SS–Nature of Single Sideband Signals

PART 2

This is the third article from a training course written by Collins Radio Company for personnel concerned with single sideband communications.

A single-sideband signal is an audio signal converted to a radio-frequency, with or without inversion.

To illustrate the manner and the results of this conversion simply, pure sine-wave tones will be used, rather than the very complex waveforms of the human voice. For this reason, single tones or combinations of two or three tones are generally used in the following discussion.

The Generator

The most familiar SSB generator consists of a balanced modulator followed by an extremely selective mechanical filter as shown in figure 1. The balanced modulator produces basically two output frequencies:

1. An upper sideband frequency equal to the injected IF frequency plus the input audiofrequency.
2. A lower sideband frequency equal to the injected IF frequency minus the input audiofrequency.

Theoretically, the injected IF frequency is balanced out in the modulator so that it does not appear in the output.

It should be especially noted that in any mixing operation undesirable products are generated as well as the desired products. The equipment must be so designed to minimize the generation of undesirable products and to attenuate those undesirable products which are generated. This result is attained by designing good linear operating characteristics into the equipment to minimize the generation of undesirable frequencies and by choosing injection frequencies that will facilitate suppression of undesirable frequencies.

It should also be noted that the IF carrier injected into the balanced modulator is only theoretically canceled from the output. Practical design considerations determine the extent to which the carrier can be balanced out.

Present balanced modulators, in which controlled carrier leak is used to balance out uncontrolled carrier leak, result in carrier suppression of from 30 to 40 decibels below the PEP of the sidebands.

Further suppression of the carrier by the SSB filter results in an additional 20-decibel carrier suppression. Total carrier suppression of from 50 to 60 decibels can, therefore, reasonably be expected from the transmitter system.

Generating Single-Tone Waveform

The most fundamental SSB waveform is generated from the single audio tone. This tone is processed through the SSB generator to produce a single IF frequency. The SSB signal is actually generated at an IF frequency and is subsequently converted up in frequency to the transmitted RF frequency. It is the generation of the SSB signal at the IF frequency with which we are concerned.

Figures 2 and 3 show the waveforms obtained in a filter-type SSB generator. The audio tone injected into the balanced modulator is 1 kilocycle, and the IF frequency injected is 300 kilocycles.

The output from the balanced modulator contains the 299-kilocycle lower sideband and 301-kilocycle upper sideband frequencies. These two sideband frequencies, being of equal amplitude, produce the characteristic half sine-wave envelope shown in figure 2. The repetition rate of this envelope with a 1-kilocycle tone is 2 kilocycles, the difference between the two frequencies represented by the envelope.

This IF signal, which contains both the upper sideband and lower sideband signal, is called a double-sideband signal (DSB).

By passing the DSB signal through a highly selective filter with a 300- to 303-kilocycle passband, the upper sideband signal is passed but the lower sideband signal is attenuated. The 301-kilocycle signal which remains is the upper sideband signal and appears as shown in figure 3.

Note that the SSB signal remaining is a pure sine wave when a single-tone audio signal is used for modulation. This SSB signal is displaced up in the spectrum from its original audiofrequency by an amount equal to the carrier frequency, in this case 300 kilocycles.

Figure 1. Filter type of SSB generator.

