GTE Communications Transmission Systems

DEMODULATOR

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Fault Locating part 3

DATACOM

This third and concluding article in our Fault Locating series describes the testing of vf channels used for data communications. Impairments which affect data transmissions are discussed and methods for measuring these impairments are described.

When a significant need for transmitting high-speed data arose in the 1950's, an extensive telephone network was already in place. Since data is easily transformed into electronic signals, it was logical to use this existing network for data transmission.

The telephone network was designed for analog transmission and optimized for voice communications. Speech is relatively low speed and inherently redundant. The human voice has a wide dynamic range and people have human intelligence to correct for transmission impairments.

High-speed digital data has a narrow dynamic range and is very sensitive to transmission impairments. Analog systems are not the optimum transmission path for digital data. So, people can communicate over transmission circuits which are virtually unusuable for high-speed data.

Services optimized for data communications are available. These services use techniques and equipment which are designed for digital transmission. Although this kind of service is growing, analog systems are still the predominant transmission path for data communications. Figure 1 is a simplified diagram of a datacom system using analog lines.

As used in Figure 1 and this article, the term "modem" refers to the equipment used at the transmit end to change the digital information to a form suitable for analog transmission. At the receive end, the same kind of equipment is used to return the information to digital form. In the telephone industry, datacom modems are called "data sets", to avoid confusion with the modulator/demodulators (modems) used for other kinds of information.

The analog parameters that most significantly effect data communica-

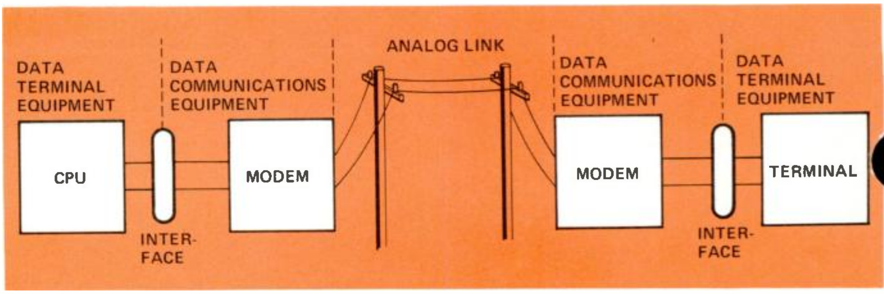


Figure 1. Datacom System

Courtesy Hewlett Packard

tions are bandwidth, transmission mode and line characteristics (impairments). The following paragraphs discuss each of these parameters in turn.

Bandwidth

Bandwidth defines the frequency range of a channel. Generally, bandwidth is defined as the difference between the highest frequency and the lowest frequency that are attenuated 3.0 dB below the average level of the band. The maximum data transmission speed is directly related to bandwidth.

Narrow band, sub-voice channels have the narrowest band-width (0-300 Hz). Transmission speeds are from 45 to 300 bits per second (bps). Narrow band is used for data only. Voice transmission is not possible.

Voice grade bandwidth (4 kHz) can accommodate data from 60 to 9600 bps. Speeds up to 19.2 kbps are possible with some modems. Two voice grade options are available, dial-up (switched) and leased.

Dial-Up

Dial-up lines use the switched telephone network. Calls are often routed through several switching offices to get to their destinations. Switches are a prime source of impulse noise.

Usually, impulse noise is a simple annoyance during telephone conversations but it can cause serious errors during data transmissions. Since a different circuit is selected each time a call is placed, the transmission path is unpredictable. This makes it impractical to condition the circuit for data signals.

The lack of conditioning limits the speed to about 4800 bps for data transmission over dial-up lines. However, higher-rates are achieved on these lines by using modems with built-in equalizers.

An advantage of the dial-up network is redundancy. If the first line is not suitable for datacom, simply hang up and dial again. The second line is almost bound to be different.

Leased Lines

A leased line is a permanent circuit for private use. The leased line is physically connected between locations, or through a central office without using the switching equipment. The same transmission path is always used so the circuit can be conditioned to improve data transmission. Data speeds up to 9600 bps can be achieved over conditioned leased lines. Line conditioning is discussed later.

Wideband lines, consisting of a number of voice channels, are available for lease. A 12 channel group has a bandwidth of 48 kHz and allows a data speed of nearly 50 kbps. Figure 2 shows the data transmission speeds for some types of data communications.

Digital Lines

Point to point digital links are most suitable for datacom. Since the entire transmission is digital, modems are not required. There are no carrier freuqencies and the signal can be completely regenerated at each repeater. These links have a lower error rate and higher data speed than analog links.

An example of a digital data link is AT&T's Dataphone Digital Service (DDS) — Registered Trademark American Telephone & Telegraph. Demand has exceeded the available circuits in areas where direct digital service is available. The limited availability is the principal disadvantage at present.

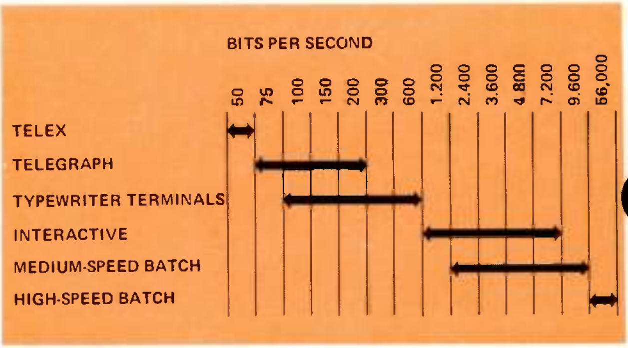


Figure 2. Transmission Speeds

Transmission Modes

The three main transmission modes for datacom are, simplex, half duplex and full duplex. Transmission on a simplex circuit is always in one direction. An input terminal can only receive. An output terminal can only transmit. Simplex is seldom used for datacom since it is not possible to send error or control signals from the receive end back to the transmit end.

Half duplex circuits can transmit and receive in both directions but not simultaneously. Half duplex is the most common transmission mode in use today.

Full duplex circuits can transmit and receive in both directions simultaneously. Full duplex provides the most efficient data transmission or "through put rate". In data communications, efficiency (through put rate) is defined as the number of usable bits received divided by the number of bits that were transmitted.

Data communications equipment (DCE) and data terminal equipment (DTE) are described by their ability to operate in one or more transmission modes. Figure 3 diagrams the three modes.

People are sometimes confused when relating duplex terminology to wire terminology. Two terms used in the telephone industry are two-wire and four-wire circuits.

Two-wire is generally used for the subscriber loop on dial-up lines. Four-wire is used for long distance calls between central offices and for leased lines. In the early days, four-wire was used to describe modems used for full duplex transmission.

Today, full duplex can be four-wire or two-wire. An equivalent four-wire circuit can be realized with a single pair by using a different frequency for each direction of transmission. Modems are available which use this principle.

Line Characteristics

As data signals pass through the telephone network, they travel through various types of transmission media and are acted upon by hundreds of components. Each type of media and component has its own characteristics. The summation of these characteristics determine the line characteristics of a particular circuit and hence its suitability for data communications. The following paragraphs discuss those line characteristics which have the greatest effect on data signals.

Loss

Circuit loss is measured end-to-end at a frequency of 1,000 Hz. The circuit is terminated in the proper impedance at both ends for this measurement. The difference bet-

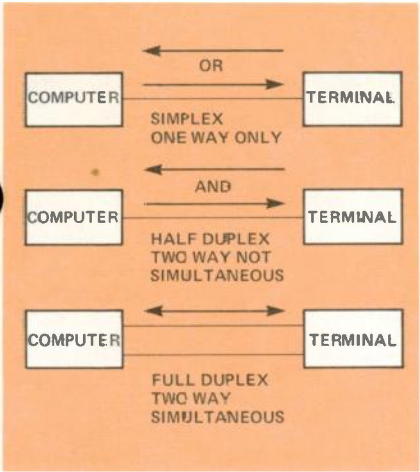


Figure 3. Datacom Transmission Modes

ween the transmitted and receive signal levels is the circuit loss.

