The Lenkurt® Demodulator

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the universal VOICE CHANNEL

Most terminal equipment for modern telecommunications service is designed to operate over a voice channel, a fraction of a voice channel, or several such channels put together. The transmission medium may be cable, open wire, or radio, but nevertheless the basic unit for defining the facility is usually the voice channel. But the question naturally arises, what is a voice channel? This article attempts an answer, which necessarily includes discussion of services other than speech transmission.

If several types of communications users, such as a telephone man, an industrial user, and a military man, were asked to define the term "voice channel" there would be as many different definitions as there were types of users. This is to be expected, since each one puts his definition in terms of his own particular needs and applications.

One man may be concerned only with the essential information content in speech transmission. Another may be concerned with the reactions of paying customers to how well the communications equipment reproduces the quality of their friends' voices. Still another man may not be concerned with speech at all, but rather with the transmission of data or some other service over the "voice channel."

But regardless of these diverse needs, and of the various qualities of the facilities required to fill them, all voice channels have a single common denominator. They are designed primarily around the characteristics of the human voice and the human ear.

Speech Characteristics

From the listener's point of view the quality of a voice channel can be measured in terms of two parameters, intelligibility and intensity, which together determine the quality of reception of sounds transmitted over the channel. Interestingly enough, intelligibility and intensity are virtually independent of each other over quite a broad frequency range. Most of the speech energy, and hence the intensity, is concentrated in the lower frequencies, while the high frequencies contribute most to the intelligibility. If *no* frequencies below 1000 cps were transmitted, articulation would be about 86 per cent perfect, but the received energy content would be only about 17 per cent of the original energy. On the other hand, if *only* the frequencies below 1000 cps were transmitted, articulation would be reduced to about 42 per cent, while 83 per cent of the total energy would be transmitted (Figure 1 illustrates this graphically). This means that any voice channel must include both the low frequencies and the high frequencies. Furthermore, some compromise is usually necessary because available bandwidth is limited.

Perhaps the most important factor in determining bandwidth, however, is the reaction of the people using the facilities. Many experiments have been performed to test subjective reaction to various transmission impairments, including restricted bandwidth. As a result of such tests, the "standard" voice channel bandwidth has come to be accepted as about 3 kc. Typically, the range of transmitted frequencies is from about 200 cps to about 3200 cps.

Another significant characteristic of speech is its redundancy. As much as 75 per cent of the information content in normal speech is redundant. If a syllable is lost (or often even a word), the listener automatically fills in the gap from the context. The result is that the requirements for speech transmission are often much less stringent than those for other types of transmission.

Services Other Than Speech Transmission

Manufacturers of equipment for the transmission of telegraph, data, and facsimile often find it convenient to design their equipment for operation over voice channels. The reason is simply that voice transmission facilities are almost universally available. However, difficulty may arise because the voice channel is designed around the peculiarities of speech transmission, while these other services may have quite different requirements. It then becomes necessary to evaluate the quality of the voice channel in terms of the technical factors which are important for the other types of transmission.

One of the first characteristics which comes to mind in defining any type of communications channel is bandwidth. The bandwidth required for voice transmission is, of course, determined by speech characteristics, as previously mentioned. The bandwidth required for digital transmission, however, is primarily determined by the speed of transmission. The higher the speed the more bandwidth required. Thus, telegraph service with its relatively slow speed requires a small fraction of a voice channel — typically, 170 cps for 100 words per minute. Conversely, high-speed data transmission, measured in kilobits and even megabits per second, may require enormous bandwidth, equivalent to many voice channels. Where a data system operates over a voice channel with a nominal 3-kc bandwidth, the transmission speed is often 1200 or 2400 bits per second.

One of the items of major concern in evaluating any communication channel is noise. But there are various types of noise, and one kind may affect a particular type of communications more than another kind. For example, white noise (background noise) measurements are usually used in evaluating a channel for speech purposes because the human ear hears this kind of noise — white noise tends to mask the desired message. Data, however, is relatively insensitive to white noise. Here it is impulse noise which causes the most trouble. Impulse noise consists of

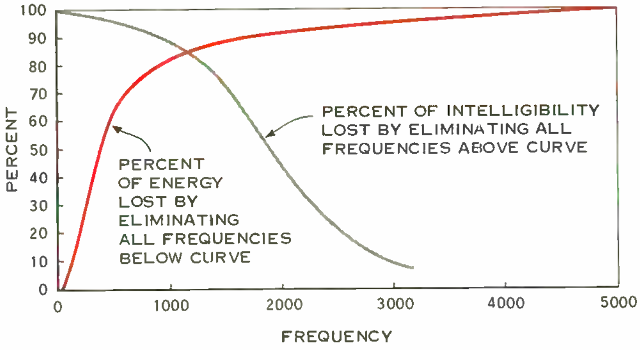


Figure 1. Curves indicate that speech energy is concentrated in the lower frequencies, while the higher frequencies contribute most to intelligibility.

short "spikes" which may reach quite a high amplitude, but which have a relatively short duration. The length is often measured in fractions of a millisecond. Even though they have a high amplitude, these spikes are often too short for the human ear to hear. Therefore, they cause very little trouble in voice transmission. In data transmission, however, they can easily reach the level of the data signal, thereby causing an error, as shown in Figure 2.

