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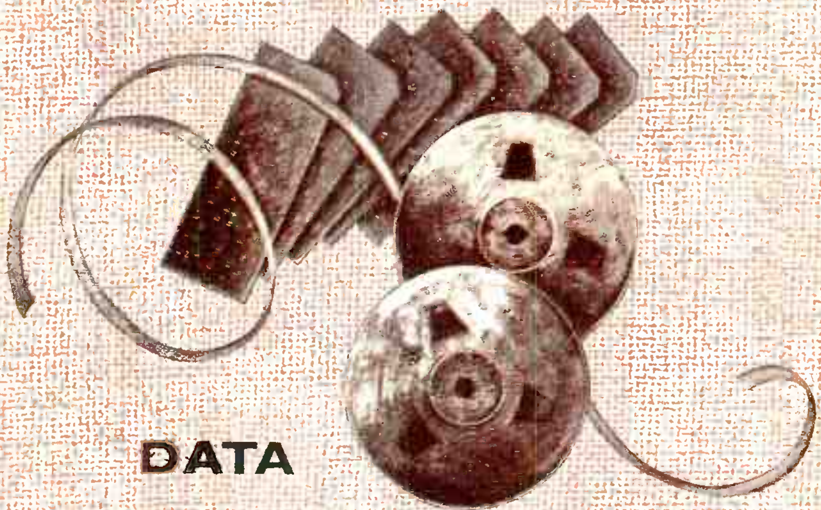
the Lenkurt Demodulator



DATA COMMUNICATIONS

Part 2

LENKURT ELECTRIC... specialists in VOICE, VIDEO & DATA transmission

Today, telephone companies are faced with the challenge of providing reliable data transmission at higher speeds over voice facilities. Because data signals do not possess the built-in redundancy of speech signals, data accuracy is very important. A single error could render a message incorrect. This article, the second of two parts, describes some of the major causes of errors and discusses some of the considerations involved in transmitting data over telephone lines.

ONE of the most important considerations in transmitting data over communications systems is accuracy. Data signals consist of a train of pulses arranged in some sort of code. In a typical binary system, for example, digits 1 and 0 are represented by two different pulse amplitudes. If the amplitude of a pulse changes beyond certain limits during transmission, the detector at the receiving end may produce the wrong digit, thus causing an error.

It is very difficult in most transmission systems to completely avoid such errors. This is especially true when transmitting digital signals over an analog transmission system designed for speech signals. Many of the inherent electrical characteristics of telephone circuits have an adverse effect on digital signals, often making the circuits unsatisfactory for data transmission—especially at high speeds. Frequently, these circuits must be specially treated before they can be used to handle data at speeds above 2000 bits per second.

Voice channels on the switched (dial-up) telephone network exhibit certain characteristics which tend to distort typical data signal waveforms. Since there is random selection of a particular route for the data signal with each dialed connection, transmission parameters will generally change, sometimes upsetting the effect of built-in compensation networks. Often data communications operations require speeds or types of signals which cannot be handled by the switched network. In addition, the switched network cannot be used for large multiple address data systems using time sharing. Because of these considerations, specially treated voice bandwidth circuits are made available for data use (see Table A). The characteristics and costs of these point-to-point *private lines* are published in documents called tariffs, which are merely regulatory agreements reached by the FCC, state public utilities commissions, and operating telephone companies regarding charges for particular types of telephone circuits. The main advantage of private or dedicated facilities is that transmission characteristics are fixed and remain so for all data communications operations.

Signal Impairments

Probably the most critical circuit qualities affecting data transmission are attenuation-frequency response, phase-frequency characteristics, and impulse noise. In addition, echoes and circuit net loss or over-all attenuation tend to degrade data pulses and usually have to be considered when selecting a voice circuit for data transmission.

The attenuation-frequency response limits the bandwidth for transmission. This characteristic causes the transmission loss of a circuit to vary with frequency. Ideally, the attenuation-frequency curve should be "flat" across the band required for data transmission. Most modern voice transmission circuits are sufficiently "flat" between 600 and 2700 hertz so that low-speed data is not seriously affected by this characteristic. However, for high-speed data transmission, it often is necessary to provide some sort of compensation in order to equalize the attenuation-frequency response.

