Chapter 8 TARIFFS

The services which a common carrier offers to the public and their prices are described in tariffs. Most of countries, the telecommunication facilities are set up by government bodies and the tariffs are directly under their control.

CATEGORIES OF LINE

1. Subvoice grade: Lines designed for telegraph and similar machines transmitting at speed ranging from 45 to 200 bps.
2. Voice grade: At present telephone channels normally transmit at speed from 600 to 9,600 bps. A speed of 9,600 is possible but requires elaborate modern design and powerful facilities for error correction.
3. Wideband: Wideband lines give speeds much higher than voice channels, using facilities that carry many simultaneous telephone calls. Speeds up to about 500,000 bps are in use, and higher bit rates are possible.

SWITCHED VERSUS LEASED LINES

The important parameter about the lines is whether they are public switched lines or not.

Voice lines and telegraph lines can either be switched through public exchanges (central offices) or permanently connected.

When you dial a friend and talk to him on the telephone you speak over a line connected by means of the public exchanges. This line, referred to as a "public" or "switched" line, could be used for the transmission of data. Alternatively, a "private" or "leased" line could be connected permanently or semipermanently between the transmitting machines. The private line may be connected via the local switching office, but it would not be connected to the switchgear and signaling devices of that office.

CATEGORIES OF SWITCHING

Telephone switching is designed primarily for connecting people to people, not machines to machines. As mentioned before machine to-machine communication has different requirements. One of the differences is in the switching. Telephone switching is slow, typically taking between 10 and 30 seconds to complete a call connection.

But new networks specifically designed for date transmission are now operating, which have high-speed switching. For these networks the tariffs can be different from telephone tariffs in that they permit shorter inter-connection and charge a proportionately low price for it. The minimum billing interval on the Pakistan telephone network is one minute. The minimum billing interval on the SPC system, a specialized common carrier switched data network in the U.S.A, is one second.

Networks which switch circuits electronically in a fraction of a second are referred to as fast-connect networks.

An alternative to the switching of circuits - the interconnection of electrical paths - is a network which interleaves the data from different users ob fixed paths. Short messages can then be transmitted economically because many users share the facility. The interleaving can be done by small network computers which briefly store the messages from different users and then transmit them on shared transmission paths. This can be done by a concentrator network or a packet-switching network. Networks which store and forward messages in this way can have another type of tariff structure in which the user is charged by the amount of data transmitted rather than by the duration of the connection.

In some data systems there is no urgency to transmit the messages in a short time. This is the case, for example, with message delivery systems. Such systems can maximize the line utilization by filing messages at the switching points until the required circuits is free, waiting if necessary until off-peak periods. This technique is referred to as message switching. Let us note here that there are four fundamentally different types of switching which lead to different tariff structures:

i. conventional telephone switching
ii. fast-connect circuit switching
iii. packet switching
iv. message switching

PROS AND CONS OF LEASED LINES

Leased voice lines have certain advantages for data transmission over switched telephone connections. Let us summarize these advantages.

1. If it is to be used for more than a given number of hours per day, the

leased line is less expensive. The break-even point depends on the actual charges, which in turn depend on the mileage of the circuit, but it is likely to be of the order of several hours per day. This factor is clearly an important consideration in designing a data transmission network.

2. Because the leased line is permanently connected, there need be no delay associated with switching times. Leased lines are therefore better than telephone switching systems for applications requiring fast access to a distant computer.

3. Private lines can be specially treated or "conditioned" to compensate for the distortion that is encountered on them. The common carriers charge extra for conditioning. In this way the number of data errors can be reduced, or, alternatively, a higher transmission rate can be made possible. The switched connection cannot be conditioned beforehand in the same way, because it is not known what path the circuit will take. Dialing at one time is likely to set up a quite different physical path from that obtained by dialing at another time, and there are a large number of possible paths.

4. Switched voice lines usually carry signaling e.g. Multi Frequency signal within the bandwidth that would be used for data. Data transmission machines must be designed so that the form in which the data are sent cannot interfere with the common carrier's signaling.

5. Mostly the leased line is less interfered by noise and distortion than the switched line. The switching gear can cause impulse noise that results in errors in data. This is a third factor that contributes to a lower error rate for a given transmission speed on private lines.

There are certain advantages which switched lines can have over leased lines, as follows:

1. If the terminal or terminals at a location have only a low usage, switched lines will give a low overall cost.

2. The ability to access multiple distant machines using a switched network gives great flexibility. Many different machines offering services and with different data bases may be dialed by a terminal user.

3. If a leased line fails, its user may be cut off from the facilities it connects to. With a switched system the user can redial and may obtain an alternate path to the facilities.

4. If a computer is overloaded or under repair, its user might be able to dial an alternative computer.

5. Simplicity. Leased-line system often become complex because of the techniques used, such as polling, concentrators, and multipoint lines, which allow separate users to share the line.

LINE CONDITIONING

As has been mentioned private leased voice lines can be conditioned so that they have better properties for data transmission. Tariffs specify maximum levels for certain types of distortion. An additional charge is made by most carriers for lines which are conditioned. Different frequencies suffer different attenuation and different signal delay. So, the conditioning attempts to equalize the attenuation and delay at different frequencies.

Chapter 9 NOISE AND DISTORTION

On any telecommunication link there will be noise and distortion of some degree. A variety of techniques can be used in engineering the link to reduce this to an acceptable minimum.

Unfortunately, the acceptable criteria for data transmission differ from those for other uses of the link, such as voice. There can be a considerable degree of impulse noise and distortion on a voice line without the speech becoming unintelligible, annoying, or too unnatural. Similarly, teletype lines can be noisy; if telegram arrives with a few incorrect letters, it is still basically readable and understandable to a human being.

Voice transmission differs from data transmission in two fundamental ways. First, with voice we have intelligent agency at each end of the line. If a burst of noise or other failure prevents the listener from hearing a word, he can either guess what the word should have been or else ask the speaker to repeat it. If he cannot hear, he will ask the speaker to speak louder or clearer. This highly flexible intelligence does not exist when machine talks to machine, so rigid, precise procedures are devised which are as fail-safe as possible. Second, the information conveyed by the human voice is at a very much slower rate than at which we want the machines to "talk." The normal rate of speaking is equivalent to something like 40 bps of written words.

On the other hand, data transmission over voice channels can take place at 4,800 bps and higher. It is because of the low rate of information in speech and because of the adaptability of the human ear that distortions and noise levels damaging to data transmission have been quite acceptable in the telephone lines.

SYSTEMATIC AND FORTUITOUS DISTORTION

We may classify the disturbances into two types: systematic and fortuitous. Systematic distortion is that which occurs every time we transmit a given signal over a given channel. Knowing the channel we can predict what is going to occur. The pulses may always be narrower or always be distorted in a certain way. Given frequencies will always have a certain minimum phase delay. Fortuitous distortion is something which occurs at random, and so it is not predictable, except in terms of probability. Examples of fortuitous distortion are white noise, impulse noise, chatter from the switchgear, crosstalk, atmospheric noise, sudden changes in signal phase, and brief losses of signal amplitude. Fortuitous distortion refers to transient impairments rather than continuing conditions on the line.

Systematic distortion is then something which might possibly be compensated for electronically so that its effects are eliminated. Fortuitous distortion is more difficult to compensate for, though steps can be taken to minimize its effects and repair the damage it does. It may be possible to correct systematic distortion so that it never actually damages data. Fortuitous distortion, on the other hand, is occasionally going to produce an extra large noise burst or impulse which destroys or creates one or more bits at random.

Box summarizes the main types of transmission impairments of concern. We will first discuss the various types of fortuitous distortion.

Box 9.1 Types of impairments which afflict communication circuits

Systematic Distortion (Static Impairments)	Fortuitous Distortion (Transient Impairments)
Loss	White noise
Attenuation distortion	Impulse noise
Delay distortion	Crosstalk
Harmonic distortion	Intermodulation noise
Frequency offset	Echoes
Bias distortion	Changes in amplitude
Characteristic distortion	Radio
	Line outages
	Radio fading
	Changes in phase
	Phase jitter
	Dropouts

WHITE NOISE

White noise is the random hiss that forms a background to all electronic signaling. It cannot be removed, and so it sets a theoretical maximum on the performance of any communication link and on the various modulation methods. The amplitude of the signal after attenuation must be kept sufficiently far above the white noise background to prevent an excess of hiss on radio or telephone circuits or an excess of errors in data transmission. On the majority of lines the signal-to-white-noise ratio is better than 30 decibels, although occasionally a dialed call will encounter a much worse ratio than this.

Occasionally there will be spikes of white noise higher than the majority and peaks of other noise types such as crosstalk which may add to the white noise.

IMPULSE NOISE

Unlike white noise and the various types of systematic distortion, impulse noise can have peaks of great amplitude which saturate the channel and block out data. Impulse noise is the main source of errors in data. The duration of the impulses can be quite long relative to the speed of data transmission-sometimes as long as 0.01 second, for example. This would be heard merely as a sharp click or crack to a human listener and would not destroy and verbal intelligence, but if data were being transmitted 75 bps, one bit might be lost. For speeds of 4,800 bps a group of 50 or so bits would be lost.

Often an impulse noise removes or adds two or more adjacent data bits, and this means that odd/even parity checking may not detect the error. A more sophisticated form of error-detection code is needed.

Figure 9.1 gives an illustration of the effects of a burst of noise. In this example the signal-to-noise ratio is low, as is indicated by the amplitude of the white noise ratio is low, as is indicated by the amplitude of the white noise relative to that of the signal. In this diagram the sampling is shown taking place at one instant. Some systems take a series of samples throughout the intended duration of the pulse and thus lessen the probability of short noise spikes giving incorrect results.

There are many causes of impulse noise, some of which can be controlled but most of which cannot without a complete reengineering of the telecommunication facilities. Some impulse noise is audible during telephone conversations and some goes unnoticed. Stray clicks and crackles are all too familiar. Impulse noise comes from a variety of different sources. It may come from within the communications channel itself or from a source external to the channel.

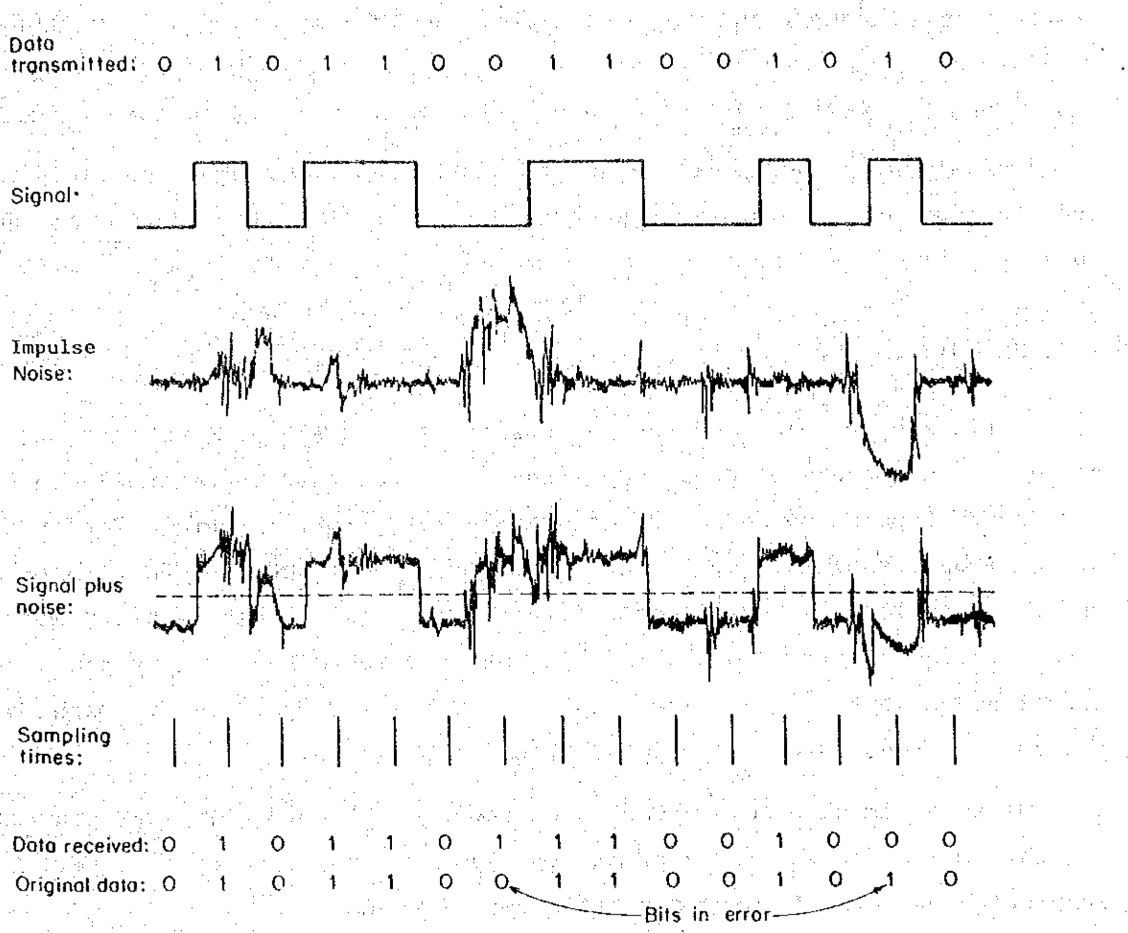


Fig.9.1 Transmission in the presence of bad noise

External noise is picked up by induction or capacitance effects. Sharp voltage changes in adjacent wires or equipment induce noise spikes in the communication channel. Many of the audible clicks which are of high amplitude and which damage one or several adjacent bits come from switchgear and telephone exchanges. Sometimes one can hear the rapid sequence of clicks generated by another person dialing. All relay operation is potential source of noise if the shielding or suppression is not adequate. Any switches or relays which make or break circuits carrying current cause a sharp voltage change and so can induce an equivalent voltage change in nearby sensitive circuits. Sometimes the power supply may induce hum or higher-frequency components into the communication channel.

The inductive or captive coupling through which noise is induced may be in the exchange. It may be coupling between adjacent cable pairs which are physically close. The noise generated by relays and switches may travel down wires a long way before reaching the low-level transmission signals, and may come from plant in separate buildings. Exchanges with step-by-step, like EMD switches, are worse than those with crossbar switches. Often these switches are the most important noise source on a line. Figure shows characteristics of a typical noise burst caused by step-by-step selection switches. Amplitudes of 100 millivolts are common in such disturbances. They can occur at points where the signal strength is low and may last up to several hundred milliseconds. The periodicity is caused by the step-by-step motion of the selector.

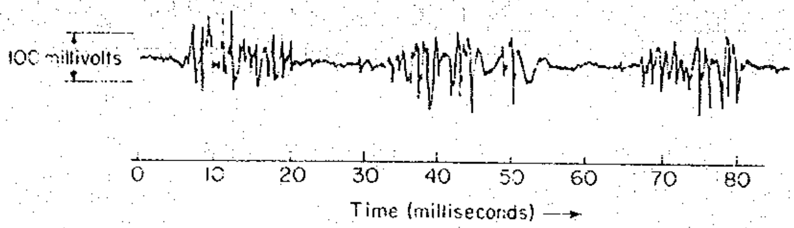


Fig. 9.2 Typical example of impulse noise caused by Strowger switches in a public exchange

Noise usually causes errors in data to occur in a cluster rather than singlely.

In years to come, when the present exchange equipment is largely replaced by electronic exchanges, the noise from this source will probably be much less. Today, however, lines going through public switching offices are more apt to have noise than private lines. Other sources of impulse noise on public lines are the various ringing tones that are used, and the devices used during maintenance such as breakdown test-sets and buzzers.

External noise can also come from atmospheric sources. Open-wire pairs hanging between telegraph poles can pick up atmospheric static. They are affected by distant lightning flashes and sometimes by contacts with trees or other foreign objects. Sometimes power lines or radar transmitters can cause interference. Radar interference has caused trouble with computer systems installed at aiports. Electric trains or electrical machinery sometimes cause periods of severe noise.

Impulse noise also originates from within the communication channel itself. This may be caused by circuit faults such as poor quality soldering and dirty relay contacts and jacks. Nonsoldered twisted joints may cause noise due to changes in temperature and slight movements of the joint. In all these cases a variation in the contact resistance causes a fluctuation in voltage.

Many circuits carry more than one channel, in such systems there is usually a small amount of crosstalk between one channel and another. An extra strong signal or impulse on one channel will exceed an overload point in the amplifiers and other devices, and various effects of this will be felt in the other channels. The signal which causes the overload may be a noise impulse, or it may result merely from the fact that all the signals being transmitted by the same multiplex device happen to be at a peak at that instant. The sum of the peaks extends for capacity of the channel for distortion-free transmission. This fortuitous adding together of peaks occurs in most multiplex systems, especially during busy periods. Harmonics and modulation products spill into other channels.

INPULSE NOISE REDUCTION

There are many different means of reducing the effects of impulse noise. First, good screening can be used and careful planning of the circuit paths to minimize induction, especially from switching equipment and relays. Second, multiplex systems are designed so that crosstalk and peak overloads are minimized. Third, the amplifiers, filters, repeaters, equalizers, and other equipment on the line are designed to lessen the noise effects sufficiently for voice. For data transmission, further equalization can be employed in the form of conditioning. Fourth, the choice of modulation method has an effect on the accuracy of transmission.

CROSSTALK

"Crosstalk" refers to one channel picking up some of the signal that is traveling on another channel. Occasionally you hear faint fragments of somebody else's conversation on the telephone. 1) It occurs between cable pairs carrying separate signals. 2) It occurs in multiplex links in which several channels are transmitted over the same facility. 3) It occurs in microwave links where another antenna picks up a small reflected portion of the signal for another antenna on the same tower. In case No. 2) and 3), the level of crosstalk noise is very small, because the system is designed with strict criteria for the maximum allowable crosstalk.

Often the strongest source of crosstalk is induction between separate wire circuits. Any long local telephone circuits running parallel to each other will have crosstalk coupling unless they are perfectly balanced, which is not the case in practice. Crosstalk between wire circuits will increase with increased length of circuit, increased proximity, increased signal strength, or increased signal frequency.

