Topic 2: Existing Telecommunication Networks (cont'd).

Lesson C_3: Wire-line Transmission Links

Session 1

Contents

Lesson Outline

Session 1 of 3

3.1 Telephone Transmission Basics for Voice

In the introductory stages of communication engineering, you learned that the channel is the fundamental component of the whole communication system. The telephone system is a special communication system optimised for the transfer of voice-frequency information. We thus call the telephone channel the voice channel.

The voice channel occupies a special spectrum irrespective of the chosen medium. For a telephone instrument equipped with an ideal transducer the voice frequency spectrum would stretch from 20 Hz to 20 kHz. The maximum aural sensitivity of the human ear is for signals from 30 Hz to 30 kHz. If we agree that the primary content of a voice is energy level and emotion, then the essential band is only 100 Hz to 4 kHz. On the basis of this understanding, the ITU-T defines the voice frequency band as occupying 300 to 3400 Hz. Similarly, the Federal Communications Commission (FCC) of the United States defines it over the band 0 to 4000 Hz.

Obviously, the bandwidth and the spectral content are the essential parameters of the voice channel. The other main parameters are outlined below. For an in-depth treatment of wire-line transmission, see chapters 5 to 9 of [1]

3.1.1 The Other Main Parameters

The other essential parameters to consider in designing wire-line transmission systems are:

- distortion due to attenuation,
- distortion of the phase due to the finite time of signal transmission over the channel,

- changes to the signal power level or intensity referenced to a planning chart level, and
- the noise level whose significance is determined by the signal-to-noise ratio.

Other parameters such as singing, stability, echo, and reference equivalent are rather network-specific and will not be treated in this lesson.

3.1.1.1 Attenuation Distortion

The frequency response (or attenuation distortion) of the channel is frequency selective, meaning that certain frequencies are attenuated more than others. On wire pairs, the channel attenuates higher frequencies more than the lower frequencies. Band-pass filters used on carrier equipment attenuate the most towards the edges of the band. The response of a channel is *flat* if when tested from input to output it varies by not more than several decibels. For voice channel modems, it should vary by at most ±0.5 dB.

Normally, we select a reference frequency to measure attenuation against. Attenuation at the reference frequency is normalised to zero dB. Africa, Europe and some Hispanic American countries use 800 Hz as the reference frequency while North America prefers 1000 Hz. An example of the use of the reference frequency would be the requirement that the signal level between 600 and 2800 Hz should not vary by more than –1 dB and +2 dB respectively from the level at 800 Hz. The negative index indicates gain while the positive index is for attenuation. *Apply the limits to a –10 dBm signal.* A typical attenuation versus frequency response profile is shown in **Fig. 3.1.1**.

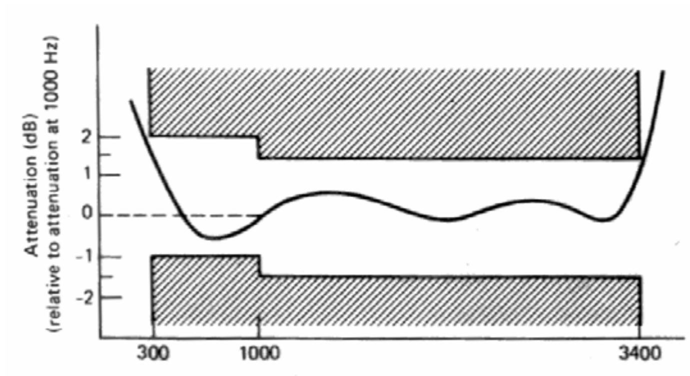

Figure 3.1.1 A typical voice frequency attenuation distortion profile within specified limits.

3.1.1.2 Phase Distortion

Like any other channel, the voice channel acts as a band-pass filter. The finite time a signal takes to pass through a filter is called *delay* and is a function of the velocity of propagation (which in turn is a function of the medium characteristics and frequency). The velocity is greatest near the centre of the band. See **Fig. 3.1 2**.

Delay at the reference frequency is called absolute delay. Signal components at other frequencies experience relative phase shifts or distortion compared with the delay at the reference frequency. Distortion occurs in modulated signals only due to the non-linear phase shift with respect to frequency. In other words, phase distortion depends on the phase linearity of the system elements. If we measure the partial derivative of the phase shift with respect to frequency, we obtain the parameter *envelope delay distortion* (EDD) in milliseconds or microseconds. EDD is the maximum difference on the derivative (slope) over any given frequency interval within the pass-band.

