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An Experimental Modulation-Demodulation Scheme for High-Speed Data Transmission*

Abstract: An experimental low-cost system was designed to determine speed and reliability limitations on transmitting binary data over private telephone lines. A brief review of alternative approaches is given, with a description of the laboratory model. Performance of the equipment is reported with the reliabilities experienced at 600, 1000, 1600, and 2400 bits per second.

Introduction

World-wide telephone facilities seem attractive as a potential high-speed data-transmission carrier. Considering the need for high-speed transmission of data for business, industrial and defense applications, it is evident that the telephone network could eventually become one of the most important means for data transmission.

There are two fundamental questions to be answered in an evaluation of the telephone network and of its capacity as a vehicle for data transmission:

1. How many bits per dollar can we transmit?
2. How reliably can we transmit data at a reasonable cost?

An attempt is made in this paper to answer the question of how many bits per second one can expect to transmit over the telephone network. Reliability considerations have also been made, based upon limited experience gained with a data set which was developed for experimental purposes. Simplicity of the terminal equipment (modulation and demodulation) is shown to be one very important factor in designing a system that is both reliable and economical.

Theoretical and practical transmission speeds

Theoretically, according to Hartley, Nyquist, Shannon and others, one should be able to serially transmit 6000 binary bits per second without interbit interference in a frequency band of 3000 cps. At first it might seem that the useful bandwidth of a telephone channel for data transmission is from 300 to about 3300 cps, and therefore a speed limit of 6000 bits per second is expected.

It has been demonstrated† that it is practical to transmit data at a rate of 2W bits per second through a low-pass filter of the effective bandwidth of W cps. Using metallic circuits, which by their nature have low-pass characteristics, speeds of 2W bits per second are therefore theoretically and practically possible.

Actually the telephone line in general is a band-pass filter of a very special kind. In addition to its restrictive properties with respect to the frequency spectrum, it has the property of slightly shifting the whole spectrum by an arbitrary and continuously changing amount of up to approximately $\pm$ 20 cps. This continuous drifting of the frequency spectrum is caused by single-sideband suppressed carrier systems when both ends of the carrier link are not in rigid synchronism. Such drifts are not noticeable in speech transmission, yet they make the data transmission problems quite difficult.

Phase distortion, tolerable for the human ear in speech transmission, makes the useful bandwidth for data transmission much less than what it is for speech applications. At the present time the highest practical speed limit was achieved by the Collins Kineplex 202 System and it is

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†Philco demonstration at the 1956 IRE National Convention in New York City.

3000 bits per second.[IV:2] To achieve that goal, quite complex and sophisticated terminal equipment is necessary, along with a high-quality private line. The bandwidth of a telephone line is one of the most important factors in speed considerations. In telephone circuits which are in operation and one can talk over, it appears that the bandwidth varies between 1400 and 3400 cps. (The lower limit exists only on a few emergency channels remaining from World War II.) Considering the actual useful bandwidth for data transmission, a speed of 600 bits per second is quite reasonable.

As was emphasized, simplicity and cost of the terminal equipment are important factors in our considerations. Our research activity was therefore aimed at the promising serial-by-bit transmission techniques, which seemed relatively unexplored.

The challenge was to find out how fast and how reliably one can transmit data with relatively simple terminal equipment. The assumption was made that slight adaptation of the telephone plant to data transmission needs can be expected so that one does not have to degrade the prospective services to match the weakest link in the present telephone network.

If one considers the problem of high-speed data transmission in assuming the above restrictions, the situation looks quite promising. Transmission speeds in the order of the capacity of parallel, frequency multiplexed telegraph channels can be achieved with a single high-speed, serial-by-bit channel. The prospective reliability of such a service can be made sufficiently high so as to become attractive for computer as well as message-transmission applications. A transmission speed of 1000 bits per second seems to be practical without phase equalization and without special noise-reduction effort. For private-line applications, 1600 to 2400 bits per second are possible, depending on the line facilities which can be made available.

The problems

Let us consider now the problems one has to face in attempting to use the telephone network for high-speed binary information transmission. The telephone network is designed for speech transmission and therefore is adapted to the human information receptor. In order to achieve greater efficiency, the telephone plant designer takes advantage of the peculiarities of the human speech perception mechanism. Unfortunately, sophisticated techniques used with great success in speech communications cannot be applied to binary information transmission systems because of the specific differences between the two information entities.

What are those specific differences which are of relevance to binary information transmission and make data transmission problems so difficult?

• *Impulse noise*

The human ear is much less sensitive to impulse noise disturbances than it is to white noise and constant tones. The telephone noise measurement sets, such as the Western Electric 2B and the Psophometer (recommended by CCITT*), are therefore adapted to the human ear and the telephone receiver. Those measurement sets would not indicate that a noisy line becomes intolerable for data transmission.

The impulse noise is produced by dc interruptions in dialing operations and is induced into the adjacent unshielded pairs of telephone cables. In general, no special attention was paid to careful shielding and other means of impulse noise elimination since occasional impulse noise spikes, in the order of a millisecond or less duration, are not objectionable for speech communications.

In serial binary data transmission, the bits of information are in the order of a millisecond or less. Impulse noise interference spikes of approximately the same duration, similar in shape and sometimes up to 20 db greater in magnitude than the desired information signal, represent one of the main problems and limitations in speed and reliability of data transmission over telephone lines. (See Fig. 12.) Of little use, if not detrimental, are ingenious devices such as compandors, which are used with great success in making too-frequent impulse noise and other interference considerably less objectionable for speech communication. One can successfully trick the human ear by allowing the noise to occur only during the time when speech is on the line. A compandored telephone line sounds quiet, since the noise is suppressed between syllables and words. The apparent signal-to-noise ratio improvement of compandored lines is approximately 20 db. It is obvious that such tricks do not apply for binary transmission. That is why lines quite acceptable for speech transmission sometimes present difficult problems if used for data transmission.

