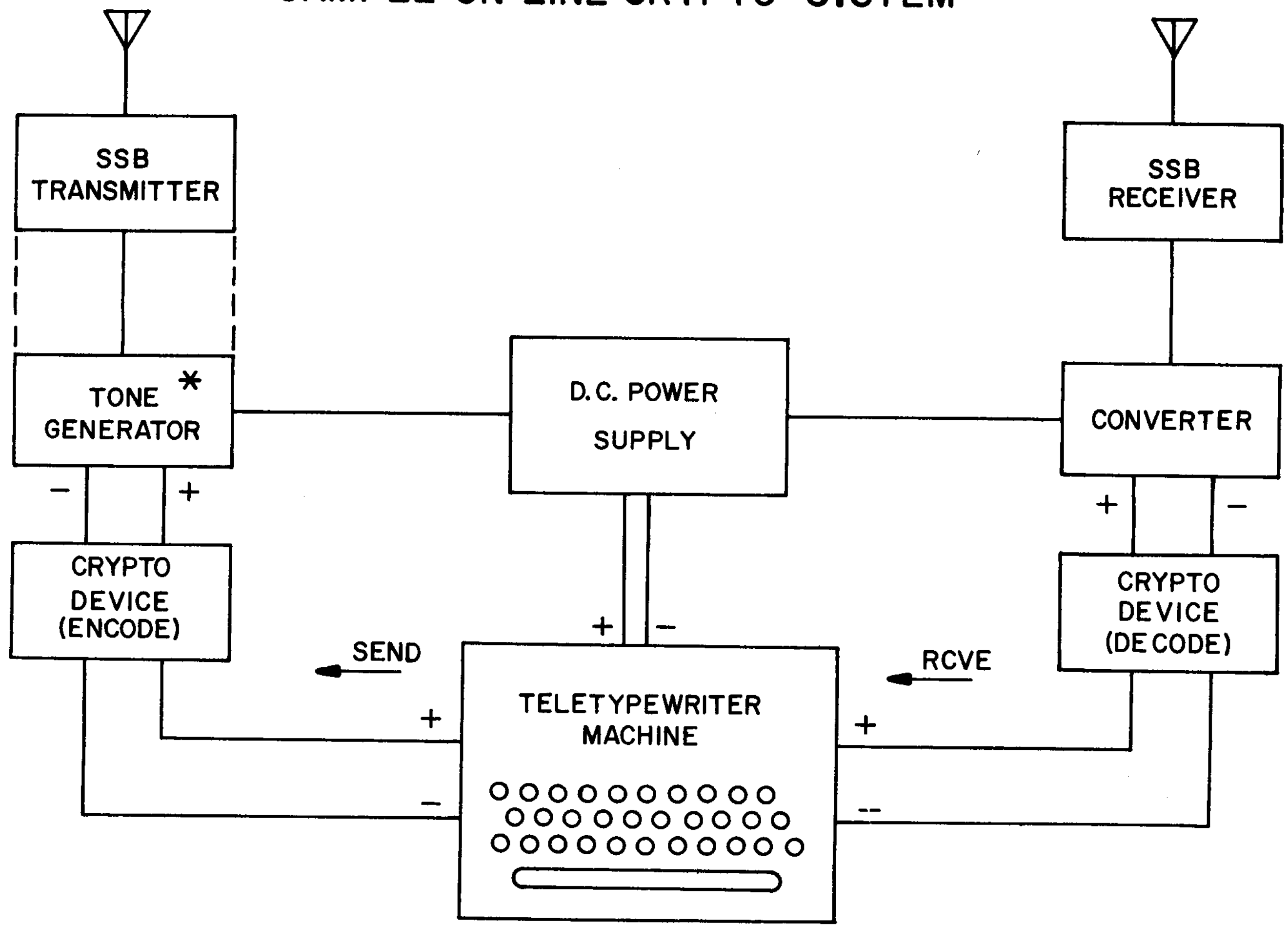
## 2.8 R. F. I. GENERATION DUE TO IMPROPER OPERATION OF SSB TRANSMITTERS

About 99% of all RFI problems associated with the operation of SSB/ISB equipment can be traced to the distortion products created by non-linear amplification. Since just about any amplifier can be operated in either a linear or nonlinear fashion (depending on its operating parameters), this reflects upon the operator of the equipment. In practical terms it goes back to the overdriving of the equipment which causes the RF amplifier to shift to the curved or nonlinear portion of its input/output characteristic graph. The design engineer cannot be at fault here because he intended the RF amplifiers in the transmitter to operate on the linear portion of the input/output graph for a certain range of input signal levels. When you exceed this range you have exceeded the design limits.

The careless operator produces RFI by a mere flick of the wrist, during transmitter adjustment. It is known that the output wave from a nonlinear amplifier will not be an exact replica of the input wave. That is, if one pure sine wave were injected into a non-linear amplifier the output wave might be a flattened sine wave, or one with jagged tops, or a lop-sided sine wave where the top is larger than the bottom. In terms of intelligibility of a voice signal this wave form distortion makes voices difficult to understand. A person will sound garbled or fuzzy even though received signal strength may be quite high. (From an efficiency standpoint, nonlinear amplifiers produce more power than linear ones for given power supply.) However a nonlinear amplifier generates several replicas of the voice signal on harmonic frequencies, making for less readable signals. You may recall reading in part 1 of this handbook that nonlinear amplifiers are used in frequency synthesizers for their high harmonic content. They can be very useful in that application as frequency multipliers. Any time you inject one sine wave into an amplifier and come out with a wave form that is more complex than the basic sine wave, you know that harmonics are contained in that output.

An additional harmful source of RFI is caused by overdriving a transmitter. (Suppose a 1 kHz tone is used to modulate a SSB transmitter (USB) whose carrier frequency is set to 3000 kHz.) Suppose also that the transmitter is over-driven into nonlinear operation. In addition to the expected sideband at 3001 kHz you would get, as a result of overdriving, harmonic sidebands at 6002 kHz, 9003 kHz, 12,004 kHz, etc. The amplitude of the harmonic sidebands are less than that of the desired sideband; (generally they are progressively smaller, the 3rd harmonic is weaker than 2nd, the 4th is weaker than the 3rd etc.); however, they can still be picked up by a sensitive receiver and may be strong enough to jam or block out a weaker signal from a more distant station. This is a very common RFI situation where the interference being generated may not be obvious to the station you are working,

## SAMPLE ON-LINE CRYPTO SYSTEM




* TONE GENERATOR IS GENERALLY PART OF TRANSMITTER

NOTES:
PATCH PANELS IN THIS R F AND DC LOOPS HAVE BEEN ELIMINATED IN ORDER TO SIMPLIFY THIS FIGURE

Figure 34. Sample On-line Crypto System

but it is blocking nearby receiving equipment which may be tuned to some integral multiple of your operating frequency.