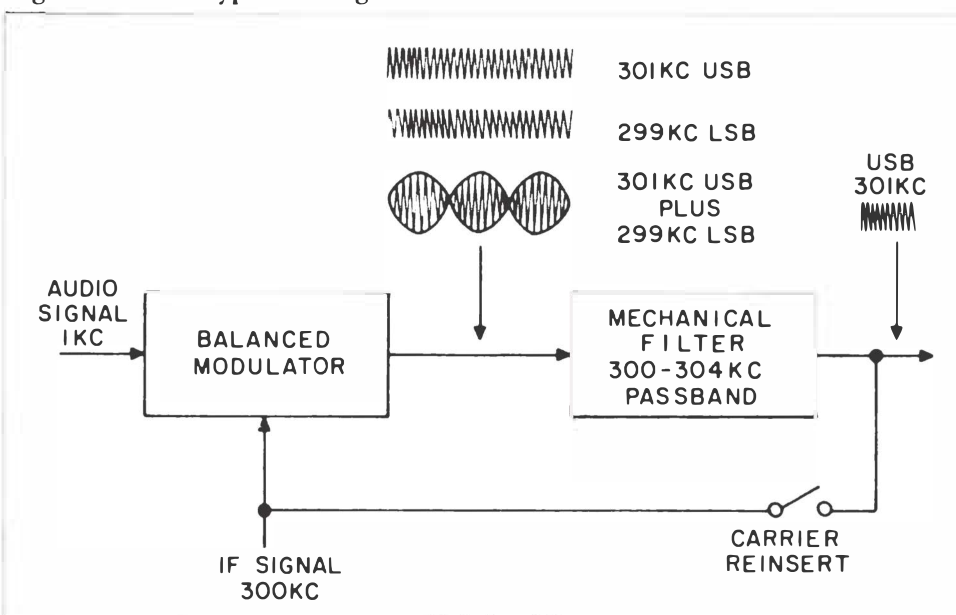


Figure 2. Single-tone, balanced modulator output.

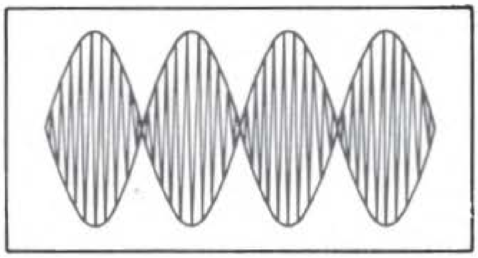


Figure 3. Single-tone balanced modulator output after filtering out the LSB.

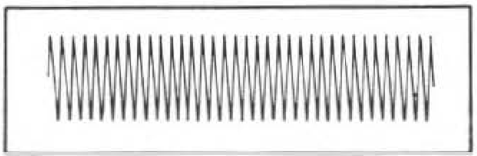


Figure 4. Single-tone SSB signal with carrier—carrier equal in amplitude to tone.

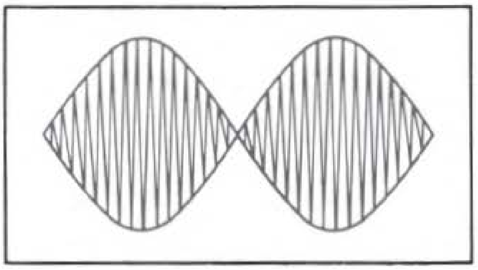


Figure 5. Single-tone SSB signal with carrier—carrier 10 decibels below tone.

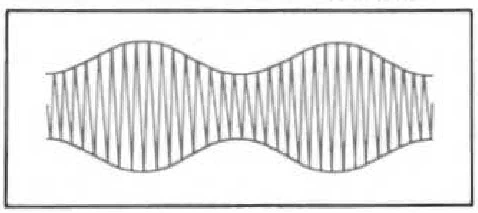


Figure 6. Two-tone SSB signal—tones of equal amplitude.

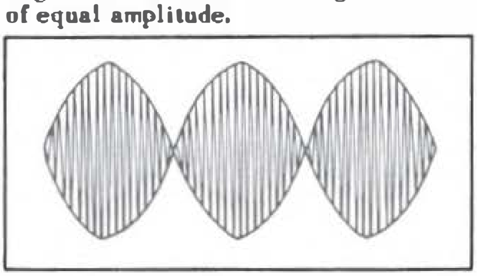
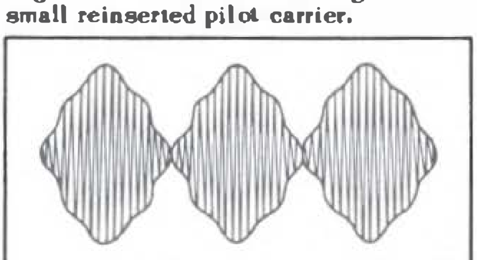


Figure 7. Two-tone SSB signal with small reinserted pilot carrier.



