LESSON 33: DATA TRANSMISSION IN PSTNS AND SWITCHING TECHNIQUES FOR DATA TRANSMISSION

Objective

To provide a detailed understanding of the concepts of data transmission in PSTN's

Introduction

In 1878, the first manual exchange was constructed in La Porte, in the United States. At its inception it served 21 subscribers and could connect any two of them together. A ringing signal sounded at the operator's switchboard when any of the subscribers turned the crank of his telephone. Upon answering the signal, the operator was asked to connect the call to one of the other subscribers, which she did manually. She also made a note of who placed the call and when it started and stopped - notes that made it possible to charge for the call.

We can say that the market forces of the early 1890s prompted the development of the first automatic telephone exchange. It was called the "Strowger switch", after its originator Almon B. Strowger. Strowger was an undertaker from Kansas City who, soon after the advent of the telephone, found himself exposed to a serious form of unfair competition. The woman who operated the manual exchange was the wife of Strowger's competitor, and she connected anyone who asked to speak with an undertaker to her husband.

Many years passed during which electromechanical telephone exchanges were developed and improved. The primary objective was still the same as that of the manual exchange: to detect the A-subscriber's call attempt, to connect him to the correct B-subscriber, and to save data about the call for the purpose of billing. The 1960s and 1970s saw the advent of telephone exchanges that were controlled by processors: stored program control (SPC) exchanges. As a result, new functions could be built into the exchanges - of benefit to subscribers and network operators. These functions enabled the introduction of new types of service and facilitated supervision, charging and the gathering of statistics.

In the 1960s, the PSTN was the network, of which the transmission components, in the form of cables and radio links (employing analog transmission), were integral parts. Today we have a variety of bearer networks and a transport network based on digital transmission, which all the bearer networks have in common to a great extent. Pure PSTN transmission exists only in the access part of the network.

Data Transmission In Pstns

Public Switched Telephone Networks (PSTNs) and Electronic Private Automatic Branch Exchanges (EPABXs) are designed to carry analog voice signals. They can however, be used for data transmission by employing suitable interfaces.

LANs can be designed around PABXs and MANs around PSTNs. In these cases, the data rates are usually limited to a maximum of 64 kb/s. Terrestrial data networks and the upcoming Integrated Services Digital Networks (ISDNs) can, however, support data rates of 1.544 or 2.048 Mbps.

Transmission of digital data signals over PSTN networks demands that the digital signals be converted to analog form at the transmitting end and vice versa at the receiving end. A modulator translates the data pulses into voice band analog signals at the transmitting end. A demodulator translates voice band analog signal to digital information at the receiving end. In other words, we can say that at the receiving end, the voice band analog signals are demodulated to recover the digital information. A combined modulator l Demodulator unit is called MODEM.

Initially, modems were used to connect terminals, located in remote places, to a central computer. Later, computer-to-computer communication was established using modems and PSTNs. A data communication scheme using modems and PSTNs is shown in Fig. 33.1

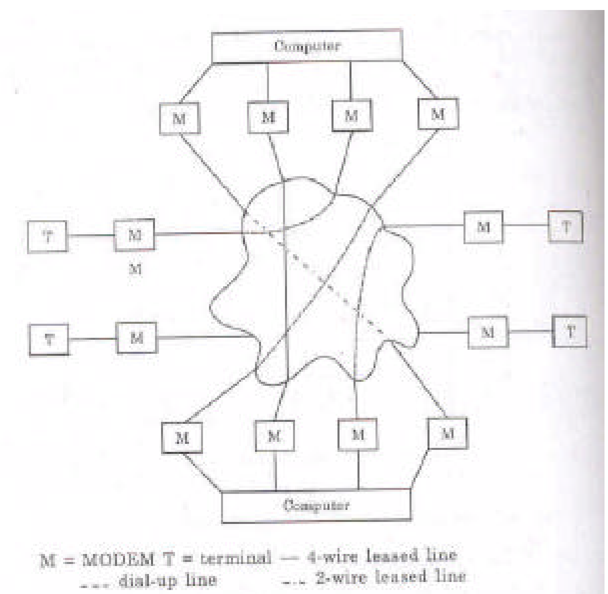

Fig.33.1 Illustration of data communication using PSTN

The digital interface of a MODEM is connected to the computer and the analog interface to the telephone network. In addition to data exchange, the digital interface permits control signals to be exchanged between the MODEM and the computer.