However, on most systems it is barely greater, and often less, than the level of white noise, and so, like white noise, it does not normally interfere with data and is notannoying during speech. Occasionally, sometimes due to faults in the exchange, crosstalk becomes louder, and it is possible to hear another person's voice on the telephone. There will also be momentary peaks in crosstalk which do interfere with data. With these exceptions, however, the level of crosstalk is known or measurable and can be treated like white noise for the purposes of selecting transmission parameters so that no interference with data will normally occur.

INTERMODULATION NOISE

There are certain undesirable types of data signals which can cause bad crosstalk. On a multiplexed channel, many different signals are amplified together, and very slight departures from linearity in the equipment cause "intermodulation" noise. The signals from two independent channels intermodulate each other to form a product which falls into a separate band of frequencies, just as two sound waves may "beat" to form a sound oscillation of a different frequency. The result of this may fall into a band of frequencies reserved for another signal. Such products arising from large numbers of pairs of channels combine to form low-amplitude babble, which adds to the background noise in other channels. However, if one signal were a single frequency, then when it modulates a voice signal on another channel, this voice might become clearly audible in a third independent channel. One telephone user in this case would hear the conversation of one other. Privacy is important on the telephone, and so the telephone company attempts to restrict the power of any single-frequency signal to a suitably low level.

The guilty single frequency could arise from data transmission in one of two ways. First, repetitive code in a data signal could cause it unless the modem were specifically designed to prevent this. Many data-processing machines send repetitive codes to each as part of their "line-control" procedures, for example to keep machines in synchronization while data are not being transmitted. Second, the modem itself, if not designed to avoid doing so, might transmit a single frequency when not transmitting data. Often one can hear this when dialing a computer. When the connection is established the apparatus at the other end will send a single-frequency "data tone" down the line to tell you that the connection is established.

ECHOES

Echoes on transmission lines are similar to crosstalk in their effects on data transmission. Where there is an change in impedance on the transmission line, a signal will be reflected so that it travels back down the line at reduced amplitude, thus forming an echo. The signal-to-echo power ratio can occasionally become less than 15 decibels, though it rarely falls below 10 decibels. It can, however, be greater than white noise or crosstalk.

Echo suppressors are used on long lines. They are not generally of value, however, in the problem of echoes in data transmission. Their action is triggered by the detection of human voice signals, and they are normally disabled when the circuit is used for data. In voice telephony, echoes of the speaker's voice become annoying when he hears them with a time delay measured in tens of milliseconds. In data transmission delays of a fraction of a millisecond are significant. Multiple reflections down a two-wire path, or echoes formed at the junction of a two-wire and four-wire circuit, reach the receiving machine or modem. If they are of sufficient volume, they can cause errors in data.

SUDDEN CHANGES IN AMPLITUDE

Sometimes also the amplitude of the signal changes suddenly. This may be due to faults in amplifiers, unclean contacts with variable resistance, added load or new circuits switched in, maintenance work in progress, or the switching to a different transmission path.

These sudden changes can have an effect on certain data transmission systems which could result in the loss or addition of a bit. The effect they have depends on the type of modem in use.

LINE OUTAGES

Occasionally a communication circuit fails to be operational for a brief period of time. These "outages" may be caused by faulty exchange equipment, storms, temporary loss of carrier on a multiplex system, or other reasons, giving a brief period of open or short circuit. Often maintenance work on the lines, repeaters, and exchanges is the cause of brief interruptions. This gives rise to two concerns. First, the data may be damaged by signal losses of a few milliseconds. Second, signal losses of 10 seconds or more may cause a serious break in system availability. Line outages are of serious concern to the designers of some types of computer systems.

RADIO FADING

Radio links are subject to fading. Many long-distance telephone circuits travel over microwave paths, and fading sometimes occurs. Small fades are compensated for by the radio automatic gain control. Large fades, however, may cause a serious degradation of the signal-to-noise ratio.

CHANGES IN PHASE

The phase of the transmitted signal sometimes changes. Some impulse noise causes both attenuation and phase transients. Transients which affect only the phase are also common, especially on long lines. Fig. 9.3 shows a brief change in phase. Sometimes the phase slips and return as in Fig. 9.3 and sometimes it slips without returning. Typically the phase change occurs in less than 1 millisecond,but sometimes a gradual rotation of phase occurs.

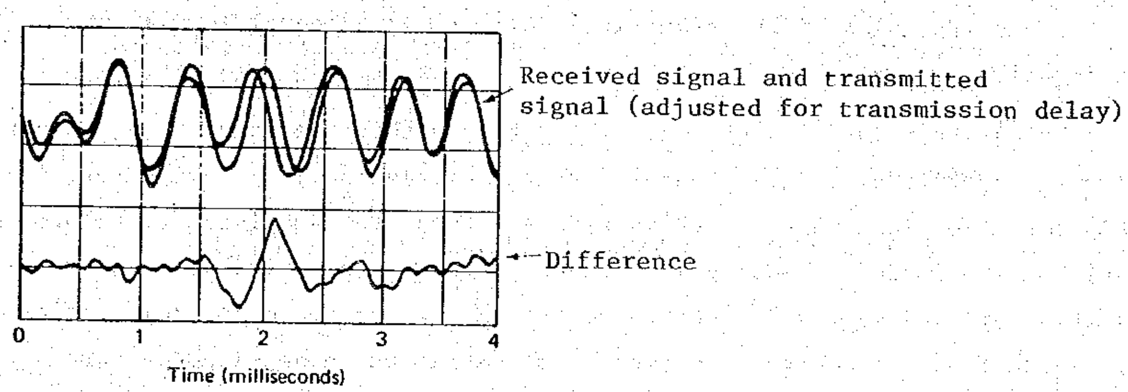


Fig. 9.3 A typical transient change in phase

SYSTEMATIC DISTORTION

The types of noise and distortion discussed above were all "fortuitous". Those below are "systematic" and so can be compensated for partially or completely in the design of the electronics.

In general, as the speed at which data are to be transmitted increases, so the need for uniformity in the transmission characteristics becomes greater.

On private leased lines, steps can be taken to ensure that this uniformity measures upto certain standards. When dialing for a connection, however, it is not certain which path the call will take, and there are likely to be certain connections in the network on which the distortion will be high.

The proportion of faulty or poor-quality sections in a public network differs from one area to another. Some networks still have a high proportion of old telephone plant in them. Some of the early cables used very heavy loading: this gives bad delay distortion, which is discussed below. In most of the major industrial countries this old plant is being replaced with modern equipment having more suitable characteristics for data transmission, but in most developing countries, the replacement rate is slow.

LOSS

The loss of signal strength on a circuit is typically about 16 decibels. It may vary because of aging equipment, amplifier drift, temperature changes, and other causes. Changes are adjusted for during routine maintenance.

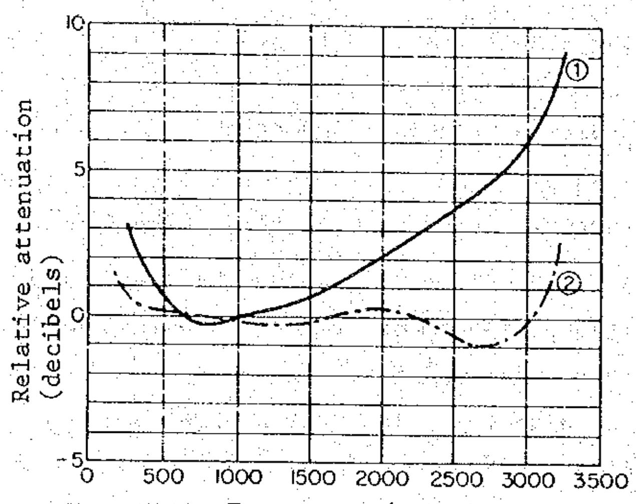
ATTENUATION DISTORTION

The attenuation of the transmitted signal is not equal for all frequencies, as ideally one would like it to be.

The attenuation of a typical cable pair, within the voice band, is approximately proportional to the square root of the frequency. To compensate for this variation in amplitude and to reduce attenuation as discussed before, the cable may be loaded by adding inductance at intervals. Similarly, on multiplex systems carrying many voice channels, filters are designed to yield a flat amplitude-frequency curve. However, some variation in amplitude across the band remains, and this is referred to as amplitude-frequency distortion. The effect of this is to distort the receivable signal slightly.

The solid line in Fig. 9.4 shows the attenuation-frequency curve of a typical telephone line. Attenuation relative to that a 1,000 Hz is shown. It is common that, as in this case, the attenuation level does not vary more than 10 decibels between 400 and 3,000 Hz. This line would be satisfactory for transmission at 1,200 bps. If, however, it were desirable to transmit data on the line at speeds of, say 4,800 bps, or greater, a further flattening of the attenuation-frequency curve would probably be needed, and to do this an equalizer might be used at each end of the line. The equalizer would give a somewhat greater overall attenuation but a flatter frequency-attenuation curve. The dotted line in Fig. 9.4 shows the effect of the equalizer. Here again attenuation is shown relative to that at 1,000 Hz.

Much worse results than these, however, are sometimes obtained on old telephone plant or equipment that is not functioning correctly. In some areas this is encountered on the switched public network.



① Is a typical attenuation - frequency curve for a leased telephone line (without equalization) with today's modems this line would typically be used for transmission at 1200 or 2400 bits per second

② Is for the same line with equalizers. This line could now be used for transmission at speeds up to 9600 bits per second.

Fig. 9.4 Variation of attenuation with frequency on telephone lines

HARMONIC DISTORTION

Harmonic distortion refers to distortion in which the signal attenuation varies with amplitude. For example a 1-volt signal may be attenuated by one half while a 5-volt signal is attenuated by two thirds. If a sine wave is transmitted on such a channel, it is flattened at the peaks. Pulse shapes are therefore not reproduced faithfully at the receiver, if harmonic distortion is considerable the demodulation operation is effected.

The flattening of the sine wave is equivalent to adding harmonics of low amplitude to the transmitted signal. If a sine wave of frequency f is transmitted, harmonic distortion would be equivalent to also transmitting a low amplitude sine wave of frequency 2f and a lower amplitude sine wave of frequency 3f. Hence it is called harmonic distortion, and is measured in terms of the relative power of these 2f and 3f harmonics. Thus a signal with harmonic distortion might have a signal-to-second-harmonic ratio of 25 decibels and a signal-to-third-harmonic ratio of 35 decibels. Keeping these ratios low is important for high-speed data transmission.

DELAY DISTORTION

The phase of the signal likewise is not transmitted linearly. The signal is delayed more at some frequencies that at others. This is referred to as phase-frequency distortion or delay distortion. Some frequencies reach the receiver ahead of others. Figure 9.5 illustrated how signals on wire pairs are propagated with different speeds at different frequencies.

It is a serious form of distortion in data transmission. It has only been corrected to a limited extent on the voice channels that we may wish to send data over, because the human understanding of speech is not greating affected by it. The ear is a relatively slow-acting organism. It is normally necessary for a sound to exist for 0.2 second delay in order to be recognized. If we have delay distortion of 0.05 second, the speech is still intelligible.

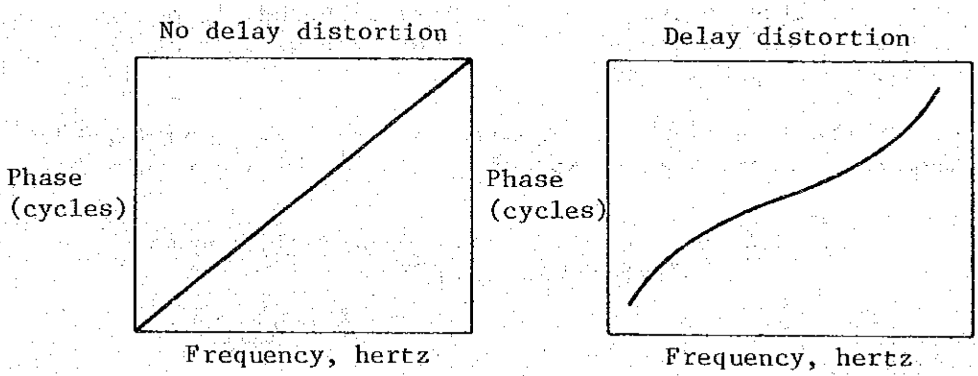


Fig. 9.5 If there were no delay distortion, the curve of phase of the received signal plotted against frequency would be a straight line. In reality it is a curve like that in the right-hand diagram. "Envelope delay" is defined as the slope of this curve and may be measured in micro-seconds.

In effect, envelope delay at a given frequency is the delay that would be suffered by a very narrow bandwidth signal transmitted at that frequency. Because there is delay distortion, it varies with frequency, as shown in Fig. 9.6. It is sometimes simply referred to as the delay at a given frequency.

Figure 9.6 shows the effect of delay distortion on the transmission of a square-edged pulse. A square-edged pulse train is, in effect, composed of many frequencies, and so the edges of the pulses begin to distort as the wave travels to its destination. Again, if parallel transmission were used on the line, in such a way that 1 bit of a character were transmitted at 800 Hz, the next at 1,000, and so on, then, because of delay distortion, some of these bits would reach the receiver before others. The main effect of delay distortion, however, is on the more elaborate forms of modulation. Delay distortion must be kept below a certain level in order that the fastest and most efficient types of modem work correctly.

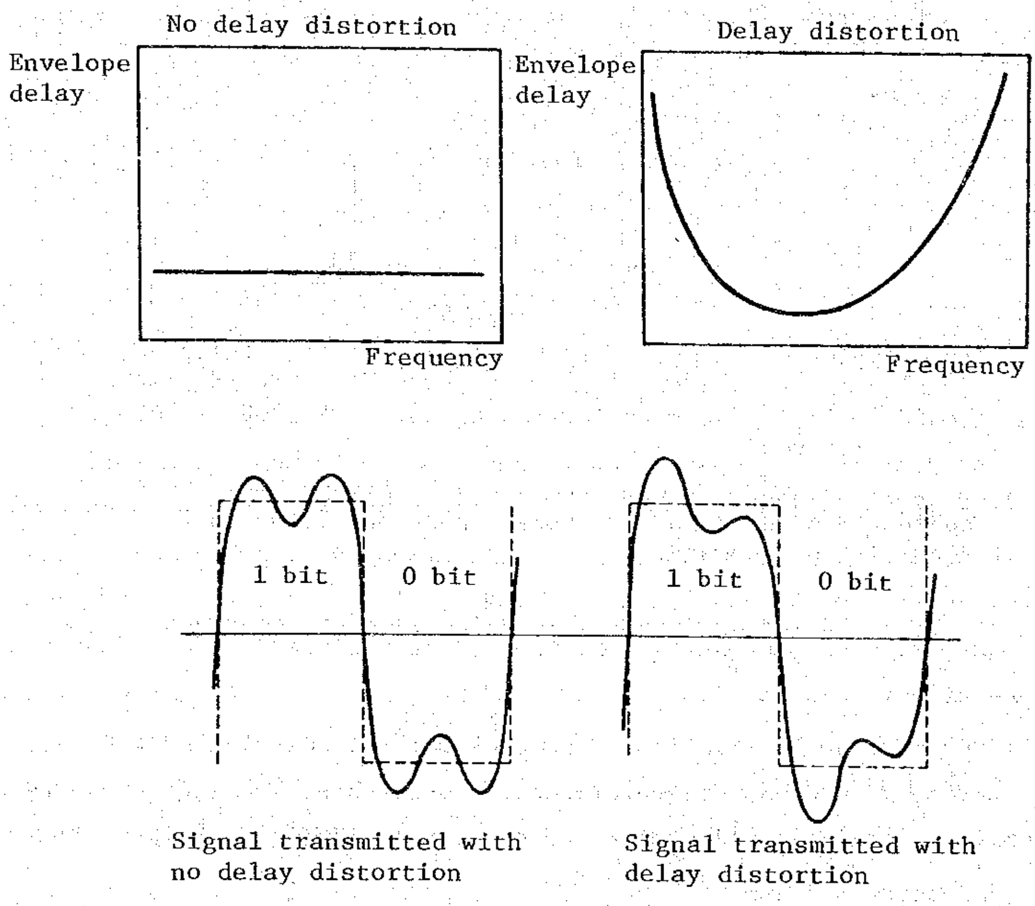
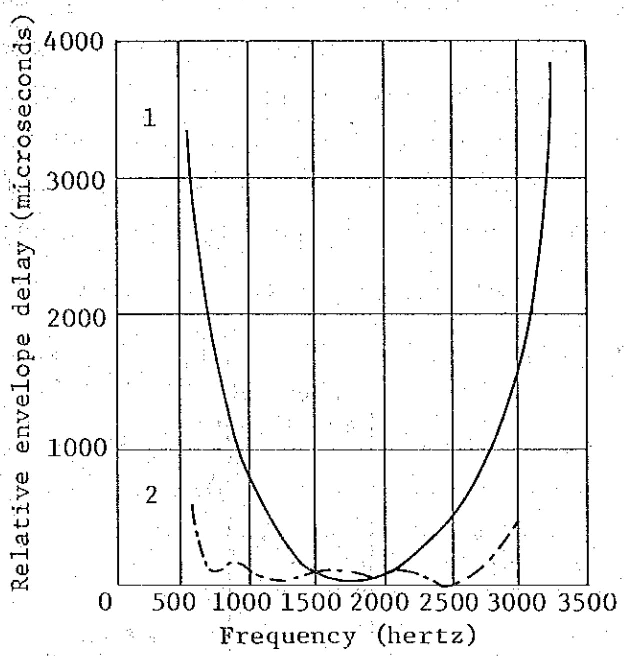


Fig. 9.6 The effect of delay distortion on the transmission of square-edged pulses over a line of limited bandwidth

Figure 9.7 shows the delay distortion on a typical telephone line. The solid curve shows the delay of different frequencies in microseconds, relative to that at about 1,800 Hz (which arrives first in this case). Just as equalizers are used to compensate for attenuation distortion, so a form of equalizer can be employed to compensate for delay distortion. The signal at 1,800 Hz is slowed down to the speed of those at the outer frequencies. A typical result of this compensation is shown by the dotted line in Fig. 9.7.



① is a typical envelope delay curve for a leased telephone line (without equalization) with today's modems this line would typically be used for transmission at 1200 or 2400 bits per second.

② Is for the same one with phase equalization. This line could now be used for transmission at speeds up to 9600 bits per second.

