Model 930A Communications Test Set

Applications Note #2

VOICE FREQUENCY MEASUREMENTS

7/1/91
Rev. 2

This pamphlet is intended to augment Sections III and IV of the Model 930A Operation Manual. Useful background information is provided regarding voice channel impairments and their measurement on voice frequency trunks.

9120-0930-02

Table of Contents

VOICE FREQUENCY MEASUREMENT PRINCIPLES

Introduction

The telephone network today carries voice, data, telegraph, facsimile and other services, although it was originally designed to support only voice. The modern telephone network uses both analog and digital transmission facilities.

Voice communication is two-way, at relatively slow speed and has the wide dynamic range of human speech. Because of the redundancy inherent in speech communication and the intelligence of the listener and talker, many of the deficiencies in the transmission medium can be overcome.

Digital data however, is a high speed information transfer, either one-way or two-way, and with a narrow dynamic range. Data is sensitive to the transmission medium's deficiencies since these can cause errors. Analog transmission systems are less than optimum for digital data.

To transmit or receive a digital data signal over an analog voice frequency facility, a MODEM (Modulator-Demodulator) is used. This device converts the digital data signal into analog tones which can be passed over the network, and vice versa.

Modems, or data sets as they are called, are sensitive to certain analog parameters which have no effect on speech. This sensitivity increases as the data rate increases. It is not at all uncommon for people to be able to speak over lines which are unuseable for data.

The most significant analog parameters affecting data communications are bandwidth of the channel, mode of transmission and line impairments. These parameters can be specified to some extent by the user and some are controlled by FCC tariffs.

TYPES OF SERVICE

Channel Bandwidth

The bandwidth of a channel defines its useable frequency range. The maximum speed of data transmission is directly related to the bandwidth. There are three grades of bandwidth:

Narrowband (0 to 300 Hz)
Voice band (300 to 4000 Hz)
Wideband (> 4000 Hz)

Narrowband, sub-audio channels allow transmission speeds from 45 to 300 bits per second and are used for data only. Voice transmission is impossible. Narrowband is used for Teletype, Facsimile and low speed modem data.

Voice band channels (300 to 4000 Hz) can support a wider range of data rates. Modern modems are available which can operate at rates of up to 16.8 Kbps on voice band channels. The majority of modems operate at speeds between 300 and 4800 bps, although the voice band channel can support rates between 60 bps and 19.2 Kbps. At the higher rates, most modems operate in the half-duplex mode. Voice band channels are available as either dial-up (public switched network or switched special services) or as leased lines (private, dedicated point-to-point non-switched service).

Dial-Up or Switched Services

Dial-up lines are the most common form of service and use the established switched network. A dial-up call may route through a number of switches to reach its destination. The switches may be older electro-mechanical types, modern digital machines or combinations. Electro-mechanical switches are a major source of impulse noise on dial-up lines. Since dial-up lines can have a different path for each call placed, it is impossible to condition a dial-up circuit for data signals. While impulse noise or noise alone may be annoying during a conversation, voice communication is still possible. However, impulse noise can render data communications impossible.

The lack of conditioning generally restricts the speed of data transmission to less than 9600 bps over dial-up lines and at such a high rate the distance would usually have to be short (< 50 miles) as well. This of course assumes that a modern modem with adaptive equalization and error correction coding is used. These restrictions are the reason that the most frequently used rates on dial-up lines are 300 bps and 1200 bps.

The dial-up network has the advantage of redundancy. If the first line does not permit data communication, hang up and re-dial. The second line will almost certainly be different.

Leased Lines

Leased lines, sometimes referred to as private lines or “nailed-up” circuits, are point-to-point dedicated connections between locations. These dedicated lines are used in private data networks and are independent of switching and signaling equipment in the telephone central office.

Since the circuit always has the same transmission path, it can be conditioned (shaped) to allow higher speed data transmission reliably. Data speeds above 9600 bps are possible with proper conditioning and the restrictions on distance are less severe.

Line conditioning is discussed on Page 9.

Wideband Services

Wideband lines consist of a bundle of voice channels. Data speeds of up to 56 Kbps are possible on Series 8000 service. This service is quite expensive and is not in common use. Wideband analog transmission has given way to digital (either DDS or T-1 Carrier) service.

Digital Transmission

Point-to-point transmission service eliminates the need for modems. The signal is regenerated along the path so that there is negligible noise accumulation and signal degeneration. Lower error rates result and speeds of up to 1.544 Mbps are now available to the end-user. These are no longer single voice channels but trunks capable of handling voice and data simultaneously using multiplexing techniques.

SOURCES OF TRANSMISSION IMPAIRMENTS

Types of Transmission Facility

The transmission medium is usually the source of the impairments discussed in the next section. The sources are described in this section.

The transmission path for voice frequency signals is commonly referred to as a 2-wire or a 4-wire circuit. Two-wire cable is usually found in the “subscriber loop” between the telephone company end-office or wire center and the telephone set on dial-up lines. Four-wire circuits are normally used between telephone central offices or as leased lines. Although called a 4-wire circuit, it is not unusual for a 4-wire toll circuit to be a microwave radio, coaxial cable, satellite or fiber optic facility, or a combination of these types.

Copper Wire Cable Pairs

The 2- and 4-wire circuits which are provided by copper wire cable are referred to as “metallic” facility in the telephone companies to delineate them from other types of transmission media. Metallic facility is, as might be expected, the most common transmission medium. Copper wire has resistance and pairs of wires have capacitance between them. Both of these parameters can attenuate the signal. The amount of attenuation depends upon cable length, signal frequency, proximity to power lines, and weather conditions if the line is in open air. Attenuation, or loss, is one form of transmission impairment.

Coaxial Cable

Coaxial cable consists of up to 20 “tubes” enclosed by a protective sheath. Each tube has a copper center conductor suspended by insulation. Since the tube and center conductor share a common center axis, it is called coaxial. At one time these cables were the principle means of carrying long distance (toll) traffic but were rapidly superseded by microwave radio. Coaxial cables use repeaters spaced at one to four mile intervals. The voltage which powers them is sent down the center conductor along with the signal energy. This voltage may be as much as 1600 volts. In older cables, it is not uncommon for arcing to occur between the center conductor and tube surrounding it due to deterioration of the insulation. These arcs cause noise spikes and a general increase in noise level.

Microwave Radio Relay

Microwave radio relay systems consist of stations spaced about 30 miles apart. Each station has radio and may have multiplex equipment at intermediate sites along the backbone route. Microwaves travel in straight lines so the stations' antenna towers must be within line of sight of each other. Each station amplifies and re-transmits the signal.

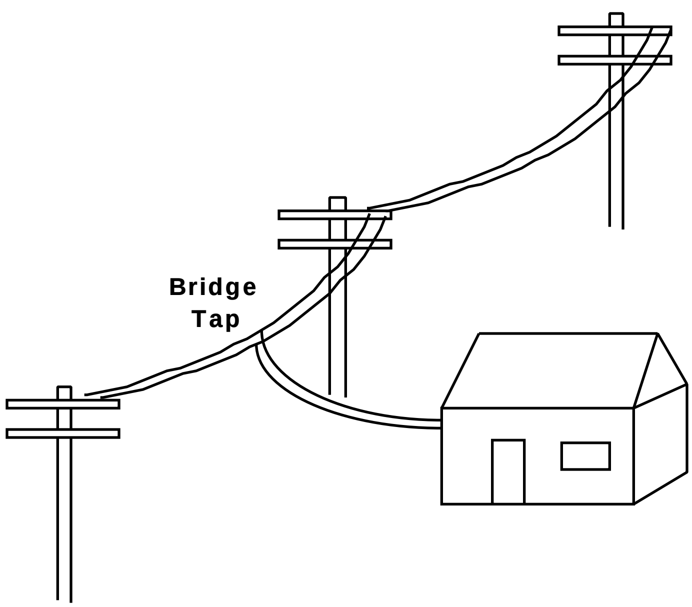
Microwave propagation is adversely affected by weather conditions and path anomalies. Changes in signal amplitude are referred to as “fading”. If the fading is severe enough, the radio’s protection switching system may switch to an alternate path or frequency. Fading can therefore result in phase distortion and dropouts.

Satellite Facility

Long distance dial-up calls may be routed over satellite facility. The delays involved in propagation over such a path may be too much for a modem. The turnaround time of the modem is almost certain to be less than the delay. The user has no control over the path chosen by the switch on dial-up calls. It is conceivable that one direction might go via satellite with the return path being on terrestrial microwave. Satellite paths may cause complete loss of data communication capability.

Bridge Taps

A bridge tap is formed by bridging across the cable pairs to bring service to a customer as shown below:



When service is disconnected, the wires are left in place but are unterminated. The capacitance these unterminated wires present to the circuit can produce significant phase distortion. Conditioned lines have all bridge taps removed. However, a regular dial-up call cannot be sure that bridge taps are not present. This limits home computer users to less than 1200 bps modems more than any other factor, with the possible exception of load coils which are discussed next.

Load Coils

Loading coils are placed at approximately 6000 foot intervals on subscriber loops. Loading coils reduce attenuation distortion by smoothing attenuation over a given frequency range to a nearly constant amount. Load coils act as filters in that they have a limited frequency bandpass. Above 4 kHz the load coils attenuation increases drastically. Data, with its sharp transitions, has high frequency components above 4 kHz. Attenuating these high frequency components distorts the shape of the digital signals. This distortion leads to increased error rates.

Repeaters

Voice frequency repeaters are amplifiers which compensate for attenuation on long circuits. VF repeaters have a gain or amplification factor of about 23 dB. Any active device such as an amplifier is inherently non-linear. This means that additional frequencies may be present at the output of the repeater that were not present at the input. This comes from the “mixing” of input frequencies to produce sum and difference frequency products at the output. This is called Intermodulation Distortion (IMD). Another form of non-linear distortion causes integer multiples of the input frequencies to appear at the output and is referred to as harmonic distortion. For more information on non-linear distortion, refer to page 22. Amplifiers also amplify the unwanted noise and distortion which appear at their inputs as well as the desired signal. This accumulation of noise and distortion on long distance calls limits the data speed and increases the error rate. Modern all digital transmission media such as fiber optic facilities do not accumulate noise because the signal is completely regenerated at each fiber optic repeater location and the signal-to-noise ratio does not deteriorate.

Hybrids

The 2-wire facility which comprises applications such as subscriber loop and PBX trunks must interface to the 4-wire toll network for long distance or interoffice calls. This is accomplished in the telephone network by using a 2W/4W hybrid network. In its simplest form, a hybrid is a transformer with a tapped winding. The conversion from 2-wire to 4-wire and vice versa usually takes place in the end-office or serving office nearest the customer. The path can be converted at intermediate points as well.

The problem with hybrids is that they present a discontinuity to the circuit. No matter how small, the discontinuity has a direct impact on transmission. A portion of the energy incident upon a discontinuity, or impedance mismatch, is reflected back. If the energy is speech, then a portion of the energy reflected back toward the talker is his own speech. This is referred to as echo. It is most discernible on long distance calls, particularly over satellite links, but may also be found on local calls where the echo makes the talker feel like he is talking into a barrel. If the discontinuity is small and the network is reasonably well matched then the reflected energy is small and not noticeable. If the mismatch is large, the talker will experience echo.

Echo Suppressors/Cancellers

To prevent, or at least diminish echo on switched long distance calls, echo suppressors and more recently, echo cancellers are used.

Echo suppressors insert attenuation into the return path to reduce the echo. Echo cancellers employ a complex feed forward technique which mixes echo path response with the incoming desired signal to produce an error voltage. This error voltage is used as an estimate of the echo. This estimate is subtracted from the echo path output thereby “cancelling” the echo.

Echo cancellers and suppressors, while they remove an impairment to voice communications, create a possible impairment to data signals. These devices must be removed from the transmission path during data communications or they attenuate the reverse direction of full duplex modems. To disable these devices during data transmission, a 2125 Hz disable tone is transmitted on the path for at least 400 milliseconds prior to data transmission. This tone is sent by the modem. During turnaround time or when data is absent, the modem transmits the disable tone to keep the suppressors/cancellers disabled. When the modem carrier tone is absent for more than 100 milliseconds, the suppressors/cancellers are re-enabled for voice.