In PSTN, a connection may be established using a dial-up facility or a dedicated non-switched leased line. A 2-wire or 4-wire leased line may be used.

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The modems usually have automatic dialing and answer facility which is useful when working with dial-up line. MODEMS are driven by a communication software package. This software package runs on the terminal or the personnel computer of the user. The transmission may be half-duplex or full duplex.

Both terminal-to-computer and computer-to-computer connections are shown in Fig. 33.1.

Data Rates in Public Switched Telephone Networks (PSTNs)

A voice channel in a PSTN is band limited with a nominal bandwidth of 3.1 kHz. Here we want to know the maximum data rate that a voice channel can support with a limited bandwidth. A first-cut estimate of this can be obtained from Nyquist's theorem which applies to noiseless channels and states:

$$R_m = 3\ B \log_2(L),\ \text{bps}$$

Where

R_m = maximum data rate

B = Bandwidth of the channel

L = number of discrete levels in the signal levels

For a 3-kHz ideal channel, and a binary signal, the maximum data rate works out to be 6000 bits/sec. In a practical channel, the maximum bit rate would come down. In a noiseless channel, the bit rate is proportional to the number of discrete levels used to represent the signal. It is important to recognise that the actual number of signal transitions is still limited to the binary level limit; the effective bit rate goes up with more than two signal levels as each signal level can now re-present a group of two or more bits.

The maximum rate of signal transitions that can be supported by a channel is known as **Baud Rate** or **Symbol Rate.**

In a noisy channel there is an absolute maximum limit for the bit rate. The limit arises because the difference between two adjacent signal levels becomes comparable to the noise level when the number of signal levels increased. Claude Shannon extended Nyquist's Work to the case of Noisy Channels affected by random or thermal noise.

Shannon's Channel capacity theorem is stated as:

$$C = B \log_2 (1 + SNR)$$

Where

C = Channel Capacity or maximum bit rate of information transmission over a channel

B = Bandwidth of the channel

SNR = Signal-to-Noise ratio

Typical SNR for thermal noise of telephone channel is about 30 dB, i.e. a ratio of 1000. Hence, for B= 3 kHz, C works out as 30,000 bits/sec. C is the absolute maximum: bit rate irrespective of the number of levels used to represent the signal.

Up to now we have discussed two aspects of data rates in a voice channel. These are:

1. The baud rate or sym1;l01 rate that can be supported in noiseless channel
2. Maximum bit rate that can be supported in a noisy channel.

The baud rate and bit rate are related as:

$$C = R_m \times n$$

Where, n = Number of bits required to represent signal levels.

For binary signals, n = 1 and $C = R_m \times 1$ or $C = R_m$ i.e. band-width required for the channel is $(R_m / 2)$ Hertz .

Now, we would like to determine the baud rate that can be supported in a practical environment. Here, there are two opposing factors:

1. On the one hand, the channel is noisy, which goes to reduce **the maximum baud rate2.**

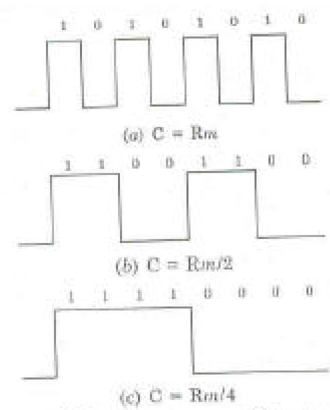

Fig.36.2 Baud rates and bit rates.