Thus, spurious signals of this kind from your ships transmitters can effectively jam your own ship's receivers. Even though the harmonics are much weaker in power than the desired emissions, they are still more powerful locally than signals being received from long distances.

This situation applies to a single tone input to an overdriven SSB transmitter. Things can get much worse when a multi-tone intelligence such as a multi-channel RATT or a voice (represented by many sine waves) signal is overmodulated. The reason for this is due to another distortion effect in nonlinear amplifiers called intermodulation (I. M.) distortion. This effect always occurs any time two or more waves are present at the input of a nonlinear device. It is more serious than harmonic distortion because the interference is closer in frequency to the desired signal. Intermodulation distortion can also appear in the form of spurious signals below the operating frequency. Figure 35 gives an example of this type of interference.

In this illustration two test tones of equal amplitude were injected into the exciter modulator of a SSB transmitter: one tone at 3 kHz and the other at 5 kHz. The transmitter is set up for USB operation at a carrier setting of 2997 kHz. If everything were adjusted properly (for linear operation) all that would be emitted by the transmitter would be a pair of sidebands at 3000 kHz and 3002 kHz. But here, in the nonlinear case, we see a number of other sidebands appearing on both sides of the (suppressed) carrier frequency. These additional sidebands that don't belong are called intermodulation distortion products. In this example, the two desired sidebands interact in the nonlinear device in every possible combination. For example, the first I. M. product will be at two times the frequency of one sideband minus the frequency of the other sideband. The next I. M. products is three times the frequency of one sideband minus two times the frequency of the opposite sideband. Theoretically this goes on forever: $2F_2-F_1$, $2F_1-F_2$, $3F_1-2F_2$, $4F_2-3F$, etc. Practically speaking, only the first few I. M. products are high enough intensity to be troublesome. They are also the ones that are too near the desired sidebands to be completely eliminated by filters or tuned circuits in either the SSB transmitter or receiver. Intermodulation products stem from three principal sources aboard ships (1) transmitter amplifier non-linearities, (2) receiver intermodulation, and (3) intermodulation produced in a non-linear element of the antenna environment; i. e. topside rigging and structure of the ship. ("Rusty Bolt" phenomena.) Items recently identified as problems in the third category include poor contacts in antennas, rigging, stays, handrails and lifelines, stanchions, ladders, booms and hoisting cables, boat davits, cable hangers, corroded cable armor, door hinges, gratings, etc.

Consider the vast numbers of intermodulation products that would result in passing the 16 tone pair signal from a multiplexed exciter through a non-linear amplifier. You now can see the magnitude of the RF problems created by carelessly operating SSB equipment. This RFI problem is severe when both USB and LSB equipment is operating on adjacent frequencies on the same ship. As seen from Figure 34, the I. M. products can cross over to the opposite sideband. The amount of attenuation or isolation depends mainly on the distances between antennas and to some extent on frequency. The problem has been more noticeable now than in days prior to the advent of SSB in the Navy because transmitter power has increased at the same time that receiver sensitivity has improved.

Recently vacuum tube and transistor amplifier designs have been invented which should reduce the amount of I. M. distortion when used under proper conditions. Figure 36 shows a graph of approximate amount of I. M. distortion produced in a typical power tube in use today. Note that using conservative power levels is still the easiest means available for achieving minimum distortion in RF amplifiers. Overdriving even the best designed power tube can create a high level of I. M. distortion. The lesson should be clear:

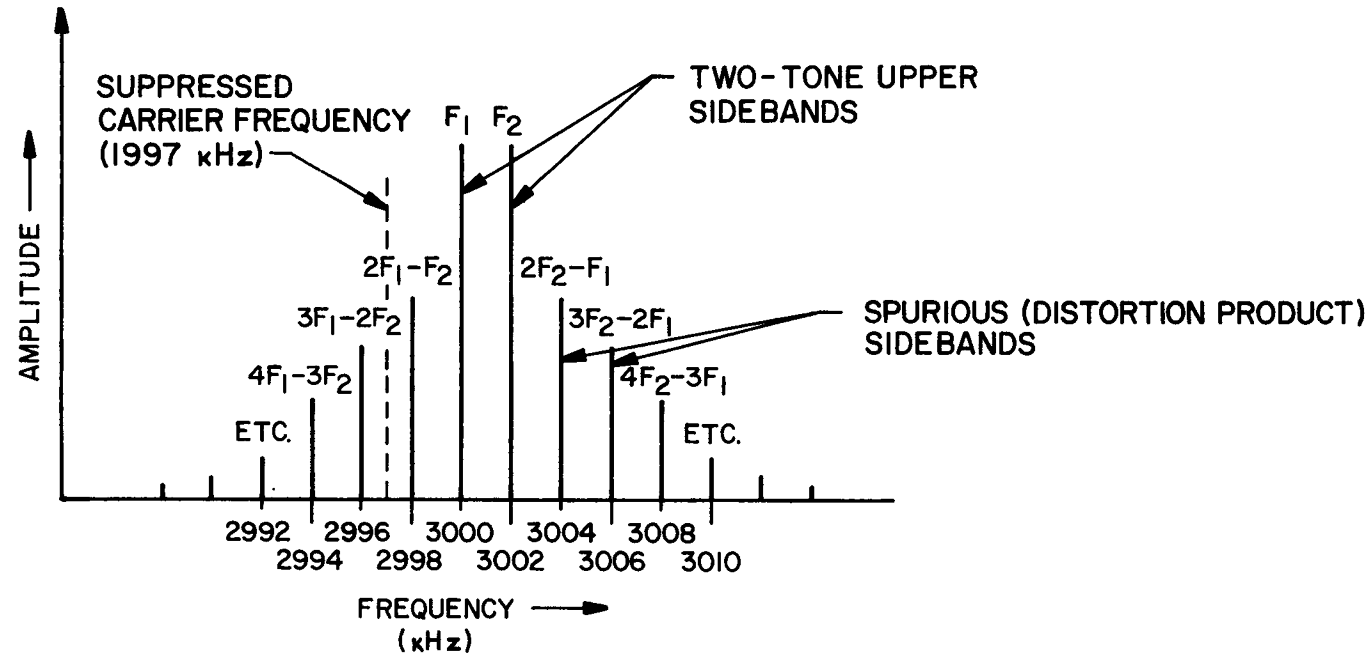
WATCH MODULATION LEVEL SETTINGS
IN ORDER TO AVOID RFI PROBLEMS.

