INTERNATIONAL ATOMIC ENERGY AGENCY
UNITED NATIONS EDUCATIONAL, SCIENTIFIC AND CULTURAL ORGANIZATION
INTERNATIONAL CENTRE FOR THEORETICAL PHYSICS
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H4.SMR/585-6

FIRST INTERNATIONAL SCHOOL ON COMPUTER NETWORK ANALYSIS AND MANAGEMENT

(3 - 14 December 1990)

DATA COMMUNICATION OVER PSTN AND PSPDN

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Lecture notes

DATA CIRCUITS

1.PHYSICAL LAYER

The lowest level of the OSI reference model defined by the ISO is the **physical layer** which defines the characteristics of the transmission channels relating to the teleprocessing system.
The services provided to the upper layer (link layer) by the physical layer ensures transparent transmission of the message bits.
Data units in this case contain only a single bit.

The physical channel between two pieces of data processing equipment can be represented diagrammatically as shown in Figure 1.1 The data circuit consists of a transmission medium and two pieces of Data Circuit Equipment (DCE) whose role is to establish the communication, to ensure that the digital data is in a suitable form for transmission on the medium and to break the communication when the transmission is terminated. If the line is a long-haul telephone line, as is most often the case, the DCE are Modems (modulator-demodulators) which perform the complex functions of modulation, demodulation and filtering in order to adapt the digital signals to the conventional telephone channel; however the name modem is often extended to all DCE even when this equipment does not perform modulation or demodulation.

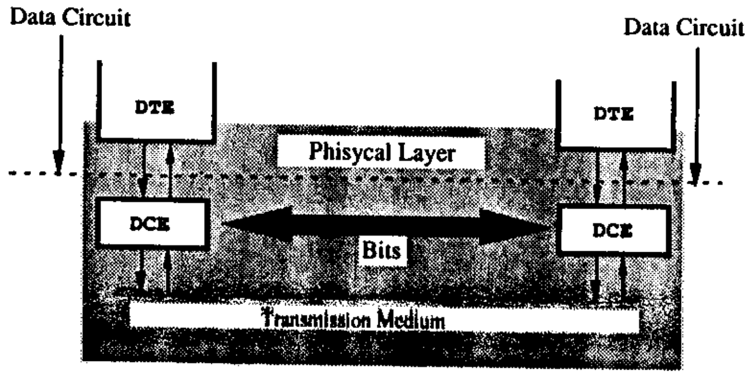

Fig. 1.1 Data circuit

A data circuit connects two or more pieces of data processing equipment, conventionally called Data Terminal Equipment (DTE). In practice DTEs in the simplest form are plain terminals; advanced DTEs are Personal Computers, Workstations or even Mainframes.

In order to ensure suitable interconnection of the equipment, which can be of many forms, it is essential that the means of digital transmission should be as compatible as possible. This has led to the **standardization** of the transmission techniques and the interfaces between the data transmission systems and the data processing equipment.
This standardization relates essentially to the types of line signals which the DCE must be capable of handling and DTE-DCE interfaces. The first series of standards defines the principal characteristics of the modem and the transmission lines which it uses. The second series of standards specifies the electrical and mechanical characteristics of the modem-terminal interface and the sequence of operations to be performed in order to exchange information (establishment, maintenance and release of the circuit).

1.1 Basic functions

Data circuits can be **simplex** , **half-duplex** or **full-duplex** according to whether they operate in a single direction, in two directions alternately or in both directions simultaneously. In general, data processing equipment handles information in the form of words which contain a certain number of bits, for example 16 or 32. The bits of a word are manipulated as a whole and processing can be considered is performed sequentially word-by-word and in parallel on the various bits of the word. Vis-a-vis data transmission is generally performed bit serially, for reasons of transmission efficiency, and above all, of economy of investments on transmission infrastructures.

The sequence of information bits transmitted on the line is, therefore, in the form of a series of blocks separated by intervals during which there is no transmission of information. In order to decode the information at the receiver it is necessary to provide a bit clock which indicates the sampling time at which a bit is considered to be valid. It is also necessary to provide a block clock or frame clock which enables the start and end of a block or frame to be recognized. The receiver bit and frame clocks clearly must operate synchronously with the corresponding clocks at the transmitter; this implies that the DCEs must exchange signals which enable them to maintain synchronism.
There are two principal methods of synchronization in data transmission, **asynchronous transmission** and **synchronous transmission**.
With asynchronous transmission, or start-stop transmission, the data is transmitted in blocks of several bits leaded by a start bit which is always 0 and followed by one or two (sometimes 1.5) stop bits which are always 1 (Figure 1.2). The spacing between blocks is arbitrary and the state of the line between two blocks is the logical state 1. It can be seen that the start of a block is identified without ambiguity by a 1-0 transition which corresponds to the

arrival of the start bit. As the number of bits in the block is constant and the spacing between bits in a block is fixed, the sampling instants of the various bits in the block are determined from the start bit. In practice, due to the unavoidable mismatch between the receive clock frequency and the transmit clock frequency, the receiver sampling instants will shift from their theoretical position in proportion to the distance from the start bit For this reason the block size used in asynchronous transmission is usually limited to a length of about ten bits, which generally corresponds to a character. This transmission technique leads to poor utilization of the line, since each group of bits (five to eight) forming a text character must be enclosed between a start bit and one or two stop bits. Moreover, correct operation is closely related to correct detection of the start bit, which sometimes may become critical in the presence of noise. For these reasons, asynchronous transmission, still used in telegraphy and for low speed data transmission, is being progressively abandoned in favour of synchronous transmission.

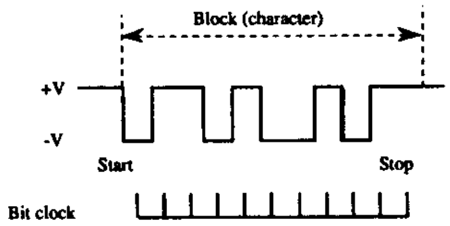

Fig. 1.2 Asynchronous transmission

The principle of synchronous transmission consists of transmitting a synchronizing signal in parallel with the data in order to lock the receiver bit clock, by sending, for example, the transmitter clock signal on a separate line.
For obvious reasons of cost, this approach is in practice used just on the DTE-DCE junction, when the cost of the interconnection means (i.e. ribbon cable) is quite negligible; otherwise a more convenient (as well as complex) technique is adopted, by either transmitting a carrier frequency on the same line as the data or, more often, by using a circuit which reconstructs the receiver bit clock from transitions of the data. This leads to an approach where the receiver continuously provides both the bit clock and the received data, thereby permitting precise sampling of the data (Figure 1.3). As the frequency of the receiver bit clock is in this case precisely equal to the frequency of the transmitter bit clock, the blocks can be of any length; this enables very long blocks to be used and also minimizes the number of block delimiters with respect to the text.

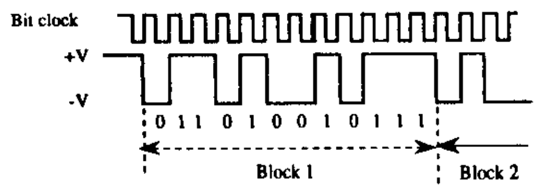

Fig. 1.3 Synchronous transmission

By consequence, in synchronous transmission, the block or frame delimiters no longer serve any purpose with respect to bit synchronization, that is, (unlike asynchronous transmission) bit synchronization and frame synchronization are completely separated. The OSI model provides for bit synchronization to be achieved at the physical layer and frame synchronization to be realized by the data link layer according to techniques which will be examined later.

2. TRANSMISSION MEDIA

The transmission of an electrical signal between two pieces of equipment requires the use of a transmission medium.
In most cases, this consists of a pair of conductors or wires, generally referred to as transmission lines; however, transmission is sometimes achieved by passing a beam of light through a piece of glass fibre or an electromagnetic wave through free space. The type of transmission medium used is important, since it determines the maximum rate, in terms of bit per second that can be transmitted. The more common types of transmission media are briefly reviewed.

2.1 Two-wire open lines

A two-wire open line is the simplest type of transmission medium: each wire is insulated from the other and both are open to free space. This type of line is perfectly adequate for connecting two pieces of equipment that have a short physical separation (less than 50 m) and a modest bit rate (less than 20 kbit/s). The signal, which is typically a voltage or current level relative to some ground reference, is applied to one wire while the ground reference is applied to the other.