• *Phase distortion*

It is a well-known fact that the human ear is not sensitive to phase distortion of the telephone network. Therefore, nonlinearities of phase characteristics caused by sharp filtering and loaded cables are not objectionable for speech transmission. The situation is reversed for high-speed data transmission where phase nonlinearities of the telephone channel represent one of the main limitations. Phase distortion makes the portion of the frequency spectrum below 1000 cps less and less suitable for binary information transmission as one approaches the lower portion of the spectrum, since it does not pay to correct the phase characteristics of that portion of the telephone channel. Therefore the useful frequency band for high-speed data transmission starts between 600 to 1000 cps, depending on the transmission speed. The upper portion of the band, say from 2500 cps up, usually has to be phase corrected if maximum efficiency of the telephone line has to be achieved. The phase correction is only practical for private-line applications. For dial-up applications, one simply has to limit the service in speed and transmit the signal in the portion of the band which is acceptable for

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data transmission. Phase linearity is important for binary transmission since phase distortion causes overshoots and other undesirable waveform changes. Phase nonlinearities therefore extend the duration of the received data pulse and produce interbit interference. The wave shape of the information bit and its duration are important factors in binary information transmission. The phase distortion of the telephone line, therefore, makes the useful bandwidth for data transmission narrower than it is for speech transmission.

• *Frequency band shift*

To illustrate the significance of a spectrum shift for binary transmission, let us assume that we are transmitting a continuous sine wave, where a bit is defined as one cycle (dipulse) of the sine wave. If 1000 such bits per second are transmitted over an asynchronous network, the number of dipulses per second can vary as much as $\pm$ 20. It appears therefore as if one would lose or gain up to 20 bits per second. What happens is that, in going through an asynchronous telephone network, one could easily lose two important factors for information recovery: the timing and the phase of the information carrier. Consequently, special care must be taken to make binary information possible in using the telephone network. One of the many ways of doing this will be shown later.

• *Predictability of binary signals*

Some of the specific differences between speech and data transmission which create difficult problems for data transmission have been considered. On the other hand, binary information has some properties which could be of great help in the attempt to overcome these problems. In binary information transmission one can reasonably well predict the wave shape of the received signal as well as the time of its arrival. This is, of course, not the case in speech transmission.

The above two properties of the binary signal can be very useful in devising detection schemes which make strong discrimination against noise possible. Ultimately, it is possible to establish reliable data communications through links which are not acceptable for speech transmission.

Choice of modulation schemes

High-speed binary-information transmission over presently available telephone lines very much depends on the choice of modulation and detection schemes. Let us therefore examine what alternatives one has in the light of boundary conditions set by the present telephone network.

• *On-off modulation or double-sideband AM systems*

The *on-off* modulation schemes were historically the first used for binary information transmission. The two binary states are characterized by the presence or absence of an information carrier frequency (subcarrier) which is usually located in the middle of the available frequency band.

The advantage of such schemes is their simplicity. They are, however, sensitive to sudden amplitude variations of the line and are relatively vulnerable to noise. Since both sidebands are transmitted, better frequency spectrum utilization seems possible.

• *Frequency modulation systems*

In low-speed data transmission applications (up to 200 bits per sec) the FM systems can tolerate about 10 db more white noise and maintain the same systems performance as AM double-sideband systems, in respect to speed or bandwidth utilization. The immunity to level variation is one of the crucial factors which paved the way for FM into telegraph carrier systems.[II: 1, 2, 3]

The same arguments do not apply entirely for high-speed data transmission, where other factors have to be considered. In low-speed applications, with many carrier cycles per bit of information, the impulse noise does not represent a problem, since the bit is so much longer in duration than the impulse noise disturbances. At higher speeds, where there is often only one cycle of the FM subcarrier per bit, the problem of vulnerability to impulse noise and to noise in general becomes severe. The FM capture effect is not effective at higher data transmission speeds.

The FM systems remain insensitive to amplitude variations, even at higher speeds, as long as the noise does not become a paramount problem. Since FM systems require both sidebands, they are like double-sideband AM systems in that they are not very efficient in bandwidth utilization.

• *Vestigial sideband on-off schemes*[III]

The vestigial sideband *on-off* modulation schemes operate with greater efficiency than double-sideband systems. Only a vestige of the upper sideband is usually transmitted, resulting in higher speeds. Its sensitivity to amplitude variations and noise, however, impose stringent and costly requirements on the telephone network in order to achieve the required reliability.

• *Phase modulation and synchronous detection*

The modulation and detection scheme most suited for telephone-line data transmission appears to be phase modulation and synchronous detection (sometimes called homodyne detection).[IV: 2, p. 727] Great immunity from noise can be achieved along with insensitivity to level variations. If vestigial sideband transmission is combined with synchronous detection techniques, high speed and reliability can be achieved with simple means. Quadrature component rejection makes the synchronous detection schemes the least sensitive to phase distortion of all the systems yet discussed.

These qualities make the phase modulation schemes very attractive for high-speed data transmission over telephone lines, but some difficult problems needed to be solved before high speed and reliability could be achieved with simple means. The main obstacle to the utilization of phase modulation schemes is the continuous shift of the frequency spectrum commonly found on telephone networks. Because the information-bearing entity of the received signal is its phase, the problem of synchronous detection is obviously difficult in an asynchronous telephone carrier system.

Another difficulty to be mentioned is the phase ambiguity of phase modulation systems, which does not exist in *on-off* or FM schemes, where the binary value is unambiguously given by the presence or absence of the subcarrier or by a predetermined frequency. The phase ambiguity of the synchronous detector does not, however, represent a serious problem, since it can be eliminated by simple starting logic.

There are two basic philosophies for deriving the detection signal (carrier or subcarrier) in phase modulation systems.

The first is a phase detection system which uses the delayed preceding bit as a phase reference.[IV: 4] This type of system is usually simple if applied on a single-channel basis. The drawback of such schemes is that single errors of the received signal can produce double errors in the system. This makes error detection and error correction more difficult.

The second type of detection system involves phase control of the subcarrier oscillator. Such schemes are used for microwave television links, for example, and are usually quite complex.[IV: 5] Relatively simple control schemes are successfully applied in conjunction with double-sideband suppressed carrier systems.[IV: 3]

An experimental phase modulation system, built in the IBM Research Laboratories in San Jose, attempted to overcome some basic limitations in high-speed binary data transmission over the present telephone network, including a microwave radio relay link.

The experimental system is designed to work on existing telephone lines with no special noise reduction treatment. Information was transmitted serially rather than in parallel, to achieve simplicity of equipment and to keep down its cost.

The experimental system

The basic principles used in our laboratory experimental system were (1) bit synchronous subcarrier modulation; (2) phase modulation and detection; and (3) clock derivation based upon the difference-frequency principle.