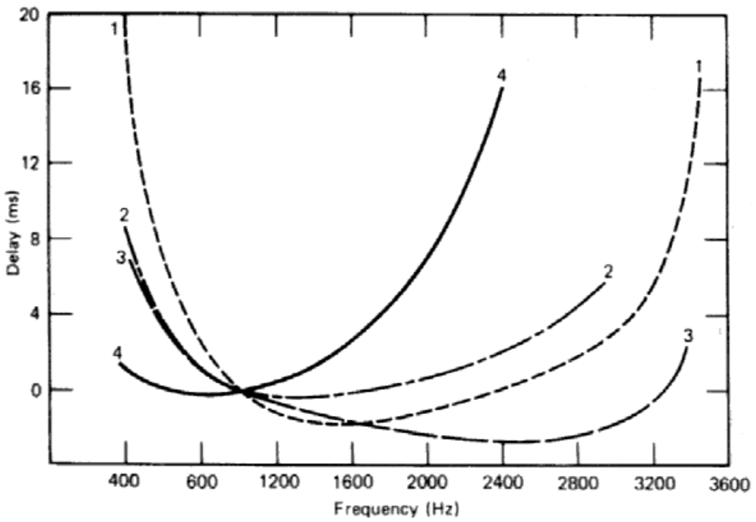

Figure 3.1.2 Envelope delay curves for various typical voice channels. Curves 1 and 3 stand for several thousand kilometres' worth of delay in a toll-quality carrier system. Curve 2 shows delay in 160 km of loaded cable. Curve 4 shows delay in 320 km of heavily loaded cable. Measurements were done by the GTE Lenkurt Demodulator, San Carlos, CA.

3.1.1.3 Signal Power Level

We measure the signal power level in telecommunications in units such as dBm, dBW or picowatts, except in video where level refers to voltage level in dBmV mostly. The crucial issue in the level parameter is the balance between a high enough level to maintain customer satisfaction and one low enough to operate below saturation (in the linear region of amplifiers). When we overload amplifiers we risk incurring intermodulation products or crosstalk.

Normally, a planning section sets system levels and establishes a level chart or reference system drawing on which a zero test-level point (0 TLP) is established. The TLP is one place in a circuit or system at which we expect to hear a given test-tone level in the process of alignment. The specified test-tone level is the 0 TLP that should register at 0 dBm the reference point. From the 0 TLP, we establish other test point using units of ±dBr (so many dB above or below the decibel reference).

3.1.1.4 Noise and Signal-to-Noise Ratio

Recall that in communication circuits we regard any undesired signal as noise and it is a major limiting factor in system performance faced by transmission engineers. It is important for you to distinguish among four types of noise in wire-line technology: Gaussian noise, intermodulation noise, crosstalk and impulse noise.

Thermal noise is a property of all communication equipment components and the transmission media as long as their temperatures are above 0 K. Random electron motion gives rise to a uniform spectral distribution of energy over the entire frequency spectrum. The respective energy levels, though assume a normal distribution, hence the term Gaussian noise. Gaussian noise then forms the lower limit of sensitivity of any receiver. It is defined in terms of the absolute device temperature, T, as

$$N_0 = kTB$$

where $k = 1.3803 \times 10^{-23}$ J/K $\equiv$ -228.6 dBW/Hz, B is the noise bandwidth for band-limited systems. In logarithmic terms then,

$$N_0 = -228.6\,\text{dBW} + 10\log_{10} T + 10\log_{10} B$$

When two or more signals at different frequencies pass through a non-linear device or medium they mix (beat together) producing spurious harmonics that may extend even beyond the band in question. These harmonics are called intermodulation products and their effect is called *intermodulation noise*. Causes of intermodulation products include:

- Overdrive of devices due to improper level setting
- Improper device alignment causing non-linear operation
- Non-linear envelope delay
- Mere device malfunction

Intermodulation noise in multi-channel systems with complex signals can be varied enough to resemble thermal noise even though its causes are different.

Sometimes signal paths undergo unintentional coupling, a phenomenon called *crosstalk*. On voice frequency cable systems, the wire pairs may become electrically coupled. Defective or poorly designed filters may lead to poor control of frequency response. At times an FDM system may act non-linearly. At worst, crosstalk may be *intelligible* when in any 7-*s* period a user is able to distinguish at least four words from stray conversations. Otherwise it is *unintelligible*. The subjective effects of crosstalk vary with the type of people using the channel, their aural acuity, traffic patterns, and operating practices of the telecommunications administration involved.

In analogue voice telephony, *impulse noise*, which occurs in short bursts or spikes of relatively high amplitude, causes imperceptible degradation. On the other hand, digital or other data networks subjected to it may suffer from intolerable *error rates* that must be controlled through *burst-error control* strategies. Error-control coding treats impulse noise effects at greater length.

In order to design a fairly noise-immune telecommunication system, it is important to determine by how much the desired signal exceeds the noise threshold within the bandwidth of interest. The minimum necessary signal-to-noise ratio (SNR) varies with the commodity being transmitted. E.g. voice requires 40 dB, video, 45 dB and data, about –15 dB. Voice and video depend on customer satisfaction while data depends on specified probability of error objectives and the modulation type.

Summary

In this session, the main point is that the telephone channel is optimised for the transmission of speech rather than data, telegraphy or facsimile. It transmits multiplexed voice band signals either in the baseband frequency band in either the Time Division Multiplex mode or the Frequency Division Multiplex mode, mostly. The essential parameters of the voice channel are the bandwidth and the spectral content. Other important parameters are attenuation distortion, phase distortion, signal power level changes, and the noise level.

In the next session we look at two-wire to four-wire transmission and multiplexing.

*************** **End Topic 2 Session 1 of Lesson C_3** ***************

References

[1] Roger L. Freeman; *Telecommunication System Engineering, 2/e*; John Wiley, 1989

[2]. Leon W. Couch II; *Digital and Analogue Communication Systems, 4/e*; Macmillan, 1993