Rely on correct meter settings as discussed here and in equipment manuals. Also check for harmonics and intermodulation products with whatever test equipment is available. A standby receiver is adequate for this purpose. A spectrum analyzer is even better if you can obtain one.

In view of the RFI possible with sophisticated SSB systems several additional practical operating hints are given here:

(1) Use the narrowest receiver bandwidth compatible with desired emission requirements.
(2) Use all receiving filters and multicouplers available in order to obtain maximum selectivity and minimum receiver front end overloading.
(3) Select receiving frequencies well separated from local transmitting frequencies--if a choice is available.
(4) Reduce, if possible, number of transmitters in operation simultaneously.

## INTERMODULATION DISTORTION (NON-LINEAR AMPLIFIER)




NOTE:

EXAMPLE SHOWN IS SSB XMTTR IN UPPER SIDEBAND, MODULATED BY 3 κHz AND 5 κHz TONES. THE SPACING BETWEEN EACH ADJACENT PAIR OF SIDEBANDS IS EQUAL TO THE SPACING BETWEEN THE DESIRED TWO-TONE SIDEBANDS.

Figure 35. Intermodulation Distortion (Non-linear Amplifier)

# INTERMODULATION CURVE FOR TYPICAL POWER TUBES

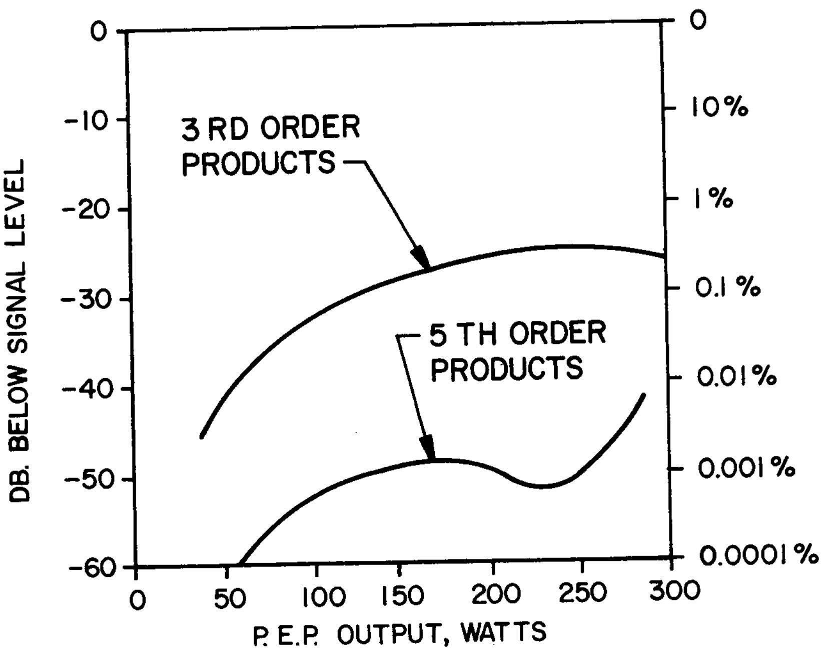



Figure 36. Intermodulation Curve for Typical Power Tubes

(5) Use transmitter power no higher than minimum required to accomplish assigned communications mission.
(6) Be sure that all connection in both receiving and transmitting antenna systems are tight and in good condition. Be careful about switches, patch cords, cable connectors, shielding braids on coaxial cables, and the antenna conductors themselves. Keep insulators clean.

## 2.9 OPERATING PROCEDURES, TESTS AND RECOMMENDATIONS

### A. UHF/HF Relay Techniques

Increased aircraft carrier operations have required that the principal communications duties of destroyers include providing the carriers with UHF/HF relays for sustained periods of time.

In the past, only certain destroyers, together with a few other ships, have worked out the details of VOX retransmission relay, and these particular ships were continually requested to perform such relay duties. So many new arrivals to WESTPAC in recent years lacked the ability to conduct a successful relay operation that the Fleet Commander called for command attention to this problem and insisted on acceptable performance from all ships. Henceforth, each destroyer must not only demonstrate a relay capability commensurate with its existing transmitter installation, but a proficiency in relay operation as well. Carriers have been requested to rotate the relay assignment among the destroyers attached to them and to report inabilities on the part of any of them. A 2100 class destroyer is incapable of providing the same degrees of relay services as the DLG class because of disparity in equipment. However, this does not excuse the 2100 class destroyer from demonstrating a proficiency in each of the modes of communications for which the ship is equipped.

Each of the following modes of relay should be worked out:

(1) Single-channel VOX relay by conversion of UHF AM voice or tone-modulated RATT signals to a HF AM voice or tone modulated teletype signal using the C-4621/SR unit.
(2) Double-channel simultaneous VOX relay by conversion of two UHF AM voice or tone-modulated RATT signals to a HF SSB voice or tone-modulated RATT signal using the C-4621/SR unit.
(3) Single-channel VOX relay operation on single-sideband by conversion of a UHF AM voice or tone-modulated RATT signal to an HF SSB voice or tone-modulated RATT signal using the C-4621/SR unit.
(4) Independent sideband two-channel VOX relay operation by conversion of one AM voice UHF signal and one tone-modulated RATT signal to one HF upper-sideband voice and one HF lower-sideband tone-modulated RATT using the C-4621/SR unit.
(5) Composite (2 or 3 channel) MUX relay operation by conversion of UHF composite MUX signal to an HF composite MUX signal on a sideband using the C-4621/SR unit.
(6) Single-channel, frequency shift teletype, relay operation by conversion of a UHF tone-modulated RATT signal to an HF frequency shift RATT signal using the AN/SGC-1A.
(7) Diversity frequency shift teletype relay consists of converting a single UHF tone-modulated RATT signal to one HF frequency shift RATT and one MF frequency shift RATT signal using the AN/SGC-1A.
(8) Use of AN/SRC-20, 21 internal retransmission capabilities.

MUX relay in the frequency shift mode is difficult to manage due to the slow response of the mechanical relay in the AN/SGC-1A. An electronic relay replacement for the mechanical relay is under consideration.

A common and erroneous belief is that the VOX (C-4621/SR) unit can be used to drive a frequency shift keyer. The VOX unit can handle only audio signals and drive a modulator. The AN/SGC-1A teletypewriter tone modulator is required to convert the tone-modulated audio signal into the mark and space bits to key the DC loop of a frequency shift keyer. Upon learning that the AN/SGC-1A is necessary to convert a tone-modulated signal to frequency shift, many ships find that they are unable to isolate the transmitter and the AN/SGC-1A in the DC loop patch panel. Both the frequency shift keyer in the transmitter and the tone-modulator have their own DC loops. Teletype systems will not operate correctly with two separate power sources. There are several solutions to this problem but the one in COMFIRSTFLT TACNOTE 1-63, which suggests isolating the AN/SGC-1 in the "miscellaneous" jack, is probably the simplest and best. Those ships that still have the TT-23/SG DC patch panel, with its versatile feature of the toggle switches, do not have this problem.