Fig. 9.7 Variation in envelope delay with frequency on telephone lines

If a leased telephone line is to be used for transmission at 4,800 or higher bps, it may be equalized beforehand. To do this, the line has attached to it circuits containing inductance and capacitance adjusted to give similar attenuation and delay at all frequencies.

STANDARD PROFILES FOR LINE DISTORTION

CCITT makes recommendations M1020, that certain types of lines should have delay and attenuation distortion characteristics lying within given profiles. To a large extent national telecommunication organizations accept these recommendations and attempt to engineer their lines to the specifications laid down. In addition, national carriers lay down distortion specifications for certain tariffs. Where a voice line is to be used for speeds higher than 2,400 bps, this has been done, because a low bit error rate would be achieved only if the line did in fact meet its obligations for low-distortion operation.

FREQUENCY OFFSET

Signals transmitted over some channels suffer a frequency change. That is, if 1,000 Hz per second are sent 999 or 1,001 Hz might be received. This is sometimes caused by the use of multiplex systems for carrying each voice band at a different frequency. The oscillators, used for generating the carrier supplies for modulation and demodulation are not precisely at the same frequency. The CCITT recommendation is that frequency offset should be limited to ±2 Hz per link, and most circuits conform to that. A circuit with five links in tandem might then have a frequency shift up to ±10 Hz. Under faulty conditions, this value may be exceeded for short periods, but rarely exceeded by enough to interfere with data transmission.

BIAS AND CHARACTERISTIC DISTORTION

Repeaters are sometimes used to reconstruct pulses, producing a new, clean, square-edged pulse out of what by that time had become a distorted pulse. Similarly, the output of modulation systems are "sliced" to give square-edged pulses. If systematic distortion occurs, this will result in all the pulses being lengthened or shortened. This is referred to as "bias distortion". If all the 1-bit pulses are lengthened, it may be called "positive bias," or "marking bias," and if they are shortened, it may be called "negative bias," or "spacing bias,". Bias distortion changes sign when the one and zero bits are interchanginged. Bias distortion may be caused by a decision threshold for the pulse regeneration being set at the wrong value and thus can usually be adjusted.

A similar type of systematic distortion is called "characteristic" distortion. Here the effect is not reversed when the 1 and the 0 are interchanged.

For example, a single 1 or 0 may be shortened in transmission, whereas a long mark or space representing adjacent 1s or 0s may lengthen. This may be caused by some nonlinear characteristic of the transmission, possibly caused by bandwidth restriction.

CONDITIONING

Many of the impairments which effect circuits can be controlled so that they do not exceed certain limits. Table 9.1 indicates which of the impairments are controlled. Some are controlled by telephone company internal practices, which may limit the impairments to levels in the CCITT recommendations. An additional degree of control is sometimes applied to leased voice-grade circuits to improve their properties for data transmission. This, mentioned earlier, is referred to as conditioning. A monthly charge is made for conditioning in addition to the cost of the circuit.

Modern high-speed modems, for example the Paradyne M-96 automatically equalize the circuit they operate on, effectively doing the same as a conditioning. With such modems a conditioning does not help performance. Less expensive modems can benefit from a conditioning.

The effects of attenuation distortion and delay distortion are thus reversible. The effects of noise and harmonic distortion cannot be reversed by the modem. Consequently American T&T introduced a second type of conditioning, D-conditioning, which specifies limits to the permissible noise-to-signal ratio and harmonic distortion. D conditioning is intended for use with modems which transmit 9,600 bits per second over voice-grade lines.

D-conditioned lines will meet the following specifications:

Signal to c-notched noise: 28 dB.

(C-notched noise is a standard measurement relating to the energy levels of a typical telephone signal).

Signal to second-harmonic distortion: 35 dB.
Signal to third-harmonic distortion: 40 dB.

D1 conditioning relates to point-to-point channels.
D2 conditioning relates to 2- or 3-point channels.

To achieve D conditioning, the routing of the leased circuits will be chosen so as to avoid noisy or old-fashioned equipment, and to avoid step-by-step switching offices. In PAKISTAN T&T it would be impossible to do this because there is much old equipment and most of the switching offices use stronger switches (EMD).

Table 9.1 Circuit impairments which can be controlled either by *conditioning* or by telephone company practices

ATTENUATION DISTORTION	BY "C" CONDITIONING	CONTROLLED BY TELEPHONE COMPANY INTERNAL PRACTICES
DELAY DISTORTION		
SIGNAL NOISE RATIO	BY "C" CONDITIONING	
HARMONIC DISTORTION		
IMPULSE NOISE		
FREQUENCY OFFSET		
ECHOES		
PHASE JITTER		
RADIO FADING		NOT CONTROLLED
CHANGES IN PHASE		
ATMOSHPHERIC GAIN HITS		
DROPOUTS		

SUMMARY

Data transmission is afflicted by both static impairments of the circuits (systematic distortion) and transient phenomena such as impulse noise and brief variations in phase (fortuitous distortion). A modem well designed for a given line with appropriate conditioning if specified can withstand substantial amounts of the static impairments such as attenuation and delay distortion without producing errors.

Chapter 10 DATA ERRORS

Because of the noise on communication lines especially impulse noise, there will be a number of errors in data transmitted. We cannot prevent all the errors occurring; all we can hope to do is to detect them and somehow correct them. In this chapter we will summarize the quantity and nature of errors that can be expected.

Many measurements of transmission errors have been made on communication lines. Telecommunication companies throughout the world have statistics about the error rates and patterns that can be expected on different types of lines.

The data communications systems designer or analyst needs to have some knowledge of the error rates and patterns his system will encounter. On some systems this has been obtained by taking measurements on the network that will be used. Often, however it is not practical for the user to test the lines he wants to use. Systems analysts constantly complain that they do not know what error rates to expect and cannot obtain any figures for this.

1. TELEGRAPH AND TELEX CIRCUITS

ERROR RATES

The CCITT Study Group on Data Transmission has made and analyzed tests on the world's 50-baud leased telegraph channels and on the international telex circuits of many countries. The results are as follows:

Most Probable Error Rate

1. In point-to-point service:

   On elements: one to two-bit errors in 100,000 transmitted.

   On characters: one-to eight-character errors in 100,000 transmitted.

2. On switched telex circuits:

   On elements: one-to two-bit errors in 100,000 transmitted.

   On characters: four-to five-character errors in 100,000 transmitted.

BURSTS

A considerable proportion of the errors were found to by different authorities. Here a burst of errors was defined as elements in error separated by less than 10 non-erroneous elements. The proportions of such bursts were found to be as follows:

isolated errors on elements: 50 - 60%

bursts with two errors: 10 - 20%
bursts with three errors: 3 - 10%
bursts with four errors: 2 - 6%

These figures are important for predicting the effectiveness of different error-detecting codes.

Time periods in which the "start" condition remains on the line for more than 300 milliseconds are referred to as "dropouts", the line being regarded as temporarily out of service. These are not included in the preceding figures.

The CCITT recommends that the above performance be taken as that of standard 50-baud telegraph and telex circuits.

BLOCK ERROR RATE

Where data are to be sent in blocks, it is desirable to know how the error rate varies with the block length so that an estimate can be made of the number of blocks that have to be retransmitted and so that the systems can select an optimum block length. The bottom chart of Fig. 10.1 plots the probability of a block error against the block lengths.

2. SUB-VOICE-GRADE CIRCUITS

200-BAUD CIRCUITS

The character error rate over the world's 200-baud circuits, is generally somewhat better than over telex circuits. Here the voice channel is typically divided into 480-Hz spacing (360 Hz in some countries), as opposed to 120 Hz for 50-baud transmission. The slightly better performance is attributed to the fact that the error-free intervals are about the same and that more characters are transmitted during these periods.

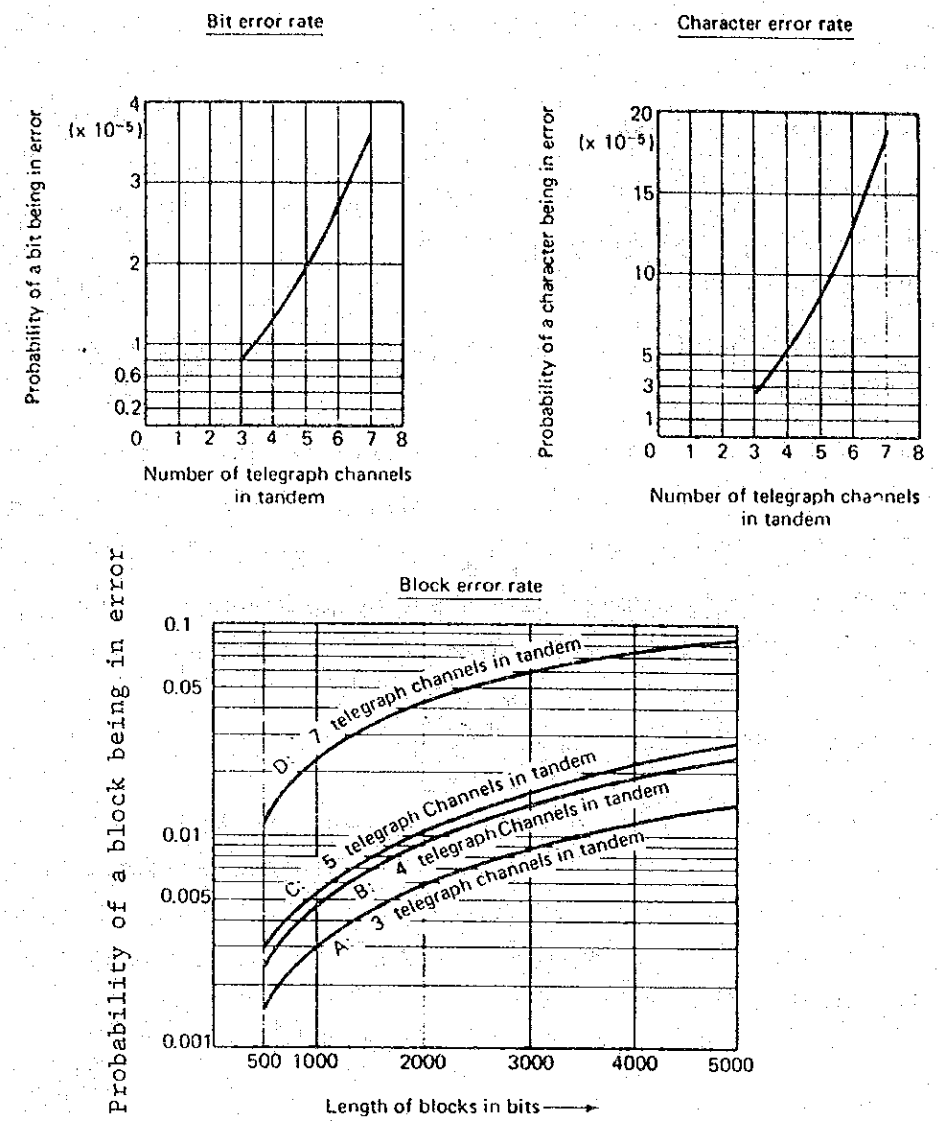


Fig. 10.1 Mean bit, character, and block error rates on Telex circuits - from a survey of Telex connections in Germany

3. HIGHER SPEED USE OF TELEPHONE CIRCUITS

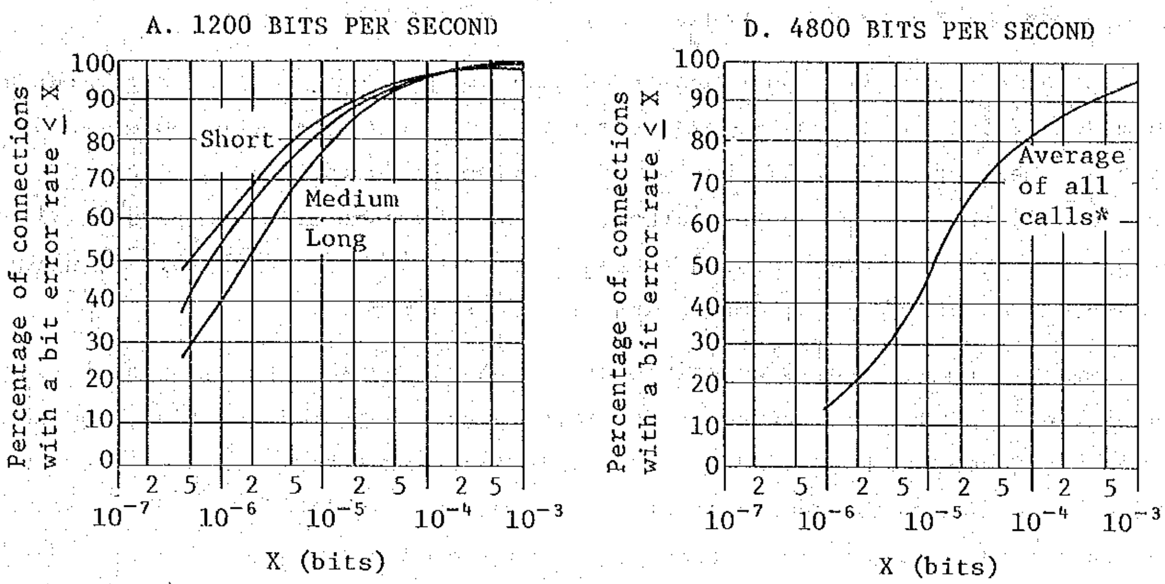
When data are transmitted over telephone circuits, the error rate tends to be greater at higher transmission speeds. The error rate tends to increase rapidly as transmisssion speeds are raised above 4,800 bps, and at such speeds is so variable that reliable statistics are not published. Below 4,800 bps a voice-grade line with a good modem typically encounters about 1 bit error in 100,000 bits transmitted. Lines with error rates worse than 1 bit in 100,000 usually have an audible level of noise which can be a nuisance in the reception of speech. The mean error rate is worse than this typical error rate because a small proportion of lines behave badly.

At speeds of 9,600 bps transmitted over voice lines, typical error rates on modems which do not use automatic error correction have been quoted to be as high as 1 bit error in 2,000 but much variation is found from one line to another and from one modem to another. The performance at high speeds will probably improve as modem design improves and can be made acceptable by using a combination of error correction and error detection.

Again, errors tend to come in bursts. When trouble comes, it comes fast and furious. Bursts of error are the rule rather than the exception, and they sometimes go on for hundreds of bits. This clustering has both a good effect and a bad effect. The number of error-free messages which do not have to be retransmitted is much higher than if the same number of errors occurred at random. On the other hand, the clustering makes it much more difficult to devise safe error-detecting or error-correcting codes.

Figure 10.2 shows the percentage of calls have different eror rates. The calls are again categorized as short-distance (0 to 180 miles), medium-distance (180 to 725 miles) and long distance (greater than 2,900 miles), with 1,200 bps transmission, about 80% of all calls bad error rates better than 1 error bit in 100,000. Half of the short calls had error rates better than 1 bit in 5 million. However, more than 1% of the calls had error rates worse than 1 bit in 1,000. The error rates become higher as speed is increased. At 4,800 bps about half of the calls have fewer than 1 error bit in 100,000, but about 5% of the calls are worse than 1 error bit in 1,000.

Many of the errors that occur in practice are generated on local loops and on subscriber premises. A few local loops and associated equipment such as PBXs are exceptionally noisy. Sometimes badly routed wiring on a subscriber's premises consistently generates noise.



* Not broken down into short, medium, and long distance calls because of inadequate sample size.

Fig. 10.2 Error rates on Bell System voice-grade trunks at different transmission speeds using AT&T modems

BURSTS OF ERRORS

As we found with lower speed transmission the errors are highly clustered. There tend to be lenghy trouble free intervals interspersed by bursts of bit errors. Consequently, the numbers of blocks or messages that contain an error cannot be deduced from mean bit error rates for a connection. Figure 10.3 shows the block error rates for different block sizes. These curves may be used by a systems analyst in determining what would be an economical block size to use - avoiding too much retransmission.