The 400 millisecond delay in disabling the suppressor/canceller can be a serious limitation when using dial-up lines for half duplex interactive transmission. Each direction of transmission requires a disable; there would be two disable delays. Many modern half duplex modems can turn off both echo suppressors/cancellers at the start of transmission.

Echo suppressors and cancellers are not used on leased lines, only on dial-up.

Companders

A compander consists of a Compressor and an Expander. The compressor reduces the transmitter volume range by lowering high amplitude signals and raising low amplitude signals. On the circuit then, the signal has a compressed volume range during transmission. At the receiving end the expander reverses the process and restores the signal to its original range.

At the transmitter, since noise is not a significant presence, the noise level is not increased by the compressor. At the receive end, the expander restores the signal and spreads out any noise accumulated on the path thereby lowering the effective noise level.

Although this improves the signal-to-noise ratio on the line for voice traffic, it can adversely affect data signals. Amplitude modulated data signals transmitted through a compander will be severely distorted by the variable loss characteristic of the compander. Since companders are non-linear devices, frequency shift key modems can also be affected by the introduction of spurious frequencies.

Companders must be disabled before data transmission.

Equalizers

Equalizers can be either active or passive circuits. Equalizers are added to voice grade leased lines to smooth out the attenuation and delay response characteristics across the channel's bandwidth. Passive or fixed equalizers compensate for average line conditions and cable length. Adaptive equalizers in modern modems track and compensate for changes in a line's characteristics. However, rapid changes in line characteristics can cause the modem to lose sync and data because the adaptive equalizer cannot track and initialize on the change fast enough.

CONDITIONED LINES

Attenuation distortion and envelope delay distortion are two parameters that define the frequency response and bandpass of a channel. FCC Tariff No. 260 establishes the different limit levels. Equalizers are used to control these parameters. Noise level and non-linear distortion are also controlled by tariffs.

Frequency shift, impulse noise, phase and amplitude jitter, and transients such as dropouts, phase hits and gain hits are controlled by internal telephone company standards but are not tariffed.

Telephone companies offer leased lines with several types of line conditioning to provide higher data rates and/or reduce error rates. Before describing impairments it is useful to discuss the conditioning limits against which impairments are measured. Conditioning is usually required for data speeds in excess of 2400 bps and should definitely be used above 4800 bps unless a modern modem with adaptive equalization is used.

The conditioning levels are specified in FCC Tariff No. 260. There are six levels of conditioning commonly used (although there are others). Tariff No. 260 provides a basic non-conditioned voice channel designated as series 3002 service. The tariff also defines other types of conditioning including five types of C-Conditioning which are commonly used. In addition to these services, some telephone companies provide an option for high speed data above 9600 bps called D-Conditioning. C-Conditioning is the name given to the channel treatment which makes voice grade lines meet the conditioning specifications of Tariff 260. These specifications apply to the attenuation and envelope delay distortion characteristics of the channel. C-Conditioning is applied to the basic 3002 service. The removal of loading coils may be included in the circuit treatment by specifying B-Conditioning. B-Conditioning is needed for short haul modems or line drivers. Table 1 shows the conditioning levels for the various types of conditioned lines.

TABLE 1. LINE CONDITIONING LIMITS

	Non-Conditioned 3002 Channel		C1 Conditioning		C2 Conditioning		C3 Conditioning (Access Lines)	
Frequency Range (Hz)	300-3000		300-3000		300-3000		300-3000	
Attenuation Distortion (Net Loss at 1 kHz)	Frequency Range 300-3000 500-2500	Decibel Variation -3 to +12 -2 to +8	Frequency Range 300-2700 1000-2400 300-3000	Decibel Variation -2 to +6 -1 to +3 -3 to +12	Frequency Range 300-3000 500-2800	Decibel Variation -2 to +6 -1 to +3	Frequency Range 300-3000 500-2800	Decibel Variation -0.8 to +3 -0.5 to +1.5
Envelope Delay Distortion (Micro-seconds)	Less than 1750 µsec from 800 to 2600 Hz.		Less than 1000 µsec from 1000 to 2400 Hz. Less than 1750 µsec from 800 to 2600 Hz.		Less than 500 µsec from 1000 to 2600 Hz. Less than 1500 µsec from 600 to 2600 Hz. Less than 3000 µsec from 500 to 2800 Hz.		Less than 110 µsec from 1000 to 2600 Hz. Less than 300 µsec from 600 to 2600 Hz. Less than 650 µsec from 500 to 2800 Hz.	
Signal to Noise (dB)	24		24		24		24	
Non-Linear Distortion Signal to 2ND Harmonic (dB)	25		25		25		25	
Signal to 3RD Harmonic (dB)	30		30		30		30	

C3 Conditioning (Trunk)		C4 Conditioning		C5 Conditioning		D Conditioning
300-3000		300-3200		300-3000		
Frequency Range	Decibel Variation	Frequency Range	Decibel Variation	Frequency Range	Decibel Variation	
300-3000	-0.8 to +2	300-3200	-2 to +6	300-3000	-1 to +3	
500-2800	-0.5 to +1	500-3000	-2 to +3	500-2800	-0.5 to +1.5	
Less than 80 µsec from 1000 to 2600Hz. Less than 260 µsec from 600 to 2600 Hz. Less than 500 µsec from 500 to 2800 Hz.		Less than 300 µsec from 1000 to 2600Hz. Less than 500 µsec from 800 to 2600 Hz. Less than 1500 µsec from 600 to 3000 Hz. Less than 3000 µsec from 500 to 3000 Hz.		Less than 100 µsec from 1000 to 2600Hz. Less than 300 µsec from 600 to 2600 Hz. Less than 600 µsec from 500 to 2800 Hz.		
24		24		24		28
25		25		25		35
30		30		30		40

C-Conditioning on multipoint circuits applies to all of the links on the circuit. The same circuit cannot have a mixture of different types of C-Conditioning. That is, one leg of a bridge circuit cannot be a 3002 line while the other links are C2.

Basic 3002 channels and channels with C1 and C2 Conditioning may be ordered in point-to-point, multipoint and switched configurations.

C3 Conditioning applies only to private switched networks and is specified in terms of access lines and trunks. C3 Conditioning ensures an overall C2 Conditioning level end-to-end on switched connections involving up to four trunks and two access lines in tandem. C3 Conditioning is used on Common Control Switching Arrangement (CCSA) or switched circuit automatic networks which are independent of the regular dial-up network.

C4 Conditioning can be ordered only for two, three, or four point operation. C5 Conditioning can only be ordered for two point operation. C5 Conditioning is meant to achieve C2 Conditioning end-to-end on multiple link connections. The principal use of C5 Conditioning is on two point private lines which extend overseas.

D-Conditioning is an option introduced by the telephone companies to handle 9600 bps and above modems. D-Conditioning controls signal-to-noise ratio and non-linear distortion more closely than C-Conditioning, which focuses mainly on attenuation and envelope delay distortion.

When using modems with adaptive equalizers, C-Conditioning may not help performance. This is because unconditioned or lightly conditioned lines have smooth envelope delay curves, while C2 and C4 lines have steep slopes and ripples in the passband which make it difficult for adaptive equalizers to stabilize.

TRANSMISSION IMPAIRMENT DEFINITIONS

The impairments which affect data transmitted over voice channels have limits specified in FCC Tariff No. 260 or in telephone company internal standards.

The tariffed parameters include attenuation distortion, envelope delay distortion, noise and non-linear distortion. Those parameters not specified in Tariff 260 include return loss, impulse noise, gain hits, phase hits, dropouts, and phase and amplitude jitter.

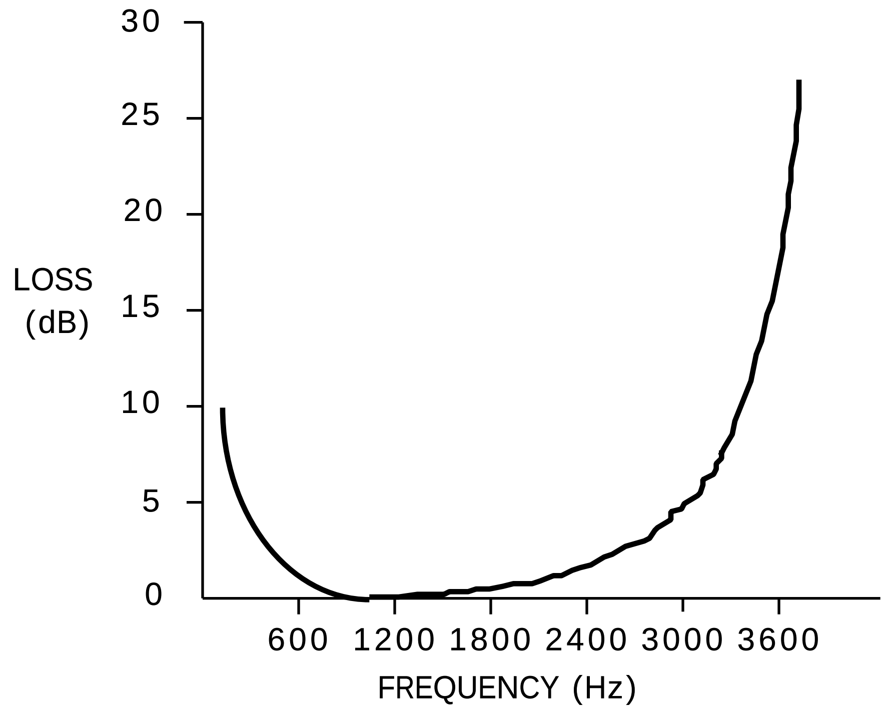
In this section the various impairments and their sources are discussed.

Loss and Attenuation Distortion

Loss is the measure of attenuation between two points. The effect of loss is to lower the signal level and thereby decrease the signal-to-noise ratio. This makes the data signal more susceptible to errors. Loss is measured by sending a tone at a known level from one end of the circuit and measuring the received level at the far-end. The difference between the level sent and the level received is the loss. Loss is usually measured at 1004 Hz with transmitted levels between 0 dBm and -16 dBm, depending upon the point at which the tone is injected into the circuit.

Different frequencies will experience different attenuation. Since this loss will not be uniform across a band of frequencies, a form of amplitude distortion can be introduced. Amplitude distortion is defined as the loss at any frequency relative to the loss measured at 1004 Hz.

Loss and attenuation distortion are calculated as positive numbers in the telephone companies, while gain is a negative number. This is opposite to engineering practice but has endured over the years. For example, if the loss at 804 Hz is 4 dB more than the loss at 1004 Hz, then the channel has an attenuation distortion of +4 dB at 804 Hz. A plot of loss in dB versus frequency is shown in Figure 1 for a typical voice channel.



ATTENUATION DISTORTION VERSUS FREQUENCY
FIGURE 1

C-Conditioning involves setting equalization to keep attenuation distortion within tariffed limits.