This SSB signal can be demodulated at the receiver only by converting it back down in the frequency spectrum. This demodulating is done by mixing the signal with an independent 300-kilocycle IF signal at the receiver.

Generating Single-Tone Waveform With Carrier

From the single-tone SSB signal without carrier, it is a simple step to generate the single-tone SSB signal with carrier. This is done by reinserting the carrier after the filtering operation, as shown in figure 1.

When the carrier reinserted is of the same amplitude as the SSB signal, the waveform shown in figure 4 results. Note that this waveform appears the same as the double-sideband signal obtained directly out of the balanced modulator, as shown in figure 2. However, the frequency components of the two waveforms are not the same.

The frequency components of the SSB signal with carrier are 301 kilocycles and 300 kilocycles when a 1-kilocycle audio signal is used. The SSB signal with full carrier can be demodulated with a conventional diode detector used in AM receivers without serious distortion or loss of intelligibility.

If the reinserted carrier is such that the carrier level is less than the level of the single-tone SSB signal, the waveform shown in figure 5 results. To demodulate the signal successfully, the carrier must be separated, exalted, and reinserted in the receiver, or locally supplied.

The separate carrier amplification should be enough to raise the reinserted carrier to a level greater than the level of the sideband signal. The waveform shown in figure 5 represents the waveform used in the SSB with pilot carrier systems. The exalted carrier technique is used to demodulate such a signal.

Generating Two-Tone Waveform

The two-tone SSB waveform is generated by combining two audio tones and then injecting this two-tone signal into the balanced modulator. One sideband is then suppressed by the filter, leaving the SSB waveform shown in figure 6.

This two-tone SSB signal is seen to be similar to the single-tone DSB signal as well as the SSB signal with full carrier. However, the two-tone SSB signal contains a different two frequencies than either of the other two.

In the two-tone SSB signal shown in figure 6, 1- and 2-kilocycle audio signals of equal amplitude are injected into the balanced modulator. After filtering, this action results in a two-tone SSB signal containing frequencies of 301 and 302 kilocycles.

If a pilot carrier is reinserted with the two-tone test signal, the pilot carrier will be indicated by the appearance of a sine-wave ripple on the two-tone waveform. This waveform is shown in figure 7.

The generation of this two-tone envelope can be shown clearly with vectors representing the two audio-frequencies, as shown in figure 8.

When the two vectors are exactly opposite in phase, the envelope value is zero. When the two vectors are exactly in phase, the envelope value is maximum. This generates the half sine-wave shape of the two-tone SSB envelope which has a repetition rate equal to the difference between the two audio tones.

The two-tone SSB envelope is of special importance because it is from this envelope that power output from an SSB system is usually determined. An SSB transmitter is rated in peak-envelope-power output with the power measured with a two equal-tone test signal. With such a test signal, the actual watts dissipated in the load are one-half the peak-envelope-power. This measurement is shown in figure 8.

When the half sine-wave signal is fed into a load, a peak-reading, r.m.s.-calibrated vtvm across the load indicates the r.m.s. value of the peak-envelope-voltage. This voltmeter reading is equal to the in-phase sum of $e_1 + e_2$, where e_1 and e_2 are the r.m.s. voltages of the two tones.

Since in the two-tone test signal e_1 equals e_2, the PEP equals $(2e_1)^2/R$ or $(2e_2)^2R$. The average power dissipated in the load must equal the sum of the power represented by each tone, $e_1^2/R + e_2^2/R$, $4e_1^2R$ or $4e_2^2/R$.