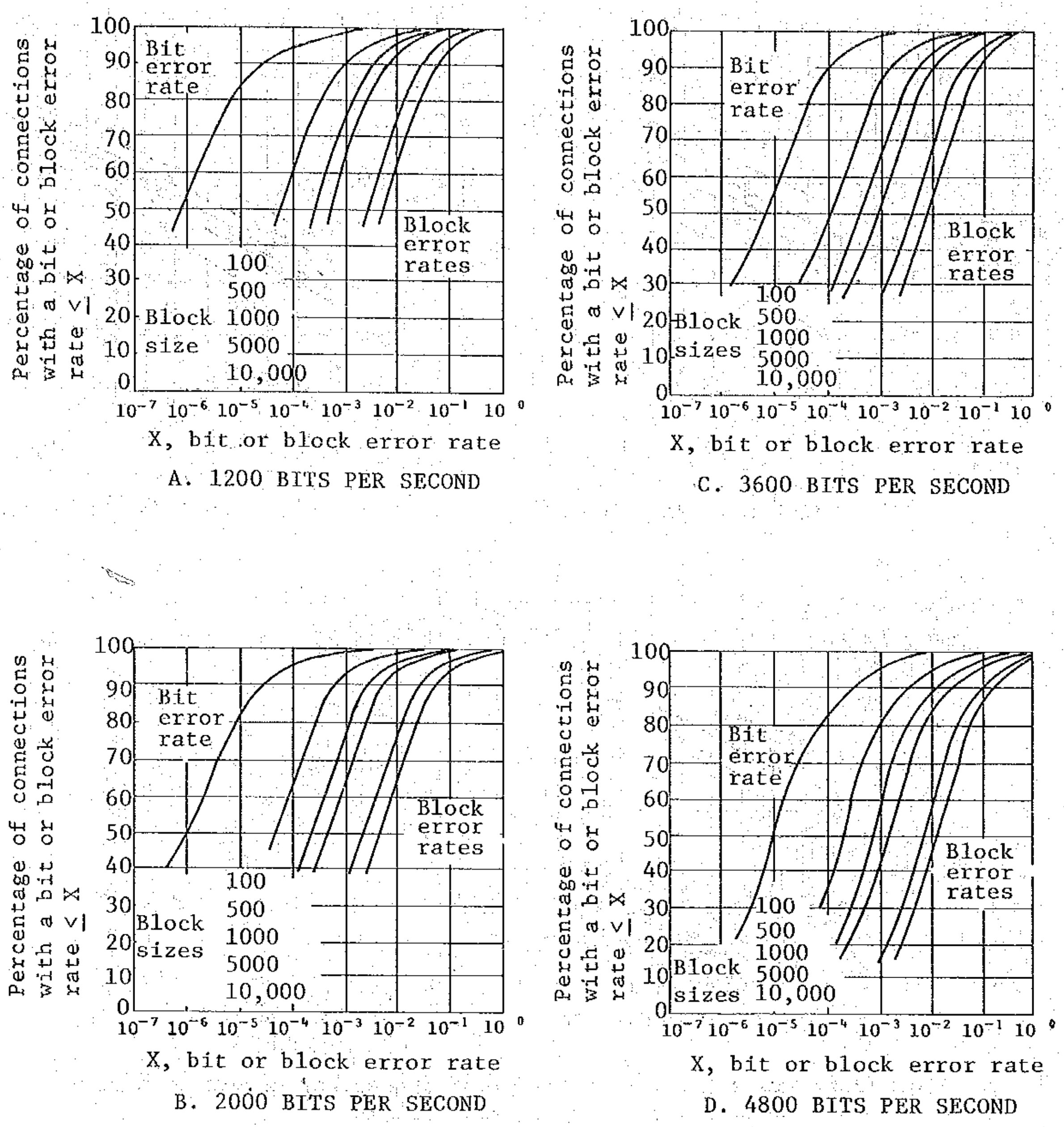


Fig. 10.3 Curves showing how block error rates vary for voice-grade trunks on the Bell System

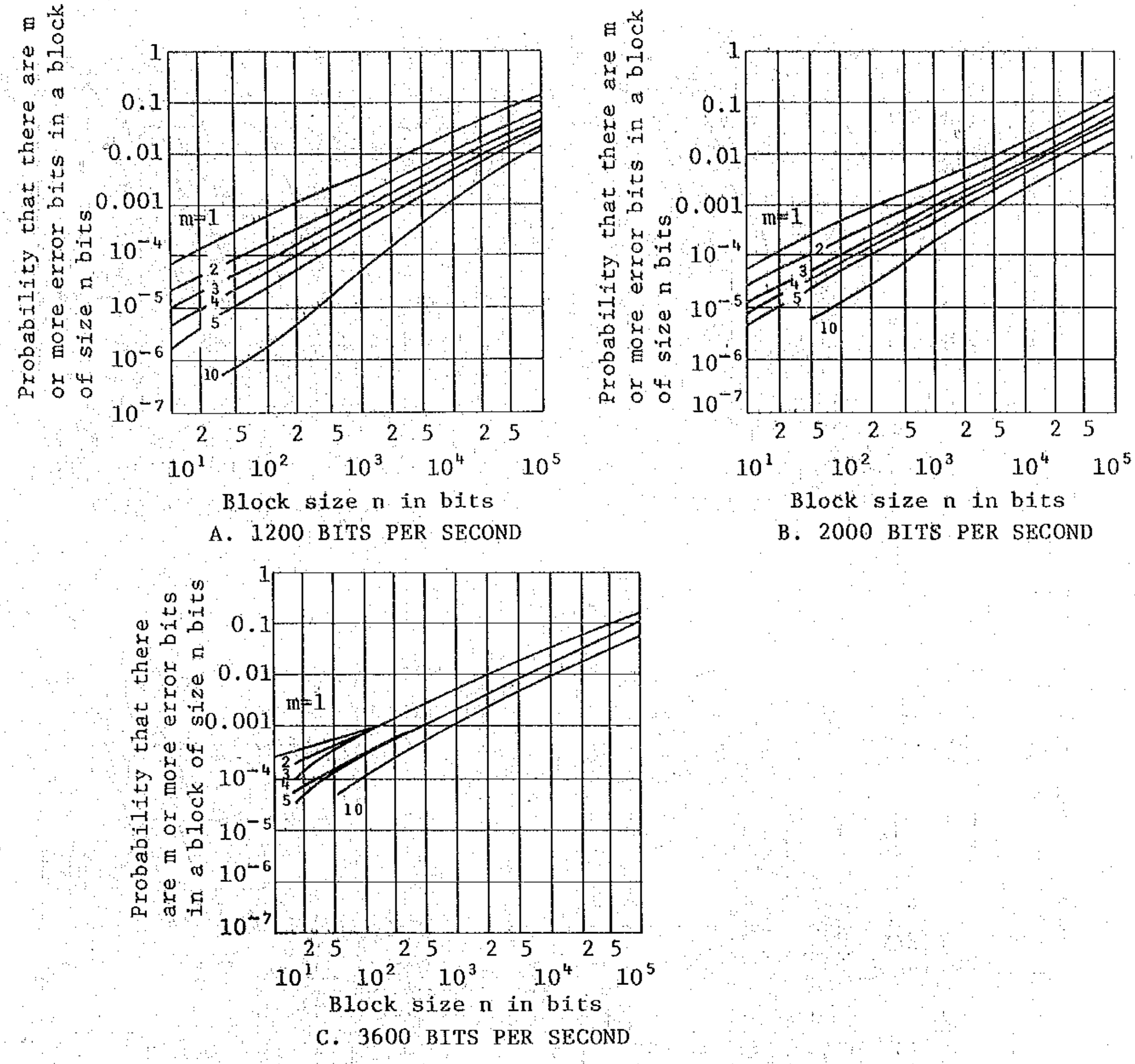


Fig. 10.4 Probabilities of different numbers of errors in a block on Bell System voice-grade trunks

Figure 10.4 shows the probabilities of having different numbers of errors in a block of a given length. The top curve, the probability that the block contains one or more errors, is approximately of the form.

$$P \text{ error} = An^{B}$$

where n is the number of bits in the block and A and B are constants.

The approximate equations for the three transmission speeds are

$$1{,}200 \text{ bps:} \quad P \text{ error} = 1.2n^{0.81} \times 10^{-5}$$

2,000 bps: $P\ error = 1.0n^{0.80} \times 10^{-5}$

3,600 bps: $P\ error = 2.5n^{0.77} \times 10^{-5}$

CARRIER FAILURES

The error statistics do not take carrier failures into consideration. Many quoted data error statistics do not indicate the carrier failures or dropouts.

Each of the modems used was equipped with a carrier detector which indicates an ON condition during normal reception of signals. An OFF indication conditions typically last from about 10 to 300 milliseconds. They cause many messages to have to be retransmitted but are not included in the error statistics.

If the carrier failures were uniformly distributed, the probability of a message being faulty because of a carrier failure, P failure, would be

1,200 bps: $P\ failure = 0.012n \times 10^{-5}$

2,000 bps: $P\ failure = 0.0084n \times 10^{-5}$

3,600 bps: $P\ failure = 0.0028n \times 10^{-5}$

Carrier failures are much less subject to clustering than bit errors. They are in fact, related to the prolonged periods of disturbance that cause bit error clustering. As the figures in the above equations are small, these probabilities of carrier failures may be added to the error probabilities to give a probability of faulty messages, P failure:

1,200 bps: $P\ fault = (1.2n^{0.81} + 0.012n) \times 10^{-5}$

2,000 bps: $P\ fault = (1.0n^{0.80} + 0.0084n) \times 10^{-5}$

3,600 bps: $P\ fault = (2.5n^{0.77} + 0.0028n) \times 10^{-5}$

It is suggested that a systems analyst use these equations (in absence of anything better) to evaluate probabilities of retransmission and hence to determine optimum block sizes.

MULTIPLE ERRORS

A systems analyst is concerned not only with the probability of having to retransmit a message but, also, and perhaps more seriously, with the probability that an error will fail to be detected. Their effectiveness in catching errors depends on the probability of different numbers of errors occurring in the same message.

LINES IN DIFFERENT COUNTRIES

The performance of telephone lines is fairly similar throughout the world. Most telephone administrations claim that an errors rate less than 1 bit in 100,000 is obtained "under normal conditions" over a leased circuit and over most switched circuits. In all countries, a few bad circuits have error rates much worse than his objective. The proportion of bad-quality circuits varies from one area to another. A few areas, usually with obsolete or EMD telephone plant, are exceptionally poor.

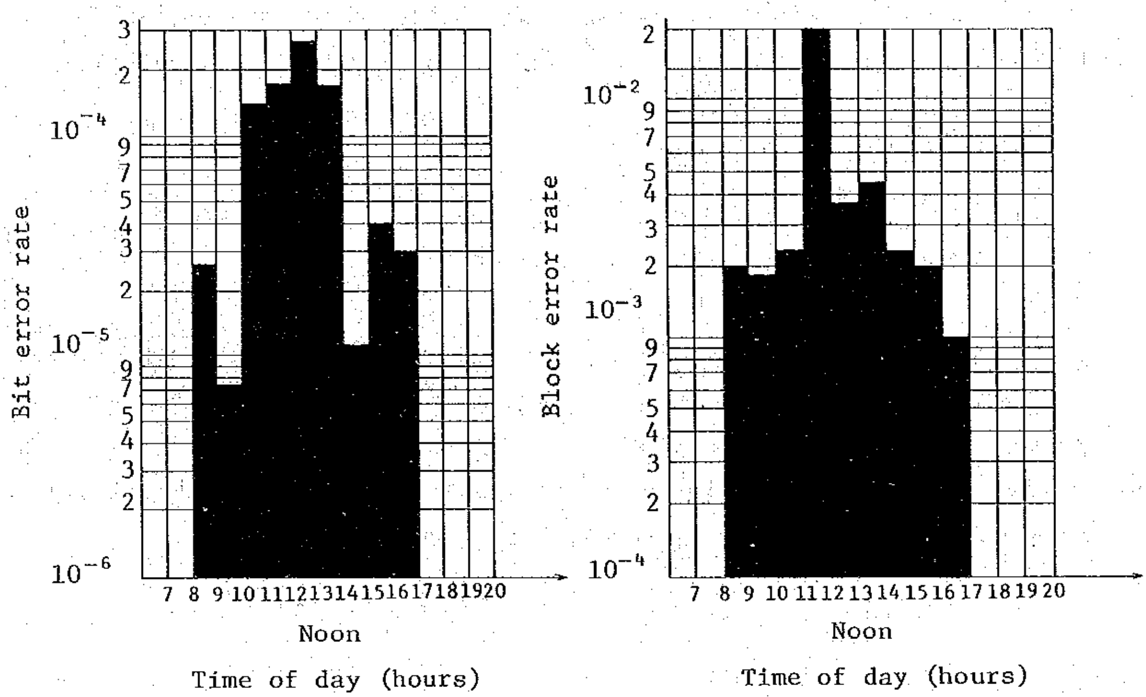


Fig. 10.5 Hour-by-hour variation of error rates encountered. Both illustrations use transmission at 1,000 bits per second over voice-grade lines. ((1) Extract from CCITT contribution COM Sp. A/No. 7, July 1961, by the Chile Telephone Company. (2) Extract from CCITT contribution COM Sp. A/No. 16, September 1961, by N.V. Philips Telecommunicatie Industrie.)

HOUR-BY-HOUR VARIATION IN ERROR RATES

When transmitting data over the switched public telephone network considerable variation in the error rate is found from one period of time to another. A typical illustration of his is shown in Fig. 10.5. This shows the result of transmitting block of data over the local public network of Stuttgart, seen that occasionally there are periods when the error rate is several times higher than the average. The rror rate is generally higher during periods of high traffic intensity. Measurements on other telephone networks show similar results. The error rates are highest when the circuits are most heavily loaded.

While traffic peaks cause a change in error rate which is somewhat predictable, maintenance work on the lines or in the exchanges can cause sudden and unpredictable peaks. These frequently cause a greater error rate for a period than the normal daily variations.

WIDE BAND DATA CHANNEL

The error performance of the lines has generally been good and only a small proportion of the packets transmitted have to be resent because of errors. Fig. 10.6 shows the result of seven days continuous measurement of the network at a time a when 86 wideband lines were in use. The measurements counted the numbers of packets that had to be retransmitted because of errors. The packets during the period in question had an average length of 218 bits. The average error rate was one faulty packet in 12,880 (12,880 packets) contained 2.8 million bits). Fourteen of the lines had no detected errors during the seven days. On the other hand 6 lines had error rates worse than 1 faulty packet in 1,000. The worst line had 1 faulty packet in 340.

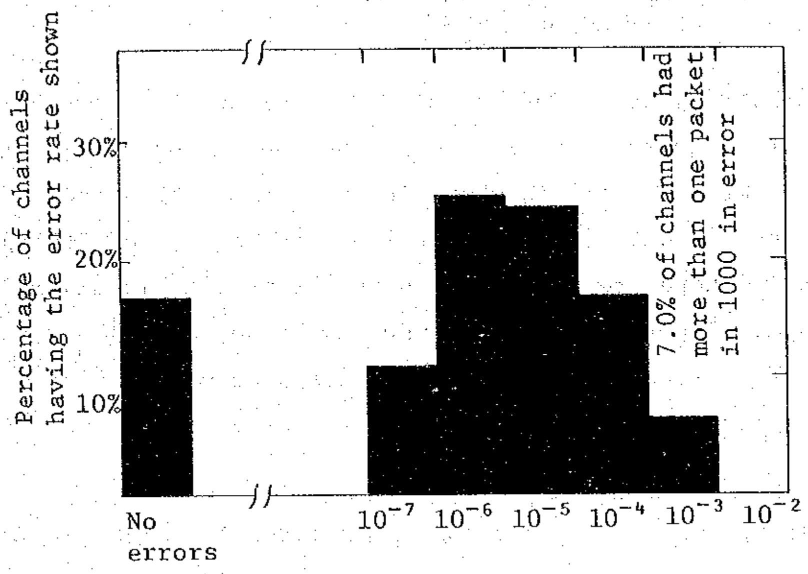


Fig. 10.6 Error rates on 50 kbps lines used on ARPANET measured during 7 days of continuous operation (average packet length: 218 bits).

Chapter 11 DATA TRANSMISSION TEST

To find out the characteristics of the telephone network of the country, relevant for data transmission circuits, a series of measurements have to be carried out. For the carrier who provide the Data transmission circuit to the customers, and the customers who want to introduce data communication system, want to know the quality of the circuit to design their own computer system.

This chapter is aimed at describing the detailed measurement method for Analog Data Transmission Test.

11.1 Flows of measurements

Usually the carrier engineers take initiative to carry out the measurement and the procedure of test is as follows:-

1) arrangements of necessary Measuring Equipments,

2) arrangements of vacant circuits including spare circuits.

3) despatch the technical staffs to the site where the test will be carried out, if necessary.

11.2 Type of measurements

The measurement test is classified according to the Transmission data speed.

Table 11.1 Classified test item

Class / Test item	V1-A PIX, FAX	V1-B up to 2400 bps	V2 up to 4800 bps	V3 up to 9600 bps
Insert loss	○	○	○	○
Loss/freq. attenuation	○	○	○	○
Group delay		○	○	○
White noise	○	○	○	○
Impulse noise			○	○
Phase jitter			○	
Frequency error			○	○
Harmonic noise			○	○
Amplitude hit				○
Phase hit				○
Drop out				○

The measurement test should be carried out as follows

1) Attenuation at 1,000 Hz
2) Attenuation distortion
3) Group delay distortion
   * Conditioning   Adjust (if necessary)
4) White noise
5) Impulse noise
6) Phase jitter
7) Frequency errors
8) Harmonic noise
9) Gain hit
10) Phase hit
11) Drop-out
12) Go-to-return cross talk
13) Error bit ratio

11.3 Signal level

The signal level setting is important for the test, to avoid interference to other used circuit and to collect test result.

TLP (Transmission level point) level which is standardized by the transmission media, and Data transmission level is always at -13 dBmo. to avoid interference to the nearest circuits.

```
        TLP
OdBmo    |- - - - - - - - -Normal telephone transmission level
         |
         |       -13dB
         |
-13dBmo  |- - - - - - - - -Data transmission level
```

So Data Transmission (send) level will be fixed with TLP level and attenuation loss.

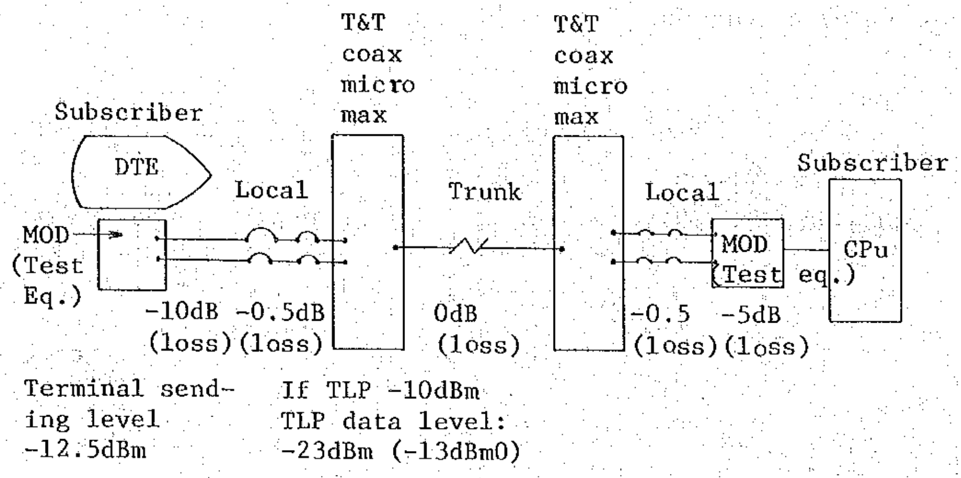


Fig. 11.1 Concept of TLP

For example, when TLP level is -10dBm at the Mux, TLP Data level is -23dBm or (-13dBm0).

So considering the local line attenuation loss (-10dB, -0.5dB), the terminal sending level becomes -12.5dBm.

1) Loss/frequency atttenuation

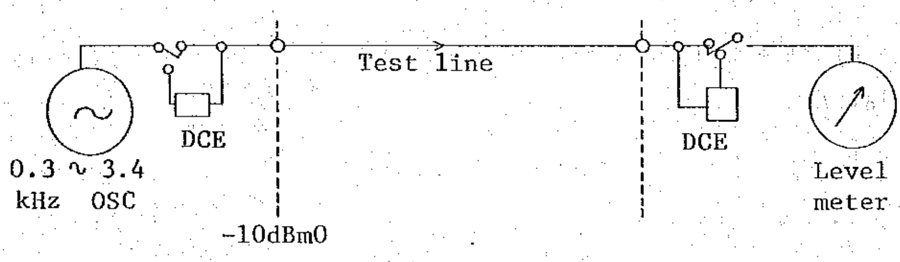


Fig. 11.2 Loss/freq. attenuation test

The sender sends the single frequencies in the band 0.3 to 3.4 kHz in step of 100 Hz, and the receiver plots the level and draws the chart. See Fig. 11.5.