The loss versus frequency characteristic describes variations in circuit loss as a function of frequency. If the attenuation loss across the bandwidth of a channel is not fairly uniform, a form of amplitude distortion is introduced. This is known as attenuation distortion. For a standard voice channel, it is the loss at any frequency relative to the loss at 1,000 Hz. A channel with a 6 dB greater loss at 2,800 Hz than at 1,000 Hz has 6 dB attenuation distortion.

Load coils are used to reduce attenuation distortion. Figure 4 shows the attenuation versus frequency curves for a loaded and unloaded pair.

From the curves, it is apparent that load coils decrease the attenuation and keep it substantially constant at frequencies up to about 4 kHz. However, at frequencies above 4 kHz, the attenuation of the loaded line increases rapidly and soon exceeds that of the unloaded line. This is a particular disadvantage to high speed data transmission.

Return Loss

Return loss in a circuit is a measure of energy reflection due to mismatches. It is the ratio of the reflected voltage to the incident voltage. Expressed in dB:

Return loss = 20 log Zt − Zc/Zt + Zc

Where: Zc = Characteristic impedance

Zt = Characteristic impedance

Reflections manifest themselves as echoes in telephone circuits. For voice transmissions, the "talker echo" is most troublesome because the reflections are delayed. For data transmission, "listener echo" is most troublesome in time and interfere with the desired receive signals, particularly at transmission speeds above 2400 bps. Telephone circuit reflection paths are shown in Figure 5.

Hybrids are one cause of echoes. A hybrid interfaces between a standard, 2-wire subscriber loop and the 4-wire trunk between central offices. Echoes are produced if the hybrids are not balanced (impedance matched). All dial telephone networks convert to 2-wire at the end or serving offices. They may also convert to 2-wire at intervening points.

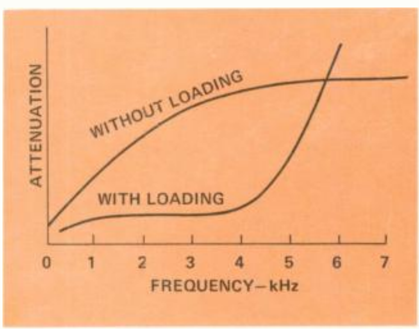


Figure 4. Attenuation Versus Frequency for Loaded and Unloaded Line

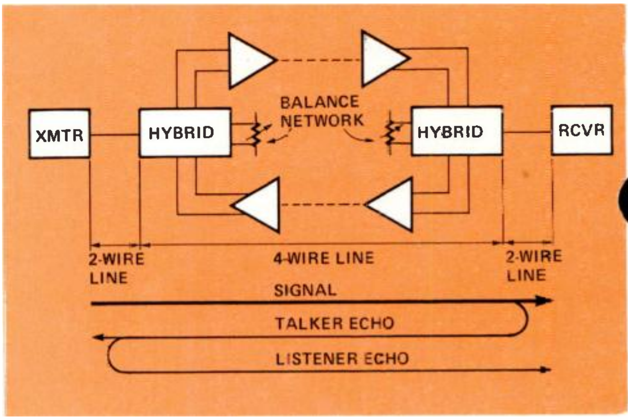


Figure 5. Echo Paths in a Typical Telephone Circuit.

Echo suppressors are often used in long distance voice circuits to suppress talker echo. During data transmission, echo suppressors must be disabled or they will prevent return communications in the half duplex mode.

Transmitting a 2000 to 2250 Hz tone for 400 milliseconds will disable an echo suppressor. This is usually done by the modem. Echo suppressors are reenabled when the carrier tone is absent for 100 milliseconds. Most modems will transmit a tone during turn-around time or when data is absent, to keep the suppressors disabled.

The disable time can be a serious disadvantage to using dial-up lines, for interactive data communications in the half duplex mode, because each interaction requires two disable delays. Some modems can turn-off both echo suppressors at the beginning of a conversation. Echo suppressors are not used on leased lines.

A device which could replace echo suppressors is undergoing extensive testing. The device is known as an Echo Canceller.

Adaptive echo cancellation is accomplished by replicating the echo path response and using an involved method for combining the replica with the incoming signal. The resulting echo estimate is subtracted from the echo path output.

Transmission is not interrupted in any way. Full, two-way information transmission can be accomplished with little or no echo.

Echo cancellers could have substantial application to full-duplex data transmission over the dial-up, switched network. Cancellers would be installed in the modems. Only one circuit would have to be dialed-up instead of two as generally required at present.

Since they are very-large-scale, integrated circuit devices on a single chip, cancellers are relatively inexpensive and do not require much space. The subject is discussed more thoroughly in a paper by Sondhi and Berkeley, published by the IEEE (see bibliography).

Envelope Delay

Envelope delay is a measure of the phase versus frequency response of a voice channel. Relative envelope delay is the difference between the delay at a given frequency and the delay at a reference frequency.

The propagation time along a pair of wires varies with frequency. This is equivalent to a relative phase shift.

If the shift is non-linear, distortion will occur.

The main reason why propagation time varies with frequency is the capacitance between wires. This capacitance and consequently the phase distortion can be increased by unterminated bridge taps.

A bridge tap is made when an installer bridges across a cable pair to bring service to a customer's premises. When the service is disconnected, the customer loop is often cut and the loose ends are left bridged across the cable. A large number of these "danglers" can substantially increase phase distortion. Switching a microwave path to an alternate path, either microwave or cable, can also produce phase distortion and dropouts.

Load coils also contribute substantially to delay distortion. The effect is noticeable at frequencies well below 4 kHz. The resulting distortion is one of the main causes of trouble in data transmission.

Phase distortion does not significantly affect voice transmission. However, it can seriously affect data. The effect of relative envelope delay is to cause the data bits to smear out in time and overlap each other. This is known as intersymbol interference. It is particularly troublesome at transmission speeds above 2400 bps over unconditioned voice channels.

Equalizers can be added to voice grade lines to improve the attenuation and delay response across the channel bandwidth. Equalizers are usually installed on conditioned lines.

Some modems have built-in equalizers. Faster data speeds can be used on unconditioned lines by using these modems. Their auto-equalizers are good at tracking and compensating for slow changes in line characteristics. However, fast changes frequently cause the modem to lose synchronization and data for a few seconds.

Frequency Shift

Frequency shift is the difference between the transmitted frequency and the received frequency. It is often caused by a lack of synchronization of the carrier reinsertion oscillators in single-sideband, suppressed-carrier, frequency-division multiplexed systems.

Carrier pilots are used to minimize frequency shift. The pilot is transmitted along with the signal. It is the fundamental frequency used to generate carrier frequencies at both ends of the system. Telephone company standards call for ± 5 Hz maximum frequency shift end-to-end. Frequency shift degrades data transmission because it changes the relationship between the signal carrier and sidebands.

Non-Linear Distortion

Non-linear distortion results from the introduction of frequencies not present in the original signal. Harmonic distortion and intermodulation distortion are two of the effects of non-linear distortion.

Harmonic distortion produces frequencies which are integer multiples of the original frequencies. If f_1 and f_2 are the original frequencies, harmonic distortion through the third order will produce frequencies at $2f_1$, $3f_1$, $2f_2$ and $3f_2$. Intermodulation distortion produces frequencies equal to the sum and difference products of the input frequencies and their harmonics:

$f_1 + f_2$, $f_2 - f_1$, $2f_1 + f_2$, $2f_1 - f_2$, $2f_2 + f_1$ and $2f_2 - f_1$.