White noise has little effect on data transmission until the noise reaches a very high level because data is not affected until the noise peaks reach the detection level of the signal. Thus, even though the ear may hear a significant amount of white noise, the chances are slight that it will have much effect on data.

Attenuation and Delay Distortion

Services other than voice transmission may be transmitted in three forms. They may use a serial digital transmission arrangement with one information bit transmitted after the other; they may use a parallel digital transmission arrangement where more than one bit is transmitted simultaneously; or transmission can be in the form of analog signals, without the "quantizing" of digital techniques. But regardless of the way the information is transmitted, there is one thing in common: the various frequency components which make up the complex signal bear a very definite relationship to each other both in magnitude and in time. This means that all frequencies within the passband should suffer the same loss if *attenuation distortion* is to be avoided. It also means that all frequencies within the passband should propagate through the transmission medium at the same speed to avoid *delay distortion* — an altering of the phase relationship between the frequency components.

Attenuation distortion is perhaps equally important in either speech or data transmission. Typically, the higher frequencies in the passband are attenuated more than the lower ones. For example, on nonloaded wire pairs the attenuation is usually proportional to the square root of frequency within the voice band. Inductance loading is used to reduce both attenuation distortion and overall loss on most cable pairs longer than about three miles. This is fine for speech transmission, but it makes the line resemble a lowpass filter with a cutoff frequency. As this cutoff frequency is approached, phase or delay distortion increases rapidly.

Phase distortion is relatively unimportant in voice transmission because the components of speech need not

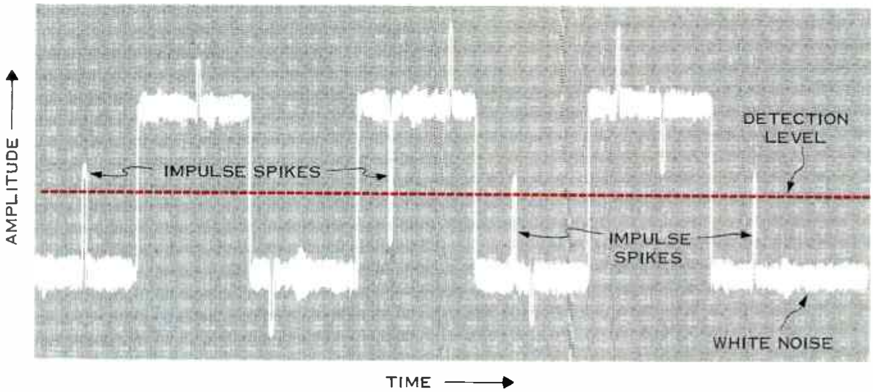


Figure 2. Unlike speech transmission, data is relatively unaffected by noise below the signal detection level. Thus, errors are quite likely to be caused by impulse "spikes," while white noise has little effect on error rate.

bear an exact time relationship to each other for intelligibility. Furthermore, the large amount of redundancy in normal speech tends to mask the effect of any phase distortion. For these reasons phase distortion is rarely considered (unless it is extreme) in evaluating a voice channel for speech transmission.

In a typical channel the frequency of minimum delay is approximately 1600 to 1700 cps. A curve of relative delay plotted as a function of frequency usually reaches a minimum in this region and forms a more or less symmetrical "U" about the midpoint. The two ends of a typical curve reach a relative delay of perhaps a millisecond or more at about 400-800 cps on the low end and at about 2700-2900 cps at the high end. Figure 3 shows the characteristics of a typical carrier system voice channel. The channel shown has a "one-millisecond bandwidth" of approximately 3000 cps and it has a half-millisecond bandwidth of 2500 cps. A relative delay of one millisecond will have little affect on speech transmission, but it may well render a voice channel completely unusable for data, particularly at higher transmission speeds. For this reason data transmission systems seldom use the entire voice frequency bandwidth, but are placed near the center of the frequency band where distortion is lowest. (Data is often transmitted in the 1000-2600 cps band).