CRITERIA of CCITT. M1020, M1030.

2) Group delay distortion

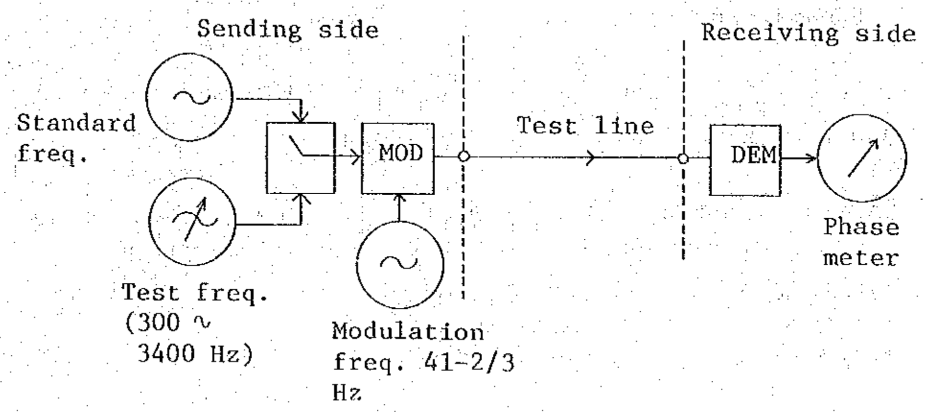


Fig. 11.3 Group delay test (CCITT method)

Note: Sending signal output level is -10dBm0.

Note: Zero is the value which equals the smallest value between 500 Hz to 2,800 Hz in the measurement. See Fig. 11.6.

3) White Noise

We measure Weighted Noise, Flat Noise.

In cancelation, weighted noise is 2.5dB better than the Flat noise. See Fig. 11.7.

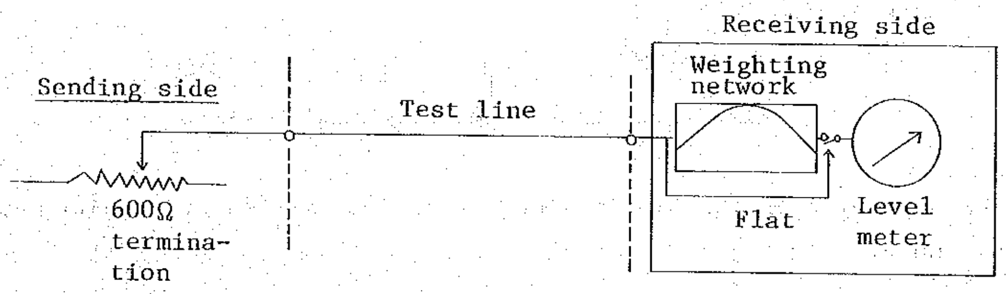
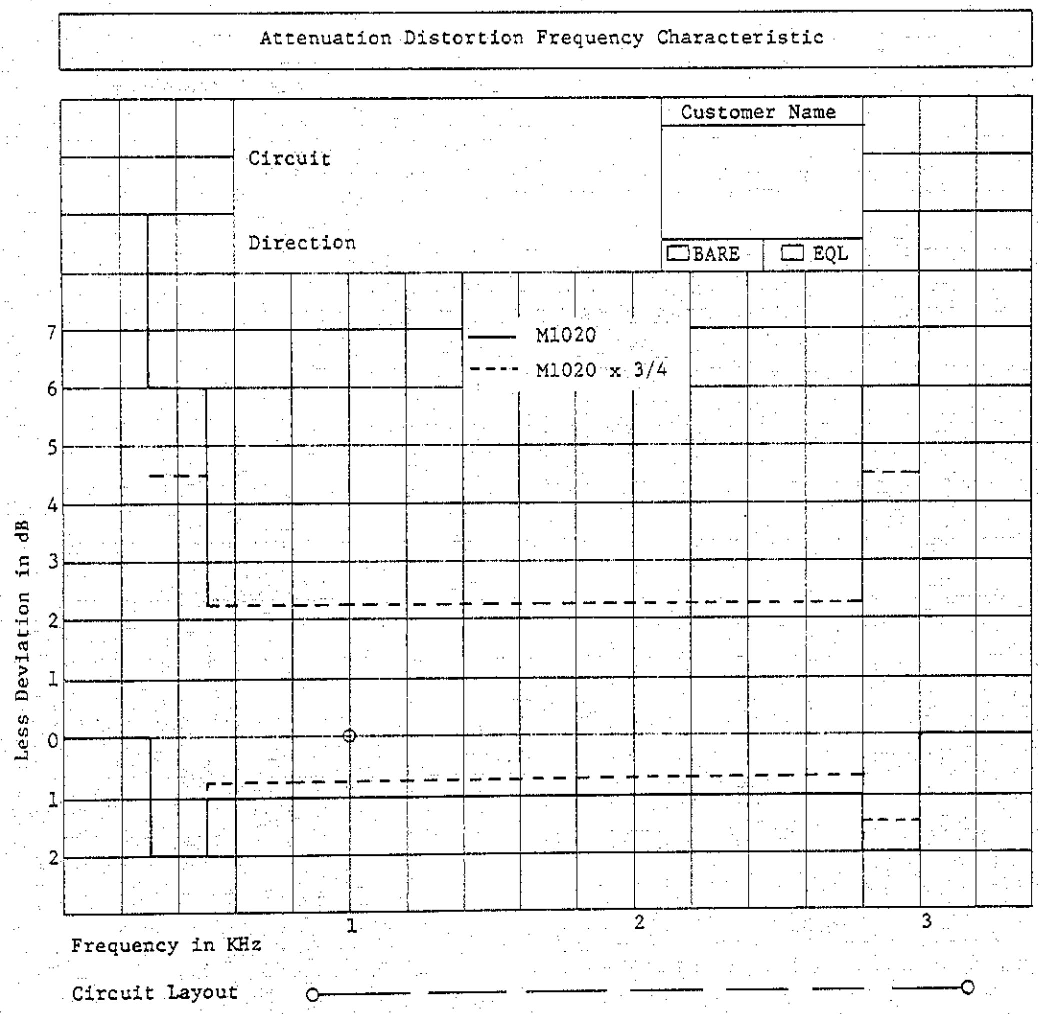
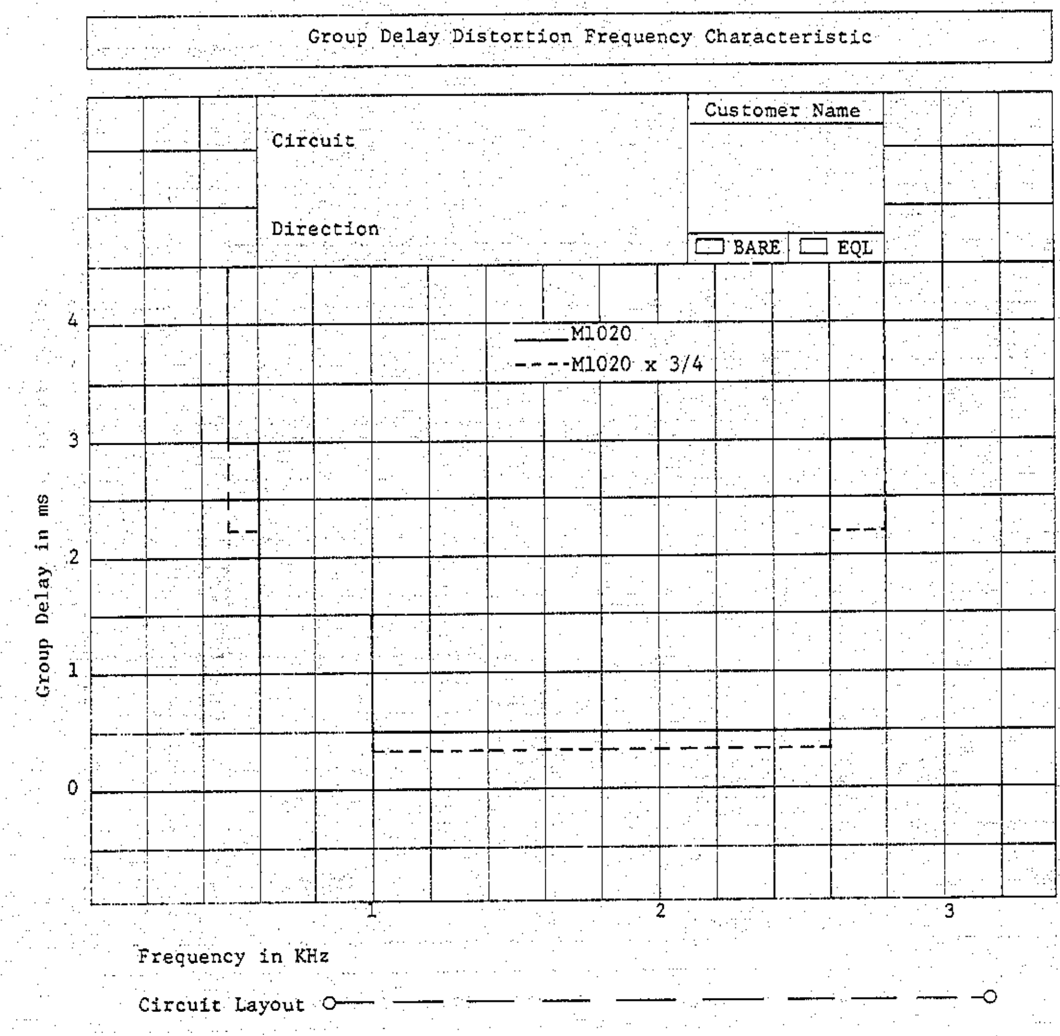


Fig. 11.4 White noise test



Date		19			
Name		N.E.			
		D.E.			
Rcv. 1 KHz		Relative			dBm0
Level: KHz		Absolute			dBm
Frequency Response					
Freq. (KHz)	Lev.D. (dB)	Loss.D. (dB)	Freq. (KHz)	Lev.D. (dB)	Loss.D. (dB)
			1.9		
0.3			2.0		
0.4			2.1		
0.5			2.2		
0.6			2.3		
0.7			2.4		
0.8			2.5		
0.9			2.6		
1.0			2.7		
1.1			2.8		
1.2			2.9		
1.3			3.0		
1.4			3.1		
1.5			3.2		
1.6			3.3		
1.7			3.4		
1.8					
Noise	Flat				dBm0
	Weighted				dBm0p
Harmonic Distortion			2nd dB		3rd dB
Freq. Error					Hz
GoToR TN. Cross Talk					dBm0
Remarks					

Fig. 11.5



Date	19	
Name	N.E.	
	D.E.	

Group Delay Distortion					
Freq. KHz	Group Delay (ms)	(ms)	Freq. KHz	Group Delay (ms)	(ms)
			1.9		
0.3			2.0		
0.4			2.1		
0.5			2.2		
0.6			2.3		
0.7			2.4		
0.8			2.5		
0.9			2.6		
1.0			2.7		
1.1			2.8		
1.2			2.9		
1.3			3.0		
1.4			3.1		
1.5			3.2		
1.6			3.3		
1.7			3.4		
1.8					
Phase Jitter	deg p-p (20∿300Hz)				
	deg p-p (Hz)				
Count Time	J(Min)				
Impulsive Noise	counts(dBm0)				
	counts(dBm0)				
	counts(dBm0)				
Phase Hits	counts(deg)				
Gain Hits	counts(dB)				
Dropouts	counts(dB)				
Remarks					

Fig. 11.6

Recommendable test equipments are as follows

1) HP 4940A TIMS
2) HALCYON 520 B2 UNIVERSAL TEST SYSTEM (Bell Type)
3) HALCYON 521 A UNIVERSAL TEST SYSTEM (CCITT Type)
4) HEKIMIAN 3901L COMMUNICATION TEST SYSTEM
5) HP MODEL 1645 A DATA ERROR ANALYZER
6) BRADLEY MODEL 2A/2B LINE SIMULATOR
7) IDS MODEL 1310 TDM MODEM TEST SET
8) HP TRANSMISSION TEST SET 3552A

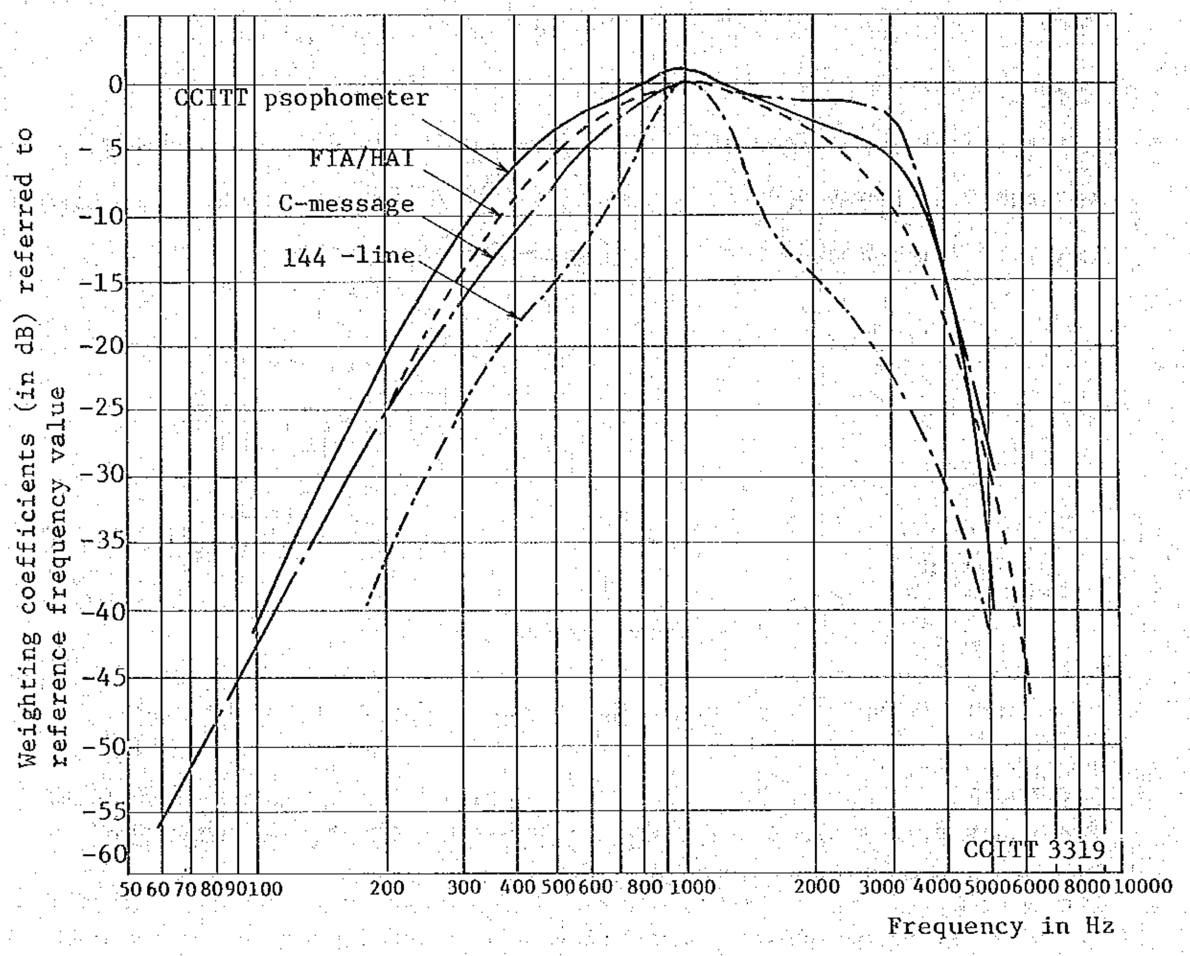


Fig. 11.7 Weighting coefficients

4) Impulse noise

We measure the number of occurrence event of impulse noise exceeding certain threshold level along with CCITT 0.71 Recommendations.

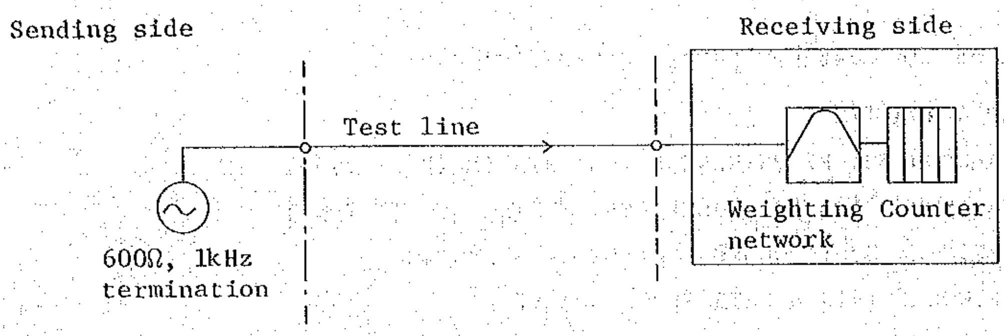


Fig. 11.8 Impulse noise test

Criteria: Occurrence of Impulse Noise should be less than 18 times during 15 minute test period (CCITT 0.71)

Note: Sending data level -13dBm0, 1kHz.

5) <u>Phase Jitter</u>

A Phase Jitter is something like a phase shift, and is created mostly by the noise of power supply for FDM Carrier System.

CCITT, 091 recommends the measuring method as follows:

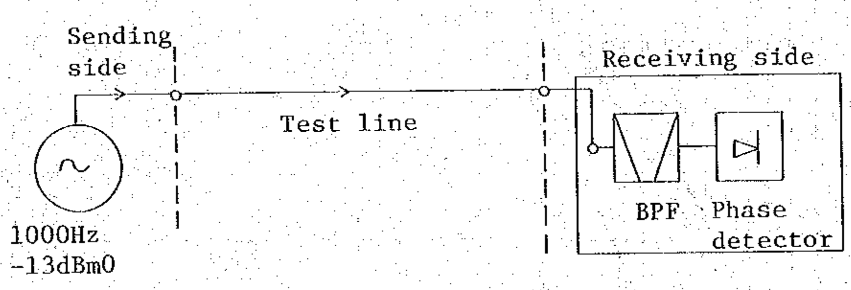


Fig. 11.9 Phase Jitter test

Criteria: Phase Jitter should be within 15 degrees/Peak to peak.