Amplifiers, modulators, demodulators and compandors are non-linear devices. Non-linear refers to the fact that their output is not linear with respect to their input. See Figure 6. The flattening effect on the sine wave is equivalent to adding low amplitude signals at frequencies which are harmonics of the original.

Repeaters are essentially amplifiers which compensate for line attenuation. They sometimes cause non-linear distortion. Modems contain amplifiers and detectors which are inherently non-linear and contribute to distortion. Compandors also contain amplifiers which may cause non-linear distortion.

Compandors may have another adverse effect on data. If amplitude modulated data signals are passed through a compandor, some of the pulses may be badly distorted, due to the varying loss characteristic.

The principal cause for non-linear distortion in PCM systems is the quantizing process. This process uses a logarithmic compression law to provide more steps per volt for small speech samples than for large samples. This action maintains a substantially constant signal to quantizing distortion ratio over a wide

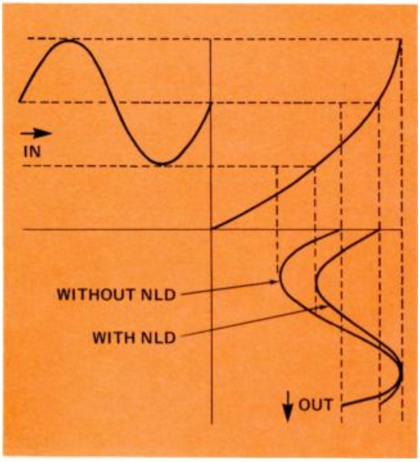


Figure 6. Effect of Non-Linear Distortion

dynamic range but the devices which produce it must be non-linear. Therefore, they produce non-linear distortion.

Steady Noise

White noise or steady noise is always present in a communications system. It is the result of thermal action in amplifiers and active components. It also is a result of radio frequency interference, crosstalk and various other interferences. A high level of noise is quite annoying in voice transmission but more serious in data transmission. A poor signal to noise ratio at the receive modem will cause a high error rate. Compandors are effective in improving the signal to noise ratio but penalties are incurred in the increased possibility of non-linear distortion.

Phase Jitter

Phase jitter is also referred to as incidental fm. Although it is relatively unimportant for voice transmission, phase jitter can seriously affect data transmission.

Phase jitter causes variations in the zero-crossings of data pulses. These variations may result in the pulses moving into time slots allocated for other data pulses.

A principal cause of jitter is power supply ripple appearing in the master oscillator signal of long-haul systems and being multiplied by many stages. In short-haul systems, insufficient filtering of image sidebands may produce phase jitter. In digital carrier systems, the 20 Hz ringing and 60 Hz power frequencies plus their harmonics are causes of jitter.

Transients

Four main types of transient impairments are; impulse noise, gain hits, dropouts and phase hits. Common causes of these impairments are switches, microwave path fading and noise transients.

Impulse Noise

The commonest cause of impulse noise is the action of electro-mechanical switches and relays. Although these devices are being replaced by solid state units — a large number, if not the majority, of telephone offices still use step-by-step electro-mechanical switches. Other causes of impulse noise are installation and maintenance activities and weather disturbances (lightning).

Bell Laboratory's studies show that impulse noise spikes have a duration of less than one millisecond and all significant effects are dissipated within four milliseconds. On voice circuits the effects are simply irritating "clicks" and "pops".

On high-speed data circuits, a single noise pulse can wipe out many data bits. Also, it is difficult for the receiver to distinguish between noise and data pulses so errors occur. At slow data rates, it is easier for the receiver to distinguish between data pulses and noise pulses so fewer errors occur.

The width of a noise pulse may closely approximate the period of a data pulse on a high speed system. The noise pulse will look like data to the receiver.

The noise pulse energy is averaged over a longer data pulse period in a low speed system so, the energy may not be sufficient to overcome the zero-one threshold. Also, the low-pass characteristics of slow speed systems reject the higher frequency energy portion of the noise pulse.

Gain Hits

Gain hits are sudden increases or decreases in receive signal level. By definition a gain hit is a level change less than 12 dB and lasting longer than four milliseconds. In fact, gain hits often last for periods of time measured in hours. A gain hit will look like data to modems that use amplitude modulation.

Dropouts

Dropouts are defined as signal decreases greater than 12 dB and lasting longer than four milliseconds. Dropouts interrupt the signal flow and data is lost. When the signal is restored, some modems require additional time to reequalize, so more data is lost.

Phase Hits

Phase hits are a sudden, large change in the receive signal phase or frequency, lasting longer than four milliseconds. Many high speed datacom systems use phase or frequency modulation. Errors are caused in these systems by phase hits because they look like data.

Phase hits may occur when switching from one carrier supply to another or when switching to alternate transmission facilities with different propagation times. These changes may cause all data to be in error until the out-of-phase condition is corrected.

Phase hits can also occur in space diversity microwave systems when the receivers are switched. Differential absolute delay equalization can prevent this occurrence.

Conditioning

Telephone companies offer several types of line conditioning to provide higher data rates and/or reduce data errors. Conditioning is usually required for speeds above 2400 bps, although speeds up to 4800 bps are sometimes possible without it. Conditioning requirements are a function of speed and distance. Even if conditioning is not required, it can give a modem a safety margin to decrease its vulnerability to transmission impairments.

Five levels of voice grade conditioning for data transmission are available from telephone companies. These conditioning levels are tariffed under FCC Tariff number 260. Figure 7 lists various parameters for the different levels.

Tariff No. 260 provides a basic voiceband channel (3002) for data and facsimile transmission and five types of C-conditioning. Some phone companies also offer D-conditioning for high speed data, such as 9600-bps operation.

C-conditioning is the name given to the circuit treatment furnished by a telephone company to make voice-grade lines meet Tariff No. 260 specifications. The specifications apply only to the envelope and attenuation delay characteristics of the line. The removal of loading coils and addition of other filter attachments may be included in the circuit treatment, particularly for short haul modems. This is known as B-conditioning.

[1] CHANNEL CONDITIONING	ATTENUATION DISTORTION (FREQ. RESPONSE RELATIVE TO 1004 Hz		ENVELOPE DELAY DISTORTION	
	[2] FREQUENCY RANGE (Hz)	[4] VARIATION (dB)	[2] FREQUENCY RANGE (Hz)	VARIATION (μSECONDS)
BASIC	[3] 500-2500 [3] 300-3000	−2 TO +8 −3 TO +12	[3] 800-2600	1750
C1	1000-2400 300-2700 [3] 300-3000	−1 TO +3 −2 TO +6 −3 TO +12	1000-2400 [3] 800-2600	1000 1750
C2	500-2800 300-3000	−1 TO +3 −2 TO +6	1000-2600 600-2600 500-2800	500 1500 3000
C3 ACCESS LINE	500-2800 300-3000	−0.5 TO +1.5 −0.8 TO +3	1000-2600 600-2600 500-2800	110 300 650
C3 TRUNK	500-2800 300-3000	−0.5 TO +1 −0.8 TO +2	1000-2600 600-2600 500-2800	80 260 500
C4	500-3000 300-3200	−2 TO +3 −2 TO +6	1000-2600 800-2800 600-3000 500-3000	300 500 1500 3000
C5	500-2800 300-3000	−0.5 TO +1.5 −1 TO +3	1000-2600 600-2600 500-2800	100 300 600

NOTES:

[1] C-CONDITIONING APPLIES ONLY TO THE ATTENUATION AND ENVELOPE DELAY CHARACTERISTICS.