Equalization

A common technique to increase the percentage of bandwidth usable for data transmission is known as *equalization*. A channel is equalized by introducing a network which produces either attenuation or phase shift characteristics (or both) opposite to those already inherent in the channel. Figure 3 illustrates how the equalizer characteristics add to the channel characteristics to produce a relatively smooth curve.

Of course it is seldom possible to obtain a truly flat curve in this way. Equalization is used simply to get the attenuation and phase delay within the acceptable tolerance for data transmission. Ripples in the attenuation and delay characteristics frequently represent echoes caused by impedance mismatches. The edges of the band normally exhibit more pronounced ripples because more discontinuities, and hence more reflections, occur here. This is one more reason why often only the center of the bandwidth is used for data transmission. Echoes from "close-in" discontinuities usually result in ripples which can be equalized, whereas echoes from remote transmission discontinuities often cannot be equalized. The reason is that an individual transmitted pulse is usually affected primarily by its own echo if the source of the echo is close in. Echoes from remote discontinuities tend to affect later transmitted pulses. Thus, the effect is random for a train of pulses, and equalization is not generally effective.

Compandors and Echo Suppressors

The interfering effects of noise, crosstalk, and echo on speech are often reduced by the use of compandors and echo suppressors. These devices take advantage of certain peculiarities of human conversation. Unless they are removed from the transmission path, however, they often render a circuit unfit for many types of data transmission.

Compandors give an apparent reduction in noise on voice circuits by increasing the circuit loss during speech pauses — between syllables and words. If amplitude-modulated (on-off) type data signals are transmitted through a compandor, some of the pulses may be badly distorted by the varying loss characteristic. However, frequency-shift data signals are not appreciably distorted by a compandor since their power level remains essentially constant.

Echo suppressors are frequently used on long circuits to prevent the echo of reflected speech from annoying the talker. They are simply devices which inhibit the return path when a person is talking. Most echo suppressors in use on commercial telephone circuits today permit transmission in only one direction at a time. During data transmission they must be removed from the transmission path or they will not permit a data receiver to ask for the repeat of a message in which an error has been detected. This "disabling" of the echo suppressor is accomplished by a device which is sensitive to a tone of 2000-2250 cps generated by the data transmitting equipment. When the disabler receives this tone for some specified length of time, it holds both directions of transmission open while data is being transmitted.

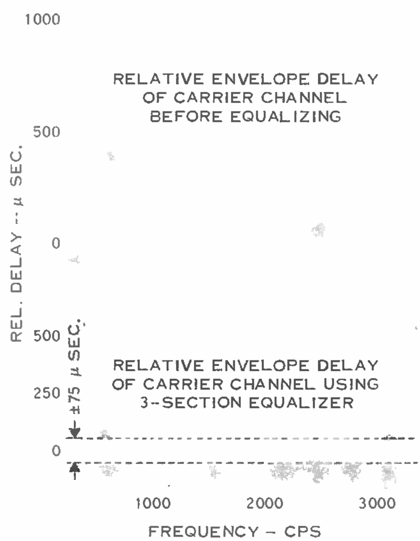


Figure 3. Top panel shows the envelope delay characteristics of a typical carrier channel before equalization. Addition of a 3-section equalizer reduces relative envelope delay to ±75 microseconds over a large portion of the bandwidth.

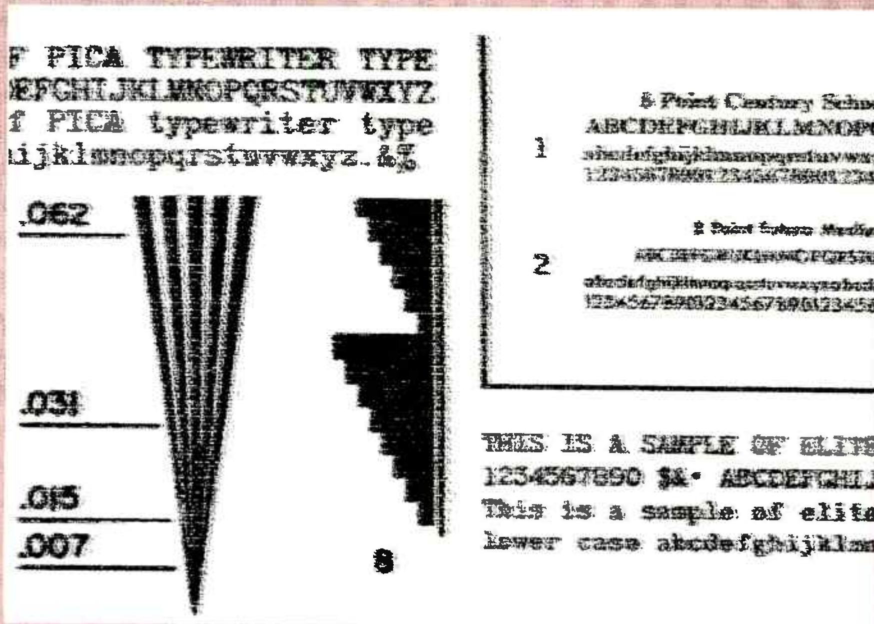


Figure 4. Actual sample of facsimile transmission over a circuit not corrected for delay distortion. Note the fine echoes following each transition from black to white or white to black. This same effect occurs in high-speed data transmission, causing errors to increase.