6) <u>Frequency errors</u>

Frequency errors are created due to the adaption of the independent carrier synchronization in FDM Carrier system.

CCITT 0.111 recommends the measuring method and its criterias as follows:

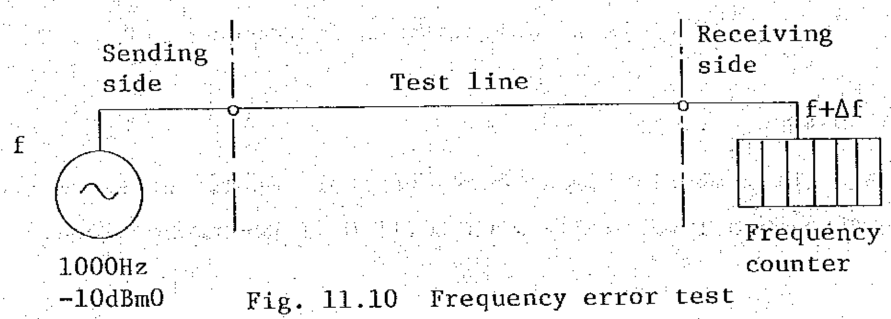


Fig. 11.10 Frequency error test

Criteria: Frequency offset should be within ±15 Hz.

7) Harmonic noise distortion

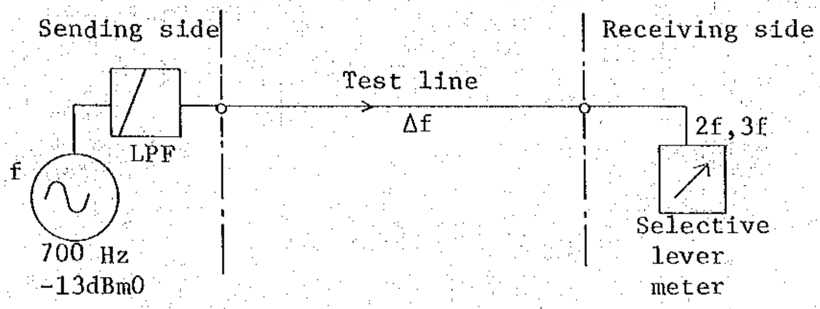


Fig. 11.11 Harmonic noise test

Criteria: Each harmonic distortion receiving level should be less 25dB than the receiving level of standard frequency (700 Hz).

8) Gain hit

Gain hit means an instantaneous level shift, and mainly occurs when a FDM system changes to its spare system, or due to fading in Microwave Communication systems.

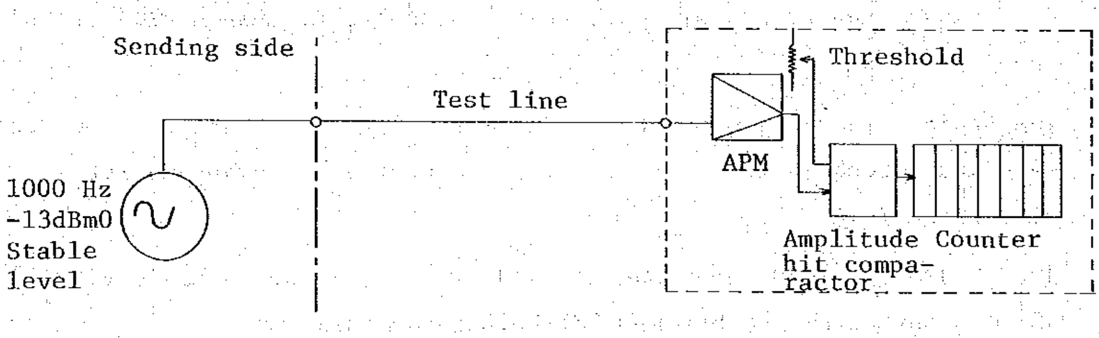


Fig. 11.12 Gain hit test

The Sender sends 1,000 Hz - 13dBm0 in very fixed level, and Receiver receives with fixed threshold level ±3dB for 15 minutes.

Criteria in KDD: Gain hits should be less than 8 times for 15 minutes.

9) Phase hit

Phase hit is same as an instantaneous phase Jump, caused by the same reason as the gain hit.

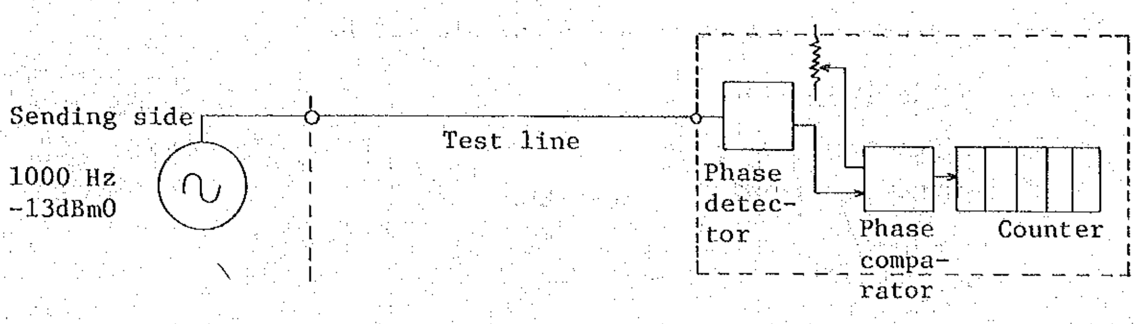


Fig. 11.13 Phase hit test

Sender sends 1,000 Hz, - 13dBm0 and Receiver receives the signal with its threshold fixed at 20 degree, this is continued for 15 minutes and the occurrence of phase jump is counted for this period.

Criteria in KDD: Phase hit should be less than 8 times for 15 minutes.

10) Drop-outs

This test measures the numbers of line break which exceeds the fixed threshold level.

CCITT 0.61, 0.62 recommends the width of the line break, and most of the Measurement Equipments are designed to detect more than 4 mili second of the line break.

The Sender transmits 2,000 Hz, with -13dBm0 output level, and the Receiver sets its threshold level at -12dB, and measures the drop-out for 15 minutes.

CCITT, M 1020, 1050 Recommendations does not fix the specification limit of this test item yet but KDD CRITERIA is as follows:

KDD criteria: The drop-outs should be Zero for 15 minutes, but once occurred, then continue the test for another 45 minutes, and confirm the numbers of drop-outs does not exceed more than 2 for one hour.

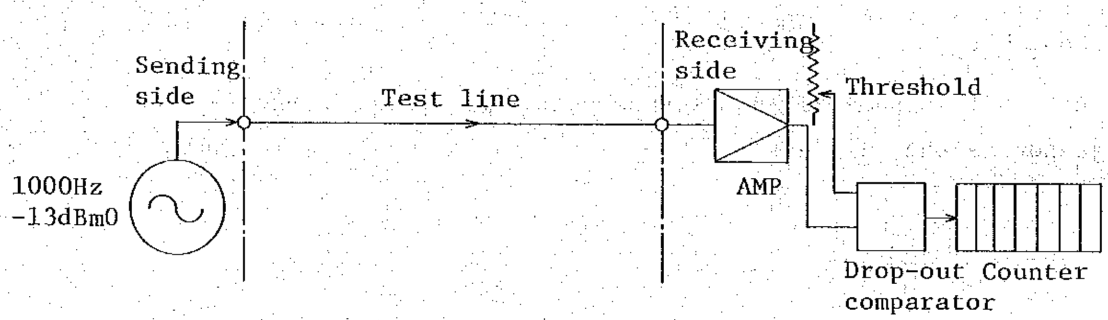


Fig. 11.14 Drop-outs test

11) Go-To-Return cross talk

Go-To-Return cross talk means a CROSS-TALK of the nearest line in the same channel.

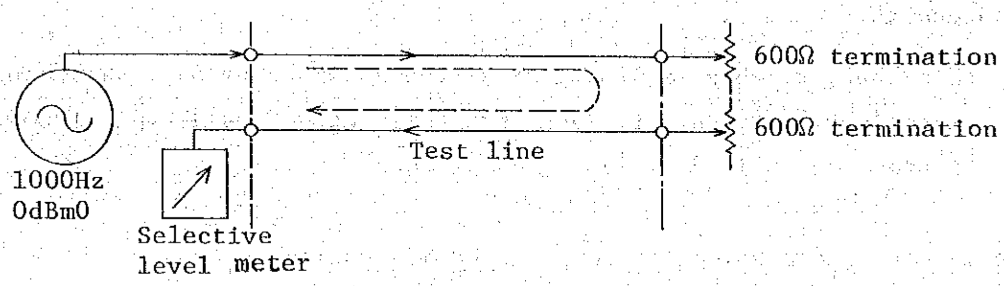


Fig. 11.15 Go-To-Return cross talk test

KDD criteria: go-to-return CROSS TALK should be within -43dB.

Chapter 12 PACKET SWITCHING

The words "Data communications" has been getting more popular, and progress of technology to meet the demand towards Data communications has actually advanced enough.

In this chapter, this is aiming at giving an image of Data switching and network technology, particularly about switching network and typical communication procedures, which is usually called protocol.

12.1 Demand and generation for computer communications

From the beginning of the 1960's demand for terminal communications to enable terminal users to utilize the resource of host computers from distant locations (apart from host premises), has increased remarcably.

Also, when more than one computer exist within one communication group, host-host communications enable resource sharing and load sharing among those hosts are also often requested.

A typical data communications example, T.S.S. (Time Sharing System) for the job of culculations, data retrievals and data storage etc. are used through the public, conventional telephone network in developed countries.

Fig. 12.1 shows the T.S.S in the public Telephone network.

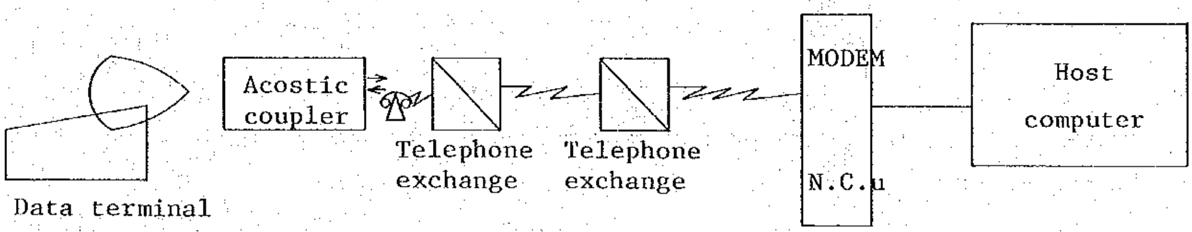


Fig. 12.1 T.S.S in the public Telephone network

Through one pair of telephone wires, the date communications can be realized in the voice frequency band signal, and forms sequencial bit stream.

12.2 Bit sequencial data communications protocol

To communicate properly, it is necessary to have a protocol (a kind of communication rules) between the data sender and the receiver, like same data format, fixed send/receive procedures.

Figure 12.2 shows typical Asynchronous Data format.

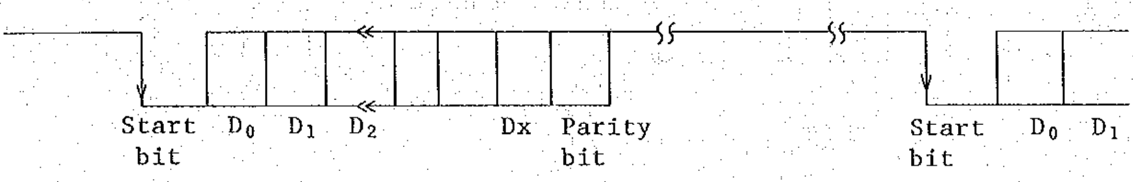


Fig. 12.2 Data Format of Asynchronous

For example, in the above case, the following matters must be regulated.

1) Parity bit, Even or Odd

2) The numbers of data bits

3) Used code (ASCII code is commonly used)

4) The numbers of stop bit

5) Communication speed (bit rate)

and other considerable matters

1) Communication method (half or full duplex)

2) Terminal interface (RS 232C, centronix, 20mA current loop, General Purpose Interface, etc.

12.3 BSC (Binary Synchronous Communications)

There are mainly two kinds of communication ways for data communications, one is asynchronous, mentioned above, and other is synchronous communication.

BSC, developed by IBM is a typical synchronous communication method, and calls BISYNC (Binary Synchronous Protocol).

12.3.1 Synchronizing

In data communications system to setup of synchronization, there are two kinds of synchronizations. One is called Bit synchronization, and the other is Character synchronization. A bit synchronization is carried out by setting a bit position as a signal of synchronization , using "leading PAD" by 55 or AA. A character synchronization is carried out by binary data synchronization, using 2 characters of <SYNC> code. At the end of data or control character of transmission, trailing PAD (FF) signal is given.

PADL	SYN	SYN	SYN	SYN	Data or control character	PADT

Fig. 12.3 Setup format of synchronizing

12.3.2 Basic specifications for BSC control

a) speed: 1,200 bps 9,600 bps

b) mode: Half duplex (4W or 2W)

c) synchronizing: Independent synchronizing

d) connection mode: Contention

e) used code: EBCIDIC (Non transparent)

f) error control: CRC check $(X^{16} + X^{15} + X^2 + 1)$

g) response: <ACK$_0$>/<ACK$_1$>, <NAK>

h) block length: within 256 Byte

i) applied circuit: Non exchange, point to point (Leased or Local Network)

12.3.3 Message format

BSC has the following regulations of message formats and using EBCDIC 8 units code. Transmission control character such as SYN, SOH, STX etc. the fixed code, are used for separation of the data block, message block or control signal, and start/end of the data transmission.

PADL	SYN	SYN	SYN	SYN	SOH		ETB	CRC	CRC	PADT

Heading Format

PADL	SYN	SYN	SYN	SYN	STX		STX		ETB	CRC	CRC	PADT

Data Block Format

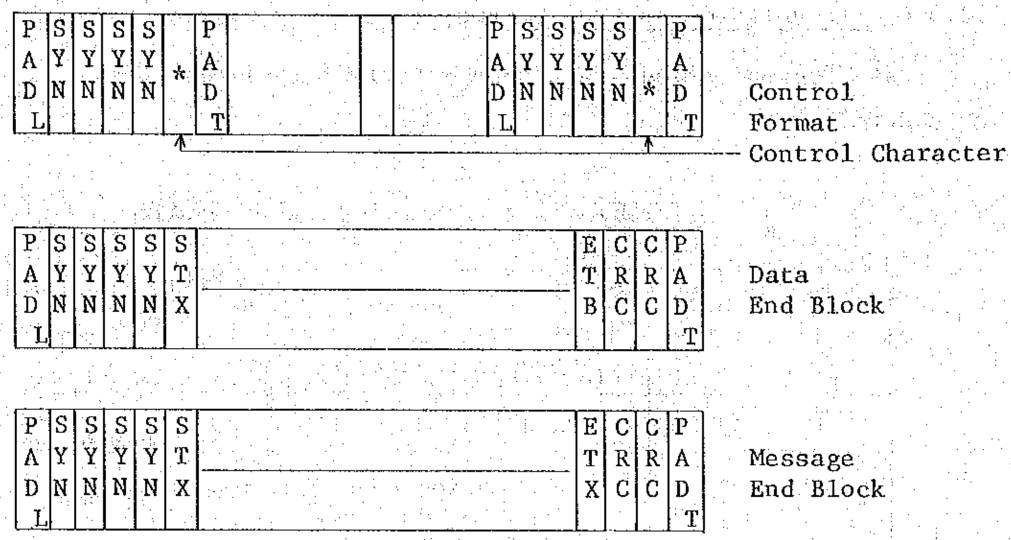


Fig. 12.4 Message Format

a) Header starts with (SOH), end with (ETB) or (STX).

b) Block stars with (STX) end with (ETB) or (ETX).

c) Message contains numbers of blocks and ends with (ETX).

d) A transmission contains numbers of message blocks and ends with (EOT).

Table 12.1 Transmission control character

Code	Name	EBC code	Code	Name	EBC code
DLE	Data link escape	10	ENQ	Enquiry	2D
SYN	Synchronous idle	32	NAK	Negative acknowledge	3D
SOH	Start of heading	01	ACK_0	Acknowledge 0	10, 70
STX	Start of text	02	ACK_1	Acknowledge 1	10, 61
ETB	End of trans-mission block	26	WACK	Wait acknowledge	10, 6B
ETX	End of text	03	RVI	Reverse interrupt	10, 7C
EOT	End of trans-mission	37	TTD	Temporary transmission delay	02, 2D
PAD_L	Leading PAD	55	PAD_T	Trailing PAD	FF

12.3.4 Transmission control procedure

Data Communications require the following Basical Transmission Control Procedure.

Table 12.2 Communication procedure

Phase No.	Basic function
Phase 1	Circuit connection phase
Phase 2	Data link setup phase
Phase 3	Information data's transmission phase
Phase 4	Data link end phase
Phase 5	Circuit disconnection phase

BSC protocol, at 12.3.2 BSC control shows that non exchange point to point. In another words, a circuit is already setup. So phase 1, phase 5, are not required.

(1) Data link setup

a) Send phase II

The data link can be setup only when a calling sequence is reached to Receive side and Sender side received a positive esponce <A CK>.

A calling sequence is <PAD_L><SYN><SYN><SYN><SYN><ENQ><PAD_T>.

b) Receive phase II

Receiving side station answers as follows against its calling <ENQ>.

<ACK_0> Returns signals when receiving side can receive and receiving buffer memories are empty.

<WACK> Returns signal when receiving buffer is full, or collision of calling sequence.

(2) Data transmission phase

a) Send phase IV

After the data link is setup, the sending side starts to send a "Data Block". At tne end of the Data Block (including data Block), the Sending-side puts ETB and if the reply from the Receiving-side is positive, it continues to send the "Data Block" or "EOT", as the End transmission. If the reply from Receiving-side is negative <NAK>, Send-side quickly retransmits the data blocks.

b) Receive phase IV

After the data link is setup, the Receiving-side starts to receive "Data Blocks". If the receiving of the Data block is normal, the Receiving-side replys with the positive return <ACK_0> or <ACK_1>. If the receiving-side received abnormal data block, including CRC error, it quickly replies the negative signal <NAK>.