[2] MEASUREMENT FREQUENCIES WILL BE 4 Hz ABOVE THOSE SHOWN. FOR EXAMPLE, THE BASIC CHANNEL WILL HAVE −2 TO +8 dB LOSS, WITH RESPECT TO THE 1004 Hz LOSS, BETWEEN 504 AND 2504 Hz.

[3] THESE SPECIFICATIONS ARE NONTARIFFED ITEMS, ALL OTHERS ARE TARIFFED.

[4] + MEANS LOSS WITH RESPECT TO 1004 Hz.
− MEANS GAIN WITH RESPECT TO 1004 Hz.

Figure 7. C-Conditioning Parameters

C-conditioning on multipoint channels applies to the complete channel; a mixture of C-conditioned and basic segments, or different grades of conditioning on the same channel, is not permitted. The basic channel and channels with C1 and C2 conditioning may be ordered in two-point, multipoint and switched configurations.

C3 conditioning applies only to private switched networks and is specified in terms of access lines and trunks. The intent of C3 conditioning is to provide C2 conditioning end-to-end on most switched connections involving a maximum of four trunks and two access lines in tandem. C3 conditioning is used on common control switching arrangements (CCSA) or switched circuit automatic networks (SCAN).

C4 conditioning can be ordered for two, three, or four-point operation only. C5 conditioning can be ordered two-point only. The intent of C5 conditioning is to achieve C2 conditioning end-to-end on multiple link connections. Its principal use is for two-point private lines between customer control points in networks which extend overseas.

The limits for attenuation and envelope delay distortion on a basic, unconditioned, 3002 channel are controlled by telephone company practices, not by tariff. Limits for the different levels of C-conditioning are shown in Figure 7. Each successive level provides tighter specifications on attenuation and envelope delay distortion.

C-conditioning can be provided for various configurations of equipment and channels. However, a mixture of C-conditioned and basic line segments, or different grades of conditioning on the same channel, is not permitted, except for use of C3 and C5 in access lines for special applications.

The degree of conditioning required depends on the desired error performance and bit rate. Modem manufacturers usually specify the necessary grade of conditioning in their performance data. Some modems have automatic equalizers which compensate for attenuation and delay distortion. Therefore, C-conditioning may not help their performance very much and may even increase their error rate. This is because an unconditioned, or lightly conditioned line usually has a smooth envelope delay curve (Figures 8 and 9). The delay curves for a C2 and C4 conditioned line may have steep slopes and ripples (Figures 10 and 11) which make it more difficult for an automatic equalizer to operate properly.

D-conditioning is an option introduced by telephone companies to handle 9600-bps operation. Either D or C, or both types of conditioning, can be provided on the same 3002 channel to limit impairments. The C-conditioning controls attenuation and envelope delay distortion. The D-conditioning controls signal-to-noise ratio and non-linear (harmonic) distortion. A high-speed data set with automatic equalizer may perform best on a 3002 facility with D-conditioning only.

Such channels meet the following specifications:

Signal-to-C-Notched noise ratio	28 dB minimum
Fundamental-to-second harmonic ratio	35 dB minimum
Fundamental-to-third harmonic ratio	40 dB minimum

Figure. 8 Envelope Delay Distortion Limits — 3002 Unconditioned Channel

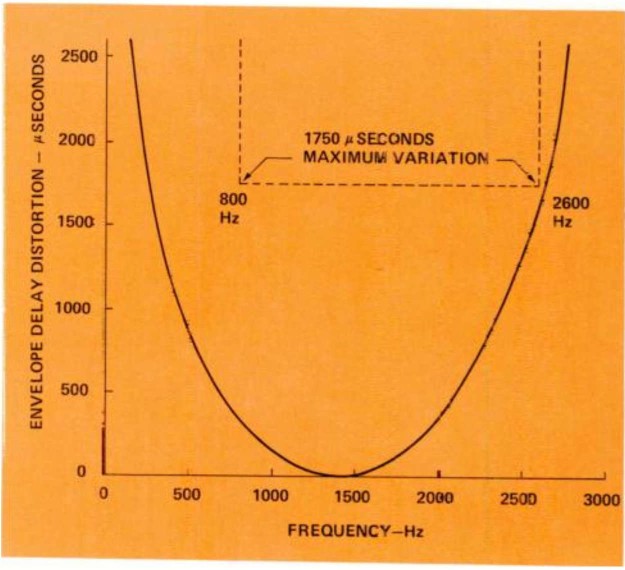
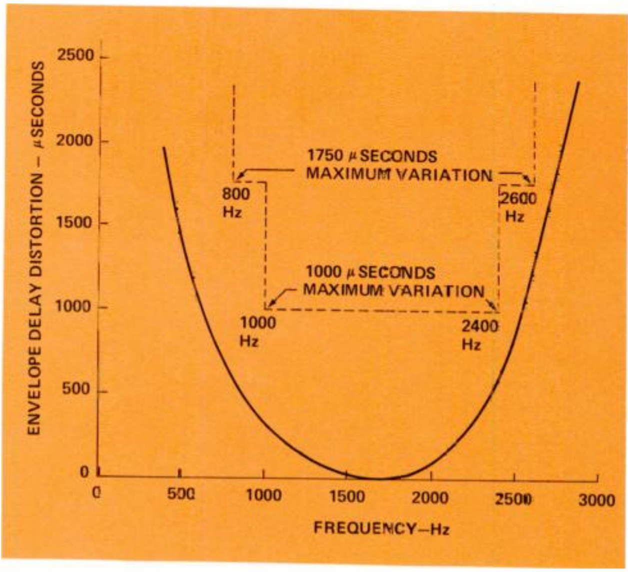


Figure 9. Envelope Delay Distortion Limits — C1 Conditioned Channel



Measurements

So far our discussion has been primarily concerned with identifying various transmission impairments and describing their effects on data transmission. The measurement of these impairments is discussed in the following paragraphs.

Loss Measurements

Loss is essentially a measure of the attenuation between two points. Loss is measured by transmitting a signal of known strength and frequency from point A and measuring the amount of signal received at point B. Telephone company practise is to