The phase-frequency characteristics of a circuit cause what is known as *delay distortion*. (See *The Lenkurt Demodulator*, June, 1965.) Delay distortion results from the capacitive and inductive reactances common in all communications circuits, causing the propagation rate of signals to vary non-linearly with frequency over the desired bandwidth. This is not a problem in speech transmission because the human ear is not very sensitive to phase-frequency varia-

TABLE A. Specially Treated Voice Bandwidth Circuits for Data Use

	Use	Interstate Tariff FCC No.
Schedule 2 Telephoto	Alternate voice/facsimile (telephoto) or FAX only	140
Schedule 2 Telephoto with Special Conditioning	Alternate voice/facsimile (telephoto) or FAX only	140
Remote Operation & Control (FAA)	Simultaneous voice/remote operation and control or remote operation and control only	135
Schedule 4 Type 4 Data	Alternate voice/data or data only	237
Schedule 4 Type 4A Data	Alternate voice/data or data only	237
Schedule 4 Type 4B Data	Alternate voice/data or data only	237
Schedule 4 Type 4C Data	Alternate voice/data or data only	237
Schedule 5 Data	SAGE data circuits digital data	237

tions. But for data, such variations limit the speed of transmission and reduce the margin for error.

Delay distortion becomes more critical as data speeds increase. Typically, higher data speeds are achieved by shortening the width (duration) of each pulse. Because of the shorter pulse, slight shifts in relative phase between frequency components have a greater effect in distorting the signal.

Some type of compensating network or *delay equalizer* must be employed when the phase-frequency characteristics of a circuit are unsuitable for data transmission (see Figure 1). Such devices introduce a controlled amount of phase shift into the circuit at various frequencies.

The most difficult type of signal impairment to overcome is *impulse noise.* Such noise is extremely unpredictable and is commonly caused by electrical storms and the operation of switching equipment in the telephone plant. Quite often impulse noise will raise or lower the amplitude of a data pulse above or below a fixed detection or slicing level, causing the wrong binary symbol to be indicated at the receiver. One way of overcoming the problem of impulse noise would be to raise the signal power. However, communications systems have limited power handling capabilities which cannot be exceeded.

White noise, on the other hand, has a relatively uniform distribution of energy. Caused by the thermal agitation of electrons in resistances, white noise is always present in electrical circuits and cannot be eliminated. Since this type of noise is predictable, its effect can usually be overcome in the design of a data communications set.

Signal impairment also may be caused by variations in the carrier frequencies between the transmit and receive terminals of a single-sideband suppressed-carrier multiplex communications system. In such systems, carriers removed at the transmit terminal must be reinserted at the receive terminal. Any change in frequency between the transmit and receive carriers will shift the various frequency components of the data signals. If the frequency difference is great enough, the data signals can become so distorted that they cannot be correctly detected. This problem is

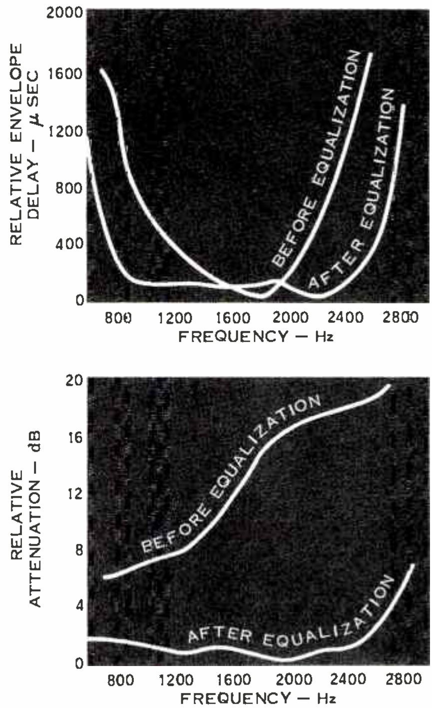


Figure 1. Both delay and amplitude distortion affect the quality of data transmission. Top panel shows the relative envelope delay of a typical voice channel both before and after equalization. The result of amplitude equalization is shown in bottom panel.

overcome by synchronizing the carrier generators of the two multiplex terminals, usually with a *pilot* signal which is transmitted through the system. Also, the use of a subcarrier in the modulation scheme of a data set helps to minimize the effect of frequency shift when data signals are fed through multiplex systems.

Error Detection and Correction

As data signals leave the business machine or computer and enter the data transmitter, they are essentially free from error. Since telephone systems were originally designed for voice transmission, various characteristics of the telephone circuit can impair the quality of a data message, possibly changing its context and rendering it unusable. Therefore, from the standpoint of service, the actual error rate or probability of error, which is determined statistically, is the most important factor in evaluating the overall performance of a data set.