Therefore, with a two equal-tone SSB test signal, the average power dissipated in the load is equal to ½ of the PEP, and the power in each tone is equal to ¼ of the PEP. Peak-envelope-power can be determined from the relationship PEP = V^2_{vtvm}/R; the average power can be determined from the relationship $P_{average} = \frac{1}{2} V^2_{vtvm}/R$.

This relationship is true only when the vtvm used is a peak-reading, r.m.s.-calibrated voltmeter. Similar measurements can be made using an a.c. ammeter in series with the load instead of the vtvm across the load.

The above analysis can be carried further to show that with a three equal-tone SSB test signal, the power in each tone is ⅑ of the PEP, and the average power dissipated in the load is ⅓ the PEP. These relationships are true only if there is no distortion of the SSB envelope, but since distortion is usually small, its effects are usually neglected.

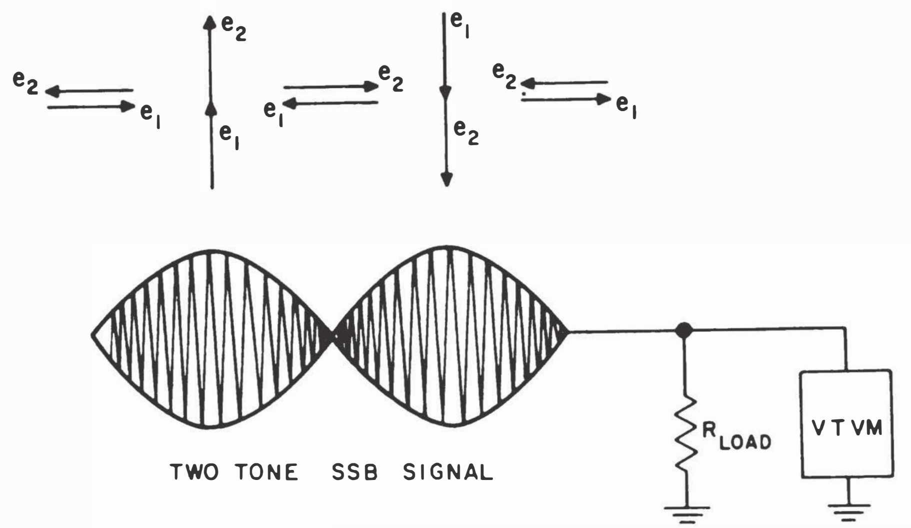
Generating Square Waveform

Transmitting an audio square wave at a radiofrequency imposes severe requirements on any transmitting system. This is true because the square wave is composed of an infinite number of odd-order harmonics of the fundamental frequency of the square wave.

Therefore, to transmit such a signal without distortion requires an infinite bandwidth, an infinite spectrum. This requirement, of course, cannot be met because tuned circuits will not pass an infinite bandwidth.

The idealized SSB square wave, when all frequency components are present, shown in figure 9, indicates that the SSB signal requires infinite amplitude as well as infinite bandwidth. This condition occurs because the harmonically related SSB components will add vectorally to infinity when the modulating signal switches from maximum positive to maximum negative and vice versa.

This infinite amplitude is not present in an AM envelope, because the AM envelope contains both sidebands with the frequency components in one sideband counter-rotating vectorally from the frequency components in the other sideband. The result then, when the resultant amplitude of one sideband is plus infinity, is that the resultant amplitude of the other sideband is minus infinity, which produces a net amplitude of zero.



$$V_{vtvm} = (e_1 + e_2),$$

with e_1 and e_2 in phase and rms values

$$PEP = V^2_{vtvm}/R_{load} = 4e_1^2/R \text{ or } 4e_2^2/R,$$

where $e_1 = e_2$

$$P_{average} = e_1^2/R + e_2^2/R = 2e_1^2/R \text{ or } 2e_2^2/R$$

Therefore: (1) $PEP = V^2_{vtvm}/R$

(2) $P_{average} = 1/2\ PEP$

(3) $P_{tone\,1}$ or $P_{tone\,2} = 1/4\ PEP$

Figure 8. Power measurements from two-tone SSB test signal.