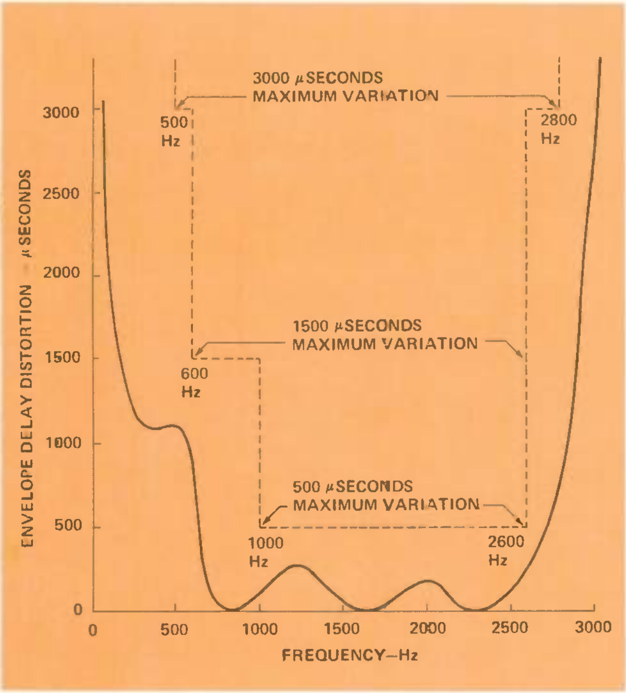


Figure 10. Envelope Delay Distortion Limits — C2 Conditioned Channel

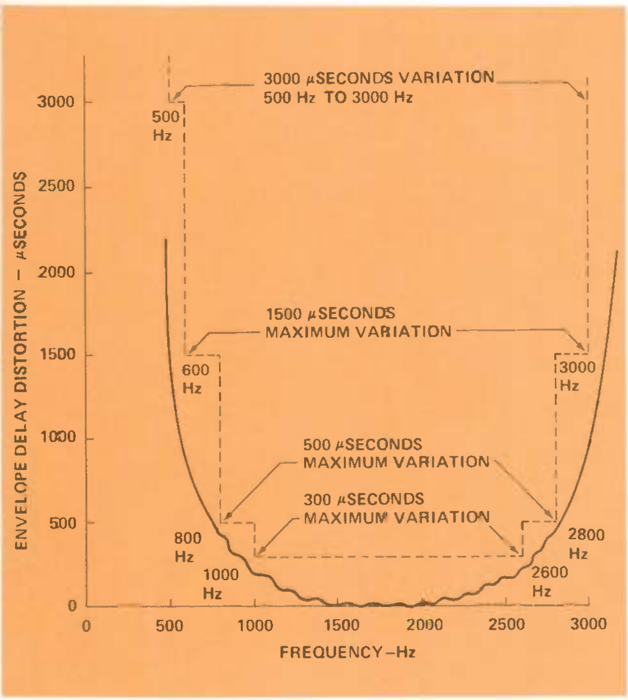


Figure 11. Envelope Delay Distortion Limits — C4 Conditioned Channel

measure loss at 1,000 Hz and with the circuit properly terminated at both ends. That is the oscillator output impedance must be matched to the line and the line must be terminated in a nonreactive load equal to its characteristic impedance (usually 600 ohms).

In actual practise the test signal is 1,004 Hz transmitted at data level. The 4 Hz offset avoids test frequencies wich are submultiples of the T-carrier sampling rate. The accuracy of the measurement is also affected by the receive end termination. A resistor with a 1% tolerance is recommended.

The meter used should be calibrated to read across the terminating impedance. As with all terminated measurements, care should be used to avoid measuring across a double termination which will make the signal appear 3 dB down from its actual value.

Attenuation Distortion

Attenuation distortion can be measured with the same test set-up used for loss measurements. The oscillator frequency is varied in convenient increments. The loss is computed for each incremental frequency and plotted against frequency to provide a curve showing loss against frequency.

This measurement can be greatly simplified by using a test set specifically designed for this purpose. These sets contain a frequency swept oscillator which eliminates the necessity for incremental settings. The more sophisticated models have an oscilliscope calibrated in dBm which automatically plots the response curve to eliminate the necessity for calculations and plotting.

Frequency Shift

Frequency shift is measured by transmitting a test tone at a known frequency and measuring the frequency at the far end. Any difference in transmitted and received frequency is a frequency shift.

Frequency shift testing must be done end-to-end. Looped-back tests are not valid because the frequency shift in one direction may be cancelled by the shift in the other direction.

Return Loss

A method for measuring return loss is diagrammed in Figure 12. As shown in the figure, a signal is applied through a hybrid to the two-wire circuit under test. A level meter is connected to the other hybrid port. The hybrid is carefully balanced and the hybrid losses are compensated by calibrating the meter.

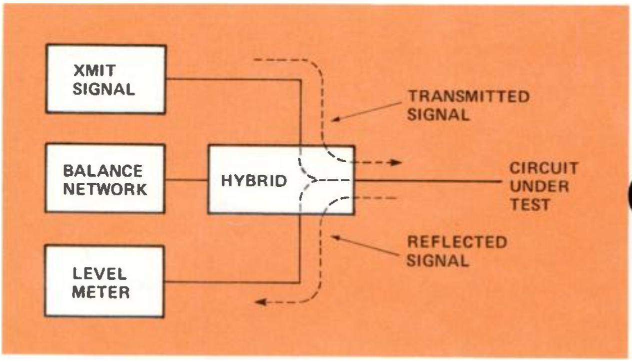


Figure 12. Typical Test Arrangement for Measuring Return Loss

The return loss test is made by setting the signal generator to discrete frequencies within the band of interest and observing the meter readings. The meter reading is the reflected energy at the frequency of the signal generator. The use of a time domain reflectometer to measure return loss is described in the May/June 1980 issue of the Demodulator.

Envelope Delay

Amplitude modulating a carrier frequency (f_c) with a lower frequency sine wave (f_m) produces the familiar waveform shown in Figure 13.

The envelope is the outline of the peak excursions of the AM signal. The frequency spectrum of the AM signal is shown in Figure 14. It consists of the carrier frequency f_c, a lower sideband $f_c - f_m$ and an upper sideband $f_c + f_m$. Because it is at a higher frequency, the upper sideband has a greater phase shift than the carrier; the lower sideband has less phase shift because it is at a lower frequency.

The phase shift difference between the upper and lower sidebands causes the envelope to undergo a phase delay during transmission of the AM signal. If the transmission circuit does not have any phase distortion (i.e. its phase shift is linear with frequency) the delay for a constant f_m will also be constant as f_c is changed. This is true because the frequency difference between sidebands is constant for a constant f_m.

Figure 15 shows the phase shift and resultant envelope delay for a circuit with a linear phase characteristic. Figure 16 shows the same information for a circuit with a non-linear phase characteristic.

One technique for measuring envelope delay is shown in Figure 17. A fixed modulation (83 1/3 Hz) is used to amplitude modulate a carrier

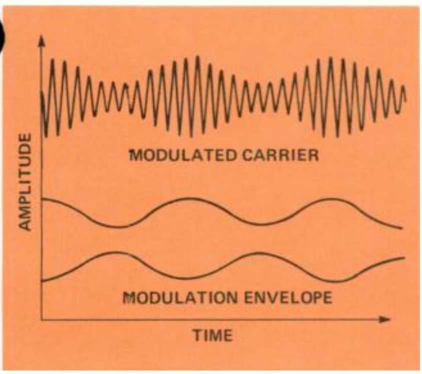


Figure 13. Amplitude Modulated Signal

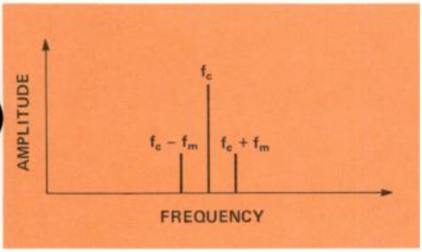


Figure 14. AM Signal Frequency Spectrum

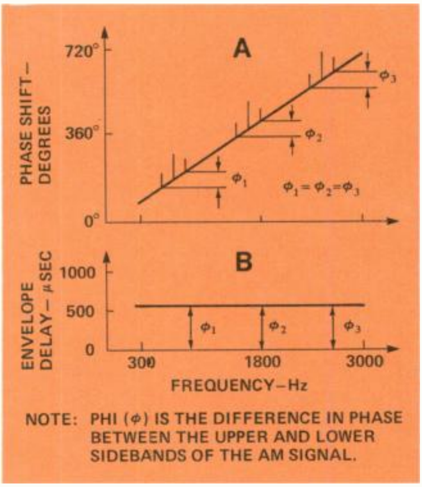


Figure 15. (A) Linear Phase Characteristic with Superimposed AM Signal Components (B) Envelope Delay Characteristic of A

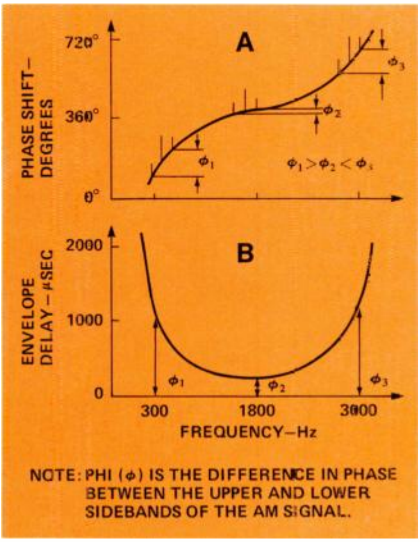


Figure 16. (A) Non-linear Phase Characteristic with Sueprimposed AM Signal Components (B) Envelope Delay Characteristic of A

which is varied across the channel bandwidth (300 - 3004 Hz). This signal is applied to the channel under test.