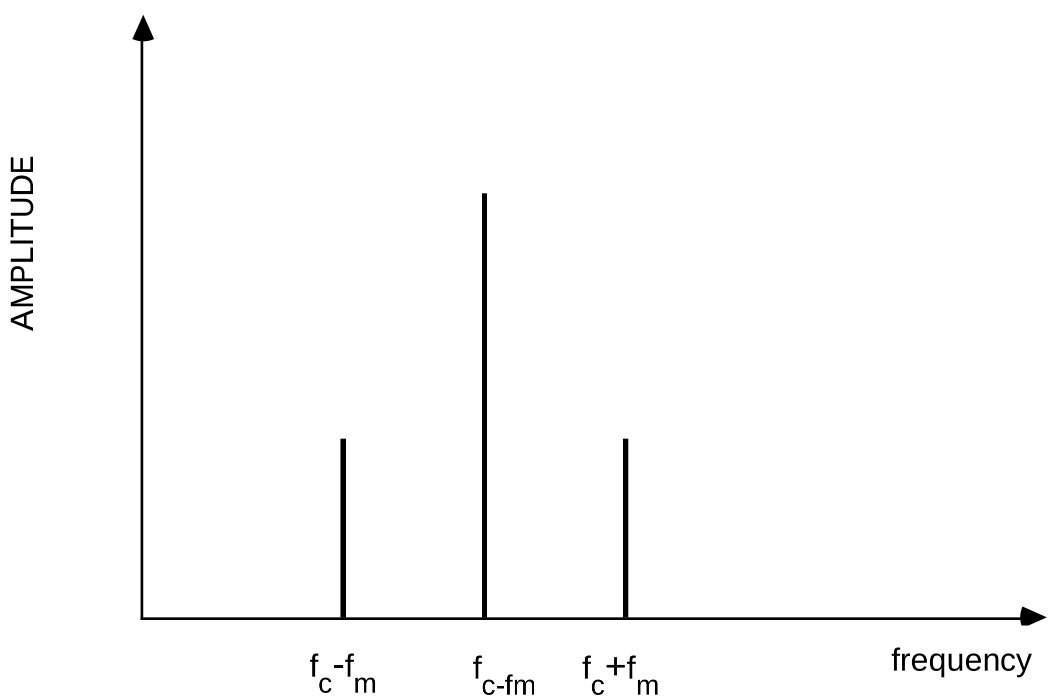
Envelope Delay Distortion

Envelope delay is a measure of the phase linearity versus frequency response of a channel. The propagation time for a signal along a pair of wires varies with frequency. This variation is equal to a relative phase shift. If the shift is non-linear, distortion of the signal results.

Propagation time varies mainly due to the capacitance between wire pairs. This capacitance is increased by bridge taps. Other sources of phase distortion are switching microwave paths to alternate paths and load coils, which also add substantial delay at low frequencies.

The effect of relative envelope delay is to cause data bits to smear out in time and overlap each other (intersymbol interference). At data speeds above 2400 bps, delay distortion inhibits transmission over unconditioned lines because the receiving modem cannot read the smeared bits and errors result. Modems with adaptive equalizers, or equalizers on the lines themselves, smooth delay distortion and permit higher data speeds.

Measuring the phase characteristic of a circuit directly is not practical due to the difficulty involved in establishing a phase reference at both ends of the circuit simultaneously. However, there is a way to measure the phase shift between two tones transmitted down the circuit. Examine the spectrum of a pure, undistorted Amplitude Modulated signal as diagrammed below:

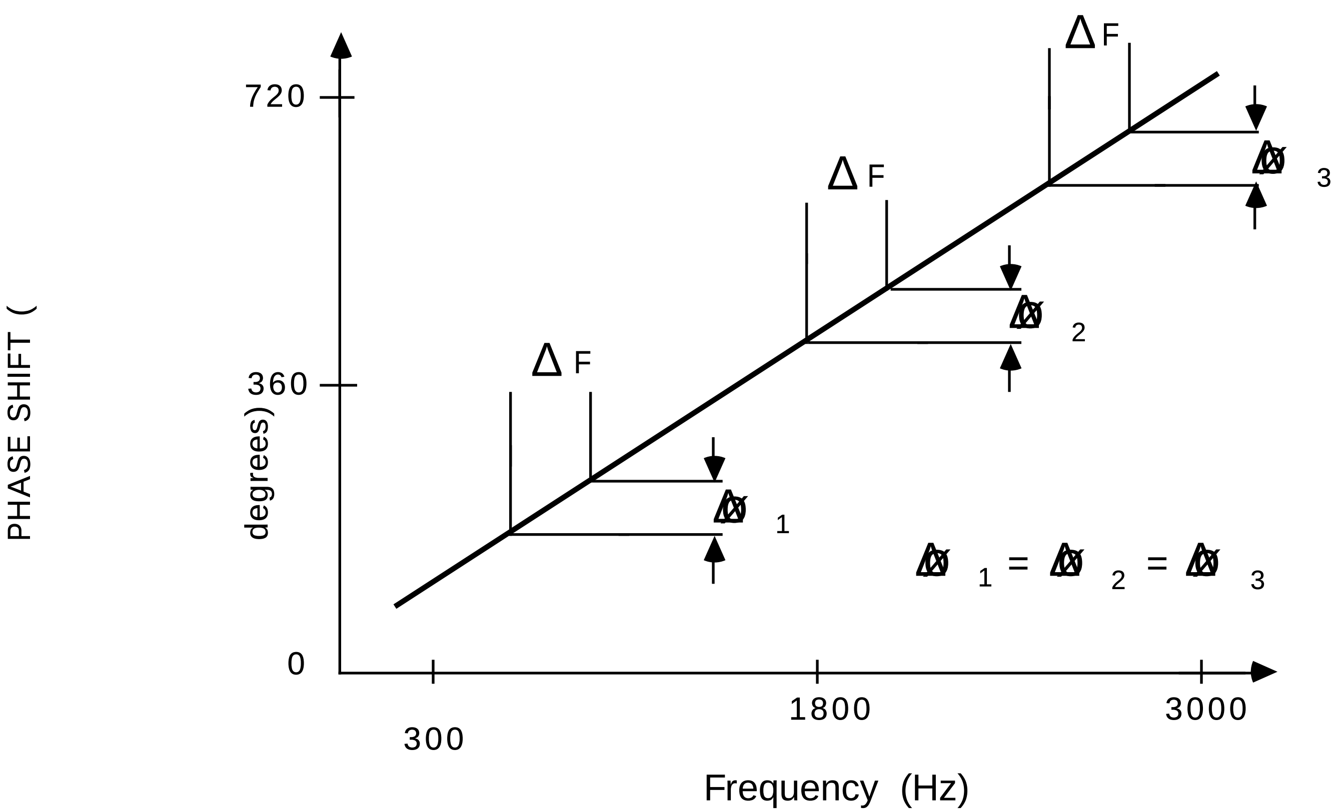


Amplitude Modulation produces a spectrum, such as that shown on the previous page, composed of the carrier frequency (f_c) plus an upper and lower sideband (f_c+f_m, f_c-f_m). The modulation contains the information being transmitted by the AM signal. In this case, the information is the phase shift or delay difference between the modulation envelopes of the upper and lower sidebands.

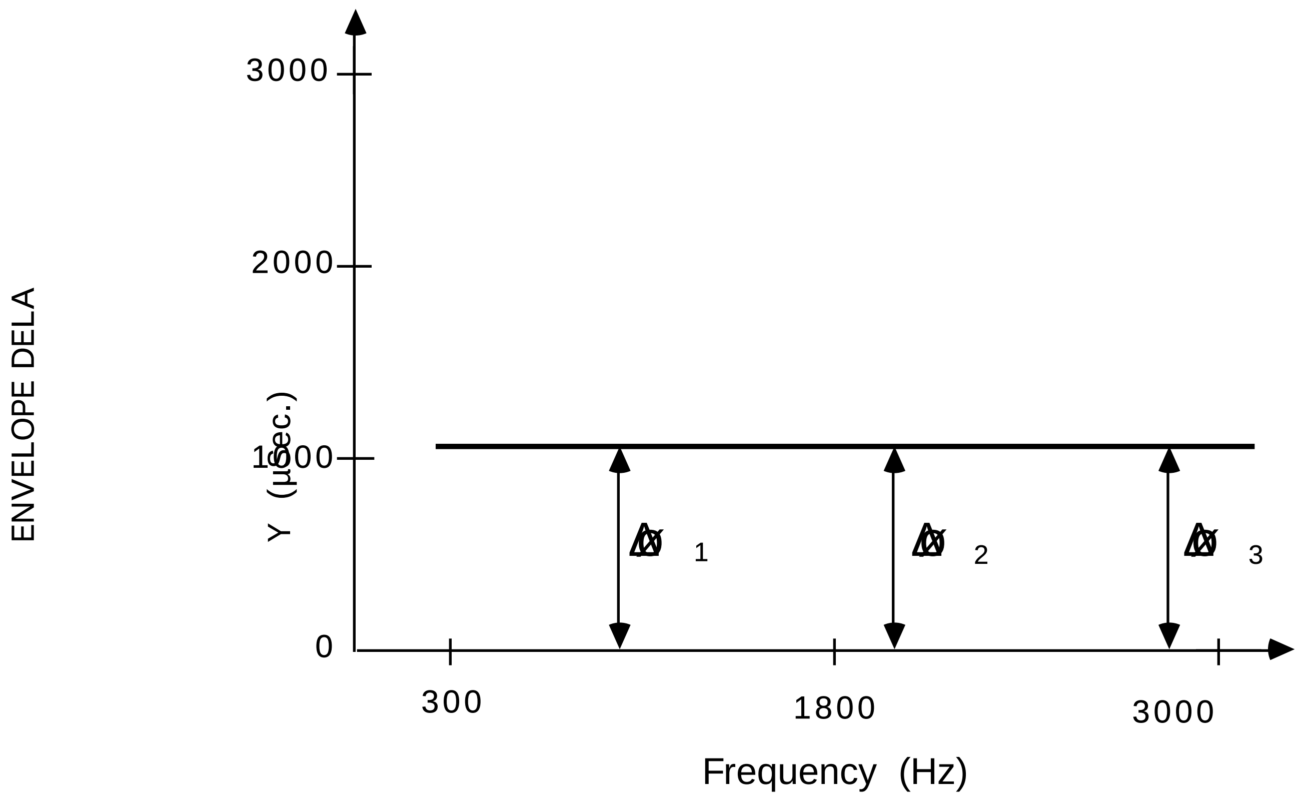
If the modulating frequency (f_m) is held constant, there should be no frequency difference between the upper and lower sidebands, even if the carrier frequency (f_c) changes. Similarly, if there were no phase distortion, then the delay for a constant f_m would also be constant even for a changing f_c. This forms the basis for the envelope delay distortion measurement.

If an AM signal is transmitted over a telephone circuit, the phase shift resulting from this less than perfect transmission medium can be used to determine the delay in the envelope of the modulation and thus the amount of distortion.

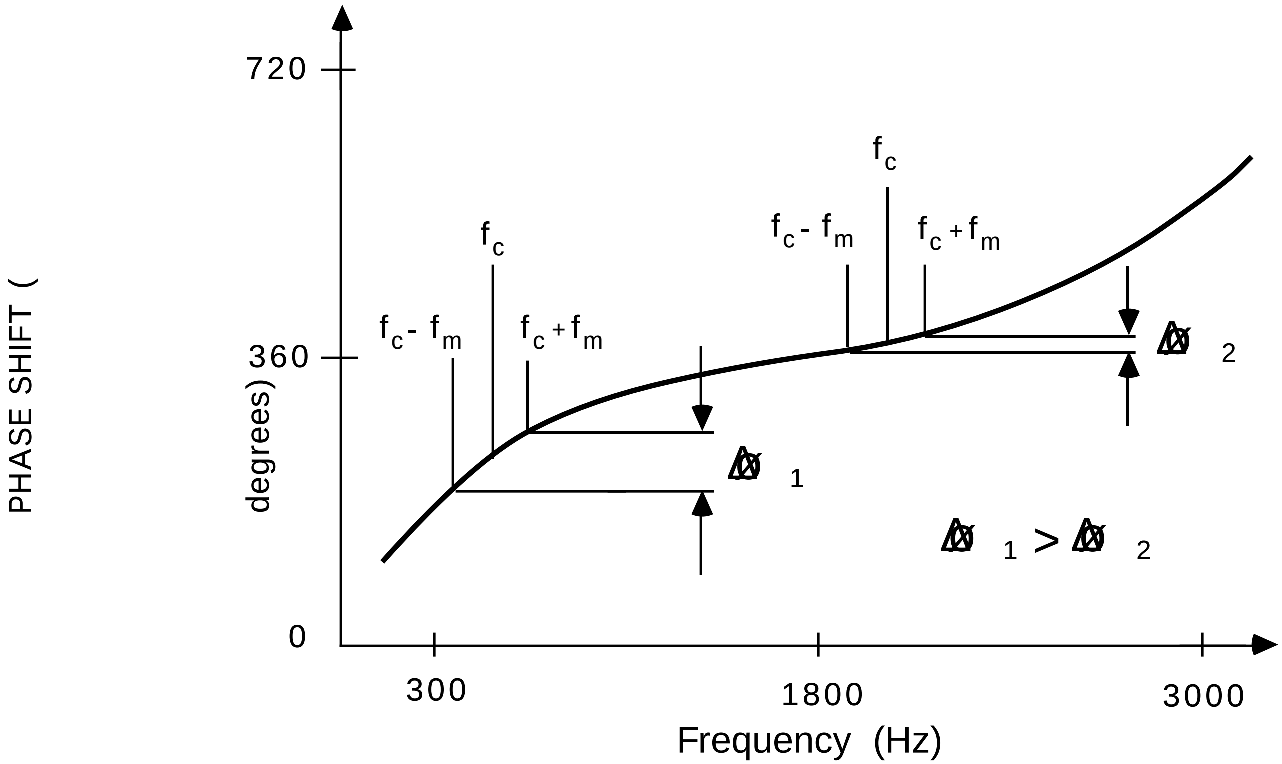
Consider first an ideal or perfect circuit with no non-linearities. In such a circuit the phase characteristic is linear. That is, the phase shift stays constant as frequency changes. A linear phase diagram is shown below:



The phase difference $\Delta \o$ for a constant change in frequency ΔF is the same anywhere along the above characteristic. In such a case, the envelope of the modulation would be delayed by a constant amount over the same frequency range as the carrier frequency was changed. The ΔF in the above diagram represents the bandwidth of the AM spectrum or $2f_m$. A diagram of envelope delay for a linear phase characteristic circuit is shown below.