As the bit rate approaches the speed of the subcarrier frequency, increased modulation jitter is produced in the system. This becomes one of the speed-limiting factors in the available data transmission systems. It is highly advantageous, therefore, for the bits of information and the subcarrier frequency to be in rigid synchronism. Bit synchronous subcarrier modulation thus eliminates one of the technological speed barriers outside of the channel itself, with its physical speed limitations of bandwidth, frequency shift, level changes, delay distortion, and impulse noise.

Then, in order to get the greatest possible rejection of the interference using relatively simple circuitry, a signal pulse is sent if either a mark or space bit is transmitted. Basically, the information is sent in reversing or not reversing the phase of the bit synchronous subcarrier by 180°. In order to cause an error, the disturbance has to override the signal, or, in other words, it has to be of the same order of magnitude. In addition, the synchronous detector will more or less reject any disturbance out of phase with the expected bit of information. Maximum interference rejection is achieved when the disturbance is $\pm 90°$ out of phase with the received signal. In order to create an error, the disturbance has to be greater than the signal and 180° out of phase with the information bit. The proposed system is basically a phase modulation and detection system with practically achieved ruggedness and disturbance rejection, as was predicted theoretically.

Level variations are basically no problem in the system since the detector has a positive or negative value at the bit-sampling time. A level change affects only the magnitude of the sample and not its polarity, which carries the information.

The advantages of phase modulation and detection systems in interference rejection have already been discussed, but the difficulties of phase detection with an asynchronous carrier arise when the bit duration approaches that of a single cycle of the subcarrier, and the bit is transmitted through a medium which shifts the original spectrum as in the case of single-sideband suppressed carrier telephone systems. Even though synchronously transmitted, the phase of the received information changes continuously at the rate of the frequency spectrum shift.

Another problem which is created by shifted carrier operation is the problem of bit synchronism, or the problem of transmission of timing information over asynchronous carrier lines. At the transmitting end, the information-carrying subcarrier is in synchronism with the information bit. In a synchronous carrier system one can derive the bit timing from the subcarrier frequency. In shifted carrier operation this is not the case, since the bit synchronism of the subcarrier is lost.

Our experimental system derives the bit-time information or the bit rate from the difference between the information-carrying subcarrier and the non-interfering pilot frequency which is sent 10 db below the signal. Regardless of the spectrum shift, the difference frequency and its phase stay constant and are used to clock the received information.

Along with binary information transmission a "start" signal is customary in any data transmission system. A special character can be reserved for that purpose, if the information is coded on a character basis. The problem becomes more complex if the bits of information are transmitted more or less at random. A separate level for the start bit is a solution of the problem which results in increased vulnerability to noise of the information bits and is therefore not considered as satisfactory. A practical approach which does not affect the reliability of the system is one in which the start bit could consist of a sudden reversal of pilot and signal levels for a duration of 5 bits, for example. The level reversal could be followed by 2 bits of information of one polarity and one bit of the opposite.

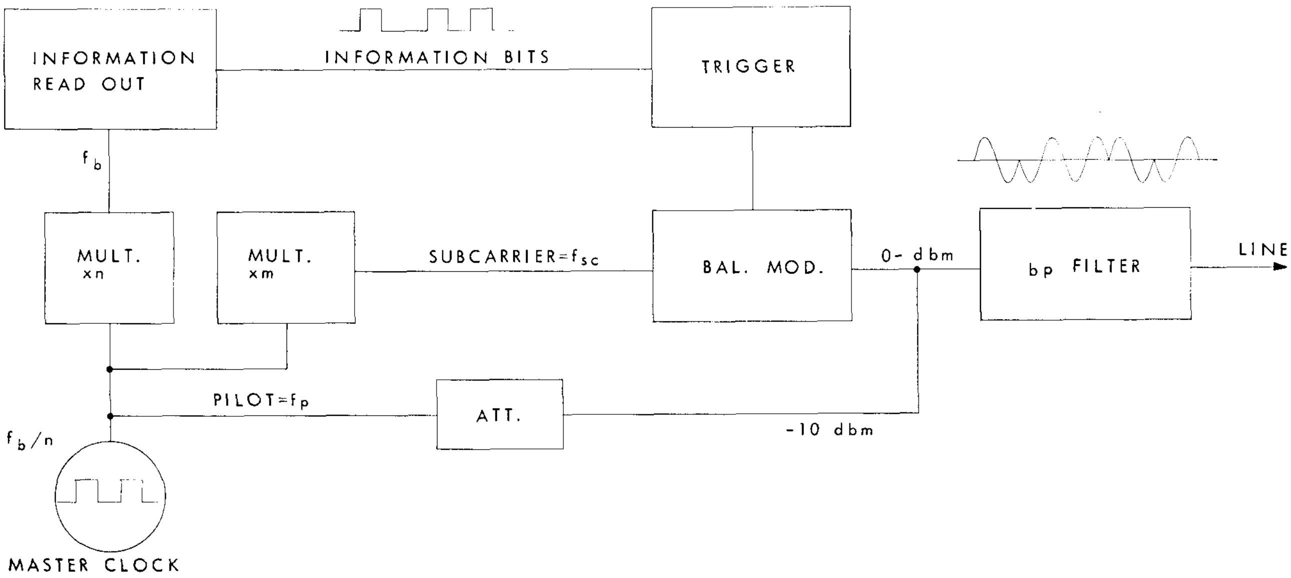


Figure 1 **Transmitter.**

The last bit could be considered as the start bit. This approach to transmitting a start signal is not an original one. It could be an integral part of a system, which, as a whole, represents a new approach to the problem of high-speed data transmission over available telephone lines.

- *Transmitter*

The master clock (see Fig. 1) which provides the clocking signal for the information readout is also used to derive the subcarrier and pilot frequencies. The information-carrying subcarrier and the bits of information are thus in phase synchronism. The pilot and the master clock are of the same frequency, whereas the bit rate and the subcarrier may or may not be the same, depending on the transmission speeds and the frequency distribution of the signal spectrum. For example, the experimental system has a master clock frequency of 800 cps, a bit rate of 1600 bits per sec, and a subcarrier of 2400 cps. Therefore, the bit rate is twice the clock frequency, and the subcarrier is three times the clock frequency. The pilot and the information signal spectrum are not interfering with each other since the bit spectrum has no energy at the pilot frequency (see Fig. 2) and the synchronous detector rejects the pilot frequency. The pilot frequency is transmitted considerably below the signal level (-10 dbm). This is done in order to allow maximum channel loading for the information signal itself.