The failure to modify the C-4621/SR unit to permit duplex operation as authorized by the field change 1-C-4621/SR limits the operational capacity of some ships.

After receiving requests to relay a two- or three-channel MUX signal, several destroyers have originated messages to the TYCOM requesting that MUX equipment be installed at the earliest possible time so that this service might be rendered. MUX relay operation does not require MUX equipment. The MUX "tone package" can be relayed without DEMUXing. Successful UHF/HF relays of multiplex signals can be achieved using the following procedures:

(1) Inactivate UHF transmitter clipping circuits.

(2) Upon receipt of a condition-one voice message on the COMM COORD NET, perform the following:

a. Tune the UHF receiver to the frequency designated

b. Tune the AN/WRT-2 or AN/URT-23 (HF transmitter) (the AN/URC-32 is not considered acceptable for this relay operation) by tuning the transmitter as usual to the frequency designated in the condition message and by making the following adjustments:

1) Set the emission selector to USB
2) Set the carrier reinsert to minus 10 db (or as indicated by condition message).
3) Set the sideband selector to the sideband indicated in the message.
4) Set the Local/Remote switch to Remote.

c. Patch the UHF receiver to the transmitter control (C-4621/SR).

d. Patch the HF transmitter to the transmitter control (C-4621/SR).

e. Send condition-two message.

(3) Upon receipt of condition-three voice message make the following adjustments:

a. When tones are heard in the UHF receiver, adjust the audio output to plus 2.5 db. (This level must be maintained).

b. Adjust the transmitter modulation level on transmitter control (C-4621) to zero db on the level meter.

c. At the HF transmitter, lock on the carrier test key and adjust the audio output of the previously designated sideband on the meter.

(4) The AN/WRT-2 should now indicate approximately 120 watts on the output power meter.

Periodic checks and adjustments of the UHF receiver audio output are necessary to compensate for the effects on signal strength and distortion when changing course, speed, and distance from the ship or station of signal origin.

B. Frequency Selection Procedures for Broadcast Reception of Ship/Shore Communications

Broadcast reception, either multichannel, single channel or CW, can be improved in most cases by a judicious choice of available frequencies. Most broadcasts provide a choice of transmitted frequencies but many ships do not always select the best frequency available for a particular communications situation. The following suggestions may assist in increasing the quality of broadcast reception:

(1) List the broadcast frequencies that are to be received and the geographical location of the transmitters. (Some broadcasts are keyed on transmitters in several different locations.) Knowing the location of the transmitter is necessary to properly select the correct frequency to be copied.

(2) Tune receivers properly on several frequencies and listen for signal distortion or interference and check the strength of the signal on the input meter of the receiver. If available, try various antennas on each frequency to obtain the best reception.

(3) If the ship is 500 miles or more from the transmitter during daylight hours, try the 12 MHz, 17 MHz, and higher frequencies. The radio frequency spectrum is less crowded in the higher frequency region and hence interference from adjacent channel stations or own ship transmitters is less likely in this portion of the spectrum. Also, the signal path attenuation will be less from the transmitter to the ship on the higher frequencies.

(4) If diversity reception is to be utilized, use transmitters from the same location and frequencies in adjacent bands such as 12 MHz and 17 MHz or 8 MHz and 12 MHz for best diversity reception.

(5) During the night hours, lower frequencies will normally be used. Use the lower frequencies as soon as reception is good and no fading is noted. Do not wait until the higher frequencies have deteriorated before sampling, tuning in, and placing at least one of the lower frequency receiver in standby.

(6) Be extremely careful in setting proper levels to terminal equipments such as the AN/UCC-1, AN/URA-8, or AN/URA-17. Follow the latest information from the manufacturer's technical manual, all

applicable changes indicated in Electronic Information Bulletins (EIB's), and any other publication pertinent to establishing proper receiver output levels.

(7) Use the narrowest receiver bandwidth adjustment permitted by the bandwidth of the received signal. This will not only reduce the chance of adjacent channel interference but will also reduce the noise level riding in on the received signal. Selecting frequencies for ship/shore communications is comparatively easy if a broadcast is being copied. To determine whether the ship will transmit on CW, or ship/shore ORESTES, simply request the broadcast band of interest. For example, if the CW ship/shore circuit is to be used and reception is best on the 12 MHz broadcast, then the 12 MHz frequency is the one that should be selected.

Tune the receiver carefully for the reception of any signal that is to be copied by the ship. A little time in tuning the receiver will pay high dividends in reliable communications. Selecting frequencies for the shore-transmit side of terminations is roughly the same as for the selection of ORESTES covered net or CW frequencies. Sample the broadcast frequencies on the air and determine which bands are received best. Observe the 12/17/20 MHz frequencies during daylight hours. Select the best band of frequencies and choose a frequency from JANAP 195 ( ) in the same band which is cleared for use by the shore station for the mode of emission to be used. Have assigned frequencies posted at the receiver for quick checks to determine if they are available and acceptable in the event communications become unacceptable on the frequencies being employed. Whenever a choice of frequencies is available make the selection as follows:

a. Choose one frequency that provides good communications and a second frequency, in the same general band, as a backup frequency. This provides the same propagation and time path from the transmitter to the receiver. During periods of propagation transition, utilize frequencies in two bands rather than one. Frequencies should be decreased in steps as nighttime approaches and increased in steps as daylight hours arrive. Once into either complete nighttime operation or daytime operation, however, it will again be necessary to utilize two frequencies in the same band.
b. Try different antennas for each frequency before rejecting it as too weak or inadequate. Try a different RF patch cord before rejecting a frequency as being unusable.

The use of higher frequencies during daylight hours minimizes the losses through the "D" layer of the ionosphere. This layer is ionized only during the daylight hours and attenuates the lower frequencies more than the higher frequencies. In addition, when lower frequencies (2 to 6 MHz) are used during daylight hours, they may follow a double-hop path and hence be attenuated by the "D" layer of the ionosphere four times prior to reception on the ship. The higher frequencies follow a one-hop path and hence only pass through the "D" layer twice. Consequently they will be attenuated far less than the lower frequencies. In many cases the lower frequencies can be detected by the ship, but the signal strength is such that it cannot override noise and local shipboard interference.