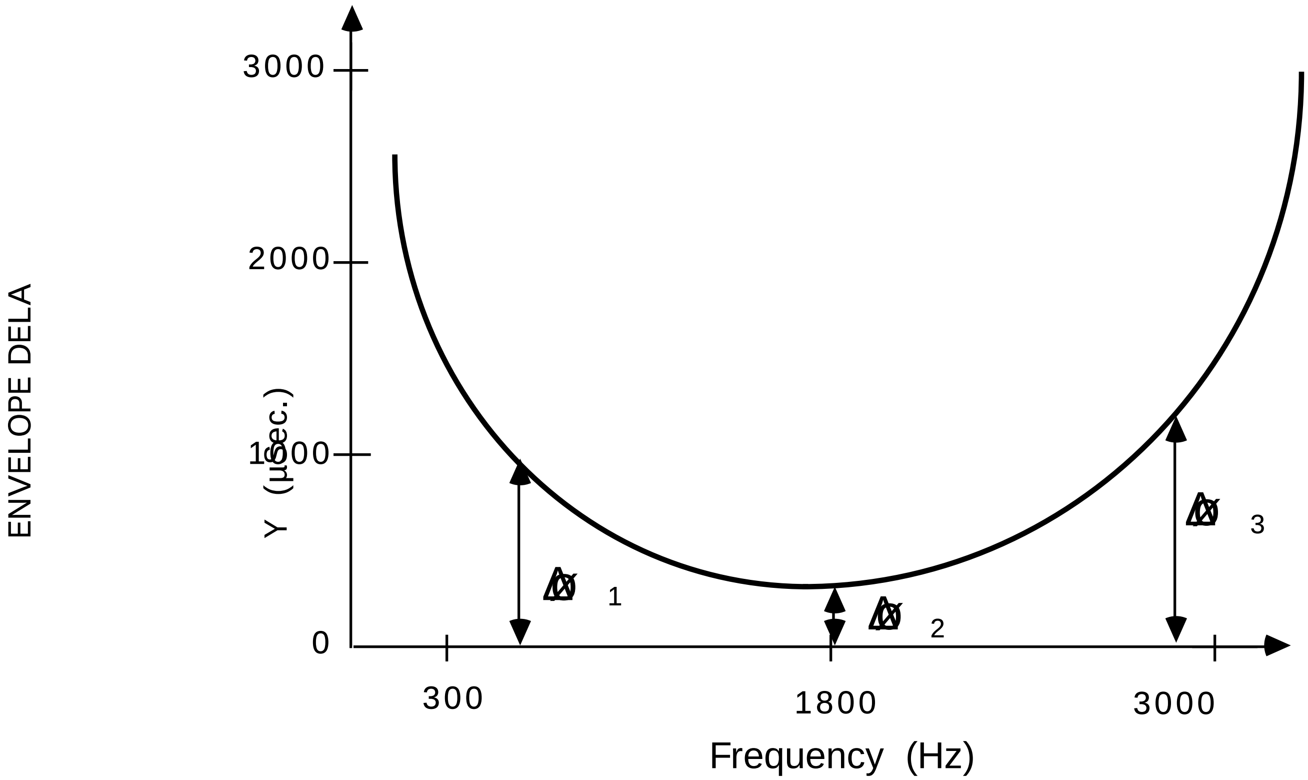


In the real world, the linear phase characteristic is not the case. In fact, there are numerous sources of non-linearities on telephone lines. A non-linear phase characteristic, typical of a real world environment is shown below:



From this diagram it is apparent that even though the modulating frequency has remained the same, the phase difference has not remained constant as the carrier frequency changed.

The delay of the modulation envelope will not be constant in this case since the phase shift is no longer linear. This provides the basis for a measurement technique. Envelope Delay is measured and used to provide an indication of the amount of phase distortion on the circuit. A typical envelope delay characteristic with phase distortion present on the circuit is shown on the following page.



If the delay of the modulation envelope is measured across the band of frequencies as the carrier frequency is varied over the bandwidth of the circuit, then this provides an indication of the amount of phase distortion on the circuit. The modulating frequency is usually $83^1/_3$ Hz and the carrier frequency is varied from 304 Hz to 3504 Hz in 100 Hz steps. Other modulating frequencies such as 25 Hz and $41^2/_3$ Hz may also be used. Envelope Delay is measured in microseconds and is referenced to the delay at 1804 Hz.

Envelope Delay is an end-to-end measurement and requires a test set at each end of the circuit. One test set transmits the test signal consisting of an Amplitude Modulated waveform. The test set at the far-end receives the AM waveform, saves only the modulation envelope, and uses it to modulate a fixed carrier frequency which is then sent back to the transmitting set. The delay introduced on the return path can be zeroed out. The transmitter compares the phase of the original f_m transmitted to that of the f_m which was re-transmitted from the far-end and the difference is the envelope delay. Since this measurement is referenced to the delay at 1804 Hz, the delay at 1804 Hz is always set to 0. All readings at other frequencies are then relative to the 1804 Hz reference. If, for example, the envelope delay reading at 1004 Hz was 350 microseconds, this would mean that the delay at 1004 Hz was 350 microseconds more than the delay at 1804 Hz.

Refer to Table 1 for the tariffed limits of envelope delay distortion for C-Conditioned lines. Envelope Delay Distortion is a required measurement when turning up C-Conditioned lines as well as during maintenance testing.

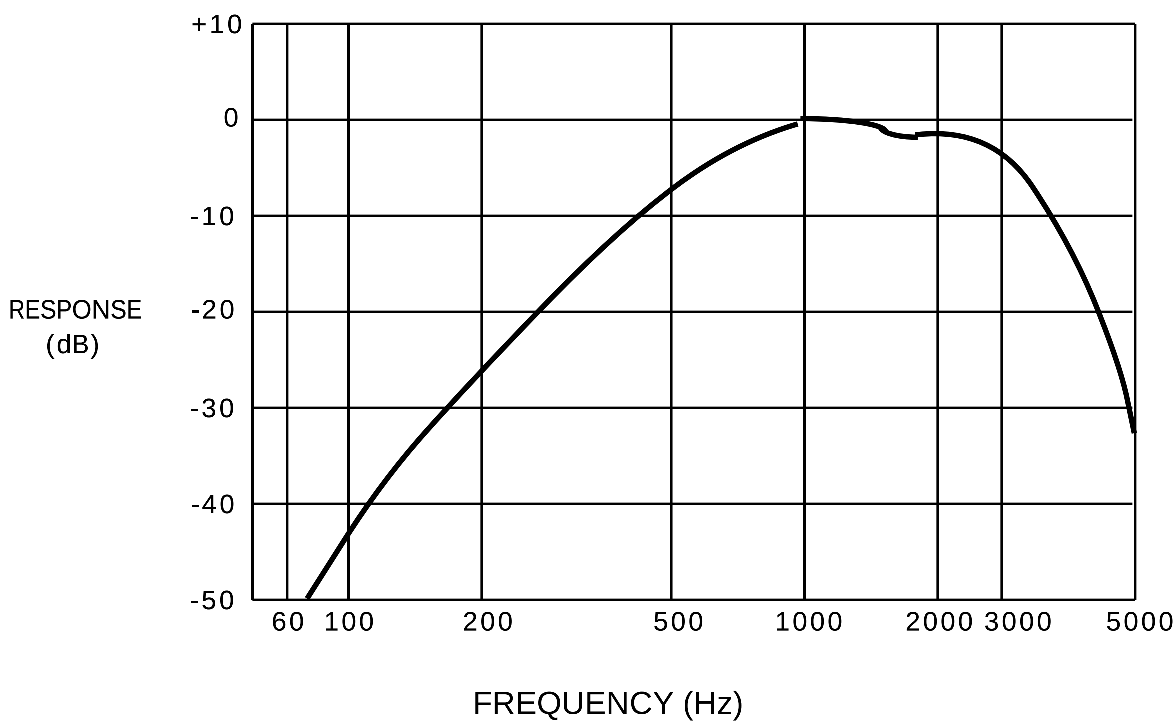
Noise

Noise is always present in communications systems and is a fundamental limit to the information transfer rate over a channel. Noise is often referred to as thermal noise or white noise since it covers the entire frequency spectrum. One source of noise on a channel is the thermal processes in amplifiers, resistances, switches and other active elements. Other sources include radio frequency interference, crosstalk from other channels, and interference from AC power line induction.

As noise increases, signal-to-noise ratio decreases and this means a higher probability of error at the receiving modem.

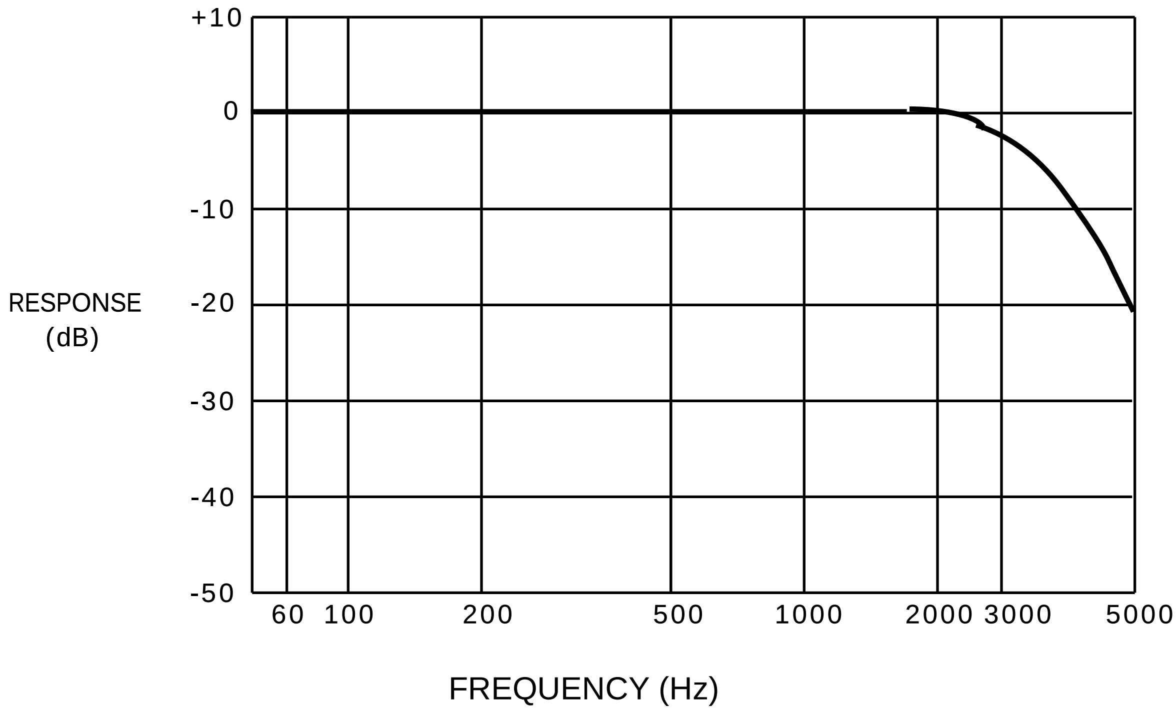
Noise on telephone circuits has traditionally been measured using weighting filters which weight the noise reading to correspond to the subjective effect of noise on voice communication. The weighting filters used in North America for 4 kHz voice channels are the C–Message and 3 kHz Flat weighted filters.