The frequency multiplication is achieved by conventional analog techniques in deriving the second and third harmonics from the clock signal.

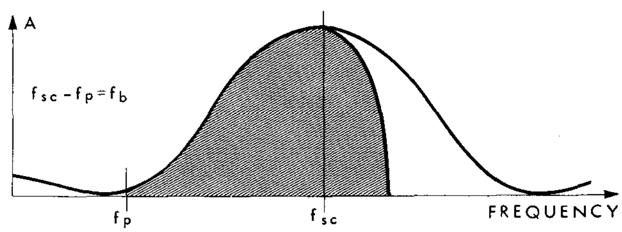
The bit-synchronous modulation occurs at the balanced modulator. Depending on the information bit, the phase of the subcarrier is switched to either 0° or 180° for the duration of the bit.

The band-pass filter eliminates the portion of the signal spectrum which cannot be adequately transmitted by the transmission channel. This filter is designed with special care in order to achieve acceptable phase linearity throughout the cut-off region. Thus a good match between the signal and the channel is achieved. Consequently, delay distortion introduced by the transmission channel is minimized, resulting in less interference between bits at the receiving end.

- *Receiver (See Fig. 3)*

To understand the synchronous detector, let us assume for the moment that we have at our disposal the shifted subcarrier frequency $f_{sc} \pm \alpha$. It is the balanced demodulator which compares the phase of the incoming bit to the phase of the shifted subcarrier during the bit time. The received bit is either in phase, or in phase opposition to the shifted subcarrier, depending on its binary value. As a consequence of the synchronous demodulator action, its output consists, during the bit time, of essentially positive or negative pulsations. The integrator integrates the demodulator output during every bit period. At the end of the bit time, the integrator output is squelched, producing a pulse of the bit polarity. The driving signal for the output trigger is generated in the following AND circuit. The output trigger finally reconstructs the transmitted bits of information.

Figure 2 **Signal spectrum.**



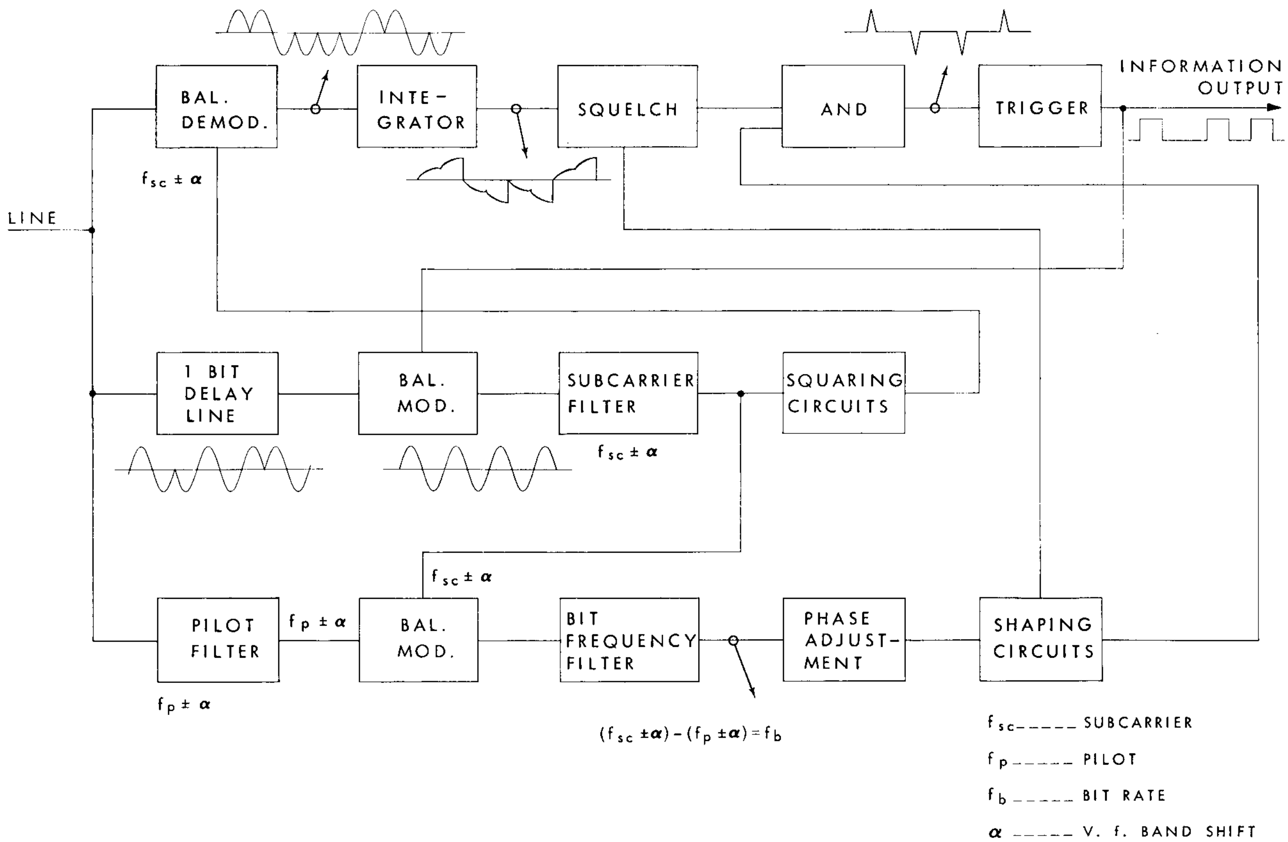


Figure 3 **Receiver.**

In order to make the synchronous detector work, two additional pieces of information have to be derived from the received signal: (1) The instantaneous phase reference for the incoming modulated bit of information, which is the shifted subcarrier; and (2) The bit-sampling time, which is derived from the bit-rate frequency.

These essential detection signals are derived as follows: The received signal consists primarily of 180° reversals of the shifted subcarrier, which is therefore suppressed with respect to its steady-state condition. In order to reconstruct the suppressed carrier from the information train, a feedback loop is used. The received signal is delayed for one bit time in order to give the synchronous detector a chance to decide if, during the bit time in consideration, the subcarrier should be left alone or shifted 180°. This information is derived from the output signal. The received signal is then left unchanged or reversed in phase, depending upon the received bit of information.