Because of the significance of errors in data communications and since it is not practical to build error-free transmission circuits, the usual course is to equip data sets with some type of error control. Typically, error control systems in use today are capable of both error detection and correction, or error detection only.

Techniques which only detect errors are generally less complex than those which detect and correct errors. The simplest and most widely employed method of error detection is simple *parity check coding*. This technique uses redundant bits of information inserted into the digital message so that there is always an odd or even number of mark or space bits transmitted. Parity check coding, though vulnerable to

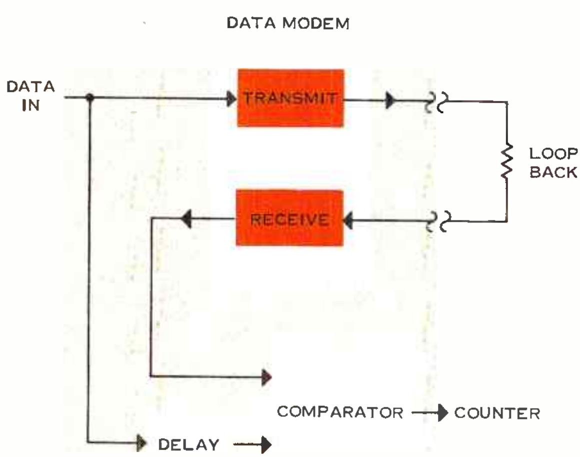


Figure 2. Typical method for measuring error performance of a data communications set. Data signals are looped back from transmitter to receiver. Delay corresponding to the amount in the path is added to the original data signals which are then compared bit-by-bit with the received data signals. An error counter keeps track of the number of errors.

many kinds of multiple errors which overcome the capacity of the coding method, has found wide use because the data receiver can be arranged to check parity without the need for complex circuits.

Error correction techniques use redundancy on a larger scale than the parity check method. The amount of redundancy determines the maximum number of errors that can be corrected. Error correction arrangements are extremely complicated, and the reduction in effective transmission speed necessary to accommodate redundant bits can become excessive. Furthermore, a substantial percentage of errors tend to come in bursts, so the utility of many correction schemes may be somewhat limited.

Data transmission systems using the Lenkurt - developed Duobinary technique (see *The Lenkurt Demodulator*, February, 1963) provide error detection without the need for adding redundant information. Instead of adding redundant bits to the message which, in effect, lowers the transmission rate, Duobinary coding follows a systematic pattern that provides a more powerful error detection method than the simple parity check.

For business purposes, data systems normally transmit information in sections or blocks. When errors in a block are detected, the system automatically keeps retransmitting the block until there are no errors. For real-time systems, such as high-speed telemetering, where data is obsolete almost immediately after it is received, duplicate or diversity transmission techniques are about the only practical solution to offset the problem of errors.

Error rate may be determined by comparing each and every bit of the received message with the transmitted message. This can be accomplished by either looping-back the received signal to the originating transmitter, or by transmitting a predetermined pattern of data signals known to the receiver. When the data signal is looped back (see Figure 2), a variable delay, corresponding to the transmission time of the data message in both directions, is inserted into the original message before it is applied to a bit-by-bit comparator. By

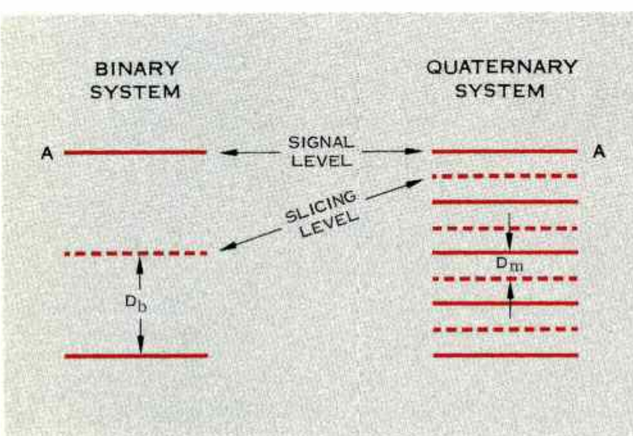


Figure 3. Determining noise penalty compared to binary. Because multilevel systems require more slicing points than binary systems, they are more susceptible to errors caused by noise.