The significance of the SSB square wave lies in its relationship with conventional clipping techniques used to limit the modulation level.

Figure 10 shows the SSB envelope that results from severely clipping a 300 c.p.s. sine wave. The clipping level is such that the modulating signal is essentially a square wave.

In generating the SSB envelope from the modulating signal, all harmonics above the ninth are removed by the highly selective SSB filter. Figure 9 shows that speech clipping, as used in AM, is of no practical value in an SSB transmitter because the SSB envelope is so different from the audio envelope.

In an SSB transmitter, automatic load control, rather than clipping, is used to prevent overdriving the power amplifier by holding down the modulation level. It is possible to use a significant amount of clipping in an SSB transmitter if the

Figure 9. Square wave SSB signal—all frequency components present.

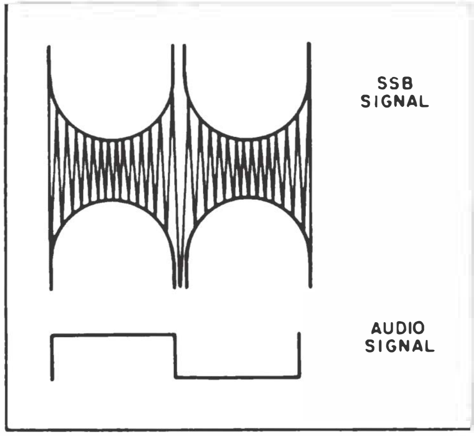


Figure 10. SSB envelope developed from 300 c.p.s., clipped-sine-wave (harmonics above 9th attenuated).

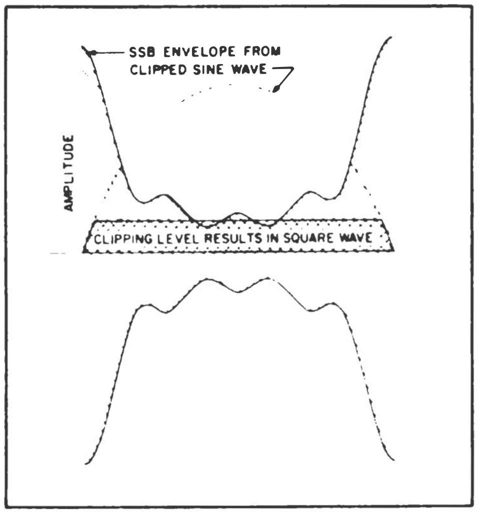


Figure 11. Voice signal at audio frequency ā sound.

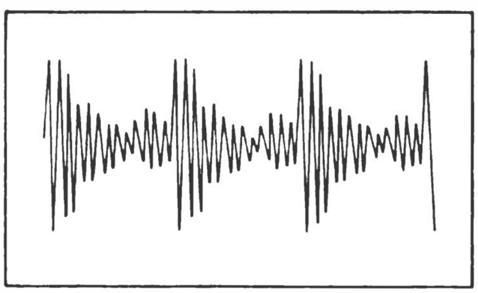
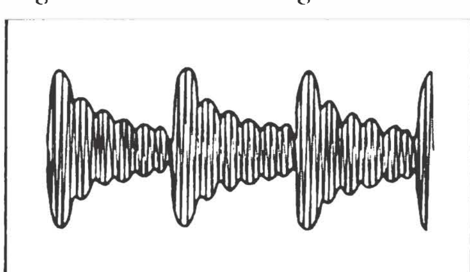


Figure 12. SSB voice signal ā sound.



clipping is performed on the IF SSB signal rather than on the audio signal.

If clipping were performed at this time, additional filtering would be required to remove the harmonic products caused by the clipping. However, clipping at this stage is satisfactory, because the harmonic products produced are not in the passband of the filter and only small intermodulation products are generated in the passband.