(3) Data link end phase

a) Send-Receive phase V

The Data Link closes when the sending side sent a <EOT>, or the receiving-side received <EOT>. The <EOT> are transmit in following conditions.

i) Sending-side, in case, there is no data to be sent.

ii) The system's failure, or abnormal condition prevents it to send a data block.

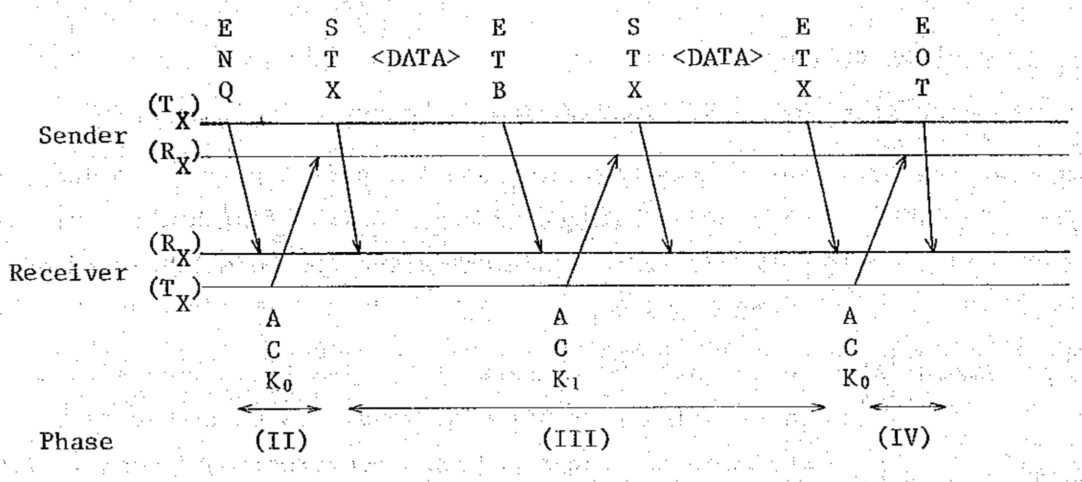


Fig. 12.5 Transmission sequence chart (Normal)

12.3.5 BSC Demerit

BSC protocol is mainly used in General Purpose Computer's data communications, but there are following demerits.

1) The contents of data should be, even within a data block, always of the EBCDIC code. It means the Randam pattern data, such as a computer memories damp can not be sent.

2) There is a limitation of the block length. (in efficient for high speed)

12.2 HDLC (High Level Data Link Control)

High Level Data Link Control (HDLC) is a message exchange procedure between sending and receiving data terminals, and is devised to accomplish error-free and efficient communication. SDLC (Synchronous Data Link Control), the IBM's adapting procedure is very similer to HDLC.

Various procedures laid down include remote activation of distant terminal equipment, notification of receiving status, and automatic error collection by signal retransmission. The standardization of HDLC was started around 1971 by ISO, and the international standards were established in 1979.

In this control system, unfailing transfer of information is made possible by controlling both sending and receiving terminals in such a manner that the receiving terminal responses to the message from the sending terminal which in turn transfer the next message after confirming this response. To achieve this, information signals to te transmitted are required to be divided into frames of a certain length, each of which is transmitted as a unit. Message transmission in frames is also a prerequisite to error correction by retransmission.

12.2.1 Frame structures

The frame consists of the following five elements;

1) Flag sequence (F): Set at the beginning and the end of a frame to indicate separation between frames. As the flag has a definite pattern, frame synchronization can be established by detecting it.

2) Address field (A): Represents the address of a frame to be transmitted, or the sending terminal's address. As described above, messages are exchanged in a conversational manner, i.e., the receiving terminal responses to a command from the sending terminal. Command frames are transmitted with remote terminal address, while response frames are transmitted with the local terminal address.

3) Control field (C): Contains bits which represent the type of frame (information frame, supervisory frame, etc.) a sequence number given to the frame, the initiation of transmission and the indication of final frame.

4) Information field (I): Stores data to be transferred. The length varies according to the source of information. Usually, a length of about 1,000 bits is used.

5) Frame checking sequence (FCS) : Includes code word for error detection, which checks whether the contents of address, control and information fields have been correctly transferred or not.

Flag sequence	Address field	Control field	Information field	Frame check sequence	Flag sequence
F	A	C	I	FCS	F
01111110	8 bits	8 bits	N bits	1 bit	01111110

Fig. 12.6 Frame construction in HDLC

So the HDLC Frames flow as follows;

F A C I .... R R F .... F A C I .... R R F ....

12.2.2 Characteristics of HDLC

(1) HDLC is a system that connects a main station and several secondary stations. The connections are multipoint connections or loop connections. See Fig. 12.7, 12.8

Secondary stations can transfer messages only when it has a permission to send. By the multipoint connection, the communication mode is full duplex, so the transfer of data can be carried out A to B and C to A at the same time. By the loop connection, the communication mode is half duplex, so secondary stations must wait until to get the permission.

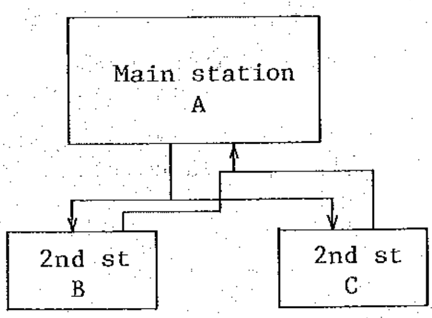


Fig. 12.7 Multipoint connections

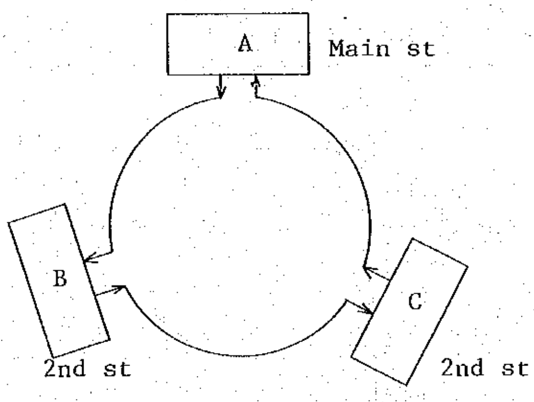


Fig. 12.8 Loop connections

(2) HDLC is a Request/Reply system in basis, but if there is no errors or Request command in transmission, system can continously send up to 8 Frames. And the control Field of the frames contains a "Reply" to other stations, so it means the HDLC can send information and Request/Reply simultaneously.

(3) Each secondary (2nd) station has its own address and common address. When the main station sends the messages to the secondary stations with the common address, all the stations which have the common address receive the messages. On the contraly, when the main station send the pole messages (message Request to the main station) to the secondary stations, all secondary stations can send replies simultaneously.

12.2.3 Control field

The HDLC is designed to accommodate expanding its function in SDLC's Address fields and control fields. In another words, HDLC fields length is not limited in 8 bit, by fixing 0 in the beginning of the field, it can expand the field to another 8 bit unit.

The HDLC has 4 kinds of frame: 1) the command frame (Main to secondary), 2) the Reply frame (from secondary to main), 3) the condition frame (both direction) and 4) the information frame (both direction).

Fields	Form	Originator
Command	c c c P c c c c	Main station
Reply	r r r F r r r r	Secondary st
Status	Nr P s s s s Nr F s s s s	Main st Secondary st
Information	Nr P Ns 0 Nr F Ns 0	Main st Secondary st

Fig. 12.9 Control field

Note

- c,r,s: Confines the meaning of command, reply, status
- p : P stands when Main st Requests to the Secondary st (1), not Requests any (0).
- F : F stands the last Frame of Reply from Secondary st (1), if not last Frame (0).
- Nr : Nr is next expecting frame number.
- Ns : Ns is the number of transmitting Frame.

12.2.4 Communication method and command

(1) Initialize (SIM, SNRM, UA, RIM command)

Fig. 12.11 shows the procedure of initialization of B station from Main A station. After the electric supply is fed to the system, the system is required the system initialization by using SIM command. After the system initialization, the system is in the status of an off-line and called normal disconnected mode. In this mode, the system replies only to the command of TEST, XID, CFGR, SNRM, and SIM.

Secondary stations become ON LINE when the Main station gives the command of SNRM, and by DISC it returns to OFF LINE.

B, $\underbrace{\text{RR,}}_{\text{Command}}$ P (1)

Called Station name → B,
Pole bit → P
Nr → (1)
(1) : P
(0) : $\overline{\text{P}}$

C, I (2), P (3)

Information Frame → I
Frame Number 2 (Ns) → (2)

$\underbrace{\text{B, RR, P (1)} \quad \text{C, I (2), P (3)}}_{\text{Main station's control field}}$

B, $\underbrace{\text{RR,}}_{\text{Command}}$ F (5)

F bit → F
Nr → (5)

$\underbrace{\text{B, RR, F (5)}}_{\text{Secondary station's control field}}$

Fig. 12.10 Expression of frame abriviation

A st (Main)	B st (Secondary)	
B, SIM, P ⟶		; Initialize Request A to B
	⟵ B, UA, F	; Positive Reply B to A
B, SNRM, P ⟶		; Normal mode request to B
	⟵ B, UA, F	; Positive Reply B to A

Fig. 12.11 Initialization of HDLC

Type	Pattern	Abriviation	Contents
U	000 P/F 0011	UI	Unnumbered information
	000 F 0111	RIM	Request initialization
	000 P 0111	SIM	Set initialization mode
	100 P 0111	SNRM	Set normal response mode
	000 F 1111	DM	Disconnected mode
	010 P 0011	DISC	Disconnect
	011 F 0011	UA	Unnumbered acknowledge
	100 F 0111	FRMR	Frame reject
	111 F 1111	BCN	Beacon
	110 P/F 0111	CFGR	Configure
	010 F 0011	RD	Request disconnect
	101 P/F 1111	XID	Exchange identification
	001 P 0011	UP	Unnumbered poll
	111 P/F 0011	TEST	Test
S	Nr P/F 0001	RR	Receive ready
	Nr P/F 0101	RNR	Receive not ready
	Nr P/F 1001	REJ	Reject
I	Nr P/F Ns0	I	Information

Fig. 12.12 Command list

(2) Transmission and retransmission of message (RR, I)

Main station can send the Information frames having specified the destination address to the secondary station; at any time. Whenever P (Polle bit) of the control field is "1", secondary stations are requested to reply to the Main station. Secondary stations can send the Information frames replying the RR (Receive Ready) command from the Main station.

A (Main)	B (Secondary)	C (Secondary)	
C, RR, P (0) ⟶			; A allows C to send
⟵		C, I (0), $\overline{F}$ (0)	: C sends # 0 frame
B, I (0), $\overline{P}$ (0) ⟶			; A sends # 0, # 1 frame to B, but no reply
B, I (1), $\overline{P}$ (0) ⟶			
⟵		C, I (1), $\overline{F}$(0)	
⟵		C, I (2), F(0)	: C completed send
B, RR, P (0) ⟶			; A allows B to send requests B to reply
⟵	B, RR, F (2)		; B replys but no data to send. Informs the completion of receiving up to # 1.
	C, RR, $\overline{P}$ (3) ⟶		; A informs C the comple- tion of receiving up to # 2, and allows C to send.

Fig. 12.13 Transmission of messages

(3) Interruption of transmission

Main station and secondary stations caninterrupt the sending and receiving by the RNR frame. The main station sends the RNR frame, then the secondary station sends the Final frame with F bit (1). Restart of transmission is required by the RR command from the Main station.

12.3 Packet switching

Packet switching introduction is main subject in this chapter.

12.3.1 CCITT Definitions:

Packet; A group of binary digits including data and call control signals which is switched as a composite whole. The data, call control signals, and possibly error-control information are arranged in a specified format (1).

Packet switching; The transmission of data by means of addressed packets whereby a transmission channel is occupied for the duration of transmission of the packet only. The channel is then available for use of packets being transferred between different data terminal equipment (1).

Note; The data may be formatted into a packet or divided and then formatted into a number of packets for transmission and multiplexing purposes.

12.3.2 General

Store-and-forward switching has existed for decades in telegraphy where it is called message switching. At the end of the 1960s a new type of store-and-forward switching came into experimental use, called packet switching. As message switching is intended primarily for non-real-time people-to-people traffic, packet switching is intended primarily for real time machine-to-machine traffic, including terminal-to-computer connections, and is employed to build computer networks.

The most important difference is in the speed of the network. A packet-switching network may be expected to deliver its packet in a fraction of second whereas a message-switching system typically delivers its message in a fraction of an hour.

ADVANTAGES OF PACKET SWITCHING

The advantages of packet-switching networks derive from two different aspects. The first aspect is the switching and routing technique, the second is the storage and message manipulation offered by the interface computers. Either aspect could be retained without the other. Switching computers could interconnect compatible hosts or terminals; interface computers could be used with a different form of switching network.

The main advantages of the switching and routing technique are as follows:

1. Fast response times.
2. High availability because of distributed routing.
3. High-speed data bursts can be handled as well as low-speed requirements.
4. No blocking except when the network storage is flooded.

The main advantages of the interface computers are as follows:

1. Machines using incompatible codes and control procedures can be made to communicate.

2. Terminals operating at different speeds can be connected because of the buffering.

3. Data for a terminal that is busy can be held until the terminal becomes free, rather than a "busy" signal being given.

4. End-to-end protection is possible against transmission errors or message loss.

5. The sending of one message to a named list of destinations is possible.

PACKET SWITCHING SYSTEM

1. Creating the packets: Data from a terminal or host are placed into one or more envelopes, and control information needed for transmission is written.

2. Reassembly of data: After transmission data, often sent in more than one packet, are reassembled. For some uses the entire message may be assembled prior to delivery; for others it may be better to deliver the date a block at a time as they arrive, provided that the blocks are in sequence.

3. Host-network protocol: The interface computer will observe a protocol for communicating with the host computers to ensure that the interchange functions correctly and that no data can be lost. Different protocols may be needed for different types of host.

4. Terminal protocol: A protocol for communicating with user terminals will be observed. Some of the user terminals may be far away, connected to the interface computer by links incorporating multiplexors, concentrators, polling, public network dialing, or other procedures.

5. Special-function protocols: Special protocols may be used for functions such as the transfer of files, the use of graphics, "conference calls" in which more than that one user participates in one dialogue, mailbox services in which the network passes messages between users and holds them until the users see them, facsimile transmission, transmission of exceptionally high security, and others.

6. Session control: Many transmissions, as discussed earlier, are part of a "session" in which multiple message are sent as in a human telephone conversation. In this case the interface computer may control the session. It will store an envolope header for the session so that it does not have to be recreated for each message. It may use a session-oriented protocol with the host computer or terminal. It may allocate a high priority to

the packets of certain sessions.

12.3.3 CCITT Standard packet switching network

CCITT Standard interfaces

Regardless of the interval architecture of the packet switching network, the CCITT Recommendations define a standard means of interfacing (1) DTE with an X.25 software package, (2) non-packet mode DTE via network support of Recommendation X.3, X.28 and X.29, and (3) packet switching networks using X.75. These interfaces are illustrated in Fig. 12.13.

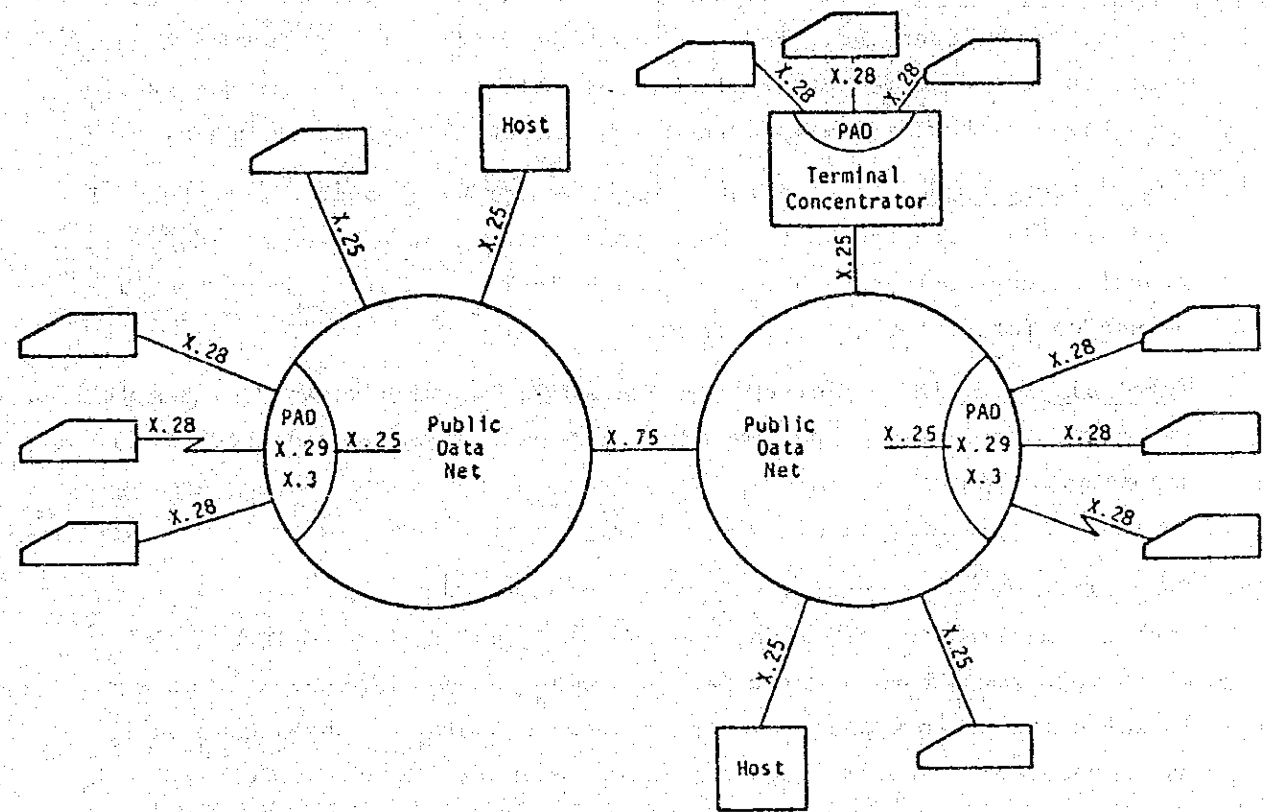


Fig. 12.13 CCITT Standard interfaces

These Recommendations are briefly described below.