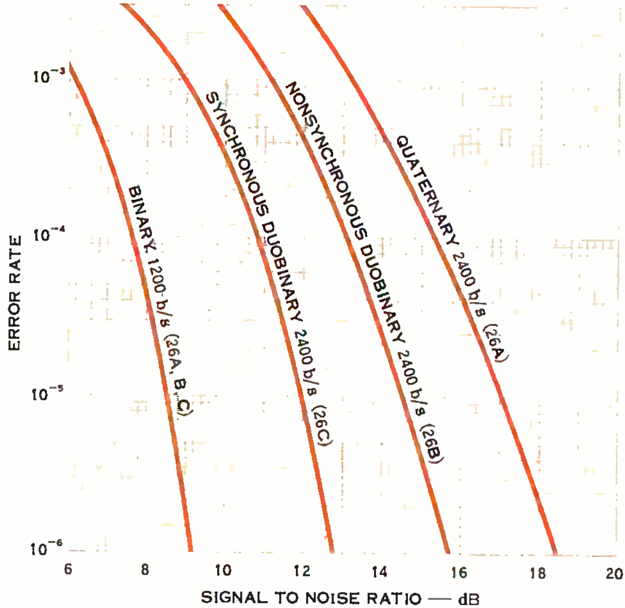


Figure 4. Comparison of error rate versus signal-to-noise ratio. Without normalization synchronous Duobinary has a noise penalty of about 3.4 dB compared to binary. This is a marked increase in performance over nonsynchronous Duobinary and quaternary.

analyzing the transmitted message and the received message together, the comparator signals an electronic counter when errors are present.

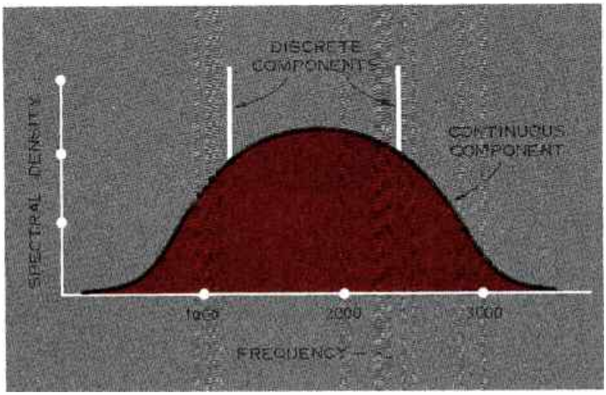
Modulation Methods

Selecting a suitable type of modulation is extremely important in the design of data transmission systems to provide simplicity and to achieve optimum performance. The three basic methods of modulation in data transmission are amplitude modulation (AM), frequency modulation (FM), and phase modulation (PM). Frequency modulation of a binary signal results in the shift of a two-state or binary signal about an FM carrier, and thus is usually referred to as frequency shift keying (FSK) rather than FM.

The most commonly used technique is binary FM (or FSK) because this type of signaling offers a good signal-to-noise ratio, and is not affected by changes in amplitude level. However, the signaling speeds achieved with binary modulation are limited and generally inadequate to meet present-day data requirements.

Vestigial sideband AM and techniques using more than two or *multilevel* signal states increase the amount of data that can be transmitted over communications facilities. Quaternary FSK and quaternary phase shift keying (PSK) are examples of multilevel systems which effectively double the data rate compared to binary that can be transmitted in a given bandwidth. But there is a greater sensitivity to noise with both vestigial sideband and multilevel systems, and error performance is usually poorer. Furthermore, distortion in the absence of noise, termed inter-

Figure 5. Spectral distribution of a binary FM signal. Discrete components present in the binary signal represent wasted power; these are not present in the synchronous Duobinary signal.



symbol interference, is increased with multilevel signals. This type of distortion is caused by the overlap of positive or negative overshoots of the past pulses into the time slots of other pulses. Experimentation on multilevel systems continues and new approaches have been investigated to improve their performance.

Correlative Technique

Correlative data transmission techniques, particularly the Duobinary principle, have aroused considerable interest because of the method of converting a binary signal into three equidistant levels. This correlative scheme is accomplished in such a manner that the predetermined level depends on past signal history, forming the signal so that it never goes from one level extreme to another in one bit interval.

The most significant property of the Duobinary process is that it affords a two-to-one bandwidth compression relative to binary signaling, or equivalently twice the speed capability in bits per second for a fixed bandwidth. The same speed capability for a multilevel code would normally require four levels, each of which would represent two binary digits.