Although a two-wire open line may be used to connect two devices (DTEs) together directly, it is used mainly for connecting a DTE to a DCE. Such connections usually utilize multiple lines, the most common arrangement using a separate insulated wire for each signal and a single wire for the common ground reference. Usually the complete set of wires is grouped into the form of a flat ribbon cable as shown in Figure 1.4

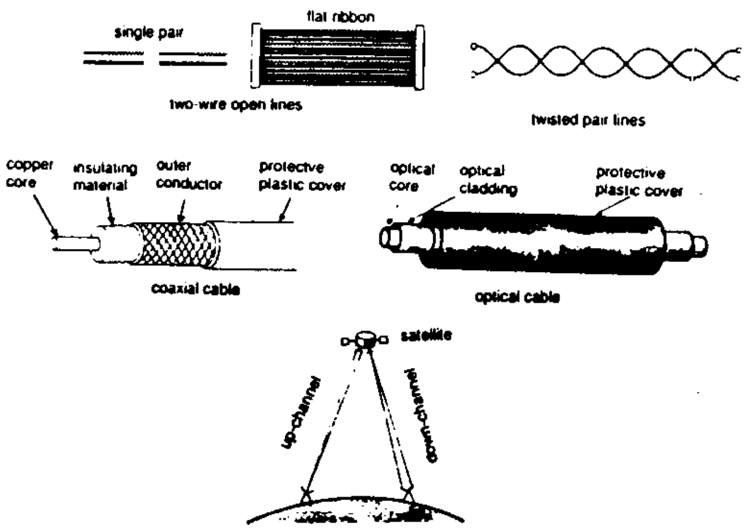

Fig. 1.4 Transmission media

With this type of line, care is needed to avoid cross coupling of electrical signals between adjacent wires in the same cable. This is known as crosstalk and is caused by capacitive coupling between the two wires.
In addition, the open structure of this type of line makes it susceptible to spurious noise signals induced by electromagnetic radiation from other electrical signal sources. As the interference signal may affect a single signal wire and not the ground wire, an additional difference signal can be created between the two wires, and hence can give rise to an erroneous interpretation of the received data signal. These factors all contribute to the limited lengths of line and bit rates that can be used reliably.

2.2 Twisted pair lines

Better noise immunity, can be obtained by employing a pair of wires that are twisted together. This is therefore known as a twisted pair line. The resulting close proximity of both the signal and ground reference wires means that any interference caused by extraneous signal sources is carried by both wires and hence its effect on the difference signal is reduced. Furthermore, if multiple twisted pairs are enclosed within the same cable, the twisting of each pair within the cable further reduces interference effects caused by crosstalk. A schematic of a twisted pair line is shown in Figure 1.4.

Twisted pair line are suitable, with appropriate line driver and receiver circuits that exploit the potential advantages gained by using such a geometry, for bit rates in the order

of 1 Mbps over short distances (less than 100 m) and lower bit rates for longer distances.

2.3 Coaxial cable

The main limiting factor of a twisted pair line is caused by a phenomenon known as the skin effect: as the bit rate (and hence frequency) of the transmitted signal increases, the current flowing in the wires tends to flow only on the outside surface of the wire, thus using less of the available cross-section. This has the effect of increasing the electrical resistance of the wires for higher frequency signals which in turn causes more attenuation of the transmitted signal. In addition, at higher frequencies, an increasing amount of signal power is lost due to radiation effects. Hence, for those applications that demand a bit rate higher than 1 Mbps, it is normal to use another type of transmission media: the coaxial cable.

In a coaxial cable, the signal wire take the form of centre conductor (usually copper) running coaxially inside a braided outer circular conductor carrying ground reference The space between the two conductors is normally filled with a dielectric insulating material.
Due to its geometry, the centre conductor is effectively shielded from external interference signals and minimal losses occur due to electromagnetic radiation and the skin effect. Coaxial cable can be used with a number of different signal types, but typically 10 or even 20 Mbps over several hundred metres is perfectly feasible.

2.4 Optical fibre

Optical fibre cable differs from all the previous transmission media because it carries the transmitted information in the form of a fluctuating beam of light in a glass fibre, rather than an electrical signal in a piece of wire. Light waves have a much wider bandwidth than electrical waves and hence optical fibre cable can be used for transmitting very high bit rates, in the order of hundreds of megabits per second. Furthermore, the use of a light beam makes optical fibre cable immune to the effects caused by spurious electromagnetic interference signals and crosstalk effects.
An optical fibre cable consists of just a single glass fibre, for each signal to be transmitted, contained within a protective cover, which also shields the fibre from any external light sources. The light beams which carry the information are confined in the waveguides formed by the fibre due to the phenomenon of total internal reflection at the interface between two media of different refractive index.
The light signal is generated by a special optical transmitter unit, which performs the conversion from normal electrical signals as used in a DTE. Similarly, at the other

end of the line, a special optical receiver module is used to perform the reverse function. Typically, the transmitter uses a light-emitting diode (LED) to perform the conversion operation and the receiver a light-sensitive photodiode or photo transistor. As the fibre is coated with a reflective film, the majority of the light produced by the LED remains inside the fibre and hence the attenuation effect is low. In general, optical fibre cable systems are more expensive than coaxial cable and, because of their construction, they are mechanically weaker, which makes them more difficult to install. It is also more difficult to join (or split) fibre cable due to the high coupling losses that occur, and hence they are only considered when either very high bit rates are required or enhanced levels of noise immunity are needed.

2.5 Microwaves

All the transmission media mentioned so far use a physical line to carry the transmitted information. However, data can also be transmitted using electromagnetic (radio) waves through free space.
Microwave links are widely used to provide communication circuits when it is impractical or too expensive to install physical transmission media(e.g. across a river or perhaps a busy highway)
Such links either implemented as terrestrial line-of-sight connections, or relayed by a geostationary satellite, may provide very large bandwidth circuits.

3. TELEPHONE NETWORK

The public telephone network provides full duplex two wire channels between subscribers with a passband suitable to carry the speech signal.
The network is of the switched circuit type with a hierarchical organization (Figure 1.5) of switching centres. Each subscriber is connected to a local centre (LC) by means of a subscriber loop which has a range of a few kilometres and uses a twisted pair of conductors. The local centre enables subscribers to be interconnected on the local network (usually referred to as **distribution network**) by establishing a temporary connection between them for the duration of their communication.

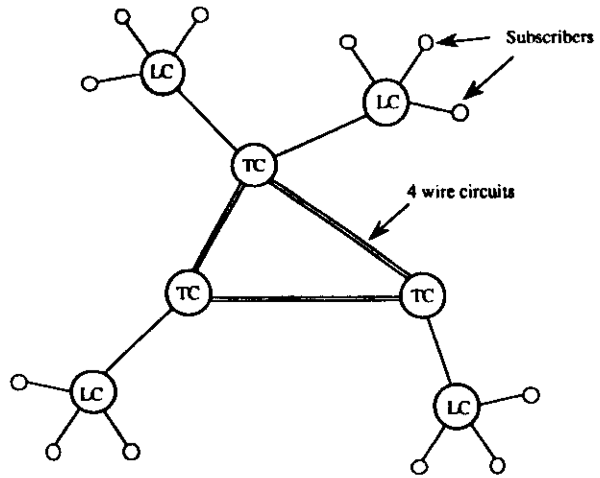

Fig. 1.5 Telephone network structure

Links between subscribers connected to different local centres are made through transit centres (TC) or toll offices which are themselves interconnected by long distance trunk lines.
To compensate for the large attenuation which occurs on long distance lines, it is necessary to use amplifiers which can be realized easily only with four wire circuits. Long distance lines are, therefore, in principle always of the four wire type and four-wire/two-wire conversion is generally realized at the transit centres using a differential transformer.
In addition all transmission media suitable for long-distance interconnections (coaxial cables, microwave, optical fibres) are very expensive, and so it is quite reasonable to exploit their bandwith as much as possible; vis-a-vis the typical bandwith required to transmit intelligible speech does not exceed 4 KHz. For these reasons long distance circuits are usually realized by multiplexing of a large number of telephone channels onto a single line

3.1 Multiplexing techniques

In contrast to the low bandwidth necessary for a speech connection through an analogue-switched telephone network, the usable bandwidth with a coaxial cable can be as high as 350 MHz . This potentially high bandwidth can be utilized in two different ways:

(1) **Baseband mode**, in which all the available bandwidth is used to derive a single high bit rate transmission channel.

(2) **Broadband mode**, in which the available bandwidth is divided to derive a number of lower bandwidth subchannels (and hence transmission paths) on a single cable.

Each mode of working will now be considered separately.

3.1.1 Baseband

The technique used to share the available capacity of a baseband transmission channel is time-division multiplexing or TDM. Two types of TDM are used:

(1) Synchronous (or fixed cycle): Each user has access to the channel at precisely defined (synchronized) time intervals.

(2) Asynchronous (or on demand): Users have random access to the channel and, once a user has acquired access, it is the sole user of the channel for the duration of its transmission.

The two alternative forms of TDM are shown diagrammatically in Figure 1.6.