C-MESSAGE FILTER CHARACTERISTIC
FIGURE 2



The C-Message filter is used to measure noise which affects voice communications and is not particularly relevant to data transmission. The 3 kHz Flat Weighted filter characteristic shown below more close ly approximates the response of modems.

3 kHz FLAT FILTER CHARACTERISTIC
FIGURE 3

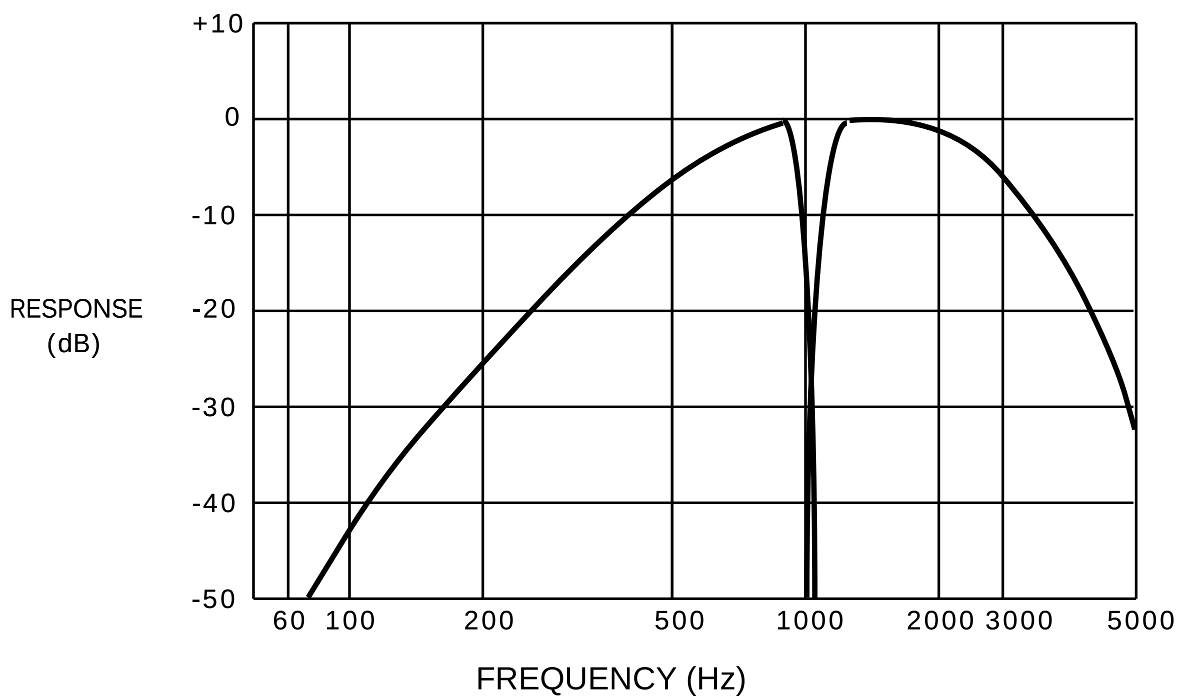


Compared to the C-Message filter, the 3 kHz Flat filter allows the low frequency noise contribution to be measured. By switching between C-Message and 3 kHz Flat filters, various sources of noise may be identified by their characteristic frequency bands.

Noise With Tone

Some elements of the telephone network such as companders and quantizers are only active when a signal is present. Their noise contribution cannot be measured unless a tone is sent to activate them. A 1010 Hz test tone is usually sent and a notch filter in the test set removes the tone prior to noise measurement. The noise filter in this case consists of a C-Message filter with a superimposed notch filter and is often referred to as a C-Notch filter.

C-NOTCH FILTER CHARACTERISTIC
FIGURE 4



Signal-to-Noise

Signal-to-noise ratio can be computed by comparing the signal level of the received tone with the C-Notch noise measurement. In reality, this is a S+N / N ratio, but if the signal is at least 10 dB higher than the noise, the noise component in the numerator becomes negligible. Signal–to–noise is a quick estimate of channel quality and is directly proportional to error performance. The higher the signal-to-noise ratio the higher the probability that the transmission will not have errors.

Non-linear Distortion

Non-linear distortion is the production of frequencies at the output which were not present in the original input signal. One form of non-linear distortion produces frequencies at the output which are integer multiples of the original input frequencies. This is Harmonic Distortion. Another form of non-linear distortion produces frequencies which are the sum and difference products of the input frequencies and their harmonics. This is referred to as Intermodulation Distortion (IMD).

That is, if there are two frequencies, f_1 and f_2, then their potential products are:

$$f_1+f_2\ ,\ f_2-f_1\ ,\ 2f_1+f_2\ ,\ 2f_1-f_2\ ,\ 2f_2+f_1\ ,\ \text{and}\ 2f_2-f_1$$

Non-linear distortion is caused by non-linear devices such as amplifiers, modulators, demodulators, companders and other active elements including switches. The principal cause of 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 signals than for large samples. This maintains a substantially constant signal-to-quantizing-distortion ratio over a wide dynamic range. However, the process is inherently non-linear and so produces non-linear distortion.

The measurement of non-linear distortion is accomplished through the use of a complex test signal. This signal consists of two pairs of tones (857 Hz, 863 Hz and 1372 Hz, 1388 Hz). The second order distortion products occur around 520 Hz and 2240 Hz while the third order products occur around 1900 Hz. These measurements are expressed in terms of dB below the received value of the desired signal. The frequency ranges of the second and third order terms show that if the distortion products have sufficient amplitude in these bands they can cause errors in modem data transmissions. IMD is not usually a problem on voice-only circuits since it would be discernible only as an increase in background noise. However on data transmissions over voiceband facilities, IMD can be a serious impairment.

Phase Jitter

Phase jitter is an undesirable dithering of the phase which appears as phase or frequency modulation. Although unimportant to voice transmission, phase jitter is particularly significant to data, especially to data modems using phase modulation techniques. Phase jitter causes variations in the zero crossings of pulses and this results in pulses moving into other pulse's time slots.

Phase jitter is measured by sending a test tone of 1004 Hz at data level over the line under test. At the receive end a phase locked loop establishes a phase reference and jitter is measured relative to this reference.

The frequency of the phase jitter characterizes its source. Phase jitter caused by ringing current in adjacent channels is at 20 Hz. AC power supply induced currents are a common source of jitter at 60 Hz. Harmonics of these frequencies up to the fifth order can be a source of jitter.

Phase jitter measurements are made over a 4 Hz to 300 Hz range in three bands: 4 Hz to 20 Hz; 20 Hz to 300 Hz; and the entire 4 Hz to 300 Hz band. For more information on phase jitter measurements, refer to page 45.

Jitter standards for data transmission are no more than 10° between 20 Hz and 300 Hz and no more than 15° between 4 Hz and 300 Hz.

Impulse Noise

Impulse noise is a transient phenomena and is characterized by short duration bursts of noise energy called spikes. These spikes are normally less than 1 millisecond in duration. The effects of an impulse noise spike usually disappear within 4 milliseconds.

Impulse noise is caused by such things as the action of electromechanical switches and relays, installation and maintenance activity, and weather disturbances such as lightning.

On voice channels the impulse spikes are an annoyance but communications are still possible. On data transmissions however, the noise bursts can cause a loss of information bits which causes errors. In slow data rate systems (<300 bps), impulse noise is less of a problem because the receiver can distinguish a data pulse from an impulse burst. As the data rate increases it becomes impossible for the receiver to distinguish between a data pulse and a noise pulse. This results in impulse noise related errors.

Gain Hits

Gain hits are another transient impairment. Gain hits are characterized as sudden increases or decreases in received signal amplitude. They are defined as being less than a 12 dB change in level and lasting longer than 4 milliseconds; they may last for hours. Gain hits may look like data pulses to modems which use amplitude modulation.

Internal telephone company standards call for no more than 8 gain hits of more than 3 dB change from the nominal received signal level in a 15 minute period.

Dropouts

Dropouts are a more severe form of gain hits. Dropouts are defined as a decrease in received signal level greater than 12 dB and lasting longer than 4 milliseconds. Dropouts interrupt the signal flow and data is lost. Furthermore, when the signal returns, the modems must re-synchronize, or re-equalize, and more data is lost.

Dropouts occur less frequently than gain hits but their effect is much more severe. Consequently, dropouts have the tightest standards of no more than 1 dropout in 30 minutes.

Phase Hits

A phase hit is defined as a sudden change in the phase or frequency of the received signal which lasts longer than 4 milliseconds. Since most modern modems employ phase shift or frequency shift key modulation techniques, phase hits look like data and cause errors.

Phase hits can occur when switching between carrier supplies in FDM systems or during protection switching from one microwave path to another with different propagation times. These changes may cause all data to be in error until the out of phase condition is cleared.

Internal telephone company standards for phase hits on data circuits are no more than 8 phase hits during a 15 minute period of 20° or more.

MEASURING IMPAIRMENTS

Loss Measurement

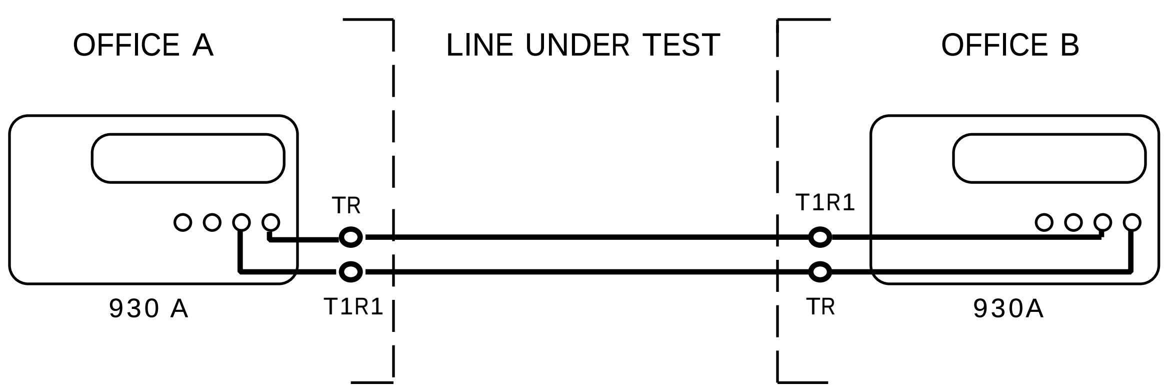
Loss, or 1000 Hz loss as it is called, is an end-to-end measurement meaning that one test set sends the 1000 Hz tone from one end and another test set receives the tone at the other end and measures the difference in level.

In practice, the tone used to measure loss is not 1000 Hz but 1004 Hz. The 4 Hz offset is required to prevent disruption of T-carrier channels since 1000 Hz is a direct sub-multiple of the 8 kHz PCM sampling rate. The circuit must be properly terminated in a non-reactive resistance equal to the characteristic impedance of the trunk under test. Common impedances are:

900 ohms on 2-wire Metallic trunks
600 ohms on 4-wire Metallic trunks
135 ohms on Wideband trunks
100 ohms on T-Carrier trunks

1200 ohm points occur on loaded cable pairs and are not generally accessible. Wideband analog circuits carrying 56 Kbps data are usually 135 ohm lines.