Noise Penalty

Generating a signal with correlated levels permits overall spectrum shaping as well as individual pulse shaping, thus minimizing intersymbol interference. However, there is a noise penalty with respect to binary systems, and this applies to level-coded correlative systems as well as other multilevel systems. Though exact mathematical calculations of such penalties are usually complicated, there is a quick method of finding an approximate value. In Figure 3, both binary and multilevel representations are shown. Assuming an equal peak voltage of A volts for both cases, the noise penalty is the ratio of the distances between any signal level and the adjacent slicing level for each of the two systems.

The corresponding distances are $Dm = A/2(m - 1)$. where m is the number of levels

and for binary

$$Db = A/2$$

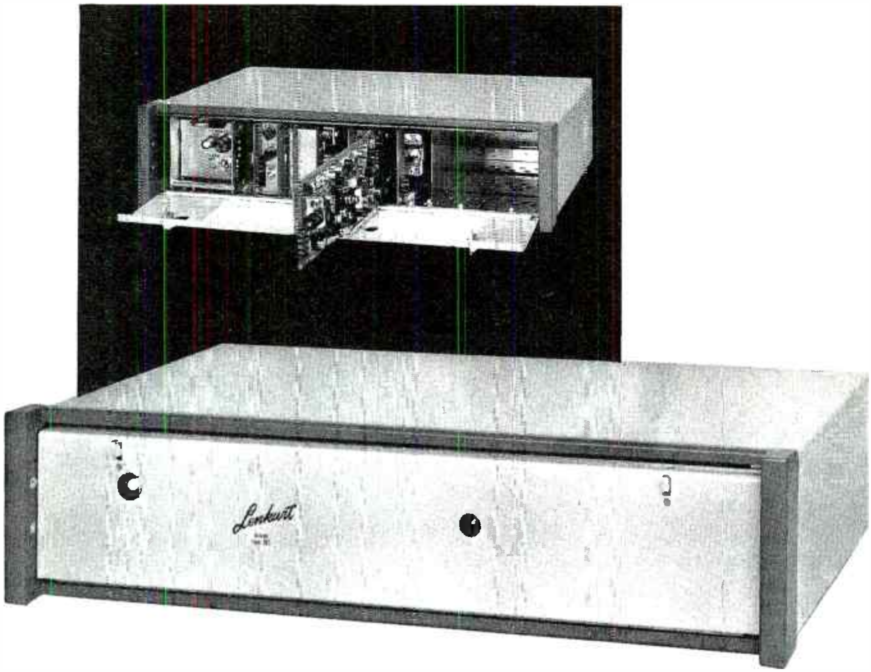


Figure 6. A typical data set shown here is Lenkurt's 26C which transmits serial digital data over standard 3-kHz voice channels at 1200 or 2400 bits per second. Other practical applications include telemetry, digitized voice or facsimile data transmission, and air-to-ground data communications over UHF or VHF radio links. Transmission at 2400 bits per second is achieved through the use of the Lenkurt synchronous Duobinary coding technique. Low error rate coupled with automatic error detection are added features of the Duobinary technique.

The approximate noise penalty (in decibels) for an m-level system relative to a binary system is then

$$20 \log Db/Dm$$

Substituting the values of Db and Dm gives

$$20 \log \frac{A/2}{A/2(m-1)} = 20 \log (m-1)$$

For a binary system, the noise penalty is 0 dB, since binary serves as a reference. The calculation for a quaternary system reveals that the noise penalty is approximately 9.4 dB relative to binary. The Duobinary signal has three levels ($m=3$). Calculating the noise penalty for a Duobinary system from the above formula results in a 6-dB value with respect to binary.

Synchronous Duobinary

A special situation occurs when data transitions are *synchronized* with the carrier phase in FM data transmission at 0°, 180°, or ±90°, as opposed to where the data transitions bear no relationship to the FM carrier phase. It would appear that because there is still the same number of slicing levels, the *synchronized* Duobinary signal has a 6-dB noise penalty. Yet, in practice, it is closer to 3 dB. This situation can be analyzed mathematically, but is more clearly demonstrated by comparative testing of actual working systems under identical conditions.

If error rate is measured as a function of signal-to-noise ratio, individual noise penalties can be determined by subtracting the resultant signal-to-noise ratio of nonsynchronous Duobinary, synchronous Duobinary, and quaternary from the signal-to-noise ratio of a binary system.