At the receive end, the signal is demodulated to recover the envelope information. The recovered envelope is used to modulate a fixed carrier (1800 Hz). This AM signal is transmitted back to the origin over a return channel. Any delay incurred in the return path will be constant and can be zeroed out.

The phase of the return envelope is then compared to the phase of the original envelope. Envelope delay is the maximum difference, in microseconds, of the envelope delay between any two frequencies. The commonly used reference is 1,800 Hz. An envelope delay of 200 microseconds at 3000 Hz means that the delay at 3000 Hz is 200 microseconds greater than the delay at 1,800 Hz.

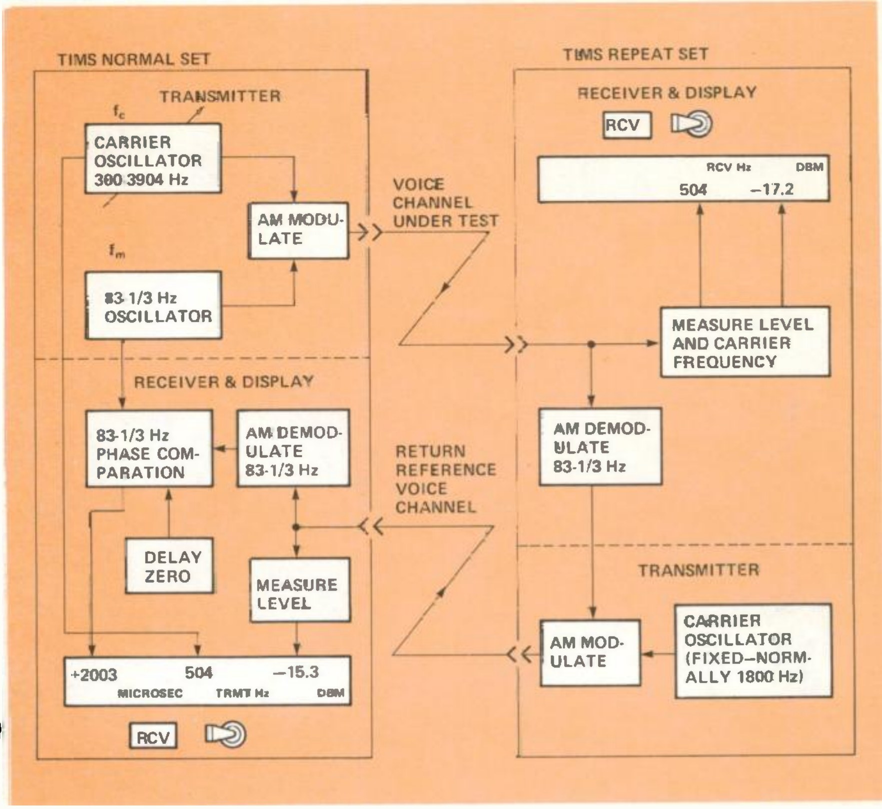
Non-Linear Distortion

The NLD measuring technique commonly used measures intermodulation distortion. The technique is licensed under Hekimian Laboratories U.S. Patent Number 3862380. Two frequency pairs are transmitted over the circuit under test. One pair is centered around 860 Hz and the other is centered around 1,380 Hz (see Figure 18). These frequencies were selected to closely approximate the non-linear distortion usually encountered by data signals while minimizing other interference and avoiding inaccurate readings from 8 kHz PCM systems.

At the receive end, the power present at the sum and difference frequencies (2,240 and 520 Hz) is measured and compared to the total receive power to determine the second order distortion. The power present at 1,900 Hz ($2f_2$ - f_1) is measured and compared to the total receive power to determine the third order distortion. As stated in the discussion of D-conditioned channels, second order distortion must be at least 35 dB down and third order distortion must be at least 40 dB down.

An older method for measuring nonlinear distortion measures harmonic distortion. A single frequency, typically a 704 Hz tone, is transmitted. At the receive end, the second harmonic (1, 408 Hz) and third harmonic (2,112 Hz) received power are compared to the total received power. On ordinary, non D-conditioned, lines the minimum ratios are 25 dB for the second and 30 dB for the third harmonic.

The intermodultion and harmonic distortion techniques will yield the same results for second and third order distortion, if only one source of distortion is involved. However,



Courtesy Hewlett Packard

Figure 17. Envelope Delay Measurement

most telephone channels have multiple distortion sources which create problems in obtaining valid measurements of a particular distortion type. Laboratory studies show that the intermodulation distortion technique is least affected by these problems.

Noise Measurement

Noise power is measured by an rms detector. A weighting filter is used so that the only noise signals measured are those which are annoying to the subscriber. There are five filters currently in use:

C-Message
3 kHz Flat
Program
15 kHz Flat
Psophometric

Figures 19 through 23 show the response curves for these filters. Figure 24 shows the response curve for a notched C-Message filter. Figure 25 shows the basic noise measurement set-up.

The C-Message filter was developed to measure noise that is most objectionable on an ordinary voice circuit. It is not particularly suitable for data transmission, although it is reasonably flat from 600 to 3000 Hz, which is an important band for data transmission. C-Message filters are often used in