A typical test set-up is diagrammed below:



For 2-wire measurements only, the Tip / Ring connection is used. For 4-wire measurements the Tip1 / Ring1, as well as the Tip / Ring, connection is used. The above test set-up works for Loop Start, Ground Start, Reverse Battery and SF supervision trunks.

The use of the 930A's E&M leads and Signal Battery / Signal Ground (SB / SG) leads is only required on E&M trunks. These leads would be connected to the appropriate jacks in the telephone company office. End-users do not have E&M type trunks which are only used between telephone company central offices.

To send a 1004 Hz tone at data level (-13 dBm0) from Office A toward Office B, set the 930A in Office A as follows:

1. Press the front panel Trunk Type Function key. Choose the correct supervision type; set for 2-wire or 4-wire operation at the correct impedance as appropriate; select BATTERY (in Loop and Ground Start) or SEND-M (in E&M trunks) and then select TERM to terminate the circuit. The softkeys under the display control all of the set-ups. Refer to Section 3-4 of the 930A Manual for details.

2. Press the Send Tone Function key and then press Softkey 4 to turn OFF the tone if it is not already off. Press Softkey 1 and the flashing cursor will appear over whatever frequency was previously entered. Press 1, 0, 0, 4 on the numeric keypad and then press the ENT (Enter) key. Press Softkey 3 and the flashing cursor appears over the level previously entered. Press 1 and 3 on the numeric keypad and then press the ENT key. A level of -13 dBm will be entered. If the transmission level point is not 0 dB, but +7 dB for example, then the level sent should be -6 dBm to achieve a -13 dBm0 referenced level. Likewise, at a -16 dB TLP, the level sent should be -29 dBm to yield a -13 dBm0 referenced level. The user must know what transmission level point is at the test access location.

3. Finally, place the front panel Hook Switch in the OFF HOOK position and press Softkey 4 to turn ON the tone. Refer to Section 3-9 of the 930A Manual as required.

At Office B, set-up the receiving 930A as follows:

1. Press the Trunk Type function key. Choose the appropriate supervision type and set for 2-wire or 4-wire operation at the correct impedance. Select CONTACT (in Loop or Ground Start) or SEND-E (in E&M trunks) and then select TERM to terminate the circuit in the correct impedance. The softkeys under the display control all set-ups. Refer to Section 3-4 of the 930A Manual.

2. Press the MEASURE TONE function key and the 930A will measure the received level and frequency. The difference between the transmitted level and the received level is the loss.

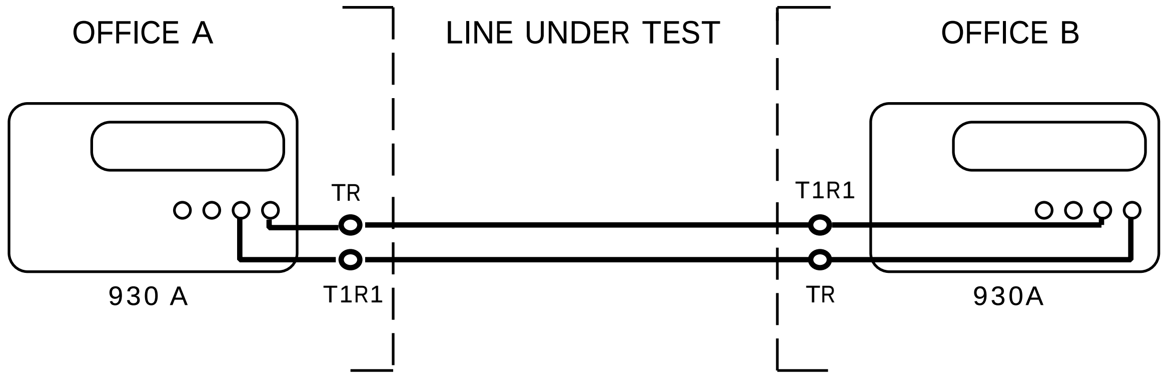
Three-Tone or Gain/Slope Measurement

The so-called three-tone or Gain / Slope measurement, while not a tariffed measurement, is a quick means of spotting potential distortion problems without doing an entire attenuation distortion sweep.

Three tones (404 Hz, 1004 Hz and 2804 Hz) are used to provide an indication of the slope of the attenuation curve for the channel under test from low to high frequency. Since 1004 Hz is usually the reference tone, the other tones are compared to the reading at 1004 Hz. It would be expected that the loss at 404 Hz would be slightly higher than the loss at 1004 Hz and that the loss at 2804 Hz would be much greater. A quick examination of Figure 1 will show why this is so.

The Gain / Slope test is made by applying the three tones in sequence at -16 dBm0 at the distant end and measuring the received tone level at the near-end of the circuit under test.

Typically, the 1004 Hz tone is sent first because the others are referenced to it. The 404 Hz and 2804 Hz tones are then sent and the difference in received tone level at 404 Hz or 2804 Hz and 1004 Hz is the Gain / Slope. The same test set-up used for the 1000 Hz loss measurement can be used for Gain / Slope.



For 2-wire measurements only, the Tip / Ring connection is used. For 4-wire measurements the Tip1 / Ring1, as well as the Tip / Ring, connection is used. The above test set-up works for Loop Start, Ground Start, Reverse Battery and SF supervision trunks.

The use of the 930A's E&M leads and Signal Battery / Signal Ground (SB / SG) leads is only required on E&M trunks. These leads would be connected to the appropriate jacks in the telephone company office. End users do not usually have E&M type trunks which are normally used between telephone company central offices.

To send the 1004 Hz tone at data level from Office A toward Office B, set the 930A in Office A as follows:

1. Press the front panel Trunk Type Function key. Choose the correct supervision type; set for 2-wire or 4-wire operation at the correct impedance as appropriate; select BATTERY (in Loop and Ground Start) or SEND-M (in E&M trunks) and then select TERM to terminate the circuit. The softkeys under the display control all of the set-ups. Refer to Section 3-4 of the 930A Manual for details.

2. Press the Send Tone Function key and then press Softkey 4 to turn OFF the tone if it is not already off. Press Softkey 1 and the flashing cursor will appear over whatever frequency was previously entered. Press 1, 0, 0, 4 on the numeric keypad and then press the ENT (Enter) key. Press Softkey 3 and the flashing cursor appears over the level previously entered. Press 1 and 6 on the numeric keypad and then press the ENT key. A level of -16 dBm will be entered. If the transmission level point is not 0 dB, but +7 dB for example, then the level sent should be -9 dBm to achieve a -16 dBm0 referenced level.

Likewise, at a -16 dB TLP, the level sent should be -32 dBm to yield a -16 dBm0 referenced level. The user must know what transmission level point is at the test access location.

3. Finally, place the front panel Hook Switch in the OFF HOOK position and press Softkey 4 to turn ON the tone. Refer to Section 3-7 of the 930A Manual as required.

At Office B, set-up the receiving 930A as follows:

1. Press the Trunk Type function key. Choose the appropriate supervision type and set for 2-wire or 4-wire operation at the correct impedance. Select CONTACT (in Loop or Ground Start) or SEND-E (in E&M trunks) and then select TERM to terminate the circuit in the correct impedance. The softkeys under the display control all set-ups. Refer to Section 3-6 of the 930A Manual.

2. Press the MEASURE TONE function key and the 930A will measure the received level and frequency.

3. Use the received 1004 Hz tone level as a reference by pressing Softkey 2 to change from dBm to the set 0 dB reference. When the 930A reads 0 dB, have the sending 930A transmit the 404 and 2804 Hz tones sequentially. The set-up is exactly the same as it was for 1004 Hz except for the frequencies. The receiving 930A will then read the difference in dB directly without any calculations.

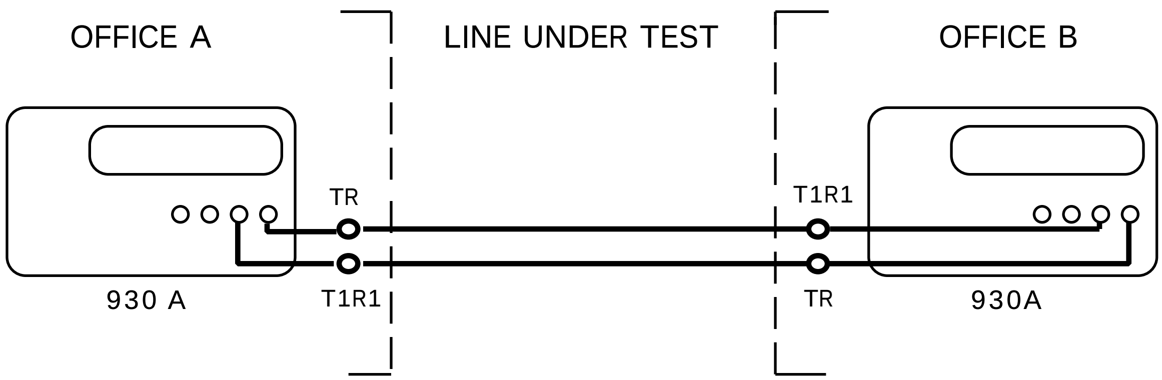
Attenuation Distortion Measurement

Attenuation Distortion is also a referenced measurement. That is, the loss at frequencies other than 1004 Hz is compared to the loss at 1004 Hz and the difference is the attenuation distortion. In this respect, attenuation distortion is similar to Gain / Slope except that there are now more tones to measure.

Attenuation Distortion can be measured using the same test set-up as that used for the Loss measurement in the previous section. In this case though, it is useful to employ the frequency sweep function of the 930A instead of the SEND TONE function. The frequency sweep function, standard on all 930A's, is located in menu option 10 under the OPTION MENU function key.

To use the frequency sweep for attenuation distortion measurement, use the same test set-up used for loss measurement and follow the step by step procedure outlined below.

At Office A:



1. Repeat the 1004 Hz Loss measurement performed in the previous section. Send 1004 Hz at -16 dBm0 from Office A.

2. Press the Option Menu function key on the 930A and then press 1 and 0 and then the ENT (Enter) key to get directly into Menu Option 10: FREQUENCY SWEEP.

3. Set-up the frequency sweep (if sweep has been set previously and no changes are needed, proceed to step 4) by pressing Softkey 2 labeled SET-UP. In this case Softkey 1 controls the upper and lower bounds of the frequency sweep; Softkey 2 controls the step size (usually 100 Hz); Softkey 3 controls the sweep time (usually 2 to 3 seconds per tone) and the level (usually 16 dB below the TLP); and Softkey 4 exits back to the main menu for testing. Refer back to Section 3-12.4 of the 930A Manual for details and pictorials on setting up the sweep.

4. To initiate testing from the main sweep menu, press Softkey 3 labeled “SWEEP” and then choose either continuous or single sweep. After the appropriate softkey has been pressed the 930A will sweep between the bounds selected.

At Office B:

1. Place the 930A in the MEASURE TONE function as it was for the Loss measurement. Have the 930A at Office A send the 1004 Hz, -16 dBm0 tone.

2. After the 1004 Hz tone has been received, press Softkey 2 until the display reads “0 dB”. The 930A should have been reading in dBm so that a single press of Softkey 2 will set the 0 dB reference. Once the 0 dB reference has been set all of the path aberrations have been zeroed out as well.

At Office A, initiate the frequency sweep over the range desired as described above.

At Office B the 930A will read the difference between the loss at the frequencies being sent and the 0 dB reference level set at 1004 Hz.

It is handy to have a printer connected to the 930A to record the received levels during the sweep.

IMPORTANT NOTE

The 930A measures loss as a negative (-) number. This is opposite to telephone company practice but is consistent with engineering practice. To compare readings it is only necessary to change the sign from - to + when checking telephone company results.