Generating Voice Waveform

The human voice produces a complex waveform that can be represented by numerous frequency components of various amplitudes and various instantaneous phase relationships. No human voice is exactly like another, but statistical averages concerning the frequencies and amplitudes in the human voice can be determined.

The average power level of speech is relatively low when compared to the peak power level. An audiofrequency waveform of an ā sound is shown in figure 11. The same ā sound, raised in frequency, is shown in figure 12 as it appears as an SSB signal. From the "Christmas-tree" shape of these waveforms, it is evident that the peak power, which is related to the peak voltage of a waveform, is considerably higher than the average power.

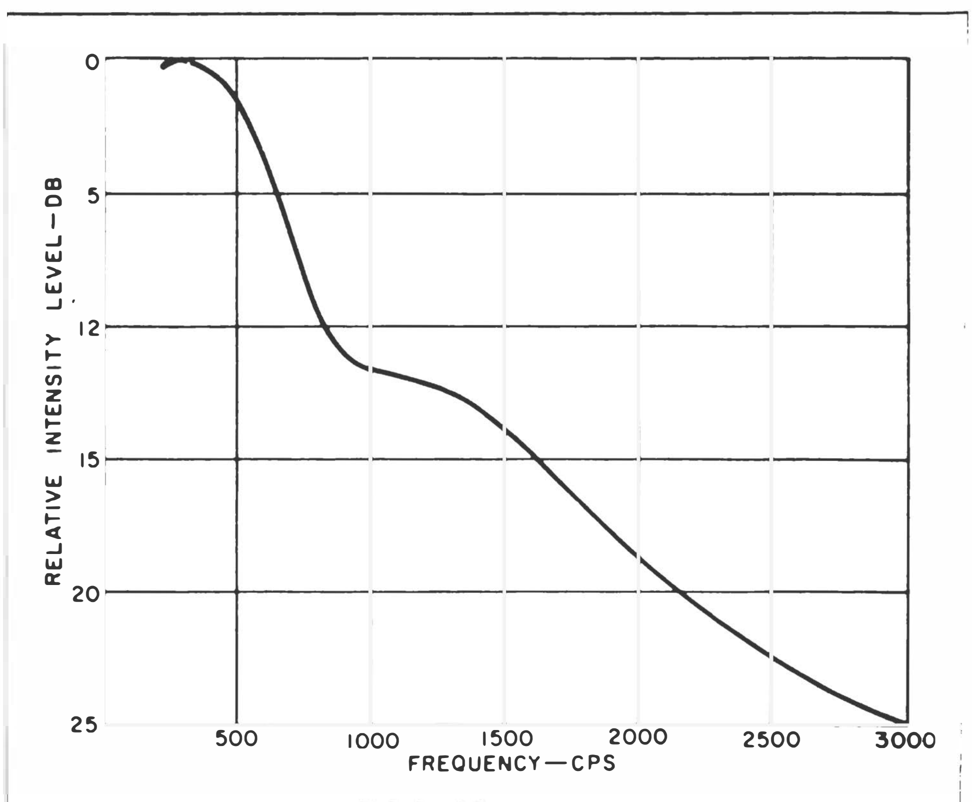
Over-all transmission efficiency depends on the average power transmitted, but transmitter power is limited to the peak power capability of the transmitter.

Therefore, for voice transmission, the transmitter designer must use speech shaping circuits that will increase the average power in the voice signal without increasing the peak power. He can do this in three different ways:

1. By clipping the power peaks.
2. By emphasizing the low-power, high-frequency components of the speech signal, and attenuating the high-power, low-frequency components.
3. By using automatic-gain-control circuits to keep the signal near the maximum capability of the transmitter.