ALCOM 79* contains guidance on channelization and frequencies for termination of multichannel ship/shore terminal equipment. It should be read and observed by all personnel concerned with multichannel terminations.

C. Tone-Modulated Teletypewriter Operating Procedures

The AN/SGC type of tone-modulated RATT circuit is a reliable type of circuit which is ideal for ORESTES-type nets within a task group or task unit. For this reason the operators and technicians should understand the use of the equipment to obtain the maximum capability of the equipment at all times. There is no better way to determine the capability of an equipment than by actual operation. This can be easily done between two ships in port, or at sea, having an authorized UHF frequency and operators who wish to become more proficient.

The AN/SGC equipment is a DC-to-audio converter in the transmit mode. It receives DC pulses from the teletypewriter and converts these DC pulses into audio frequencies for further transmission by a radio transmitter. The audio tone for a UHF teletype mark is 700 Hz and the space frequency is 500 Hz. In the receive mode, the equipment is an audio-to-DC converter. The audio from the receiver is fed into the AN/SGC equipment, and when a 700-Hz tone is received, the equipment converts the tone to a mark pulse. When a 500-Hz tone is received, it is converted into a space. The equipment has a mode selector switch that can be used to place the equipment into constant transmit or receive conditions at the XMIT or RCVE positions. This switch also has an AUTO position that permits the equipment to switch to receive whenever a signal is fed from

* ALCOM 79/68 is a message to all commands standardizing the channel frequencies for all multichannel ship shore work.

the receiver or to transmit whenever the keyboard of the local teletypewriter is operated. The operating options of these equipments make them ideal for communications net operations.

The equipment has three basic signal inputs/outputs. The DC portion of the machine is for operation with a teletypewriter. In addition, there is an audio input for receiver output tones applied to the AN/SGC equipment. There is also an audio-output which enables the AN/SGC to furnish tone modulation to a transmitter.

With two exceptions, the equipment is very simple to use. The first exception concerns adjustment of the receiver that is used to receive RF signals from another ship.

If the output of the receiver is noisy during the time a signal is not being received, the noise can trigger the AN/SGC equipment into the receive condition, and the equipment will not automatically go to the transmit mode when the local teletypewriter keyboard is operated. The output of the receiver must be squelched to a very low level when a signal is not being received. This level should be employed to provide tone output levels of approximately 0 dbm. Noise from the receiver must not trigger the AN/SGC into a receive condition when tones are not being received.

The second exception to equipment simplicity of operation results from the modulation level that has been established on the ship. If a radiophone remote having an adjustable output is set to furnish -10 dbm into the speech amplifier of a transmitter, and the output of the AN/SGC equipment is set to furnish 0 dbm to the transmitter, overmodulation will occur and result in telegraph signal distortion. If the modulation system for radiophone remotes has been set at plus 10 dbm and the output of the AN/SGC unit at 0 dbm, undermodulation will occur and the range for reliable communications between ships will be decreased. The audio output from the AN/SGC equipment should be adjusted to furnish the same level of audio to the audio transmitters that the remote radiophone units furnish.

D. Receiver RF Amplifier Section Overload Check

Receiver-processed output signals are frequently distorted due to a combination of the high level of the received signals and excessive RF gain control settings. This distortion is known as receiver overload or saturation and occurs when RF signals are sufficiently high to cause the RF amplifier stages to operate in the nonlinear regions of their plate current characteristic curves.

The following procedure is a simple test to determine if any receiver overload is occuring.

(1) Adjust the receiver for normal reception of the desired signal and note the indicated db level on the RF input meter.
(2) Insert an attenuator of known value in the receiver transmission line and note the drop in signal level. If the db level is reduced precisely by the value of the attenuator, the receiver is experiencing no overload. If the level is reduced by an amount either less or greater than the value of the attenuator the receiver was operating in an overloaded condition.
(3) In the event an overload condition is detected, repeat the test with reduced RF gain settings until a linear relationship exists between the value of the attenuator inserted and the drop in the indicated db level. For example a 10 db attenuator should provide a 10 db reduction in signal level.

E. Receiver Audio Output Signal Level Adjustment

Audio intermodulation (crosstalk) will be produced in other circuits located in proximity to leads and cables carrying high level audio signals. To minimize crosstalk, the following procedures should be employed to ensure that correct receiver audio output signal levels are applied to remote audio units:

(1) Employing AGC, adjust the audio gain control to an average or midrange position.
(2) Adjust the tuning control for a clear or unused frequency close to the desired operating frequency.
(3) Adjust the RF gain control until the ambient noise level is clearly audible.
(4) Tune to the desired signal and adjust the audio, or volume control for a maximum indication of 0 db on the audio level meter.

F. Antenna Selection Procedures

Frequently a choice of antennas can be made from either the transmitting or receiving antenna groups or both. The selections made should offer maximum separation between the receiving and the transmitting antennas of systems likely to operate simultaneously. Frequency assignments should also be considered during antenna selections. The transmitter and receiver systems employing the smallest frequency separation should be assigned the maximum distance separation.

G. Antenna Tuner Adjustment

Automatic antenna tuning or coupling devices have been noted to tune to incorrect frequencies.

Couplers, such as Antenna Coupler Group AN/URA-38, will on occasion tune to exciter unit spurious output signals or to signals from other transmitter systems located in proximity. Operators of transmitter systems incorporating automatically tuned antenna couplers must ensure that the coupling devices are adjusted in accordance with respective technical manuals prior to each transmitting operation.

## H. Single-Channel Orestes Net Operating Procedure Discrepancies

Many of the difficulties encountered on ORESTES nets (AFTS RATT) can be attributed to off-frequency situations between ships on the net. A typical teletype signal received aboard ship may already have as much as 15 percent bias distortion present. A frequency deviation from the assigned or transmitted frequency also causes bias distortion. A combination of the two distortions limits the allowable frequency difference between the transmitter and receiver.

Figure 5-5 shows the average and peak bias distortion that is generated as the frequency increases from the center or assigned frequency. Note that the amount of average bias distortion increases rapidly near the 500 Hz figure. In this particular case the signal was generated with 13 percent average bias distortion. As the slope of the distortion curve begins to rise rapidly, the peak distortion rises in a similar manner. The indications at the receiver site are manifested by garbled copy.

Absolute frequency control must be practiced by operator and maintenance personnel. Since equipments can be operated in the continuous mode, it is common practice to operate in this manner.