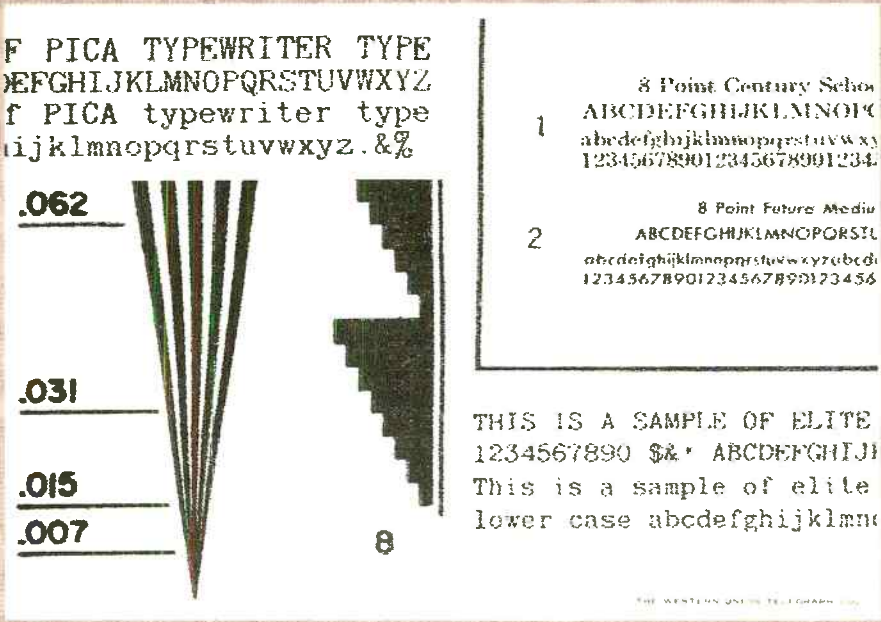
Future Trends

The history of the communications industry is one of increasingly stringent performance requirements, with no end to this trend in sight. This comes about because communications users are demanding that their transmitted information be reproduced at the receiving end with ever-greater fidelity.

But there is another important factor influencing the shift toward "tighter" specifications for voice channels. This is the changing pattern of communication. It includes not only the ever-greater quantities of information being transmitted, but also the trend toward the transmission of many kinds of information in digital form. Equipment manufacturers today must consider factors which, in the not-too-far-distant past, were only academically related to voice channel performance.

Consider delay distortion for example. Serial data at 2400 bits per second has a bit length of only 0.417 millisecond. If the delay distortion is several milliseconds, the data will be hopelessly garbled because parts of a pulse may be delayed enough to overlap with the "faster" portions of the following pulse. Newer equipment designs recognize this problem and allow for it. For example, Lenkurt's long-haul 46A carrier equipment holds delay distortion to 0.165 millisecond (without equalization) for the usual data band, 1000 to 2600 cps.

The 46A is also unique in its data-handling capabilities. Up to 65 per cent of its channels can be loaded with



Transmission is greatly improved when the delay distortion present in the transmission medium is corrected by delay equalizer. Lines and other fine detail are reproduced much more sharply. The ability to transmit finer detail by equalizing delay is similar to higher speed data transmission.

data at the same time — more than any other commercial multiplex equipment. (Older equipment may falter under as little as 25 per cent data loading.) In recognition of the fact that data-handling capacity is becoming more important, the extra power-handling capability necessary to accommodate such a load is an integral part of the design.

Thus, it is becoming increasingly apparent that the performance standards of the past do not apply to the present, much less the future. The direction of the trend is clear. A system barely adequate for today's needs may be hopelessly obsolete in a few years; whereas a system which is "over-designed" by present standards will have a working life far longer. •

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The Lenkurt 76A2 microwave equipment, operating in the common-carrier frequency band of 5925-6425 Mc, carries up to 960 voice channels. Alternatively, it carries a color or black-and-white television channel, with or without program subcarrier. The technical characteristics and advanced design features are described in Form 76A2-P4, a new publication available from Lenkurt or any Lenkurt field office.

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