This operation takes place in a balanced modulator, the output of which contains the shifted subcarrier frequency. The subcarrier filter supplies an essentially noise-free subcarrier signal to the system. The described feedback loop is a very stable one, since the loss of eight consecutive bits would not put the system out of synchronism. The reconstructed shifted carrier is clipped, shaped and fed into the synchronous detector. It is also used for the derivation of the bit timing signal.

The bit-rate frequency or clock is derived from the shifted subcarrier and the pilot frequency. Since the frequencies of both signals are shifted by the same number of cycles per second in a suppressed carrier system, their difference stays constant and is used for clocking purposes. The pilot is filtered from the received signal by the pilot filter. The difference frequency $(f_{sc} \pm \alpha) - (f_p \pm \alpha) = f_b$ is generated in a balanced modulator. After filtering, the phase of the clock signal is adjusted to the delay characteristics of the line so that the squelching and sampling of the integrator output occurs at the end of the bit time. The shaping circuits form the proper pulse shapes for squelching and for information readout.

Judging from the available private-line characteristics, it was felt that a transmission speed of 2400 bits per sec should be realizable with simple means, if the telephone carrier systems in the link are synchronous. The subcarrier frequency was 2400 cps. At that speed a bit consisted of a single cycle of the information carrier. In a synchronous carrier system the subcarrier and the clock frequencies are identical, resulting in further simplification of the equipment. Since the experimental line was not synchronous, the transmitting clock was used for received information recovery. This was possible since closed-loop tests were made.

• *Experimental setup*

In building the experimental system the emphasis was placed on basic principles rather than on miniaturization

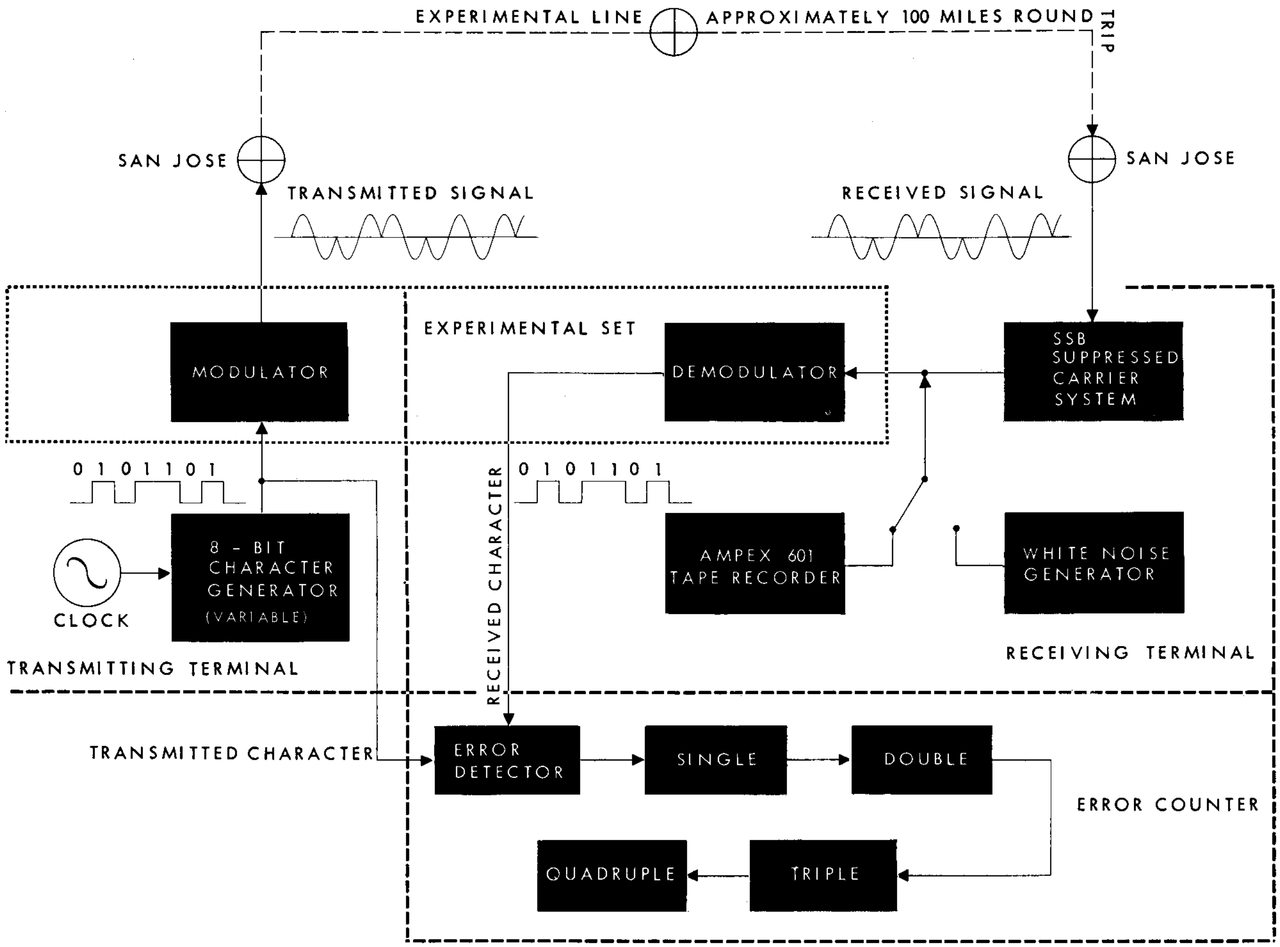


Figure 4 **Experimental setup.**

or transistorization. The system as described in this paper is a tube version and consists of 16 tubes, comprising the transmitter, receiver, and clock. Later on, the system was partly transistorized because some functions of the system could be performed better and more simply with transistor circuits. A completely transistorized version of our laboratory system consists of 15 transistors, excluding the clocking portion. It is estimated that a complete system including the clock could be built with 25 transistors.

The experimental system was designed to operate at 600, 1000, 1600 and 2400 bits per second in order to evaluate the influence of impulse noise on error rates at different speeds.

Figure 4 shows an experimental setup using a 100-mile private-line circuit having frequency and phase characteristics as shown in Fig. 5.