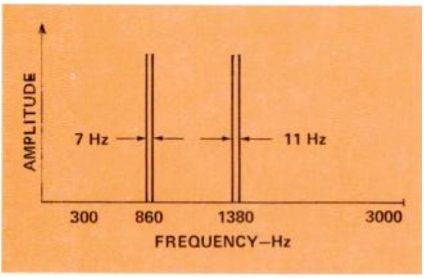


Figure 18. Non-linear Distortion Transmit Signal Frequency Spectrum

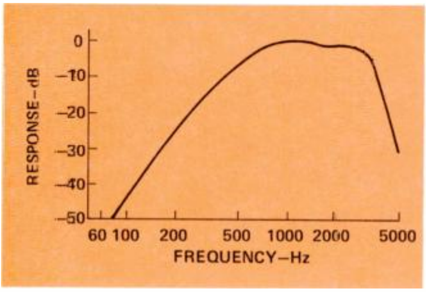


Figure 19. C-Message Filter

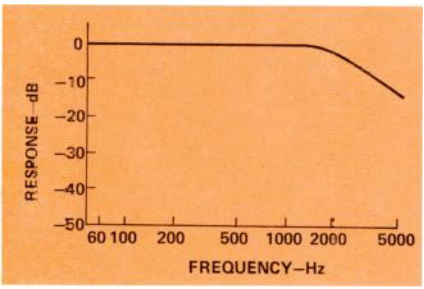


Figure 20. 3 kHz Flat Filter

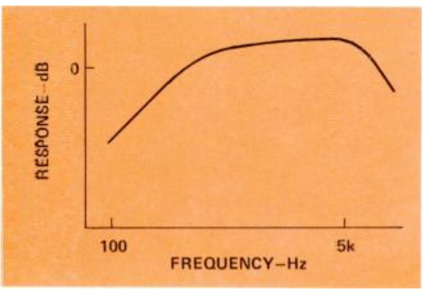


Figure 21. Program Filter

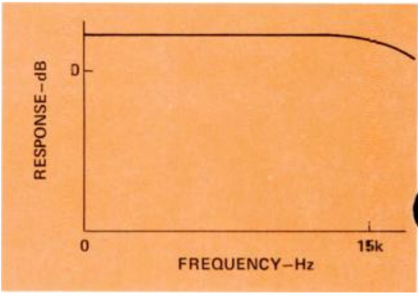


Figure 22. 15 kHz Flat Filter

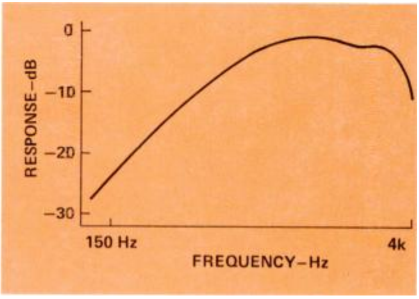


Figure 23. Psophometric Filter

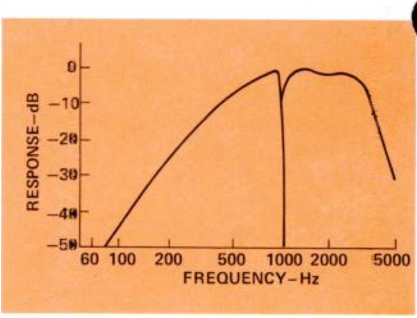


Figure 24. C-Message Filter with Notch Characteristics

conjunction with 3 kHz Flat filters to help identify a characteristic noise frequency.

The 3 kHz Flat filter has about the same frequency response as a modem. In contrast to the C-Message, the 3 kHz Flat filter provides much less attenuation to low frequencies — so the low frequency noise (20 Hz ringing, 60 Hz power

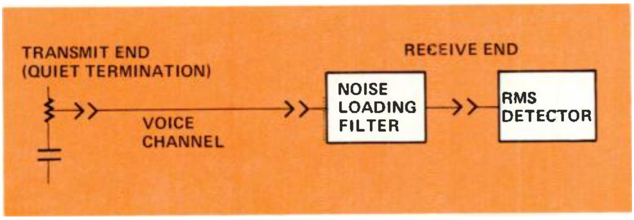


Figure 25. Basic Noise Measuring Set-up

etc.) can be measured. Noise sources have characteristic frequency bands. So, repeating measurements, using different filters, can often help identify a noise source.

The program filter is used for weighted measurements on program channels. Its principal application is on broadcast studio to transmitter links.

The 15 kHz Flat filter is also used on broadcast industry program channels. Like the 3 kHz Flat filter, the 15 kHz Flat filter's pass-band includes very low frequencies.

Some noise sources are only active while a signal is being transmitted. A tone is transmitted to include these sources' contributions in the noise measurement. A C-Message filter with a notch filter is used for these measurements.

The notch filter provides high attenuation of a narrow band (995 to 1,025 Hz). It is used to remove the test tone before the signal is applied to the noise power detector.

The signal-to-noise ratio can be computed by comparing the signal level of the tone, measured with a C-Message filter without the notch, to the noise measured with the notch filter. Since the noise is present with the tone, what is actually compared is the tone plus noise to the noise. However, the tone level is much higher than the noise level. So, the noise is not a significant portion of the tone plus noise measurement.

The signal to noise ratio is a good quick check of channel quality. If the ratio has decreased, other weighted checks can be used to isolate the cause.

Transient Noise

The ways different types of transient noise effect a receive holding tone are shown in Figure 26. Methods for measuring these disturbances are discussed in the following paragraphs.

Impulse Noise

Impulse noise is measured by counting the number of times the noise spikes exceed a preset threshold level. Telephone company standards specify a threshold level 6 dB below the receive signal level. A maximum of 15 pulses above threshold is allowed during any 15 minute interval.

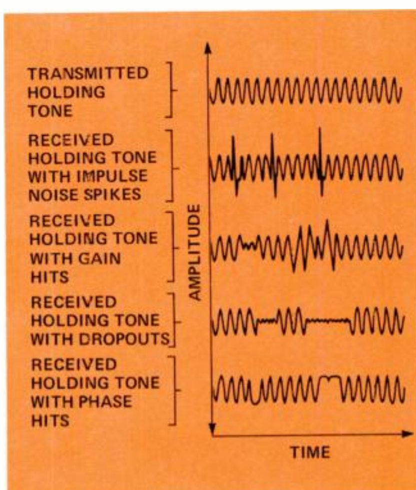


Figure 26. Impulse Noise, Hits and Dropouts Waveforms

The count rate is 7 counts per second.

The threshold level is used because not all noise disturbances have sufficient amplitude to cause problems. The count rate was originally set by the performance capability of electromechanical counters. Today's electronic counters are capable of much higher rates but years of experience and laboratory tests show the 7 per second rate yields valid data.

Gain Hits

Gain hits are sudden decreases or increases in signal amplitude that last at least 4 milliseconds. By definition the level change does not exceed 12 dB relative to the nominal receive signal level. Telephone company standards allow no more than 8 gain hits, greater than 3 dB of the received signal, in a 15 minute interval.

Dropouts

Dropouts are actually a severe form of a gain hit. A dropout occurs when the signal level decreases more than 12 dB for a period greater than 4 milliseconds. Dropouts cause more errors than any other transient. Consequently, dropouts have the tightest specifications. Telephone company standards specify no more than 1 dropout every 30 minutes.

Phase Hits

A phase hit is a suddent change in the received signal phase (or frequency) of 20 degrees or more and lasting at least 4 milliseconds. Telephone company standards allow no more than 8 phase hits in any 15 minute interval.

Phase hit measuring techniques are similar to impulse noise measurements. A phase "threshold" replaces the level threshold used for impulse noise. Phase hit measurement is complicated by the presence of impulse noise.

A solution to this problem is to use a test set which measures the transient phenomena simultaneously and is able to identify coincident situations. The set should also be able to sort out, and not count, events that occur only as a result of other phenomena.

Figure 27 is a simplified diagram of one such test set, the Transmission

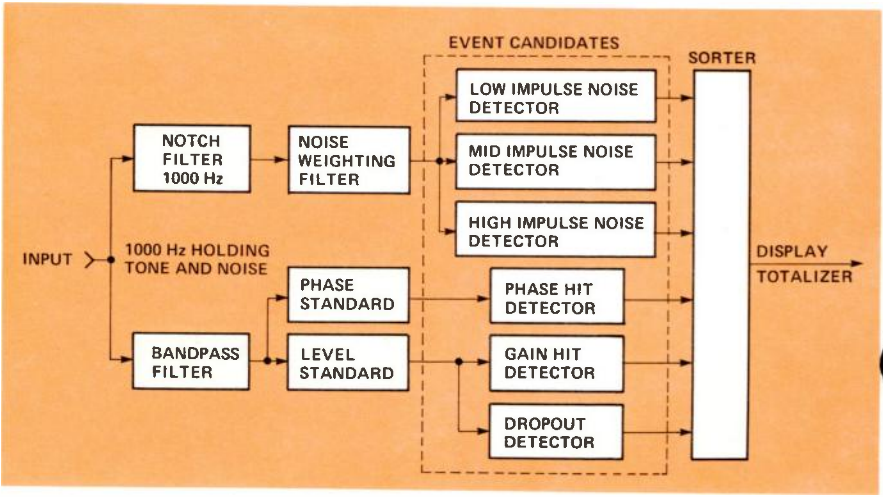


Figure 27. Impulse Noise, Hits and Dropouts Measurement

Courtesy Hewlett Packard

Impairment Measuring Set (TIMS) manufactured by Hewlett Packard. Other figures in this article, identified as "Courtesy of Hewlett Packard" are taken from publications describing HP's TIMS and Data Line Analyzers.