On the other hand, the data is rarely an alternating pattern of l's and O's so that the effective transition rate (baud rate) in the data are much lower than the bit rate. This is illustrated in Fig.33.2.

In Fig, 36.2 (a) the baud rate is equal to the bit rate. In Fig. 36.2 (b), and 36.2 (c) the baud rate is one-half and one-fourth, respectively.

It has been observed that the baud rates up to 2400 bauds may be transmitted reliably through a PSTN voice channel. By using additional signal levels, say 16, the effective bit rates may go up to 9.6 k bit / sec. The techniques of using additional signal levels to increase the bit rate are the ones used in the design of MODEMS.

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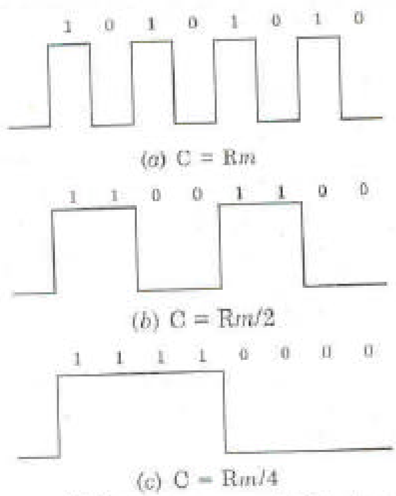

Fig.36.2 Baud rates and bit rates.

Switching Techniques For Data Transmission

Telephone networks are basically designed to carry voice traffic and there are some significant differences in the nature of voice and data traffic. Voice traffic is generally continuous (except for silence periods in normal speech). But the data traffic is bursty in nature. When a user sits at a terminal arid works with a computer, interactively, he/she spends time in thinking, keying in the query or command to the computer and waiting for a response from the computer before proceeding further. In other words, the user is in one of three states; thinking keying-in or waiting.

During thinking and waiting states no data transmission takes place, during the waiting state the computer is busy processing the user command. In order to have a low waiting time the user input must be transmitted to the system expeditiously soon after the keying-in is over. Similarly, once the response is computed by the machine, the same must be transmitted to the user at a fast rate. Thus the data traffic is bursty in nature and it calls for high bandwidth for short durations. This is true of the data traffic between two computers also. In contrast, voice traffic needs low bandwidth (3400 Hz) for long durations. In such a case, the transmission line is idle for 85·95% of the holding time in the case of data transmission and is busy for a similar period in the case of a telephone conversation.

Pauses in normal speech are considered as active transmission periods. When voice traffic is half-duplex then data traffic may be half duplex or full duplex. There is another important difference lies in acceptable error or loss rates. In data transmission, no errors or losses are acceptable.

But in speech transmission, very small amount of speech loss is acceptable. While the speech traffic always takes place in real time, the data traffic mayor may not occur in real time. We require real time or near real-time response in interactive use of computers. But; file transfer between two computers need not be in real time. Summary of differences between voice and data traffic is given in Table 33.1.

Table 33.1 Differences Between Voice and Data Traffic

Sr. No.	Voice Traffic	Data Traffic
1.	This traffic is continuous	This traffic is bursty
2.	It requires low bandwidth for long duration	It requires high bandwidth for short duration
3.	In this traffic, typical line utilization is 85·95%.	In this traffic, typical line utilisation is 5-15%.
4.	It is half duplex	It is half or full duplex
5.	It is real time	It is non-real time or near real time.
6.	Loss acceptable	Loss unacceptable
7.	Here, error is tolerable	Here, error is unacceptable.

The recognition of the diverse characteristics of voice and data traffic has led to the development of' a switching technique other than the one used for voice transmission. This technique is better suited for transmitting data traffic.

Hence, two switching techniques are prevalent for data transmission:

1.Circuit Switching

2. Store and Forward (S and F) Switching

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