Many ships have AN/URQ-9 or AN/URQ-10 frequency standards installed which are improperly utilized. Transmitting sets AN/URC-32, AN/URT-23, 24, AN/WRC-1, and AN/WRT-2 and Receivers R-1051/URR and AN/WRR-2 are designed for specialized high-accuracy frequency control and are capable of high-accuracy control if operated and maintained properly and connected to frequency standard AN/URQ-9 and AN/URQ-10 through the AM-2123/U distribution system. Unless these transmitters are operated in their stable (synthesized) modes, their dial accuracy may be so far in error (up to 7 kHz) that establishing certain RATT nets may be impossible.

Continuing reports from various levels of command indicate that operating personnel often display a disregard for standard TSEC/KW-7 circuit operating procedures. Some of the most common violations of proper operating procedures are as follows:

(1) Failure to ensure that the net is clear prior to transmitting. By this omission, the effectiveness of any ORESTES circuit is drastically reduced.
(2) Tuning the transmitter while on the air. This also serves to disrupt the entire circuit. For years standard procedures have been in effect prescribing proper procedures for tuning transmitters. Operating personnel are encouraged to strictly adhere to these procedures.
(3) Holding down the "sync." button for excessive periods of time. Even though there is a visual indication of TSEC/KW-7, synchronization and 5 seconds of "sync." time should be adequate, some operators hold down the "sync." button for 10 to 15 seconds. This results in much lost net operating time if it becomes a typical practice.
(4) Failure to remove transmitters from the air after completion of transmissions. The effect of this type of error is obvious. No one else can use the circuit until the transmitter is unkeyed.

## I. Frequency Accuracy Requirements

Equipment used for fleet communications has slowly evolved over the past years so as to increase fleet capabilities and performance. The new equipment, however, has also introduced new problems. For example, in SSB communication equipments, frequency errors of 0.02 percent, the legal limitation imposed by the Atlantic City Convention, can no longer be tolerated. Single-sideband voice requires frequency accuracies of 0.000010 percent. This accuracy translates into plus or minus 1 Hz at 10 MHz. With the advent of multichannel RATT, the requirement for frequency accuracy has increased by another order of magnitude to plus or minus one-tenth of 1 Hz at 10 MHz. Circuit performance is therefore highly susceptible to frequency errors. Frequency discrepancies result in severe degradation in message quality. Regardless of degradation, however, SSB remains the better mode for voice frequency transmission.

A SSB circuit and an AM circuit were compared in a voice intelligibility test (Fairbanks Rime) which is commonly used in the telephone circuit analysis. The test depended upon the number of words which after being transmitted over the circuit were confused with words similar except for one critical phonetic sound. From the error rate an articulation index was derived. A good telephone circuit maintains an articulation in excess of 90 percent. The SSB circuit maintained an index of 35 percent and the AM voice circuit maintained an index of 46 percent. The significance here is that if the stations on a single-sideband

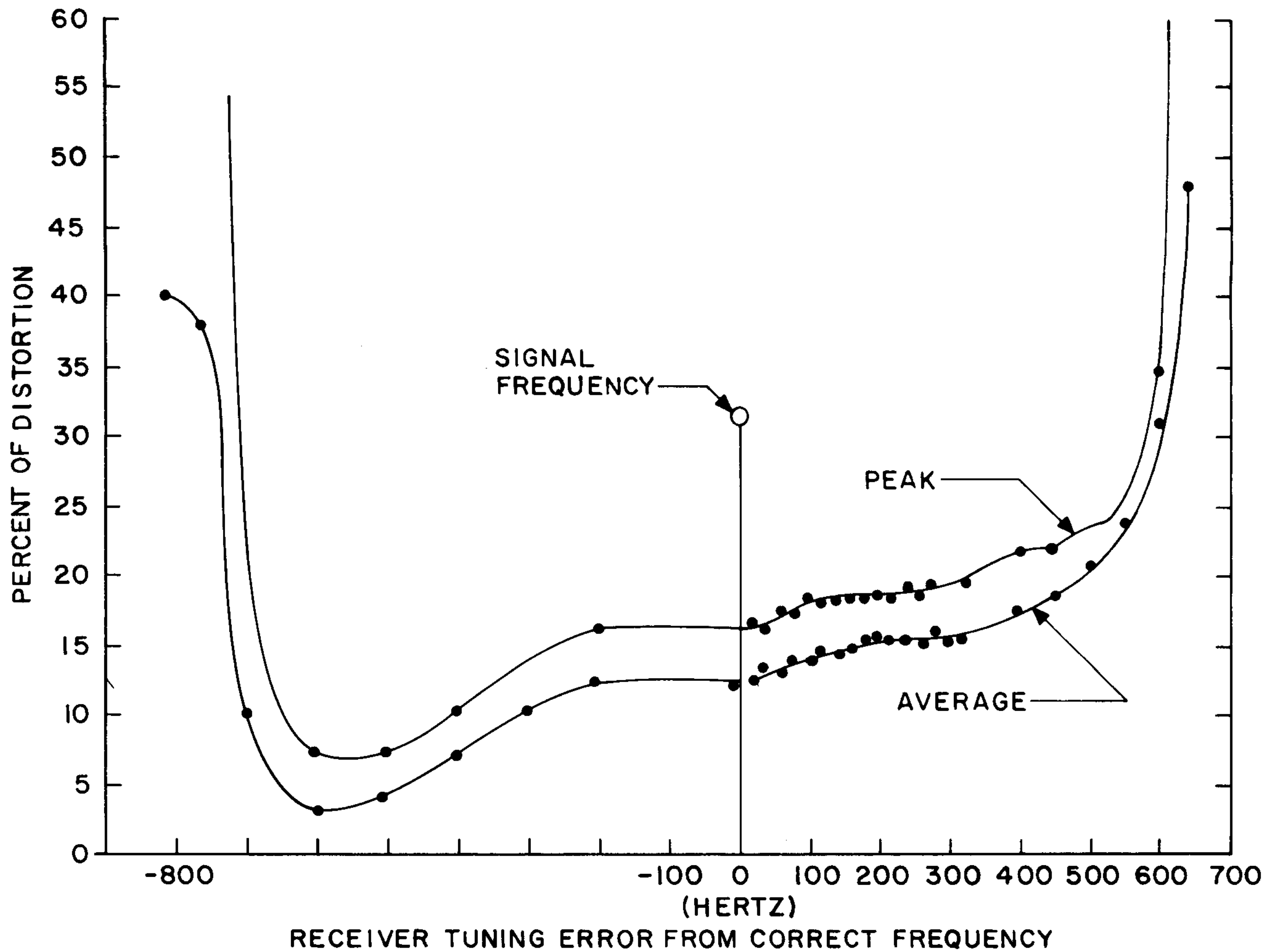



Figure 37. Radio Teletypewriter Bias Distortion Versus Receiver Tuning Error

net are not within plus or minus 10 Hz of the correct frequency, performance is penalized. This penalty results in overall degradation of the circuit which reduces the margin for any degradation which may be contributed by personnel or equipment. The above referenced articulation indexes may be translated to the percentage of sentences that could be expected to be understood. At 46 percent articulation, 98 percent of transmitted sentences would be received correctly. When the articulation index drops below 25 percent, the error rate for sentences rises sharply. In other words, the SSB circuit had only a 10 percent margin before it reached the critical point, as opposed to the 21 percent margin for the AM circuit. Most of this was due to off-frequency operation of some of the SSB net stations.