Recommendation X.25

Recommendation X.25 defines a protocol to connect a customer's DTE to a network node (DCE). The protocol requires software in both the DTE and the network.

Recommendation X.25 is based on three levels of protocol. The Physical level defines the electrical or modem interface, based on X.21 or X.21 bis. The Link level defines the link access procedures (LAP or LAPB) for transferring frames of data accurately across the access line between the DTE and the network. The Packet level protocol defines procedures for call establishment data transfer, flow control, error recovery, and call clearing. Up to 4095 calls may be simultaneously active on the access line.

Recommendations X.3, X.28, and X.29

CCITT Recommendations X.3, X.28, and X.29 are protocols defined for interfacing non-packet-mode start/stop terminals to a public data network. Software required to "packetize" the native character stream into network packets is resident in a packet assembler/disassembler (PAD).

X.3 defines parameters that specify terminal characteristics and functions of the PAD. X.28 defines terminal signalling; that is, commands typed by the terminal user that are understood by the PAD and service signals that are returned by the PAD to the terminal user. X.29 defines control procedures between the communicating X.25 DTE and the network PAD. Within the network, the PAD may be throught to function as a X.25 DTE, performing packet assembly/disassembly functions at the Packet level on behalf of the non-intelligent terminal.

Recommendation X.75

One of the features of the public data network is that the semantics of X.25 virtual circuits are maintained on an end-to-end basis. Different Administrations use unique internal network protocols (datagram or virtual circuit oriented) but in terms of an external interface, the semantics of X.25 are always maintained. Thus, procedures such as call setup, clearing, resetting, qualified data (in-band signalling), and interrupt (out-of-band signalling) are maintained from DTE to DTE.

When public data networks are interconnected the same requirement holds, so that an X.25 DTE can treat another DTE in a consistent manner, independent of whether the remote DTE is on the same public date network or on another public data network. CCITT has standardized the internetwork protocol Recommendation X.75, which, like X.25, consists of three levels: Physical, Link, and Packet. The Physical level uses a high speed digital or analog trunk. The Link level is based on LAPB, which is compatible with HDLC; the Packet level is also based on X.25, with the expection of the addition of an extra field for network utilities in call control packets. These network utilities are used for signalling network information. A procedure has recently been adopted by X.75 at the Link level to operate over multiple lines to achieve greater call reliability.

Recommendation X.121

CCITT Recommendation X.121 defines an International numbering plan for identifying public data network addressing. Each ITU member is assigned one or more data network identification codes (DNICs). A DNIC is a four digit number; the first three digits identify the country and the fourth identifies a particular public data network in that country. A particular DTE is identified by a DNIC followed by up to 10 digits assigned by the PDN on a national basis. Thus, X.121 provides the necessary elements for routing between PDNs by specifying the DNIC and also permits each PDN to structure unique addressing schemes.

Recommendations X.1, X.2, and X.96

Recommendations X.1, X.2, and X.96 provide general information to the users on access data rates supported on PDN's, network services, optional user facilities, and network generated call progress signals. Some of these items are essential (i.e., every network has to provide them), while others are marked as optional or for further study.

3.2.2 Layered levels of protocol

We can view packet switching networks as having several layers of communications. At each level, protocols have been specified to provide well-defined rules and procedures for exchanging messages (see Fig. 12.14 below).

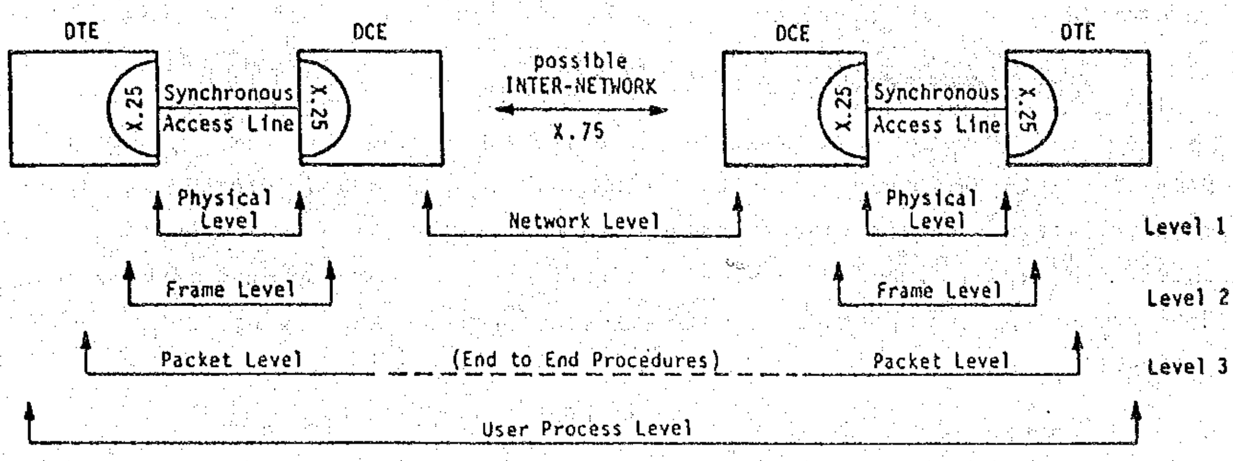


Fig. 12.14 Layered levels of protocol

At the lowest level (L0), messages are exchanged between network nodes for (1) transmission error detection and correction, (2) flow control to avoid message congestion, and (3) adaptive routing based dynamically on traffic volume in different parts of the network.

The next level of communication (L1) is the correct exchange of data set signals to control actual transmission of data over the physical access lines.

At the next level (L2) is communication between the network (DCE) and the DTE (Data Terminal Equipment, which may be Host computer, a Frotn End, a terminal cluster concentrator, etc.) to maintain and provide error and flow control on the physical connection to the network.

At the next level (L3) is communication between the network (DCE) and the DTE to create and maintain virtual connection paths between DTE's and to control flow on the virtual circuits.

On the highest level (L4), messages pass between DTE's to provide communication between processes running on remote DTEs (where one end may be a user sitting at a terminal).

12.3.4 Advice from author

Packet switching is an appealing way for the developing countries to build a public network which can attract a very high volume of users as well as low volume users. It is not economically attractive today for a medium-sized corporation or organizations to build its own private network especially long trunk link involving in the developing countries.

At the end of this chapter, typical Packet switching system is shown as a reference.

VENUS-Packet switching system

Kokusai Denshin Denwa Co., Ltd. (KDD), which is the sole common carrier providing international telecommunications services in Japan, started its long-planned international packet switched data transfer service in the beginning of this fiscal year. The new service will fill the need for improved international data communications to handle the growing quantity and variety of traffic diversification.

(1) Outline of the service

The international packet switched data transfer service (VENUS-P) is an international public telecommunications service which allows transmission of digital data by means of public telecommunications facilities, specifically packet switching exchanges and telecommunications circuits.

At present, most VENUS-P users are in Tokyo and Osaka. VENUS-P service will be available to subscribers to DDX-P service, the domestic packet switched network service offered by Nippon Telegraph and Telephone Public Corporation (NTT), by the end of July.

Areas to which KDD now offers service either through direct circuits or transit facilities are France, West Germany, Spain, the U.K., and the U.S. Later, KDD plans to expand the service to other Western countries such as Canade, Switzerland, Italy, etc., and to Asian countries.

Transmission speeds of subscriber lines are 2,400 bps, to 9,600 bps.

Data terminal equipment (DTE) supported by VENUS-P are:

CCITT X.25 packet switched terminals, HDLC terminals and BSC terminals.

For subscribers' premises, KDD provides data circuit terminating equipment (DCE) for use with subscriber DTE.

Optional facilities for users include multiple logical channels for X.25 terminals, abbreviated dialing for HDLC and BSC terminals, and direct calls for BSC terminals.

(2) Technical outline

VENUS-P adopts the packet switching system. Packet networks have been generally recognized as an appropriate network for computer communications, packet switching is suitable for data transmission between computers and conversational communication between computers and terminals. Due to these characteristics, the packet net-work has been considered to be suitable for the implementation of international data network service. Moreover, since public packet switched networks are increasingly available worldwide, VENUS-P can easily become part of the international public packet switched network which interconnects various networks of foreign countries.

The Characteristics of the VENUS-P service make it possible to provide full-scale international packet switched functions which conform to international standards.

An outline of the service technology is given below.

Due to the advantages of packet switching technology, VENUS-P affords a low data error rate through error control techniques, short calling setup time, bit-transparant data transmission for X.25 and HDLC terminals, simultaneous sending/receiving of independent data among a plural number of terminals for X.25 terminals, and communications between terminals with different ransmission speeds.

Error control can be carried out, according to the transmission control procedure, between a local terminal and a local exchange office, between exchange offices, or between a remote exchange office and a remote terminal.

(3) Network configuration.

Network configuration of the VENUS-P is shown in Fig. 2.

VENUS-P and foreign packet switched networks are interconnected in conformity with the CCITT Recommendation X.75 which standardizes international interconnection protocols among networks. Interconnection of various countries' packet networks has been recently proceeding rapidly using the X.75 standard. Therefore, VENUS-P can easily be interconnected with other foreign networks further in the future.

Interconnection of DDX-P, the domestic packet switched network, and VENUS-P also uses a protocol conforming to the X.75.

(4) Transmission control procedure for terminals.

Three kinds of transmission control procedures - X.25, HDLC and BSC are available in VENUS-P. The X.25 protocol conforms with CCITT Recommendations. The transmission control procedure for HDLC terminals conforms with the high-level data transmission control procedure (class BA:

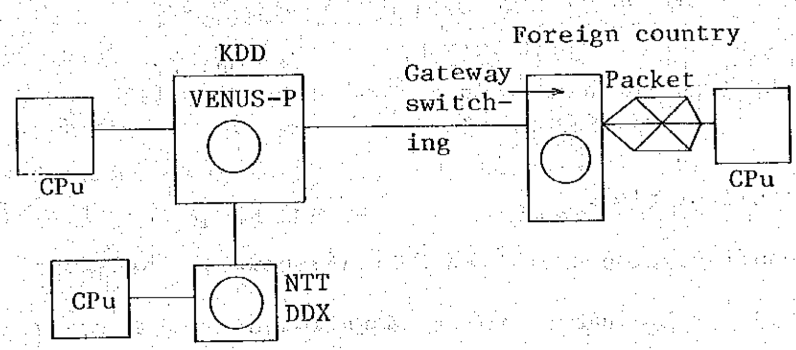
asynchronous balance type) of ISO. Moreover, the transmission control procedure for BSC terminals, which conforms with the basic-mode transmission control procedure, adopts half-duplex communication mode, SYN synchronization, and contention mode.

(5) Communications compatibility between transmission control procedures.

Combinations of the three current transmission control procedures with those used with accessible terminals abroad are shown in Table 2. As shown in Fig. 2, in foreign systems, X.25 and X.28 (control procedures between asychronous terminals and the network) are ordinarily used and, in addition, a BSC control procedure is also used in the U.S. and Canada.

Flow control is applied to the terminal according to the transmission speed of the other terminal and the degree of congestion in the packet switched network. Typical examples of possible connections using the service are shown in Fig. 12.15.

(1) Computer-Computer (File, Program transfer, Load share)



(2) Computer-Terminals (Remote job, TSS acces) or Terminals-Computer

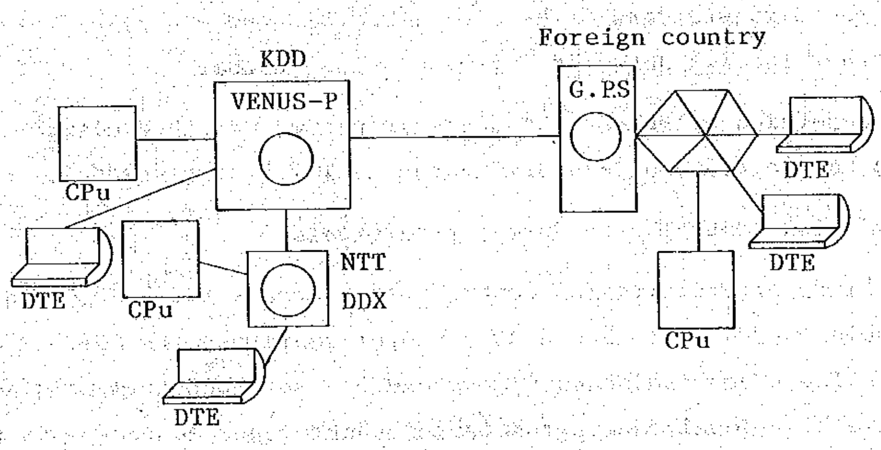
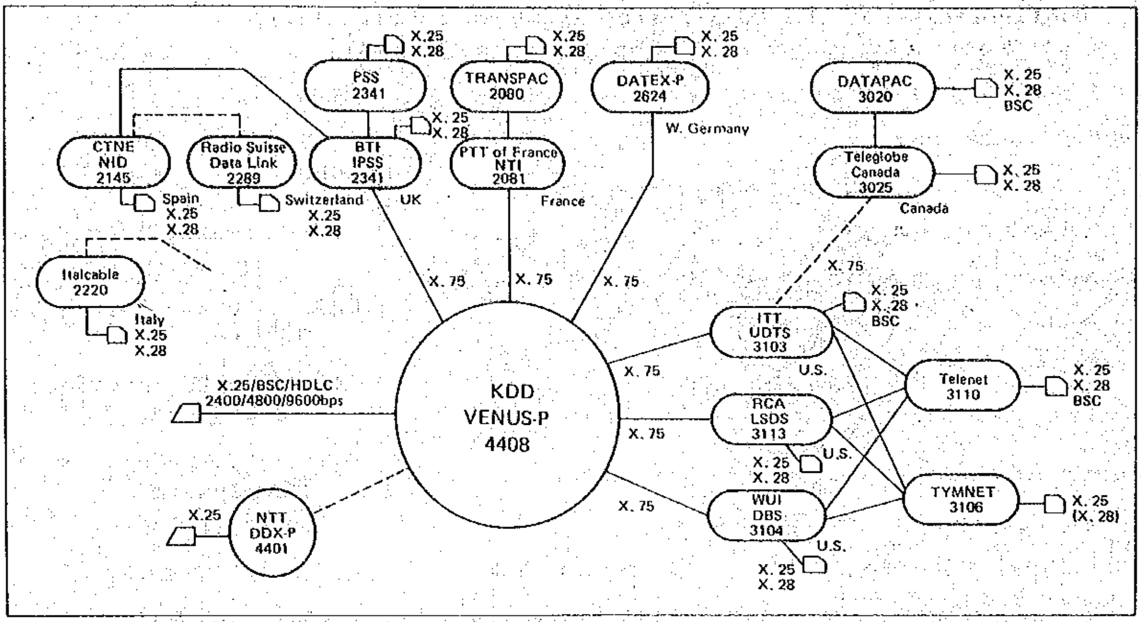


Fig. 12.15 Typical usage



Note: X.25: Interface between a packet terminal and a packet network; X.28: Interface between an asynchronous terminal and a packet network; X.75: Inter-network interface signaling system; BSC: Basic Communications Control Procedure; HDLC: High-level Data Link Control Procedure.

Fig. 12.16 Network configuration of VENUS-P service

(6) Numbering plan for VENUS-P.

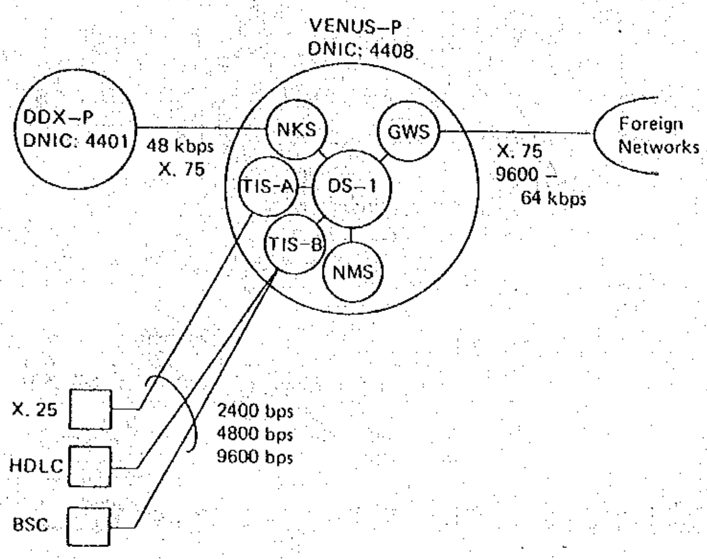
A numbering plan for public data networks is recommended under CCITT Recommendations X.121. According to this, as described below, the first four digits are data network identification codes (DNIC), and the successive digits (to a maximum of ten) are network terminal numbers (NTN).

$$\underbrace{X_1X_2}\underbrace{X_3X_4}\underbrace{X_5X_6 \text{ -------------- } X_{14}}_{NTN}$$

In the case of VENUS-P, 4408 has been assigned as DNIC (Japan's country number is 440), and NTN is shown by the five digit X_5 X_9. DNICs of the data network of each country are shown under the network name in Fig. 2.

(7) Packet switching system for VENUS-P.

The switching system (DS-1 data communications processing system) for VENUS-P has a distributed processing system in order to allow easy accommodation of increases in communications traffic and expansion of functions. In other words, the switching system consists of several by-function subsystems, and each subsystem is interconnected through high-speed data highways using optical fibers. Functions of each subsystem are: carrying out packet transmission with foreign packet switching nodes according to X.75; handling X.25 control procedure; handling BSC and HDLC control procedures; handling interconnection with DDX-P; managing the whole system; and connecting and controlling charging information, etc. The configuration of the DS-1 system is shown in Fig. 12.17



Note:

TIS: Terminal Interface Subsystem

GWS: Gate Way Switch Subsystem

NKS: NTT-KDD Interface Subsystem

NMS: Network Management Subsystem

Fig. 12.17 Configuration of DS-1 system