Satisfactory results were obtained at 600 and 1000 bits per sec with no equalization of the line. Figures 6 and 7 show the satisfactory operation of the system at 1000 bits per sec without phase equalization. For 1600 and 2400 bits per sec, phase equalization was necessary for satisfactory operation.

Figures 8 and 9 show the effect of phase equalization on a single bit at 1600 and 2400 bits per sec, respectively. The influence of phase equalization on system performance by reducing interbit interference is shown in Fig. 10.

A 25-minute tape recording was made from a line which is considered to be very noisy for data transmission purposes. The noise peak distribution (impulse noise on the tape) is shown in Fig. 11. Figure 12 illustrates the seriousness of the problem of binary data transmission with the presence of this noise. The noise pulses looked like the received signal; they were about the duration of the bit, and sometimes as high as 20 db above the signal.

Figure 13 presents the results of tests performed under white and impulse noise conditions. Zero db signal-to-noise ratio for white noise means a flat signal-to-noise ratio in the 3 kc band, whereas zero db signal-to-noise ratio for impulse noise represents normal operating conditions under the circumstances prevailing during the recording of the noise tape.

- *Test results*

In operation under simulated impulse noise conditions with our noise tape, error rates increase drastically as the duration of the signal diminishes and approaches that of

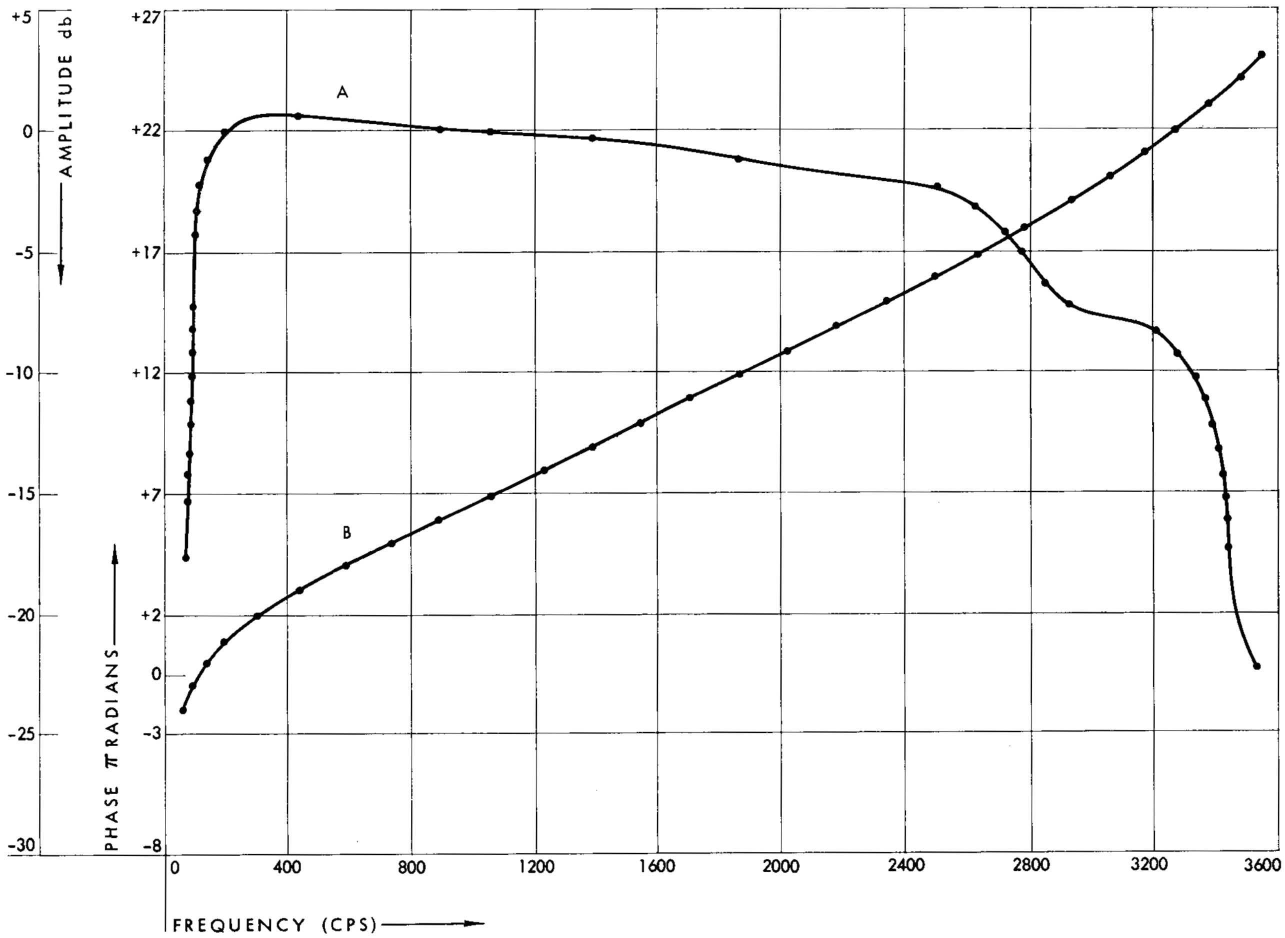


Figure 5 **Frequency (A) and phase (B) characteristics of the experimental carrier test line.**

Figure 6 **Received signal (A, C, E) at 1000 bits per second. The synchronous detection sub-carrier (B) was derived from the signal and is in phase with it. The synchronously rectified signal is integrated on a bit-by-bit basis (see Fig. 7—A, C, and E) and sampled to generate the reconstructed received binary information (D).**

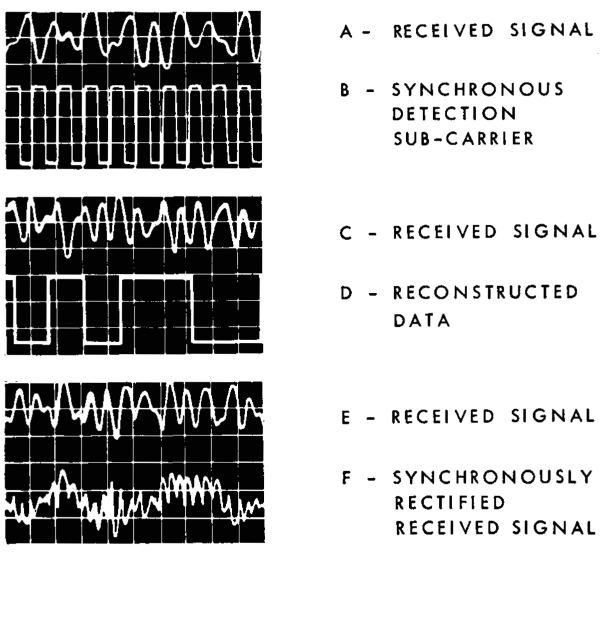


Figure 7 **Integrator output (A, C, and E) and reconstructed data (B, D, and F) at 1000 bits per second.**