On multichannel RATT each of the first ±5 Hz of frequency error translates into a 1 percent bias distortion, each of the next ±5 Hz contributes 2 percent distortion; by the time the signal is 20 Hz off frequency, the distortion would render the circuit unusable even with a nearly perfect signal. The average bias distortion of the RATT circuits was measured. It was found that when average bias distortion reached a level of approximately 25 percent, the peaks of this distortion exceeded 40 percent and the distortion was manifested in printing errors. Additionally, it was found that bias distortion on what could be considered good RATT circuits was averaging between 10 percent and 15 percent. This means that the margin for additional distortion on a good circuit is only about 10 percent. Based on a 10 percent allowable margin of distortion, it would appear that there is a tolerance of plus or minus 8 Hz for frequency error. However, this tolerance must be shared between four pieces of equipment; a transmitter, receiver, and the transmit and receive multiplex terminal equipments. By the time the multiplex terminal equipments take their allocated share of this tolerance, only 4 Hz remain to be shared between the transmitter and receiver.

To compensate for frequency errors of shipboard transmitters, an alternate method is used whereby the ship transmits a pilot frequency and the receivers ashore use automatic frequency control receivers to lock onto their signal. This method is acceptable except that the circuit becomes more susceptible to outage in the presence of poor propagation. This is caused by the necessity for the transmitter to use a part of its available power for generation of the carrier frequency. Additionally, the advantage that could be gained by the capabilities of the equipment to use frequency diversity, for selective fading propagation conditions, is degraded due to the susceptibility of the receivers (that use automatic frequency control) to this same selective fading. Further, this method permits enemy countermeasures to disrupt communications more easily.

The frequency tolerance of other types of communication circuits is not as critical as the ones mentioned so far. Unfortunately, the frequency errors observed on these circuits during tests were proportionally greater. On the SSSC amplitude modulated voice circuit there seems to be always at least one station which had an error of 2 kHz. Of all the stations on the circuit, only 70 percent were observed to be within 500 Hz of the assigned frequency.

On the itinerant ORESTES circuits, the range of frequency errors was similar to the SSSC net. The effect is that unless the receiving station had retuned its receiver for those transmissions that were in excess of 500 Hz off frequency, the transmission would have been missed. The amount of frequency shift was also in error. This means that the receiving station would also have had to readjust its converter for many of the transmissions.

There is, therefore, a problem in that frequency errors degrade communication circuits. The old procedures of zero-beating to the net control station are no longer applicable with single-sideband transmitters. Most synthesized transmitters are not capable of continuous tuning. Also, the AN/WRT-2, if operated in continuous tune mode, would cause the inherent synthesized capability of the equipment to be wasted.

Precise determination of the correct frequency throughout the Navy and the entire Military Establishment is accomplished by the U.S. Naval Observatory. This correct frequency is provided each ship from reference oscillators AN/URQ-9 or AN/URQ-10, which are corrected by Navy calibration laboratories on a periodic basis. WWV does not provide ships the capability of determining the correct frequency to the tolerance required. The reference oscillator is the only method available at present to determine the correct alignment of the internal oscillators of each piece of equipment aboard ships. There is hardly a piece of electronic equipment aboard ships that does not depend upon an internal reference oscillator for its accuracy and performance. A technical manual may assign some superlative figure to the frequency stability of an equipment, but this does not guarantee the equipment's frequency accuracy. The frequency stability only states the limit as to how quickly that oscillator will drift off frequency, and oscillators inherently tend to drift off frequency. For example, of the AN/WRT-2 transmitters inspected aboard 40 percent of the ships participating in one exercise, 87.5 percent of the transmitters were incapable of operating with the equipment frequency tolerance because their internal reference oscillators were out of frequency tolerance. Some of these errors would have resulted in transmitted frequency errors of several hundred Hz.

Errors in the internal oscillators of equipment would not account for frequency errors in the kHz range. Gross errors such as these could only be caused by misalignment of transmitter frequency generation circuits or by mistuning of transmitters when used in their continuous tune mode of operation. As an example one AN/WRT-2 transmitter measured was noted to have a 7 kHz error between the transmitter frequency output and the dial indication.

There is no reason why each ship cannot set each of their transmitters to plus or minus 1 Hz. Weekly checks of the internal reference oscillators of all single-sideband equipment by means of AN/URQ-9 or AN/URQ-10 frequency standard would ensure these accuracies. Figure 5-6 is a block diagram illustrating a typical test setup for transmitter frequency output calibration.

Frequency standards AN/URQ-9 and AN/URQ-10 provided to ships are intended for installed system use and not as a "test equipment" (see EIB 712). The AN/URQ-9 and AN/URQ-10 provide a much higher order of frequency accuracy than that provided by frequency standards included within most new equipments. The frequency standards do not measure frequency, they provide highly stable and accurate reference signals against which other signal sources such as the Translator-Synthesizer unit of Radio Receiver R-1051/URR may be compared. When an AN/URQ-9 or AN/URQ-10 is provided to a ship, it should be permanently installed as a central system (see Figure 5-7).

Equipments such as the AN/WRC-1, AN/URC-35, AN/WRR-2, AN/URT-23, AN/URT-24, AN/SRR-19, and R-1051/URR are examples of communication equipments having the capability of using either an internal or external frequency standard. The internal frequency standard unit of these equipments is intended for use in installations not having an AN/URQ-9 or AN/URQ-10 installed or for backup use in the event of any failure of the installed external frequency standard system. Where there is no installed central reference system, the equipment's internal frequency standard must be calibrated periodically against an external frequency source, such as a portable AN/URQ-10, to maintain their rated accuracy. Frequency standards "age" with time, reduction in frequency accuracy. It is also desirable to recalibrate the AN/URQ-9, 10 frequency standard itself at a Navy calibration laboratory each 6 months.

It is essential that equipments such as the R-1051/URR and AN/URT-23, when used with multiplex equipment AN/UCC-1, use an installed ships frequency standard system in preference to the equipment's internal standard for maximum circuit reliability.