Figure 13 shows a power versus frequency distribution curve for the average human voice, after filtering below 200 c.p.s. and

Figure 13. Power distribution in speech frequencies—low and high frequencies removed.



above 3000 c.p.s. This curve shows that the high-power components of speech are concentrated in the low frequencies.

Fortunately, it is the low-frequency components of speech that contribute little to intelligibility since these frequencies are concentrated in the vowel sounds. The low frequencies, therefore, may be attenuated without undue loss of intelligibility of the speech.

The low-power, high-frequency components present in a voice signal can be pre-emphasized to increase the average power level of the signal. Since it is the high-frequency components that predominate in the consonant sounds, some emphasis of high frequencies will improve intelligibility.

However, to emphasize the high frequencies enough to raise the average power level significantly would require compatible de-emphasis at the receiver to prevent loss of fidelity.

Clipping of power peaks results in flattening the waveform at the clipping level; and with severe clipping, the voice signal becomes a series of square waves.

Since an SSB square wave envelope requires infinite amplitude as well as infinite bandwidth, the audio signal must be clipped with discretion.

In the SSB transmitter, automatic load control is used to control the average power level input, rather than clipping, to prevent overdriving the power amplifier. Clipping then is used only to remove the occasional power peaks.

Speech-processing methods are being reinvestigated in relation to SSB transmission to determine the most suitable method or combination of methods. At present, several circuits are used in SSB transmitters which do some speech processing, although the primary purpose of most of them is to process the input signal to prevent overdriving the power amplifier. These circuits include:

- Automatic-load-control to maintain signal peaks at the maximum rating of the power amplifier.
- Speech compression, with some clipping, to maintain a constant signal level to the single-sideband generator.
- Highly-selective filters used in filter-type SSB exciters to attenuate some of the high-power, low-frequency components of the voice signal.

There are also several speech processing circuits under investigation which, if effective and practical, will be used to improve the efficiency of voice transmission. These circuits include:

- Increased audio clipping with additional filtering to remove the harmonics generated.
- Reduction of the power level of frequencies below 1000 c.p.s. by shaping the audio amplifier characteristics for low-frequency roll-off.
- Use of speech clipping at an IF level where the generated harmonics can be more easily filtered.

More information on the input signal processing circuits will be included in a later article in this series. The next article is on mechanical filters.

TIMELY CONSIDERATION OF INTERFERENCE CONTROL

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If naval communications and electronic control devices are to operate satisfactorily, electronic, electrical, and mechanical equipment and devices in the area must not produce interference.

Also, interference must be effectively controlled if the military and civil life of the United States is to profit from the many proposed new advantages in the conveyance of intelligence and automation.

Experience shows that interference is, in most cases, indicative of incomplete equipment design or mistakes on the production line.

Suppression Inadequate

Whenever interference control is considered as a "suppression measure" to be handled at a time past the engineering stage, increased costs and relatively unsatisfactory results usually ensue. Then reliability features are often compromised.

If proper design features are incorporated in the initial development stages and appropriate control measures are used in production, interference need not be a serious problem, nor does its control need to be costly.

The Bureau of Ships has often found equipments, supposedly ready for shipment by the manufacturer, to be deficient and thereby to have high interference characteristics. Embarrassing delays are usually experienced while corrective measures are worked out.

In practically all such cases, the delayed corrective measures are only partially satisfactory; and while incorporating the measures in the equipment, the usual reasons for interference are found to be true: Incomplete design and mistakes in production.

Responsibility

A naval contractor who procures equipment components outside his own factory should require them to be interference free. He should screen and check these components prior to acceptance to determine that interference requirements are met.

Manufacturers who do not have, in their own plants, the facilities essential to guide design, control production, and demonstrate to naval inspectors that their equipment complies with the contract requirements concerning interference, should retain the services of a consulting laboratory.

If a naval contractor is uncertain about interference requirements in a contract, or of the proper steps to be taken relative to furnishing interference free equipment, he should consult with the naval inspection office or the cognizant bureau early in the design stage of the equipment.