It is easy to establish identical conditions to determine the noise penalty by using three systems that have the same peak-to-peak FM deviation, namely, 1200 Hz, which has been proven to be optimum for binary transmission. Three such systems are the Lenkurt 26A, a quaternary system; the Lenkurt 26B, a nonsynchronous Duobinary system; and the Lenkurt 26C, a synchronous Duobinary system.

The conditions for plotting the curves shown in Figure 4 were established by supplying a random binary data signal to each system, introducing white noise with flat weighting over a 3.4-kHz bandwidth, and retiming and sampling the data with a clock derived from data transitions.

For clarity, the signal-to-noise ratio values shown are not *normalized* (put on the same speed basis), since the most significant factor is the performance of Duobinary and quaternary relative to binary rather than the absolute signal-to-noise ratio values. From Figure 4 and Table B, it can be seen that the experimental results and calculated values from the formula $20 \log (m - 1)$ are relatively similar for quaternary and nonsynchronous Duobinary when compared at an error rate of 10^{-5} (1 error in one hundred thousand bits), with a sufficiently long-time error averaging period.

For synchronous Duobinary, there is only a 3.4-dB noise penalty with respect to binary for random input data. This is because the synchronous system results in a power spectrum that does not have discrete spectral lines as does the nonsynchronous system. (See

TABLE B. Signal-to-noise ratio and noise penalty comparisons with respect to binary of synchronous Duobinary, nonsynchronous Duobinary, and quaternary systems for an error rate of 10^{-5}.

System	S/N Ratio (dB) from Figure 4	Noise Penalty (dB) relative to binary
Binary	8.5	-
Synchronous Duobinary (Lenkurt 26C)	11.9	3.4
Nonsynchronous Duobinary (Lenkurt 26B)	14.5	6.0
Quaternary (Lenkurt 26A)	17.0	8.5

Figure 5.) Discrete components are two steady tones (sinusoids) at frequencies of 1200 and 2400 Hz, respectively, and which appear in binary FM. If the ratio of frequency difference in hertz between mark and space to the bit speed in bits per second (or deviation ratio) is unity, the total power is equally divided by the continuous component and the two discrete components. This phenomenon is inherent to binary FM transmission and there is nothing that can be done about it. Unlike this situation, the synchronous Duobinary signal contains only the continuous information carrying component of spectral density, but no discrete components. Consequently, the total power can be increased by 3 dB compared to nonsynchronous Duobinary FM which contains discrete components that do not carry information.

The Future

It is universally recognized that communications is essential at every level of organization. The United States Government utilizes vast communications networks for voice as well as data transmission. Likewise, businesses need communications to carry on their daily operations.

The communications industry has been hard at work to develop systems that will transmit data economically and reliably over both private-line and dial-up telephone circuits. The most ardent trend in data transmission today is toward higher speeds over voice-grade telephone channels. New transmission and equalization techniques now being investigated will soon permit transmitting digital data over telephone channels at speeds of 4800 bits per second or higher.

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Correction: *In the September, 1966 issue, Figure 1, the lower "Transmit" and "Receive" functions within the blocks marked "Data Modem" should be reversed. Each data modem should have a transmit and a receive section.*

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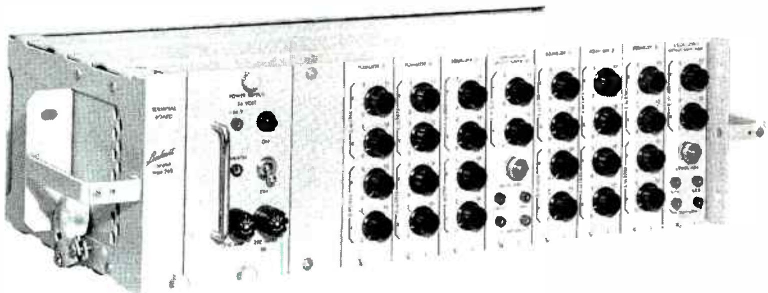

Lenkurt's 30231 Delay Equalizer corrects for both attenuation and delay distortion on voice frequency circuits used for data transmission, generally providing better equalization than fixed equalizers. Each fully self-contained Delay Equalizer consists of two sets of seven equalizer sections for two separate voice frequency lines. Adjustments are simple and do not require complicated calculations. A typical voice frequency circuit can be equalized to have a relative delay of 50 microseconds and a relative attenuation of ±1 dB between 700 and 2800 Hz.

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