The basic frequency standard system as shown in Figure 5-7 utilizes RF Amplifier AM-2123/U for isolation and distribution of 0.1, 1, and 5 MHz frequencies generated by the AN/URQ-9 or AN/URQ-10. This RF amplifier must be used when more than one transmitter or receiver is to be connected to the frequency standard. The AM-2123/U accepts the three input frequencies from the frequency standard and provides 12 isolated outputs in any combination of the three input frequencies.

When installing the AM-2123/U, an appropriate amplifier plug-in module must be installed for each individual equipment for which it is to be used; i.e. 5 MHz modules for the AN/WRC-1, AN/URC-35, AN/URT-23, AN/URT-24, and R-1051/URR; 1 MHz module for the AN/SRR-19 and 0.1 MHz for AN/URC-32, AN/WRR-2. The AN/WRC-1 equipment requires two inputs, one for the receiver and one for the transmitter. If additional modules are required to change the frequency complement of the amplifier, they may be ordered on an exchange basis from Electronic Supply Office (ESO).

NAVSEC drawing RE-F2687915 provides ship Frequency Standard System installation information. The appropriate communication equipment technical manual should be consulted regarding use of the external standard as a comparison reference in the calibration of the equipment's internal frequency standard. When calibrating an equipment's internal frequency standard, it is essential that the standard be previously continuously energized for an extended period (not less than two weeks preferred).

It is essential that the communication equipments be run continuously (operate or in "standby" condition) since the on-off cycling of the equipments' prime power will cause a degradation of the equipment's internal frequency standard unit.

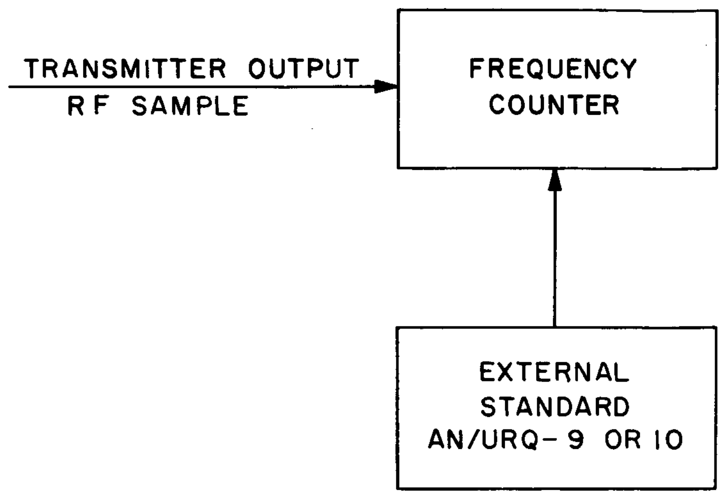



Figure 38. Block Diagram Illustrating Transmitter Frequency Accuracy Test

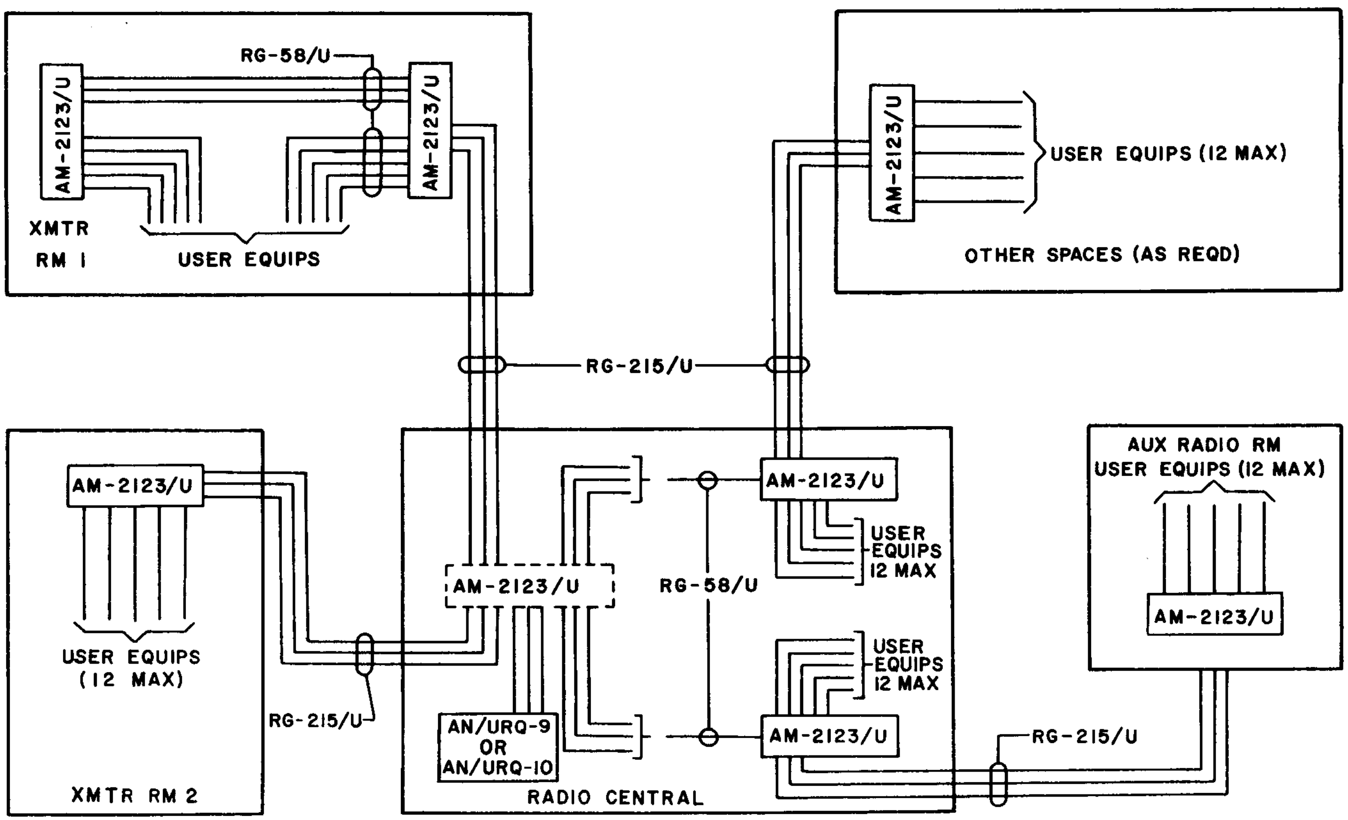



Figure 39. Frequency Standard Distribution Diagram, Typical Shipboard Installation