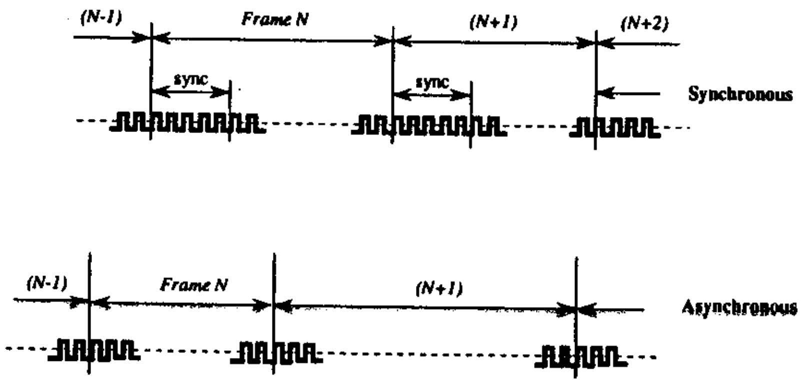

Fig. 1.6 Synchronous and Asynchronous TDM

To ensure that all systems connected to the (shared) cable transmit data at their allotted time, a special bit pattern, known as the synchronizing (or simply sync) pattern, is transmitted at the beginning of each frame. From this, each system can determine both the start of each frame and the position of the frame (frame number) in a complete cycle of frames. With asynchronous TDM, a mechanism must be employed to ensure that each system can gain access to the channel in a fair way, since each system has random access to the channel.

The use of TDM techniques implies handling of digital signals, and is becoming of great interest even for the conventional telephone network due to the widespread penetration of PCM techniques. As a matter of fact, being possible to convert, without a noticeable loss of quality, analogue telephone signals (4 kHz of band) into a digital signal at 64 kbit/s (with sampling at 8 kHz and coding using 8 bits), it is possible to use digital switching techniques with great advantages in terms of costs and performance with respect to electromechanical switching.
In Europe, PCM transmission on telephone lines has been standardized by CEPT (European Committee of Posts and Telecommunications) which has defined a 30 telephone channel system with the signal sampling at 8 kHz, coding of the samples using 8 bits and transmission on the line at a rate of 2.048 Mbit/s. Each of the 30 channels is sampled every 125 µs and each sample is coded with 8 bits. Transmission of the samples corresponding to different channels is interleaved in time within a frame of 256 bits; this is transmitted in 125 µs and contains 32 basic intervals of 8 bits which correspond to 30 telephone channels plus one channel for synchronization and supervision and one reserved for signalling.
Primary multiplexing of 30 channels in the base of a hierarchical system of time division multiplexed PCM which also contains secondary multiplexing of 120 channels operating at 8.448 Mbit/s and tertiary multiplexing of 1920 channels which combines 16 secondary multiplexings and operates at 140 Mbit/s.

3.1.2 Broadband

Using broadband approach multiple independent transmission channels are derived from a single distribution cable by using a technique known as frequency-division multiplexing or FDM. This technique provides a multiplicity of transmission channels on the same physical medium, by allocating the information channel energy around different carrier frequencies. The selected (carrier) frequency for the transmit (forward) direction is modulated with the signal to be transmitted and the selected frequency for the receive (reverse) direction is demodulated to obtain the received signal.

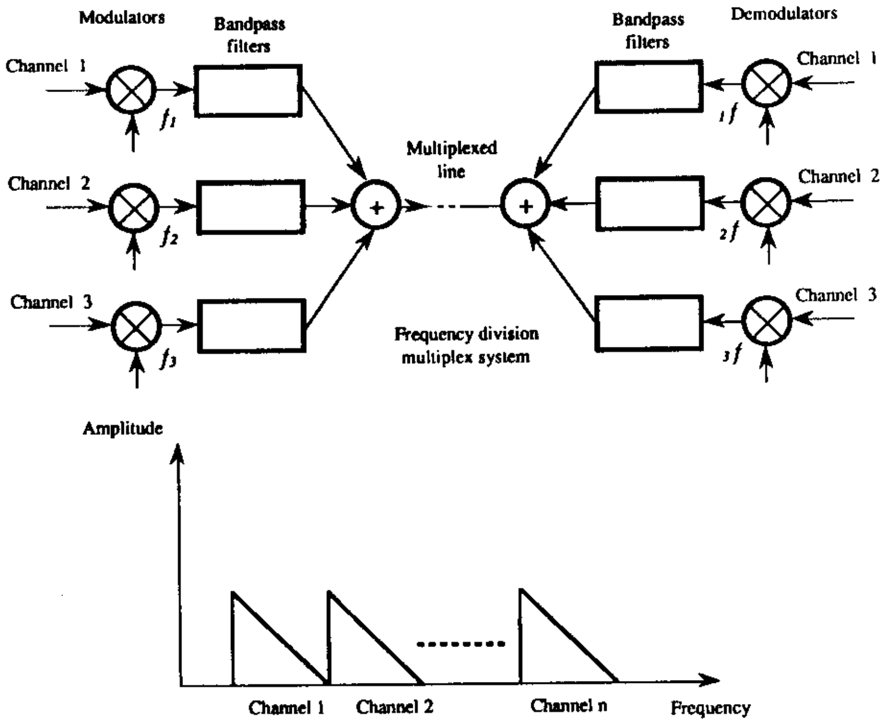

Fig. 1.7 FDM operation

The principles of FDM multiplexing and the subunits within the relevant modem are summarized in Figure 1.7. The filters shown in the figure allow just the signals associated with the assigned frequency band to be transmitted (on output) or processed (on input).
Usually frequency multiplexed systems for voice-band signals use the principle of single sideband (nominal band of 4 kHz) modulation with a hierarchical organization consisting of **primary groups** at the base which combine 12 telephone channels in the 60-108 kHz band. When the number of channels to be combined becomes larger, use is made of secondary groups (**Supergroups**) which combine five primary groups (60 channels) in the 312-552kHz band, tertiary groups (**Mastergroups**) which combine five secondary groups (300 channels) in the 812-2044 kHz band and quaternary groups (**Supermastergroups**) which combine three tertiary groups (900 channels) in the 8516-12,388 kHz band.

For some long distance links, coaxial cable transmission is replaced by radio transmission at 4 GHz (960 channels) and 6 GHz (1800 channels) using a series of line of sight station.

3.2 Telephone lines

Whatever the multiplexing technique adopted for long distance transmission lines, using the telephone network, it is possible to obtain switched two wire lines with a pass band extending approximately **from 300 to 3400 Hz**.
The quality of these lines is not constant, since the channels used on the public network differ from one call to another. For high speed data transmission, the network operators make available unswitched **leased lines**, which permanently connect two or more subscribers, available to users. The line is called point-to-point in the first case and multipoint in the second case (Figure 1.8). Special point-to-point lines can be two or four wire but special multipoint lines are in principle always of the four wire type.

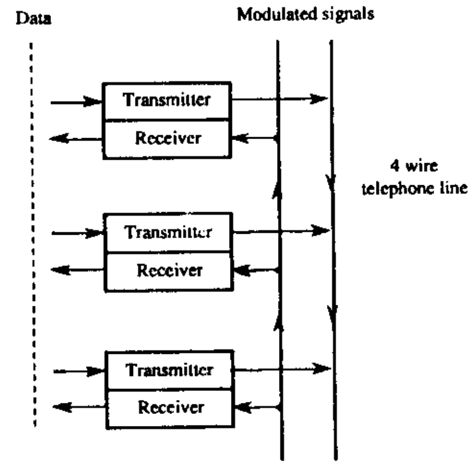

Fig. 1.8 Multipoint line

Special lines can be provided by using the network in the form of normal quality lines or, by selection of lines, equalization and amplification in the form of superior quality lines in such a way as to best satisfy the requirements of high speed data transmission. The quality of these lines is generally specified by masks fixing the maximum amplitude and group delay distortion as a function of frequency. Figure 1.9 gives the figures of the superior quality line most widely used in Europe, the CCITT M-1020 line.
For transmission at rates exceeding 9600 baud, it is necessary to use a bandwidth greater than that of the

telephone channel Access to a primary group makes a bandwidth of 48 kHz available which enables rates of 48 kbit/s or 72 kbit/s to be achieved. Similarly, direct access to a secondary group enables the rate to be increased to 250 kbit/s.

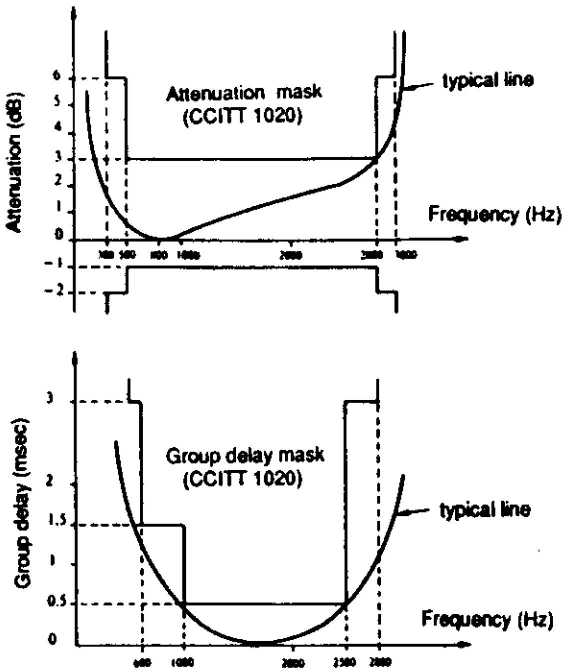

Fig. 1.9 CCITT M.1020 masks

3.3 Line impairements

To a first approximation, a transmission line can be considered as a linear filter whose complex gain G(f) is of the form

$$G(f)=|G(f)|e^{-jP(f)}$$

The phase shift as a function of frequency is generally characterized by the group delay $t_g(f)$

$$t_g(f)=\frac{1}{2\pi}*\frac{d\,P(f)}{df}$$

As from Signal theory a linear time-invariant system will pass undistorted signals if, and only if, both amplitude and group delay characteristics are a constant function in the frequency range of the signal spectrum.
A short look at Figure 1.9 clearly demonstrates how far telephone line characteristics are from the optimal behaviour. Transmitted data signals, are therefore heavily affected by a disturb, usually referred to as **symbol interference**.
Long distance telephone channels (see model in Figure 1.10) may use carrier systems with frequency division multiplexing using single sideband modulation. Frequency translation at the two extremities is achieved using two different carriers which can produce a **frequency offset** of up to 10 Hz between the spectra of the transmitted and received signals.
Transmission of the signal by carrier systems also leads to a **phase jitter** of the received signal which is due to the instability of the translation oscillators and insufficient filtering of the equipment power supplies. These impairments (frequency offset and phase jitter) have no practical effect on the spoken word but are very troublesome for data transmission at high speed, (above 2400 bit/s) on conventional telephone lines, and must be systematically compensated in high performance modems.using appropriate demodulation algorithm.