Return Loss Measurement

The problem of echo is caused by impedance mismatch and poorly equalized or compensated lines at the interface between the 2-wire subscriber loop and the 4-wire toll network.

A return loss of 30 dB or more is considered very good, while 20 dB or less is probably going to result in less than satisfactory transmission.

To measure return loss with the 930A, press the RETURN LOSS function key. Then press Softkey 1 to select either Echo Return Loss, Singing Return Loss-High, Singing Return Loss-Low, or Sinewave Return Loss. The 930A is also capable of either measuring the Trans-Hybrid Loss (THL) or allowing the user to enter the THL value, if it is known. This compensation factor, when entered or measured, is automatically subtracted from the Return Loss measurement.

Press Softkey 2 to send the momentary Echo Suppressor / Canceller disable tone prior to recording any measurements. This ensures that the line under test is clear of these devices before testing. The disable tone is usually between 2000 and 2250 Hz. The 930A uses 2125 Hz as its default value.

Refer to Section 3-6 for details and pictorials regarding the set-up of the Return Loss function.

Noise Measurement

Noise was discussed on Page 20. The measurement of noise power using the 930A is described in this section.

After selecting the correct trunk type and supervision, press the MEASURE NOISE function key and perform the following steps:

1. Press Softkey 2 to select C-MESSAGE, C-NOTCH or 3 kHz FLAT weighted filter, depending upon the type of test being performed. Select S / N if a signal-to-noise ratio test is to be performed.

2. Press Softkey 4 to select balanced or noise-to-ground measurement. Balanced noise measurements are normally performed on subscriber circuits. Noise-to-ground measurement provides a measure of the longitudinal noise present on a voice channel with reference to ground.

3. The distant end of the circuit should be in a “Quiet Termination” mode. If another 930A is connected at that end, then quiet termination is achieved by placing that unit in SEND TONE and pressing Softkey 4 to turn the Send Tone function “OFF”.

Noise-to-Ground measurements are usually made for troubleshooting purposes and to measure the magnitude of longitudinal signals. This indicates the susceptibility of a cable pair to electrical coupling from external sources such as induced AC power (60 Hz) or Ringing (20 Hz). To calculate the relative line balance, perform the following steps:

1. With the 930A in MEASURE NOISE, press Softkey 2 to select the 3 kHz FLAT weighted filter.

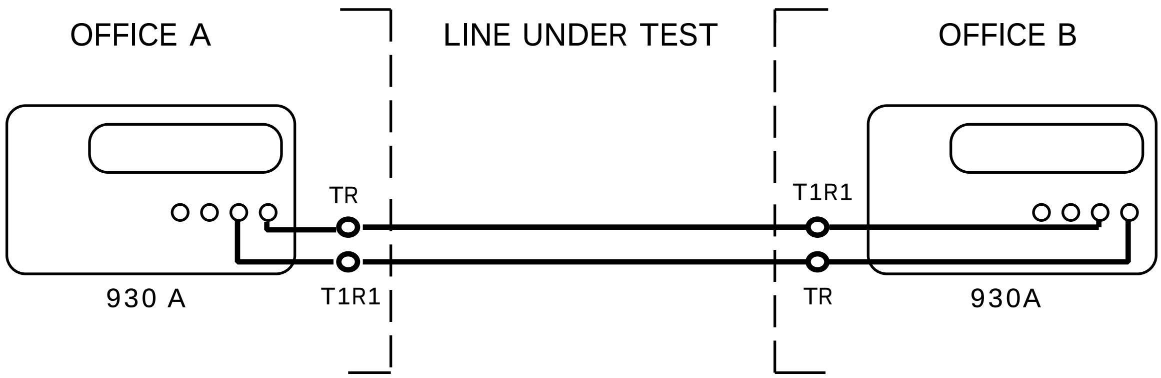
2. Press Softkey 4 to select the balanced measurement mode.

3. Record the reading of message circuit noise in dBrn.

4. Press Softkey 4 to select the noise-to-ground mode.

5. Subtract the noise-to-ground measurement value from the Balanced reading to compute the relative line balance. A smaller reading (closer to 0) indicates a better balance.

Signal-to-Noise measurements are performed with a holding tone and are end-to-end measurements. The holding tone can be anything from 995 Hz to 1025 Hz but 1004 Hz and 1010 Hz are the most common tones used and are sent at data level (usually -13 dBm0).



To set-up the 930A for signal-to-noise measurements, perform the following steps:

1. Set the 930A in Office B to the SEND TONE function and select 1010 Hz at a -13 dBm0 level.

2. Set the 930A in Office A to MEASURE NOISE. Press Softkey 2 to select S/N and record the reading.

Signal-to-noise ratio provides a measure of the separation between the desired signal and the background noise. If the separation gets too small, the receiver may be unable to distinguish the signal from the noise. Unconditioned and C-Conditioned lines have a minimum S/N of 24 dB while D-Conditioning specifies a 28 dB minimum S/N. The higher the S/N ratio is, the less susceptible a modem will be to impairments.

Noise with Tone or Notched Noise is noise measured with a holding tone present to turn on all the tone activated components in the transmission path. This provides a more accurate measure of the noise while a data signal is present. It includes the noise contributions from active elements such as companders that are only active when a signal is present.

To measure noise with tone using the 930A, the test can be set-up as an end-to-end measurement or a single 930A can be used if the distant end is looped back (4-wire circuits only).

For an end-to-end measurement of notched noise, perform the following steps:

1. Set the far-end 930A to SEND TONE and select 1010 Hz at -13 dBm0.

2. At the near-end 930A, press the MEASURE NOISE function key and then press Softkey 2 to select the C-NOTCH weighted filter.

The value of the noise with tone is given in dBrnC. The C-NOTCH filter is a C-MESSAGE filter with a notch centered at 1010 Hz. The 930A has a notch filter with a notch depth in excess of 65 dB and very steep skirts to ensure that the holding tone is removed without causing a serious discrepancy in the noise measurement.

Impulse Noise Measurements

Impulse Noise is measured by counting the number of times the noise spikes exceed preset threshold levels. The preferred measurement technique employs three threshold levels. This is the method used by the 930A.

To measure Impulse Noise with the 930A it is first necessary to select the correct trunk type, as this is the start of all test procedures for the 930A.

After setting up the correct trunk type and supervision, connecting the test cords to the appropriate jacks and ensuring that Impulse Noise is audible on the 930A speaker, perform the following steps:

1. Press the Option Menu function key and then press 1, 1 and the ENT (Enter) key to go to Menu Option 11: 3-LEVEL IMPULSE NOISE.

2. Press Softkey 2 labeled SET-UP if it is necessary to change previously entered parameters. If no change is desired go straight to MEASURE. Pressing the SET-UP Softkey will lead to a choice between the DEFAULT parameters or the MANUAL mode which allows the user to change any or all of the parameters.

3. To measure Impulse Noise on metallic trunks use the DEFAULT test set-up parameters which are:

   54 dBrnC Low threshold
   4 dB steps between thresholds
   15 minute test duration
   7 measurements per second

4. To measure Impulse Noise on T-Carrier trunks it is necessary to use the MANUAL mode to set-up the test. After pressing the Softkey labeled “MANUAL”, press the Softkey labeled “C-NOTCH” to measure Impulse Noise with holding tone present or C-MESSAGE if it is required to measure without the holding tone.

5. Press the Softkey labeled THRESHold. Enter 6, 7 and press the ENT (Enter) key to change the Low threshold to 67 dBrnC which is the T-Carrier low threshold setting. Metallic trunks have a 54 dBrnC low threshold which is the 930A's default value.

All of the other parameters would stay the same to perform a standard measurement but the user could also change step size, measurement duration or measurements per second.

6. After setting up the test parameters, press the Option Menu function key once to return to the Impulse Noise display. Press Softkey 3 labeled "MEASURE" and then press Softkey 4 labeled "START" to begin testing.

7. To stop the test at any time, press Softkey 4, now labeled "STOP?".

8. Press the Option Menu function key to exit from the Impulse Noise menu.

Impulse Noise is not a tariffed parameter and so the telephone company internal standards are the controlling documents. Impulse Noise is normally allowed no more than 15 counts above the low threshold during a 15 minute period. Also, no more than 9 counts above the low threshold + 4 dB (Mid Threshold) and no more than 5 counts above the low threshold + 8 dB (High Threshold) are allowable during the same 15 minute period. The number of measurements per second is normally 7. Refer to Section 4-5 of the 930A Manual for details and pictorials of the Impulse Noise option.

Peak-to-Average Ratio (P/AR) Measurement

The Peak-To-Average ratio provides a measure of the channel dispersion (amplitude spreads over time) due to transmission impairments. The P/AR waveform consists of 16 non-harmonically related tones with a spectral content which approximates a MODEM signal. By measuring the peak-to-average ratio of the received signal, a measure of the dispersion is obtained. By comparing P/AR readings over a period of time, deterioration of a channel can be determined.

P/AR readings are a figure of merit rating of the quality of a channel, and are directly related to the attenuation distortion, phase distortion, return loss and noise. The P/AR measurement is therefore sensitive to envelope delay distortion, noise, bandwidth reduction, gain ripples, nonlinearities such as compression and clipping, and other impairments.

If the P/AR signal were received over a perfect channel entirely undistorted, the P/AR reading would be 100. A P/AR reading of 85 would indicate a 15 percent reduction in the peak-to-average ratio.

P/AR measurement provides no clues as to the nature of a fault condition in any single case. It is a figure of merit and as such is useful only when compared to previous readings. When a channel is suspect, however, a P/AR reading can be quickly compared to the benchmark value. If the reading deviates more than +/- 4 P/AR units from the benchmark then some channel characteristic (i.e., bandwidth, phase or amplitude distortion, etc.) has changed significantly. The value of the P/AR measurement is that it can be used to quickly determine the worst direction of transmission (near-to-far or far-to-near). The parameters of the worst transmission direction can be measured first, since it is most probable that the trouble lies in that direction.

Unfortunately, P/AR readings cannot pinpoint the impairment responsible for the degradation but they do provide a quick indication of possible problems. In fact, envelope delay distortion measurements are not performed for troubleshooting in many telephone companies unless the P/AR measurement has failed to meet guidelines.

P/AR is not a tariffed measurement and so telephone company standards are used to establish the objectives.

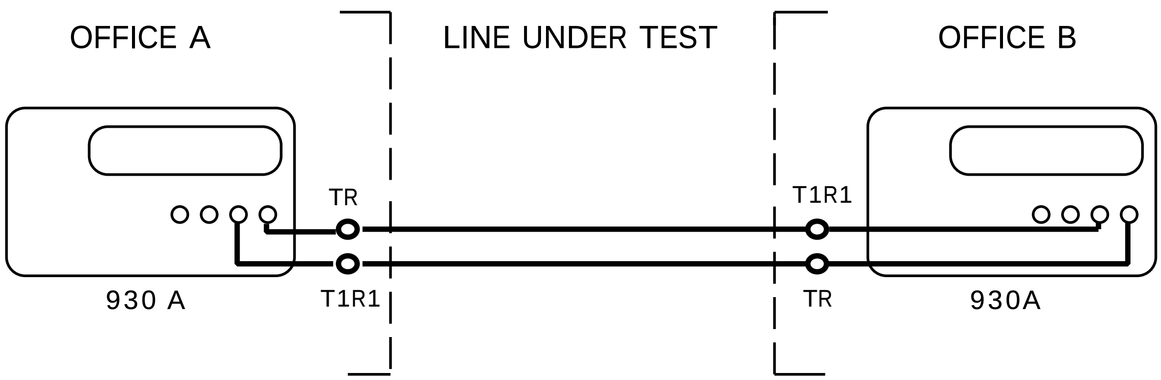
A minimum P/AR reading of 50 is required on end-to-end connections. An individual link, either an end link or mid link connection must have a minimum P/AR reading of 80.

On T1 carrier trunks, a P/AR reading of as much as 102 P/AR units is allowable while the maintenance limit is a minimum of 93 P/AR units. A typical P/AR measurement on a T1 trunk should be about 97 P/AR units.