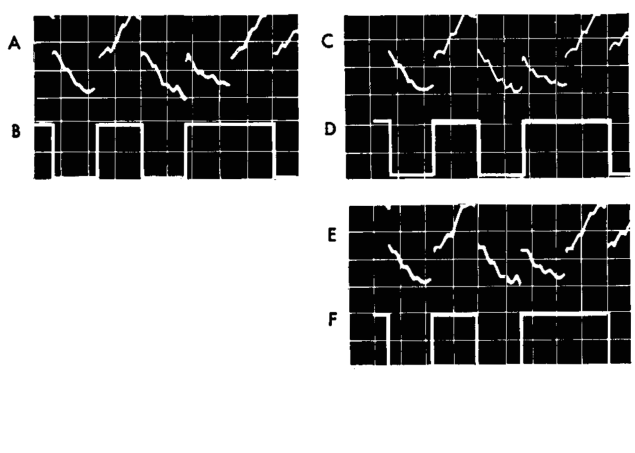


Figure 8 **Transmitted dipulse at 2400 bits per second received over the test circuit without and with phase equalization.**

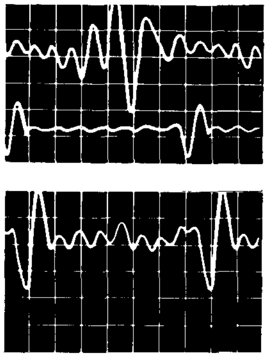


Figure 9 **Transmitted bit at 1600 bits per second (B) received over the test circuit without (A) and with phase equalization (C).**

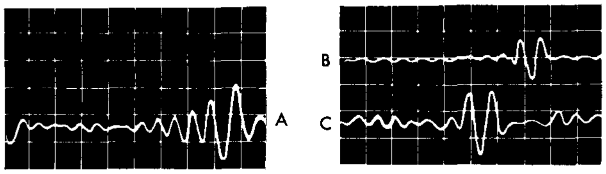
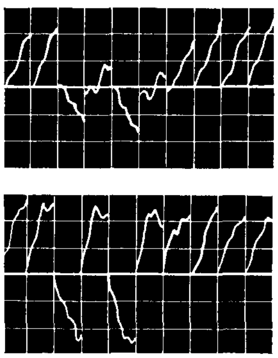


Figure 10 **Integrator output at 1600 bits per second showing considerable interbit interference without phase equalization and very little interference after the line has been phase equalized.**



impulse noise spikes. At 600 bits per sec the expected error rates are approximately one in one billion bits; this increases to one in 70,000 at 2400 bits per sec.

At 2400 bits per sec the situation was considerably worse, since the impulse noise spikes and the signal duration became of the same order of magnitude.

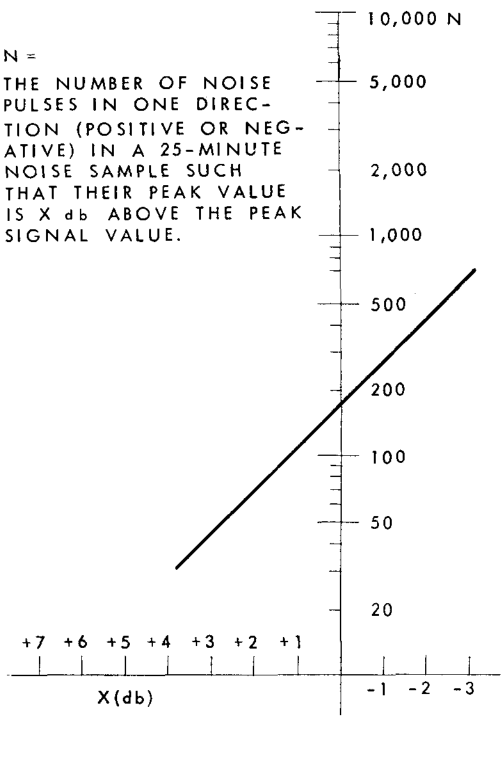
At 1000 bits per sec, one error in one million may be expected on very noisy lines. Experience to date suggests that we may expect error rates to be reduced by two orders of magnitude on average lines, as opposed to the very noisy conditions on our tape recording.

Under white noise conditions (which are important for radio circuits), at 1000 bits per sec, with a signal-to-noise ratio of 7 db, the expected bit error rate is one in one million. Our experiments indicate that satisfactory operation at 1000 bits per sec may be expected without phase equalization and at 1600 bits per sec with phase equalized lines and no special noise elimination treatment.

At 2400 bits per sec with the present experimental system, satisfactory service on private lines can only be established on lines with a noise peak distribution 6 db lower than shown in Fig. 11. Also, at the present time, such services could only be established on synchronous carrier lines and radio circuits.

The long-distance carrier network by itself does not appear to represent a serious problem for high-speed data transmission with respect to phase distortion. Excessive phase distortion for higher speed services is usually intro-

Figure 11 **Number of noise pulses, *N*, occurring on a telephone line during the twenty-five minute tape-recorded sample.**



duced, with loaded cables serving as a link between customer premises and the toll exchanges. In many cases phase correction does not represent an insurmountable problem and can be achieved with simple means and without excessive testing procedures.