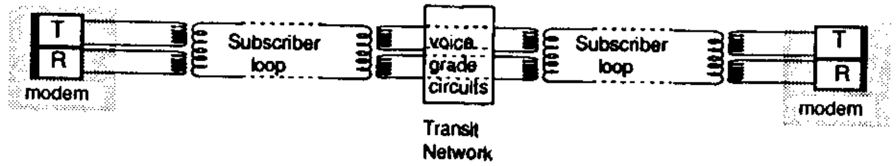

Fig. 1.10 Reference model for telephone channel

Communication can be be disturbed by interfering signals of different origins; these include: **white noise** due to thermal agitation in the system components, impulse noise due principally to electromechanical switching components, crosstalk generated by other channels and echoes caused by four-wire/two-wire conversion.
White noise generally is of low power and it is relatively easy to obtain telephone lines on which the signals to noise ratio reaches 25 to 30 dB, which is sufficient to ensure transmission in good conditions with most modems.
Impulse noise is much more troublesome since it can reach an amplitude equal to or greater than that of the signal during a period of the order of 1 to 10 ms which causes error bursts on the transmitted data. Impulse noise is the principal cause of errors on conventional telephone lines, particularly when these use the switched network.

Crosstalk introduced by spurious coupling from adjacent lines is generally negligible with respect to the impulse noise.
Echo noise is caused by a reflection of the signal due to impedence mismatching and imperfect balancing of hybrids which are differential transformers located at four wire/two-wire interfaces located just at modem output (near-end echo) and between local two wire telephone lines and long distance four wire telephone channels (far-end echo). The echo produces a return to the receiver end of an attenuated and delayed replica of the signal which it has sent. To reduce the effect of far-end echo on telephony, the telecommunication services install **echo suppressors** which make the line operate in half duplex by introducing high attenuation in the direction of transmission which corresponds to the weaker signal. Echo suppressors have a turnaround time of the direction of transmission of the order of 100 ms. When data transmission is performed in full duplex on a two wire line, the echo suppressors must be neutralized by means of a 2100 Hz tone sent on the line for more than 400 ms. For transmission on special four wire lines, it is necessary to ensure that the echo suppressors are disconnected from the line at the time of installation.
Data transmission in full-duplex at high bit rates, makes use of **echo canceller** circuits, which is an adaptive Digital Signal Processing device, able to cancel the echo (both near and far) by means of stochastic evaluation of the correlation in the received signal with respect to the transmitted one.

4. MODEM

When data are transmitted using the transmission lines from an existing PSTN, it is first necessary to convert the electrical signals output by the source DTE into a form that is acceptable to the PSTN. The latter was of course designed for speech communications which are assumed to be made up of frequencies in the range 300 to 3400 Hz.
Thus, the PSTN is said to have a bandwidth of 3.1 kHz from 300 to 3400 Hz
This means that a telephone line will not pass very low frequency signals which may arise, for example, if the data stream to be transmitted is made up of a continuous string of binary 1s or 0s. For this reason, it is not possible to simply apply two voltage levels to the telephone line, since zero output would be obtained for both levels if the binary data stream was all 1s or all 0s. Instead, it is necessary to first convert the binary data into a form compatible with a speech signal at the sending end of the line and to re-convert this signal back into its binary form at the receiver. The circuit that performs the first operation is known as a modulator and the circuit performing the reverse function a demodulator. Since each side of a data link must normally both send and receive data, the combined device is known as a modem.

4.1 Modulation

There are three basic types of modulation that may be employed for the conversion of a binary signal into a form suitable for transmission on a PSTN. These are amplitude modulation, frequency modulation and phase modulation.
With **amplitude modulation (AM)**, the level or amplitude of a single frequency audio tone is is switched between two levels at a rate determined by the transmitted binary data signal. The single frequency audio tone is known as the carrier frequency and is selected to be within the acceptable range of frequencies for use in the PSTN. This type of modulation, although the simplest, is prone to the effect of varying signal attenuation caused, for example, by varying propagation conditions as different routes through the PSTN are selected. In its basic form, therefore, this type of modulation is not often used.
With **frequency modulation (FM)**, the frequency of a fixed amplitude carrier signal is changed according to the binary stream to be transmitted. Since there are only two frequencies (audio tones) required for binary data, this type of modulation is also known as **frequency-shift keying (FSK)**. This type of modulation is the method most frequently used with lower bit rate modems (300 to 1200 baud) designed to operate with switched connections across the PSTN, and the demodulation circuitry needed is relatively simple

With **phase modulation (PM)**, the frequency and amplitude of the carrier signal are kept constant while the carrier is shifted in phase as each bit in the data stream is transmitted; this is known as **phase-shift keying (PSK)**.
The most frequently used form of PSK employs shifts in phase at each bit transition determined by the state of the next bit to be transmitted relative to the current bit. Thus, a phase shift of 90° relative to the current signal indicates a binary 0 is being transmitted and a phase shift of 270° a binary 1. In this way, the demodulation circuitry need only determine the **differential** magnitude **(DPSK)** of each phase shift rather than the absolute value.

To derive higher bit rate channels from a normal telephone line, it is necessary to use more sophisticated modulation techniques. In the examples discussed so far, the bit rate was the same as the signalling rate; that is, the number of times per second the amplitude, frequency or phase of the transmitted signal changes per second. The signalling rate is measured in baud and hence in the examples the bit rate has been equal to the baud rate. It is possible, however, when transmitting a signal across a PSTN line to utilize more than two different values, four or eight not being uncommon. This means that each signal element may contain two (four values) or three (eight values) bits of encoded

information. The resulting bit rate is then two or three times the band rate. An example is provided by V.26 modem where four different phase changes (0, 90, 180 and 270) may be employed instead of just two. Hence, the phase change of each signal can convey two bits.
The different modulation techniques may be combined to produce, for example, amplitude modulated-phase shift keying, generally referred to as **Quadrature Amplitude Modulation (QAM)**; this is the case of 9600 bit/s V.29 modem.
In addition special encoding of the transmitted bits can be performed, to force correlation between consecutive group of bits, so allowing error recovery techniques at receiving side; this is the case of **Trellis encoding** used in 9600 bit/s full-duplex V.32 modem.

4.2 The principle of operation of modems

Conceptually, the transmitter of a modem intended for operation on a telephone line can be considered to consist of the elements represented in Figure 1.11. Firstly, the code of the data to be transmitted may be changed, for example Gray code or differential encoding may be used, in order to facilitate transmission or minimize the effect of errors. The bit sequence to be transmitted then passes to a scrambler whose purpose is to make the probability of occurrence of 1 and 0 bits approximately equal regardless of the data sequence to be transmitted. The scrambler enables the power on the line to be held approximately constant and also ensures a sufficient number of 1-0 transitions to permit ready recovery of the clock at the receiver. The scrambler is followed by a modulator which translates the signal in such a way that its spectrum is centred on the passband of the line. The modulator is followed by a filter which limits the spectrum of the signal to the passband of the line and an amplifier which adjusts the level of the transmitted signal to the maximum allowed on the line. This level must not exceed a mean of - 13 dBm0 (50µW) according to CCITT recommendation V2, in order to avoid transmission equipment overload and to limit cross-talk with adjacent lines.
When a DTE wishes to transmit data, it signals this to the modem by sending it a request to send command. On receipt of this command, the modem starts to send an initialization signal whose purpose is to allow the receiver to prepare itself to receive the data. After a specified time called the turnaround time, the modem signals to the DTE that it is ready for sending the data. The DTE then sends the data to the modem which transmits it immediately.