Phase Jitter

The effects of phase jitter on a holding tone are shown in Figure 28. Telephone company standards for phase jitter call for no more than 10° between 20 and 300 Hz and no more than 15° between 4 and 20 Hz.

A 1,004 Hz holding tone is the transmit signal used to measure phase jitter. Figure 29 shows the TIMS measuring configuration at the receive end.

The receive signal passes through a band pass filter. The output of the filter is a signal with a bandwidth approximately one-fourth of the channel bandwidth. The measurement bandwidth is centered on the test tone frequency. The purpose of the filter is to reduce the effects of noise and other interference on the jitter measurement.

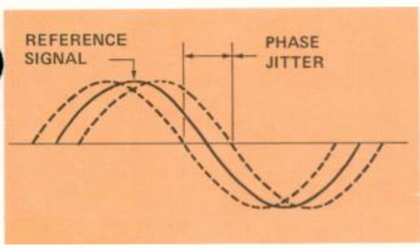


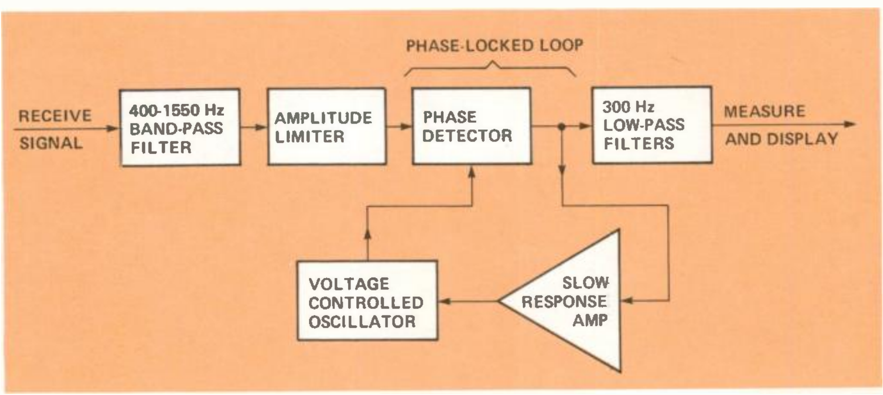
Figure 28. Effect of Jitter on Holding Tone

The output of the filter is connected to the input of an amplitude limiter. The limiter's constant amplitude output, which still contains any phase jitter components, is connected to a phase detector.

The phase detector compares the received signal to the output of the voltage controlled oscillator (VCO). The oscillator is phase-locked to the long term average frequency of the received signal. It provides a jitter free reference signal.

Phase deviations, at a rate slower than 20 Hz, are tracked by the phase-locked loop. The slow-response amplifier changes the frequency of the VCO to provide the proper frequency and phase to the phase detector. As result, the phase difference between the receive and oscillator signals are negligible. There is no error signal output from the phase detector.

The phase-locked loop will not track phase changes at a rate faster



Courtesy Hewlett Packard

Figure 29. Phase Jitter Measurement

than 20 Hz. The slow-response amplifier cannot respond fast enough to keep the oscillator frequency matched to the receive signal frequency. An error signal will be generated by the phase detector. This signal is proportional to the rapid changes in phase.

The error signal is connected to the input of a 300 Hz low pass filter. The error signal at the filter output is frequency limited to a band between 20 and 300 Hz.

This band includes phase jitter caused by 20 Hz ringing and 60 Hz power frequencies and several of their harmonics. Phase jitter rarely occurs above 300 Hz. When it does, it is accompanied by large amounts of jitter below 300 Hz, which will be detected by the measuring circuit.

Phase Wobble

Occasionally, a long-haul circuit will have a trouble indicating a transmission problem, although all transmission parameters test satisfactorily. The problem could be low frequency phase jitter, between 3 and 20 Hz. This type of jitter is known as "phase wobble". It sometimes causes temporary loss of clock synchronization.

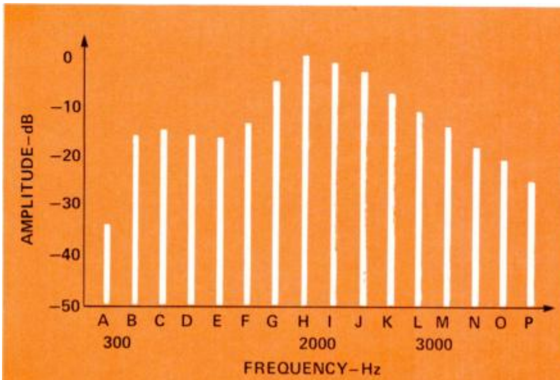
If a trouble persists, although all other parameters check good, phase wobble may be the cause. Phase wobble can be measured using several available test sets. If peaks exceeding 10 degrees are observed in a 10 minute interval, phase wobble is indicated.

Peak To Average Ratio

Peak to average ratio (P/AR) testing measures the overall spreading or smearing in time a signal experiences during transmission over a datacom channel. This spreading is representative of the combined effects of transmission impairments which cause intersymbol interference of datacom signals.

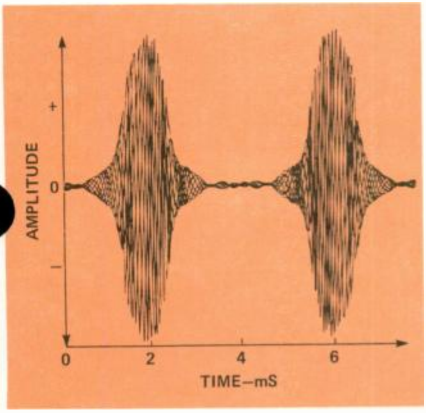
P/AR is measured by transmitting a complex signal with the frequency spectrum shown in Figure 30 and the envelope shape down in Figure 31. At the receive end, the peak-to-average ratio of the received signal is calculated and compared to the known peak-to-average ratio of the transmit signal. If the signal is not degraded during transmission, the comparison yields a P/AR value of 100.

The P/AR value is a single number rating of a channel's fidelity. It is



KEY	FREQ	KEY	FREQ
A	140.625	I	2140.625
B	390.625	J	2390.625
C	640.625	K	2640.625
D	890.625	L	2890.625
E	1140.625	M	3140.625
F	1390.625	N	3390.625
G	1640.625	O	3640.625
H	1890.625	P	3890.625

Figure 30. P/AR Transmit Signal Frequency Spectrum



Courtesy Hewlett Packard

Figure 31. P/AR Transmit Signal Envelope

quite sensitive to envelope delay, noise and attenuation distortion. A P/AR rating does not provide much information about specific transmission impairments.

However, P/AR testing provides a good quick check of channel quality. A P/AR value that shows a decrease, of more than four units from previous readings, indicates some channel degradation. More specific tests can be used to isolate the cause. The value of P/AR testing is that it can detect a problem before data transmission errors become serious.

P/AR is very sensitive to envelope delay. For this reason, lines with more stringent envelope delay specifications require higher P/AR values. Current P/AR minimums are 48 for a C1 conditioned channel, 78 for a C2 and 87 for a C4.

This concludes our discussion of analog channels for data communication. The discussion is largely from the viewpoint of a telephone company or other supplier of analog transmission facilities for datacom use.

The datacom user has a different viewpoint. His system includes elements that may be supplied by a computer manufacturer, a modem manufacturer and a common carrier. When the user has a system malfunction, he must decide which supplier to call for service. The correct decision can save him hours of down time. If he can give the supplier some clues as to where the problem is, even more time will be saved.

To decide which supplier to call, the user performs digital and protocol tests, as well as some of the analog tests we have described. Digital and Protocol testing will be the subject of a future issue of the Demodulator.

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