Minimum P / AR values for non-repeatered cable less than 18,000 feet long are 97 for non-loaded and 94 for loaded cable.

Minimum P / AR values for repeatered cable are: 90 P / AR units for non-loaded cable less than 18,000 feet long, 90 P / AR units for loaded cable less than 36,000 feet long, and 80 P / AR units for loaded cable more than 36,000 feet long.

To measure P / AR using the 930A requires one unit to send and one to receive since P / AR is an end-to-end measurement. P / AR is not performed on a loopback basis due to the possibility of cancelling certain impairments. The same test set-up used for loss measurement can be used for P / AR. A 4-wire circuit is shown below.



At Office A set the near-end 930A as follows:

1. Press the Option Menu function key and then enter 1, 8 and press the ENT (Enter) key to go to menu option 18: P / AR.

2. Press Softkey 2 labeled “SEND” and enter the transmit level of the P / AR signal using the numeric keypad.

   P / AR is usually sent at data level (-13 dBm0). If the TLP is not 0 but -16 dB, for example, then the transmitted level should be -29 dBm to yield a -13 dBm0 level.

3. Press Softkey 3 labeled “MEASURE” to begin sending and measuring the P / AR signal on 4-wire circuits or to receive P / AR on 2-wire lines.

At Office B set the far-end 930A as follows:

1. Press the Option Menu key and enter 1, 8 and press the ENT (Enter) key to enter the P/AR menu.

2. Press Softkey 2 labeled “SEND” and set the transmit level of the P/AR signal if necessary in the same manner as before.

3. Press Softkey 4 to exit back to the main P/AR menu and then press Softkey 3 labeled “MEASURE” to begin P/AR measurement.

This procedure makes a P/AR measurement in each direction. The 930A can either send or receive P/AR on 2-wire circuits but naturally it cannot do both simultaneously. When measuring P/AR on a 2-wire circuit, one end must be set to the SEND mode while the other end must be set to the MEASURE mode. After a P/AR reading has been obtained from the MEASURE end of the trunk, the two test sets should reverse which one is SEND and which one is MEASURE so as to allow a P/AR reading at the other end.

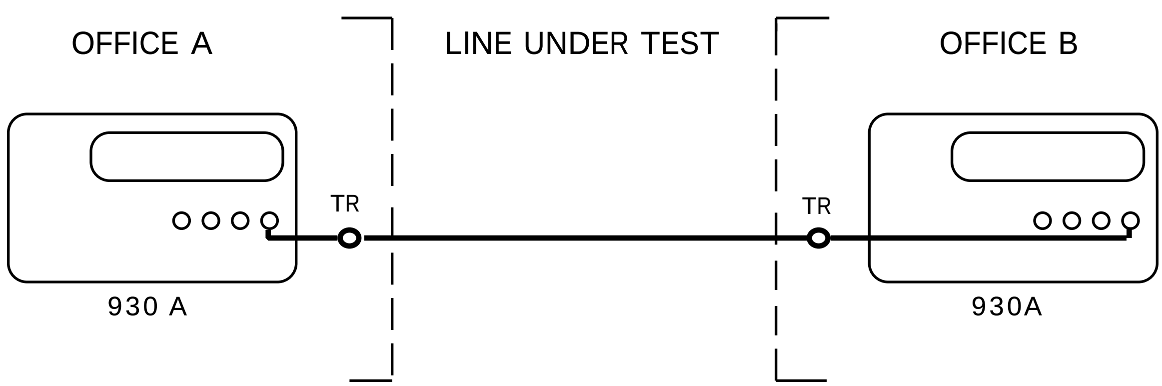
Intermodulation Distortion Measurement

Intermodulation Distortion, or Non-Linear Distortion, is described briefly on Page 22 and in Section 4-10 of the 930A Manual. The purpose of this section is to provide the user with a step-by-step example procedure for performing an IMD measurement to complement the example in Section 4-10.

Prior to performing the IMD test by sending and measuring the four test tones, the user must perform the signal-to-noise test in order to establish whether the distortion is due to overdriving the trunk under test. Most modern test sets are also able to correct their IMD measurement results for signal-to-noise automatically. Therefore, it is necessary for the test set to first perform this test and store the S/N value for use in correcting the IMD results.

The Model 930A, when equipped with Option 930A-20, enables the user to measure the S/N ratio and then make an automatically corrected IMD measurement. IMD is an end-to-end test and in this example, a 2-wire circuit will be used. It will also be assumed that the circuit is a C1 Conditioned line.

As with each measurement using the 930A, the starting point is selecting the correct trunk type and supervision. Next, the test cords must be connected to the appropriate jacks on the 930A front panel and to the circuit. The following diagram shows a 2-wire trunk with a 930A connected at each end of the circuit.



Once these steps have been performed at each end of the circuit, the operator is ready to begin testing.

To begin, press the OPTION MENU function key on the 930A in Office A and perform the following steps in order:

1. Use the numeric keypad to enter 19 and then press the ENT (Enter) key.

2. Once inside the menu option, the first screen is the main menu. The display over Softkey 2 should read "SEND". Press Softkey 2 to select the SEND mode.

3. The next step is to adjust the send level to the correct value for the trunk under test. A common value is -16 dBm and this is also the 930A default value. To change the value use the numeric keypad to enter the desired value and then press the ENT key to initiate the change.

4. Now send the signal-to-noise test tones by pressing Softkey 1 which is beneath the displayed words "SIG / NOISE".

At Office B, press the OPTION MENU function key and perform the following steps:

1. Use the numeric keypad to enter 19 and then press the ENT key.

2. Once inside the menu, press Softkey 3 which should be below the word "MEASURE". The 930A display should be reading the S / N ratio and should display the message "S / N TEST".

In order for the 930A to automatically correct the IMD measurements for signal-to-noise ratio, the 930A must have first received the signal-to-noise tones and must now be receiving and measuring the IMD tones. The displayed distortion is obtained from the following formula:

$$\text{Displayed Distortion} = Y + (-10 \log (1 - 10^{-.1X}) \text{ dB})$$

Where the signal-to-noise reading is X dB above the observed distortion Y.

At Office A, perform the following steps:

1. Press Softkey 2 under the displayed words "4-TONE" to begin sending the 4 IMD test tones.

At Office B the 930A will display the corrected values of the second and third order Intermodulation Distortion products as well as the average received level of the four test tones.

For this example, assume that the readings on the 930A at Office B are:

2nd	=	27 dB
3rd	=	29 dB

In this case, since a C1 conditioned line was assumed, the reading for signal to 3rd harmonic distortion is 1 dB low. This may or may not be a problem. It certainly indicates a marginal trunk but may not be cause for maintenance unless a customer has complained of transmission errors.

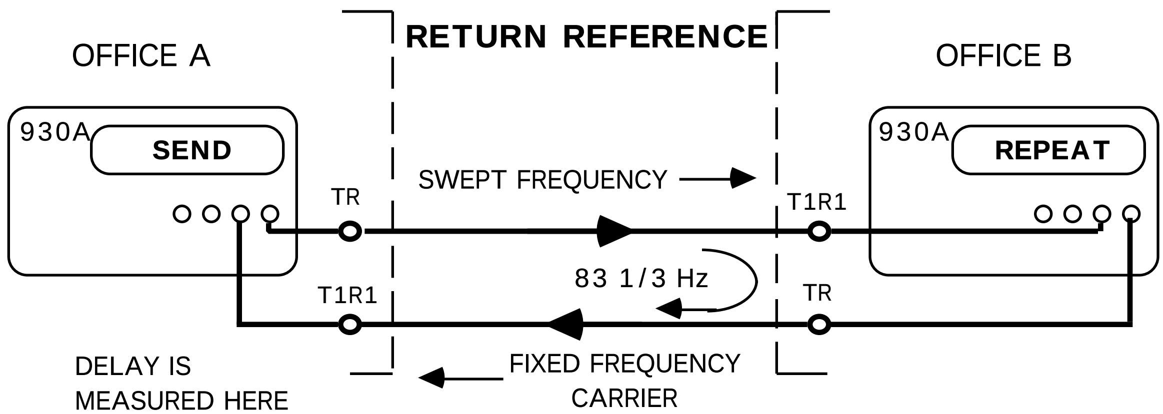
The next step is to reverse the procedures and test the line toward Office A from Office B. In this case, assume that the following readings were obtained from the 930A at Office A:

2nd	=	22 dB
3rd	=	24 dB

In this case the decision is more clear cut. This trunk would appear to have a problem which is worse in one direction than in the other and clearly fails all the criteria from Office B toward Office A. A one-way test on this trunk might not have provided cause for maintenance action but testing both directions clearly points up the need.

Envelope Delay Distortion Measurement

In this example, a Model 930A is connected at each end of the 4-wire trunk under test as shown in the diagram below:



Select the correct trunk type as described in Section 3-4 of the 930A Manual and connect the 930A as described in Section 3-4.4. Complete this step before proceeding.

Once this has been accomplished, the following steps should be performed in order to make a manual measurement of envelope delay distortion:

For each 930A:

1. Press the OPTION MENU function key.
2. Use the numeric keypad to enter the number 17.
3. Press the ENT (Enter) key to complete the numeric entry.

At the far-end 930A:

4. Press Softkey 3 to enter the REPEAT mode.

At the near-end 930A:

5. Press Softkey 2 to enter the SEND mode.
6. After the display has settled press Softkey 4 labeled “SETREF” to set the 0 µsec. reference.
7. Press Softkey 4 (now labeled “SWEEP”) again to begin a Return Reference measurement.

Since the 930A operates from default set-up values stored in memory, it is not generally necessary to change the parameters of the test. This makes the 930A easy to use on what is usually a very difficult measurement. The 930A default settings are: Sweep from 304 Hz to 3204 Hz in 100 Hz steps (with SF skip) with a time between steps of 3.5 seconds transmitting at a -16 dBm level.

Once the Return Reference sweep has been completed, the near-end 930A will stop and send the fixed reference frequency carrier of 1804 Hz at -16 dBm toward the far-end. The operator at the far-end must now press Softkey 4 labeled “SWEEP” to send the sweep frequencies toward the near-end. This is the Forward Reference measurement because the near–end is now supplying the reference toward the far-end while the delay is being measured on the return path.

Envelope delay is one of those measurements no one likes to make because of the problems in coordinating an end-to-end measurement with the people at the far-end.

Section 4-9 of the 930A Manual provides an in-depth treatment of the possible settings and options available for the 930A when measuring Envelope Delay.

Phase/Amplitude Jitter and Hits Measurement

This section expands upon the information presented in Section 4-8 of the 930A Manual. Phase Jitter is a parameter which is important to measure on circuits carrying voice band data but which has no impact on speech circuits.

Prior to making a measurement of Phase Jitter, the C-Notch Noise measurement must be made to assure that excessive noise is not the cause of the phase jitter.

The Model 930A can make three types of measurements of phase jitter. These measurements are differentiated by the frequency band over which they are performed. They are: the STD or Standard (20 to 300 Hz), the LF or Low Frequency (4 to 20 Hz) and the STD + LF (4 to 300 Hz). Phase Jitter is measured by sending a 1004 Hz holding tone at -13.0 dBm as a reference and then recovering the tone at the far-end using a phase-locked loop. The phase-locked loop error voltages, instead of being used to correct the received signal, are used as an indication of the Phase Jitter present on the incoming 1004 Hz tone reference. The 930A also measures Amplitude Jitter over the same frequency ranges as Phase Jitter.

The measurement of Hits or Transients consisting of Phase Hits, Gain Hits and Dropouts is also provided under Option 930A-18 . The results are displayed as a sub-menu of the Impulse Noise measurement.

The set-up of the 930A for the measurement of Phase and Amplitude Jitter is relatively painless and is covered in Section 4-8.1. The interpretation of the results of the measurements and possible sources of problems is also covered there. Section 4-8.2 describes the measurement of Phase Hits, Gain Hits and Dropouts as well as 3-Level Impulse Noise.

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9120-0930-02