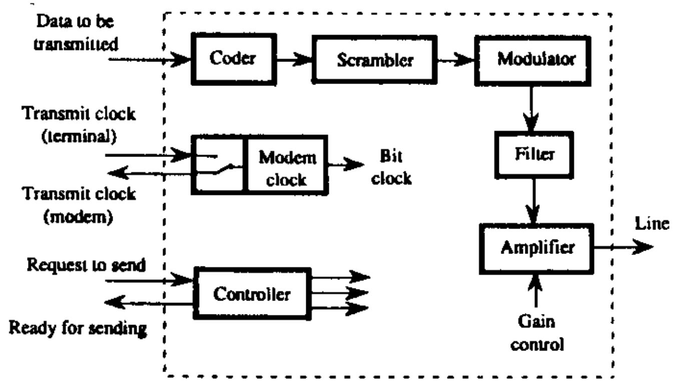

Fig. 1.11 Typical modem transmitter

The value of the turnaround time is a very important characteristic of modems, since it determines the dead time between transmission in the two opposite directions on a line used in the two-way alternate mode. The turnaround time is set to allow the receiver sufficient time to detect the presence of the signal, recover the carrier and clock, control the gain of the amplifier and perhaps adjust the equalizer. In general, the turnaround time is longer for high speed modems than for low speed modems and can vary from 10 ms for the simplest modems to several hundreds of milliseconds for very sophisticated modems which perform automatic equalization. The transmitter bit clock can be either derived from a clock signal sent to the modem by the DTE or, most often, produced by the modem itself. In the latter case, the clock signal is sent by the modem to the DTE so that it can transmit data at the rate determined by the modem.

The modem receiver is usually arranged as depicted in Figure 1.12. The line signal first passes through an amplifier whose usually including automatic gain control. The output signal from the amplifier is then filtered to eliminate out-of-band noise and then demodulated using a carrier recovered from the signal. The carrier recovery system also gives an indication of carrier detection which is sent to the receiver DTE. The demodulated signal is sent to an equalizer which compensates amplitude and group delay distortion of the line. This equalizer can be a fixed **compromise equalizer** for low speed modems, or an automatic **adaptive equalizer** for high speed modems. The bit clock is recovered from the equalizer output signal and serves to detect the data which is then processed by a descrambler and a decoder to reproduce a bit sequence identical to the transmitted sequence.

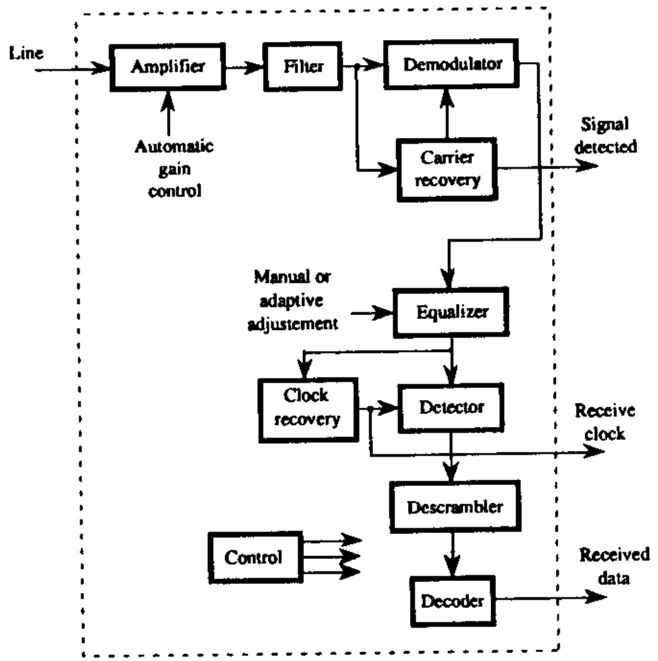

Fig. 1.12 Typical modem receiver

Arrival of the initialization signal is detected by the received line signal detector which initiates some preliminary operations within the receiver, such as adjustment of the amplifier gain, training of the equalizer and recovery of the carrier and clock. At the end of this initialization period, the modem sends an indication of signal detection, requesting DTE to accept received data.

4.4 Standardized modems

Two well established modem standard are currently used, an international CCITT (Consultative Committee of the International Telegraph and Telephone) standard, and a U.S. specific standard (Bell); full compatibility is not granted. In the following just CCITT standard will be considered.
The characteristics of the most frequently used modems are summarised in Figure 1.13, and briefly discussed in the following.

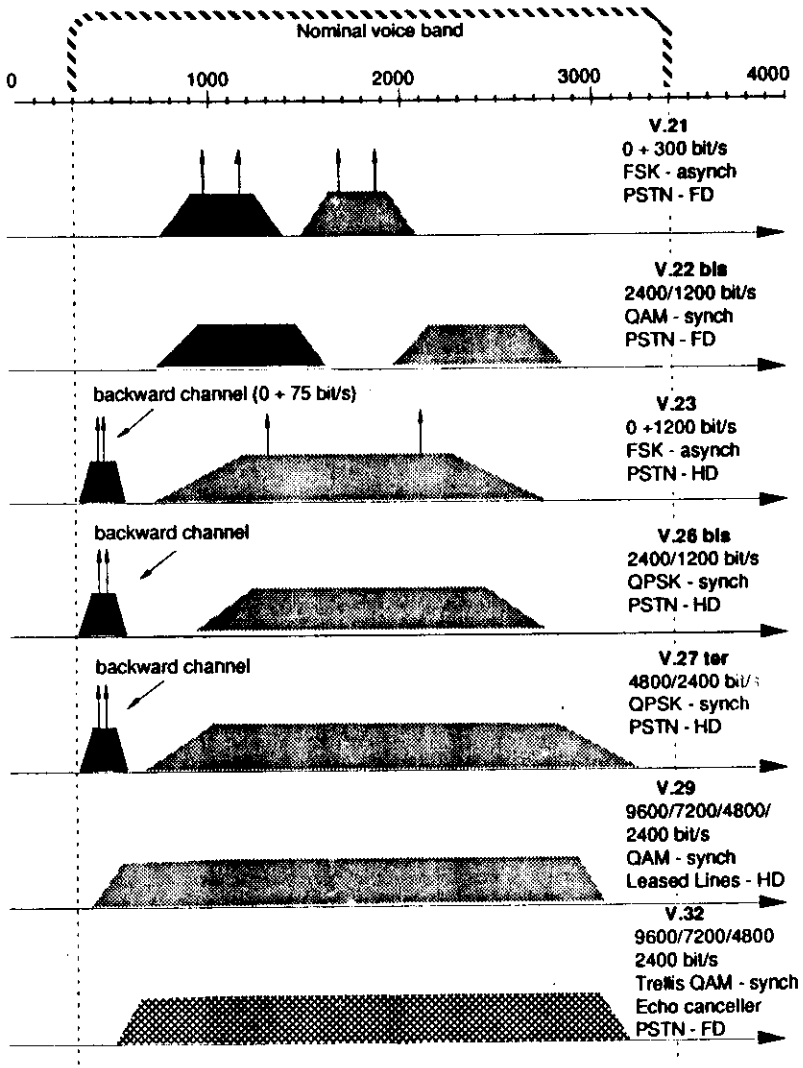

Fig. 1.13 Modem overview

V.21

The modem defined by Recommendation V.21 operates asynchronous transmission with frequency-shift keying modulation, on a two-wire line; the bit rate can reach a maximum of 300 bit/s per channel. The modem operates in full-duplex, with frequency division and can be used equally well on the switched network and special two-wire lines.

V.22 bis

It operates in full-duplex with combined phase and amplitude modulation on 16 levels over special two-wire lines or on the switched network. The two channels in opposite directions are obtained by frequency division on two-wire lines, with a 1200 Hz carrier for the lower channel and a 2400 Hz carrier for the upper channel.
This modem operates at full speed of 2400 bit/s and has a fallback speed of 1200 bit/s; though has been introduced recently it is rapidly becoming a de-facto standard for general purpose PSTN connection.

V.23

The V.23 modem operates with frequency modulation up to 1200 bit/s. It is provided essentially for asynchronous transmission in half-duplex on two-wire lines and on the switched network.The V.23 modem can be equipped as an option with as asynchronous 75 baud backward channel which operates with frequency shift keying with a carrier at 450 Hz for binary 0 and a carrier at 390 Hz for binary 1.
The main applicability of this modem is for Videotex Services, with transmission at 75 baud on the return channel in the terminal-Service centre direction and transmission at 1200 baud on the main channel in the opposite direction.

For operation in half-duplex on a two-wire line which is liable to produce echoes, the modem must be protected against echoes by holding the data received signal at 1 and the signal detector at OFF for 150 ms after the OFF condition of the request to send signal.

V.26bis

The modem specified by Recommendation V.26 bis is intended for synchronous transmission and operates at 2400 bit/s with four-phase modulation in half-duplex on special two-wire lines or on the switched network. The modem conforming to Recommendation V.26bis has a fall back rate of 1200 bit/s which permits it to operate with larger tolerances and hence to continue to provide a service at a lower speed when the line quality is poor.
V. 26bis modem can be equipped with a backward channel at 75 baud conforming to Recommendation V.23.