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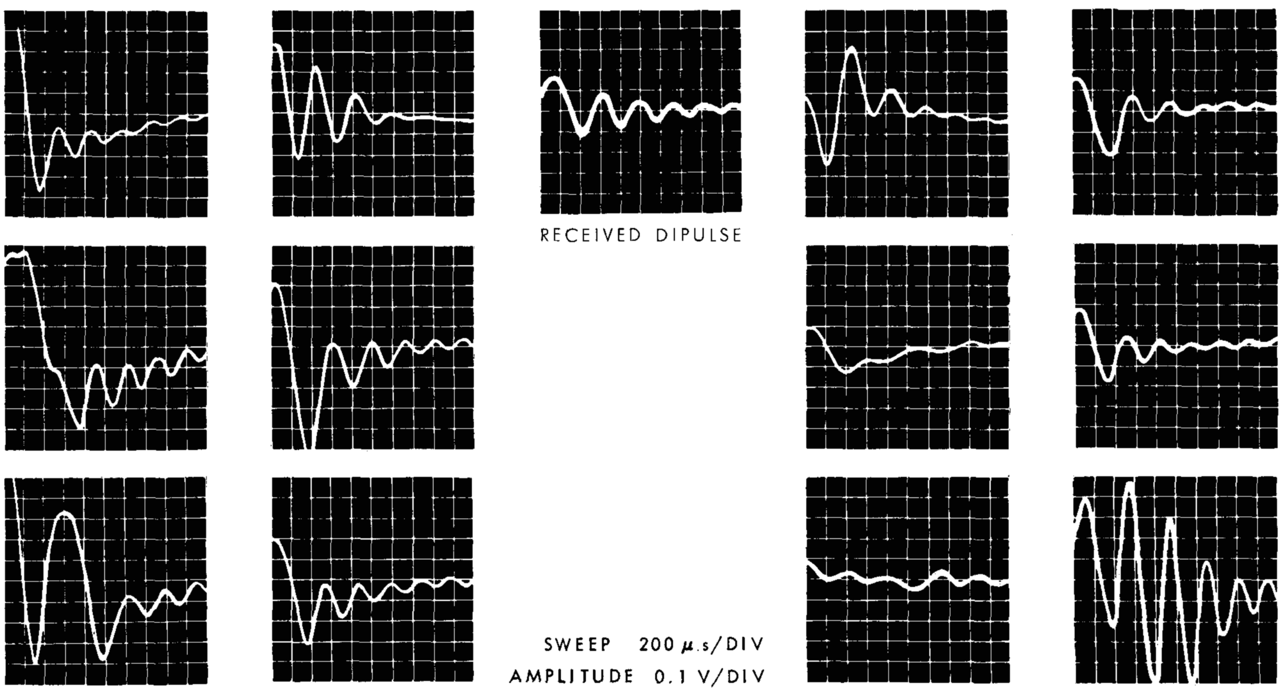
Figure 12 **A received signal without noise (center). All other waveforms are impulse noise without signal; all waveforms are shown at the same scale.**



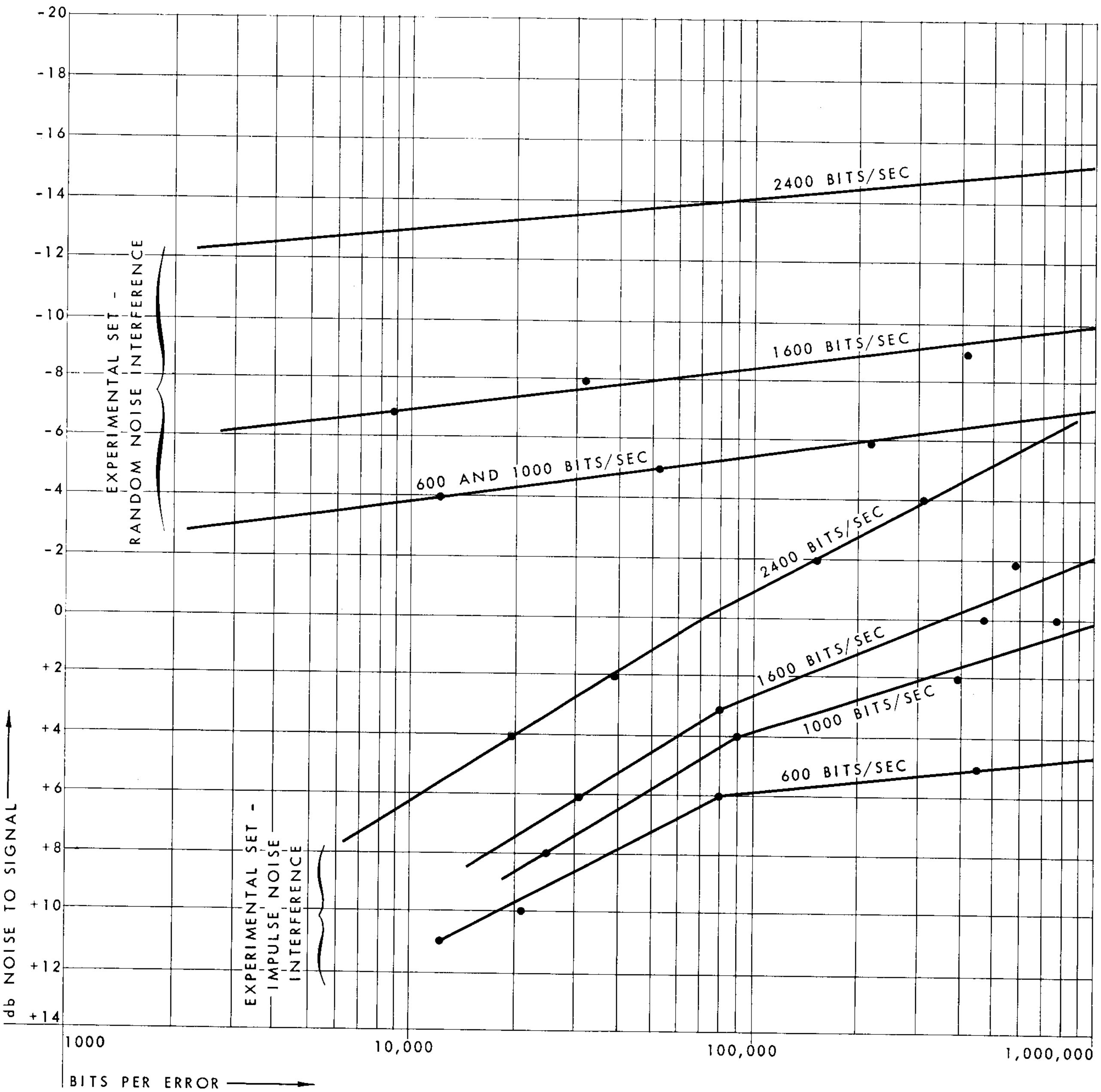


Figure 13 **Experimental data set—error rates as functions of impulse and random noise levels.**

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