V.27 ter

Recommendation V.27ter defines a modem with eight-phase modulation operating in synchronous mode on normal quality and on the switched network. The modem is equipped with a scrambler and has a fallback speed of 2400 bit/s which

allows to operate with larger tolerances. The modem can be equipped with a backward channel at 75 baud conforming to Recommendation V.23.
The turnaround times are, including protection against echo suppressors, can be in the order of 1 sec.

V.29

Recommendation V.29 defines a modem with combined phase and amplitude modulation for operation in synchronous mode at 9600 bit/s on special lines of the superior M1020 quality with automatic equalization. The modem is equipped with a scrambler and has fallback speeds of 2400, 4800 and 7200 bit/s which permit operation with larger tolerances. The V.29 modem does not have an option 75 baud return channel.
The modem-DTE junction is of the V.24-V.28 type. The turnaround time is 253 ms.
The 9600 bit/s channel can be divided into sub-channels with rates of 7200, 4800 and 2400 bit/s by time division multiplexing.

V.32

The V.32 modem is intended for operation in full-duplex on the switched network at 9600 bit/s by using a complex echo compensation technique. Trellis encoding of bits is performed at transmitter side, so leading to a forward error correction approach to be exploited at receiving side for reduce the overall bit error rate.
This modem is at present quite expensive due to the complex algorithms required; however taking into account next advances in VLSI technology, it has to be regarded as a candidate de-facto standard for the next generation of PSTN modems.

Modem for primary groups

The primary groups of long-haul transmission system using frequency division multiplexing offer a bandwidth of 48 k Hz which is around 15 time wider than that of a single telephone line. It is, therefore, easy to design modems for these channel whose bit rates are much higher than those which can be achieved on telephone lines.

Recommendation V.35 and V.36 define modems using single sideband modulation for synchronous transmission at 48 kbit/s on primary groups. Recommendation V.36 also provides for transmission at rates of 56, 64 and 72 kbit/s. The rate of 64 kbit/s is intended to permit extension of analogue channels to PCM channels standardized at 64 kbit/s. Primary group modems are generally located in telephone exchanges, in close proximity to the carrier systems. Connection with the subscriber is then realized with a baseband channel. The link can include a telephone service channel as an option.
The modem-DTE junction is of the V.36 type with a V.10-V.11 electrical interface.

Recommendation V.37 defines a synchronous 96, 112, 128 and 144 kbit/s (Class IV Partial Response) modem for transmission on the primary group. The link can include a telephone service channel as an option and the data circuit can be split int sub-channels at 48, 56, 64 and 72 k bit/s by time division multiplexing. The modem DTE junction is of the V.37 type with a V.10-V.11 electrical interface.

5. PHYSICAL LAYER INTERFACE STANDARDS

Again two set of standards for interconnecting pieces of equipment are available: the international CCITT and U.S. specific emitted from the Electrical Industries Association (EIA).
Although the standards defined by both bodies use slightly different terminology, the basic signals and their meaning are the same. In the following the international CCITT standard will be considered in more details.

5.1 Mechanical standards

The mechanical characteristics of the interface are defined by ISO standards which specify the connector used for the different types of modem
The most widely used connector is specified by the ISO 2110 standard and contains 25 pins. This connector is used for low and medium speed modems up to 20 kbit/s which are used with the CCITT V.24/V.28 or EIA RS-232-C standards.
Another common standard is specified in ISO 4903 and fixes a 15 pin connector.

5.2 Electrical Interfaces

It has already been indicated that the most frequently used electrical interface is that specified by CCITT Recommendation **V.28**. This is used particularly for low and medium speed modems up to 20 kbit/s. Transmission is over a single wire with return by a common wire (Figure 1.14). The generator must provide an open circuit voltage of amplitude less than 25 volt whose sign is positive for transmission of a logical 0 and negative for transmission of a logical 1.

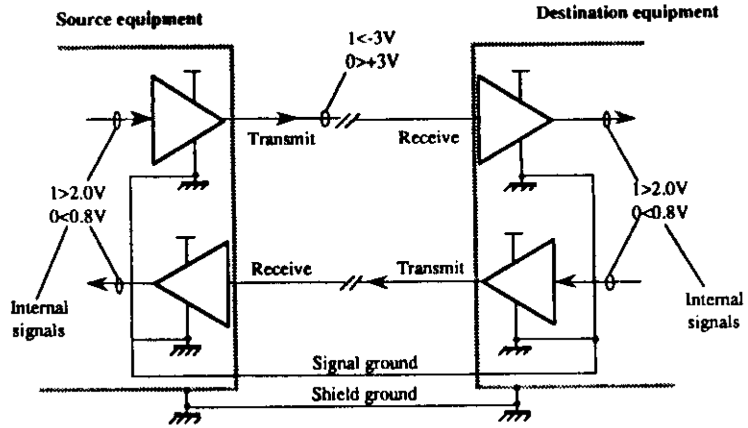

Fig. 1.14 Unbalanced electrical interface

V.11 interface circuits are arranged to be compatible with integrated circuit implementation and balanced transmission on two wire lines (Figure 1.15); this allows reduction in the cost of the interface circuits, an increased bit rate and an increase in the range of the interface cable.The V.11 link can be operated in multipoint, with a single active transmitter and the other transmitters in a high impedance state.

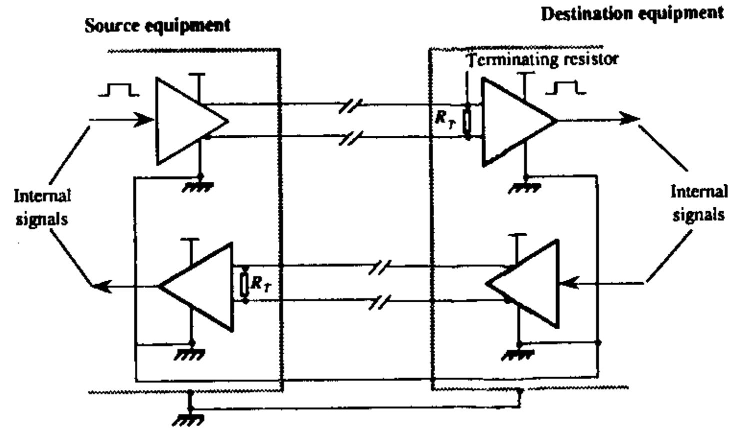

Fig. 1.15 Balanced electrical interface

To facilitate the progressive change from the V.28 interface to the V.11 interface, the CCITT has specified, in Recommendation **V.10**, electrical circuits which are compatible with both V.28 and V.11 circuits. Transmission is of the unbalanced single wire type with return by a common wire, but the receiver is of the balanced type and identical to the receiver of Recommendation V.11.

5.3 Functional interfaces

V.24 (EIA RS232C)

The various wires which are used by the V.24 interface at the DTE or modem level are represented by a three-digit circuit number whose first digit is 1 (**series 100** circuits). By convention, the active state of the control wires corresponds to a 0 on the data wires. The principal circuits of the V.24 junction are represented in Figure 1.16 and the functions which they fulfil are as follows:

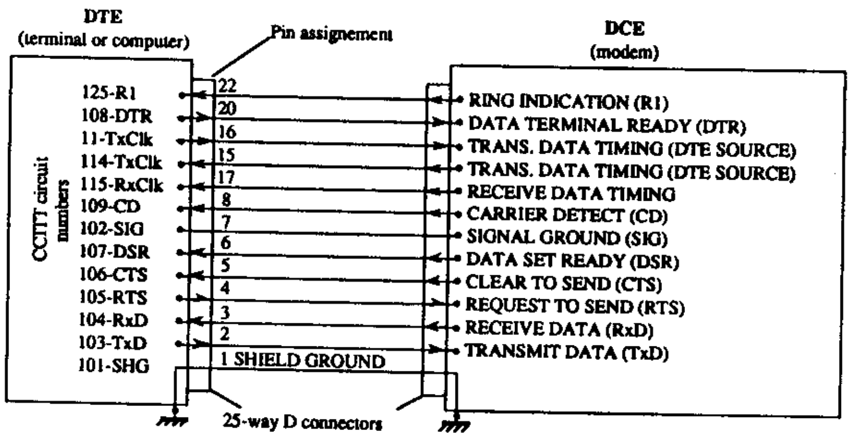

Fig. 1.16 V.24 Interface

113-114-115. Circuit 115 provides the DTE with the receiver clock from the modem. The transmitter clock can be produced either by the modem and fed to the DTE via circuit 114 or by the DTE and fed to the modem via circuit 113

103-104. Circuit 103 feeds data form the DTE to the modem transmitter. The data received by the modem is sent to the DTE through circuit 104

105-106. Circuit 105 is set by the DTE to indicate a request for data transmission to the modem. After a certain delay (turnaround time), the modem signals to the DTE that it is ready to transmit by activating circuit 106

109. Circuit 109 indicates, according to its value, the presence or absence of a carrier signal on the line

118-119-120-121-122. These circuits play exactly the same role as circuits 103-104-105-106 and 109 for the backward channel which can be added as an option to some modems

V.25 Auto-Call Interface

CCITT recommendation V.25 covers "automatic calling and/or answering equipment on the general switched telephone network including disabling of echo suppressor on manually established calls".
V.25 specifies an auto-call unit (ACU) which enables a computer to dial telephone numbers over the public switched telephone network (PSTN) and establish dialled connections with remote computer/terminals.
An ACU is connected to a computer by two 25-pin connectors. One interface uses the CCITT 100-Series interchange circuits (V.24/V.28) connected via the ACU to a modem for data transmission; the other uses the CCITT 200-Series interchange circuits (V.25/V.28) for automatic calling.
The 200-Series (EIA RS-366A) interface is used only during call establishment and disconnection. A computer dials a telephone number by sending via the V.25 interface one dial digit at a time in 4-bit parallel form. After the last digit of the telephone number has been sent to the ACU, one more 4-bit code is sent to indicate end of number (EON). When a PSTN connection is established, then data transmission is performed via the 100-Series interface.
Auto-call units have been overtaken in technology by auto-dial modems which use only one V.24 interface for both auto-calling and data transmission, according to V.25bis Recommendation.

V.25bis Auto-Call Interface

The V.25 bis recommendation covers "automatic calling and/or answering equipment on the PSTN using the 100-Series interchange circuits". Most modem manufacturers supply modems containing integrated autocalling circuitry. A computer/terminal (DTE) connects to a modem (DCE) via a single V.24/V.28 (RS-232C) interface and can perform, through this interface, both autocall/answer and data transfer operations.
As part of the automatic calling procedure, instructions (called commands) are sent by the DTE to the modem and instructions or responses (collectively called indications) are sent by the modem to the DTE. Commands and indications can be sent/received using asynchronous or synchronous operation.

Hayes interface

The automatic calling procedure is relatively complex to implement by non-specialist. The aim of Hayes interface is to eliminate almost completely this problem by offering an

intelligent modem interface, driven by ASCII commands and answering with ASCII characters.
This leads to the provision of two distinct mode of operation of the modem. The first mode of operation is purely local and corresponds to the situation where the modem interprets commands sent by the DCE in sequence of asynchronous ASCII characters, and responds in the same way; the second mode of operation corresponds to the normal use of modems on an established circuit.
Hayes commands are leaded by the "AT" string and hence this interface is referred to as "AT interface".
Using this interface is quite simple, and therefore it is steadily becoming a de-facto standard.

V.35

V.35 specifies a synchronous interface for operation with an analogue wide-band modem at a transmission rate of 48 kbps. The differential signal used mean that each line requires a pair of wires; some of the control signals are the same as those used with the V.24 standard. (DSR and DTR lines); additional Test mode lines are included.

V.36 and V.37
Are intended for use with wider bandwidth (wide-band) circuits. The latter are normally leased from the PTT authority and, because they by-pass the normal switching circuits, provide a direct point-to-point circuit (link) between two sites. Typically, this can be operated at data rates of from 48 to 168 kbps.

The X.20 and X.21 procedures

The V.24-V.25 procedures were originally designed for interfacing with analogue telecommunication networks. These networks are being progressively replaced by digital networks which provide more extensive services than analogue networks. To satisfy these new services and to provide a DTE-DCE junction better suited to the use of integrated circuits, it has become necessary to develop new DTE-DCE interfaces. This work has resulted in CCITT Recommendation X.21 which defines the junction procedure for connecting synchronous equipment to a digital network, together with CCITT Recommendations X.24, X.26 and X.27 which specify the nature and role of the junction circuits and the electrical characteristics of the interface. The electrical characteristics defined by Recommendation X.26 and X.27 are exactly the same as those which correspond to Recommendations V.10 and V.11 respectively. In the future, all interface circuits ought to be of V.11 type and hence based on transmission of the two-wire balanced type.
The procedure which specifies the connection of asynchronous equipment to a public digital network is defined by Recommendation X.20 , and the various recommendations concerning maintenance are included in X.150.

Vis-a-vis the procedures defined by the CCITT series V where each command is represented by one line, X.20 and X.21 junctions are reduced to a minimum of balanced lines on which the commands are represented by characters transmitted serially by bit (Figures 1.17 and 1.18). The control characters are coded with International Alphabet No. 5 which is the same as the 8-bit ASCII code.

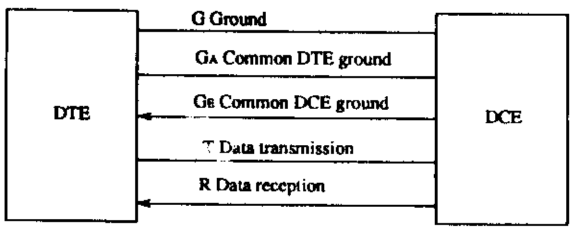

Fig. 1.17 X.20 Interface

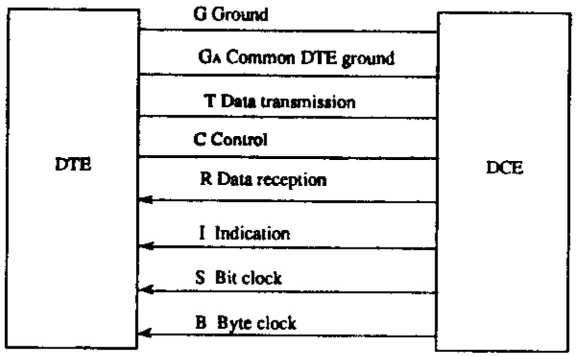

Fig. 1.18 X.21 Interface

For synchronous equipment, Recommendation X.21 specifies a junction which contains six two-wire lines and two ground lines .In this junction, the two line T and R are used for the transfer of data in the DTE-DCE and DCE-DTE directions respectively. The C and I lines serve to transmit control and supervisory information in the DTE-DCE and DCE-DTE directions respectively. Line S provides the bit clock signal from the DCE at the DTE. Finally, line B transmits a byte clock signal to the DTE which indicates the last bit of each ASCII character and enables character synchronization to be ensures. In practice, the byte clock signal is only an option since the sequence of control characters sent on

lines T and R is always preceded by at least two synchronizing characters SYN which enable the clock to be recovered for character synchronization without the help of circuit B.
The X.21 protocol can be considered to define both the function relating to the physical level of the ISO model, such as establishment of the circuit, and the functions of a higher level.

At a given time, the DTE-DCE junction can be in one of three possible phases which are the idle phase, the control phase and the data transfer phase. During the idle phase, the interface is inactive. The control phase corresponds to either establishment or release of communication. Finally, the useful part of the communication consist of the data transfer phase during which the stations exchange their messages. Recommendation X.20 and X.21 specify the various functions executed during the different phases together with the procedures used to switch from one phase to another. Recommendation X.20 and X.21 provide a complete list of the call progress signals used at the interface.

X.20 bis and X.21bis

The use of two different standards for the interface with public networks poses problems of migration, since equipment with V.24/V.25 interfaces will be used for a very long time and it will be necessary to be able to use these with new synchronous public networks. To facilitate this migration, the CCITT has provided an interim solution which enables equipment using V.25/V.25bis interfaces to be connected to synchronous public network. The corresponding procedures are defined in Recommendations X.21 bis and X.20 bis for equipment which operates in synchronous and asynchronous mode respectively.
Operation of X.21bis and X.20bis is based on the use of the circuits defined in V.24 (series 100); in particular:
- 105 Request to send
- 106 Clear to send
- 107 Data set ready
- 108 Data terminal ready
- 109 Carrier detect.

DATA LINKS

1. FUNCTIONS AND SERVICES

Data transmission channels, which correspond to the physical layer of the ISO network model, provide a means of transmitting bit serially. To cope with the error rate on the data circuit, which is often greater than that permissible in data processing, it is necessary to convert the physical channels into logical channels suitable for transmitting information reliably between data processing station. By consequence the higher levels can operate in a manner relatively independent of the characteristics of the transmission channel.This is basically the function of the link layer which correspond to layer 2 of ISO reference model.
Exchanges on a data link are defined by a set of rules which form the data link protocol or line procedure and specify the configuration of the link, the format and the sequence of operations The link layer represents the crucial means in a structured approach to the control of a data connection, which can be therefore divided in the following logical steps.
Circuit establishment phase. This phase realizes the physical connection of stations on the transmission channel and corresponds, for example, to a call on the switched network. The functions of establishing the circuit depend on the channel and therefore form part of the physical layer and not the link layer.

Link establishment phase. Once the stations are connected to the channel, they must exchange a number of control and supervisory messages before communicating the information frames. These messages permit stations to identify each other, to negotiate the conditions under which exchanges will be made and to ensure that the stations are ready to communicate.

Information transfer phase. This phase represents the core business during which the stations exchange messages of information and acknowledgement. At this stage, flow control must be provided to avoid the transmission rate exceeding the absorption capacity of the receiver and causing it to overload. In order to allow users complete freedom of the nature of the information exchanged, it is desirable that the link should be transparent to the code used during the transfer phase; that is, no constraint should be imposed on the nature of the information transmitted.

Link closure phase. This phase terminates the transfer of information by an exchange of supervisory and control messages which free the stations and cause the link to return to an idle or control state.

Circuit release phase. When the link between stations is established by a switched network, the end of communication must be marked by release of the circuit. The release phase is part of the physical layer.

The principal service is transferring data from the network layer on the source machine to the network layer on the destination machine. On the source machine there is an entity, call it a process, in the network layer that hands messages to the data link layer for transmission to the destination. The job of the data link layer is to transmit these messages to the destination machine, so they can be handed over to the network layer there, as shown in Figure 2.1(a). The actual transmission follows the physical path of Figure 2.1(b), but it is easier to think in terms of two data link layer processes communicating using **peer-to-peer** virtual path to exchange Link Protocol Data Units (L-PDU).

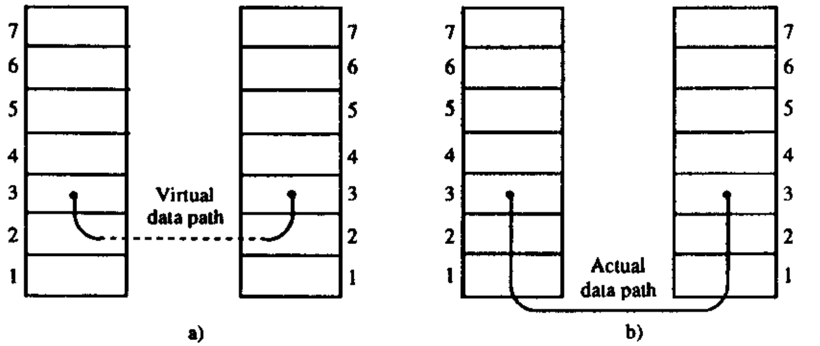

Fig. 2.1 Peer-to-peer communication

The communication between the network layer and the data link layer uses the standard OSI service primitives, which we will briefly review here.
The primitives are: **request, indication, response, and confirm.**
Request primitives are used by the network layer to ask the data link layer to do something, for example, establish or release a connection or send a frame.
Indication primitives are used to indicate to the network layer that an event has happened,. for example, another machine wishes to establish or release a connection, or a frame has arrived.
Response primitives are used on the receiving side by the network layer to reply to a previous indication.
Confirm primitives provide a way for the data link layer on the requesting side to learn whether or not the request was successfully carried out..

Figure 2.2 illustrates these four primitives in two different ways. In Figu~e 2.2(a), the layer structure is shown explicitly, as is the flow of data. In Figure 2.2(b) (from left to right) we have the data link service user on the sending side, than the data link service provider, and finally is the data link service user on the receiving side. Time runs downward, so that events higher up occur before events lower down.

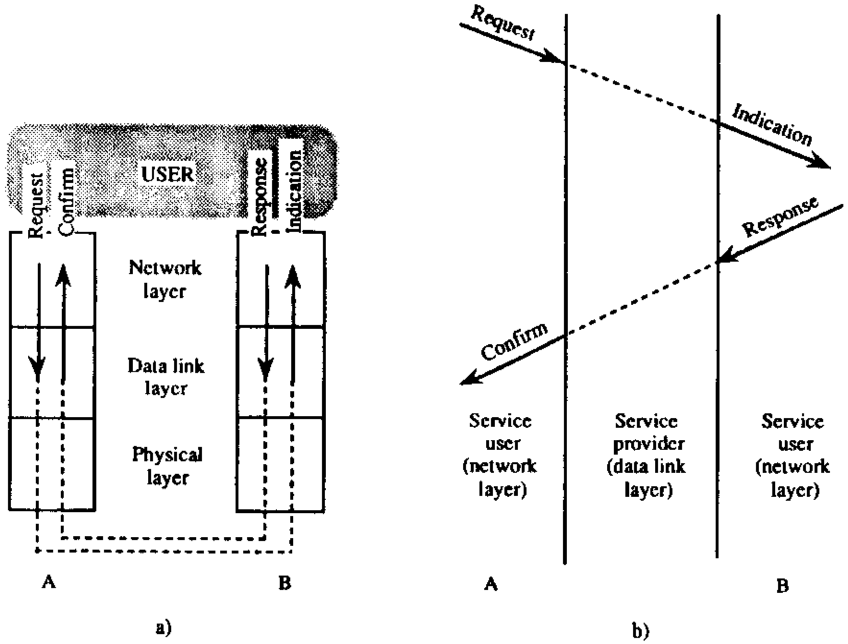

Fig. 2.2 OSI service primitives

A typical set of user services the link layer can provide to the network layer is shown in Figure 2.3, where both confirmed services (L.CONNECT and L.DISCONNECT) and uncorfirmed (L.DATA) are indicated.

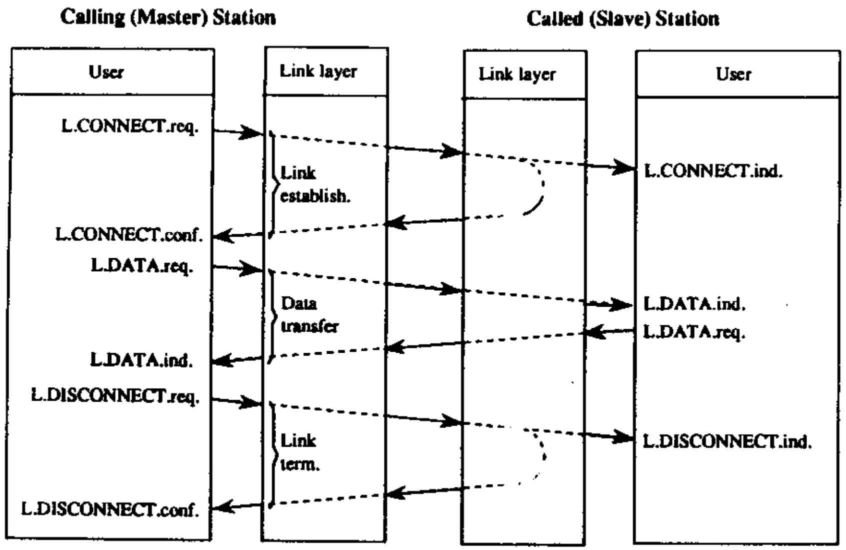

Fig. 2.3 Link layer primitives

2. ERROR DETECTION

In order to provide service to the network layer, the data link layer must use the service provided to it by physical layer. What the physical layer does is accept a raw bit stream and attempt to deliver it to the destination. This bit stream is not guarantied to be error free. The number of bits received may be less than, equal to, of more than the number of bits transmitted, any they have different values. It is up to the data link layer to detect, and if necessary, correct errors.

The usual approach is for the data link layer to break the bit stream up into discrete frames and compute the checksum for each frame. When a frame arrives at the destination, the checksum is re-computed. If the newly computed checksum is different from the one contained in the frame, the data link layer knows that an error has occurred and takes steps to deal with it (e.g., discarding the bad frame and sending back an error report).

There are two approaches available for achieving this:

Forward error control, in which each transmitted character or frame contains additional (redundant) information so that the receiver cannot only detect when errors are present but also infer from the received bit stream what it thinks the correct information should be.

Feedback error control, in which each character or frame includes only sufficient additional information to enable the receiver to detect when errors are present and then to employ a retransmission scheme to request that another, hopefully correct, copy of the erroneous information be sent.

The former strategy uses **error-correcting codes** and the latter uses **error-detecting codes**.

In practice, the number of additional bits required to achieve reliable forward error control increases rapidly as the number of information bits increases; hence, feedback error control is the predominant method used in the types of distributed system suitable for data communication.

2.1 Parity

The most common method employed for detecting error when the number of information bits is small and the probability of an error being present is low is by the use of a single additional parity bit per transmitted element. This method is particularly suitable with asynchronous transmission; the data bits in each character are inspected prior to transmission and the parity bit computed. This is then added so that the total number of binary 1s in the complete envelope is either odd or even according to whether odd or even parity is used. The receiver can then re-compute the parity for the received character and determine whether any transmission errors have occurred. The format of a transmitted envelope is as shown in Figure 2.4.

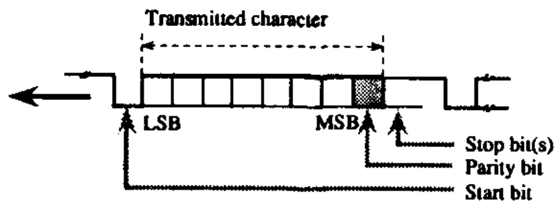

Fig. 2.4 Parity bit

The ability of a particular type of error-detection scheme to reveal errors depends strongly on the types of error that can arise on the data link used. For example, it can be readily deduced that the inclusion of a single parity bit with each character will only reliably safeguard against single bits being in error (odd errors), since if two bits are corrupted the transmitted parity bit will not indicate the error. In practice, however, if two or more (even) errors occur, and hence remain undetected by the line transmission control circuitry, the retransmission control schemes used with character-oriented transmission provide a means for detecting this.

2.2 Error correction by retransmission

Error correction codes does not generally reduce the error rate sufficiently to satisfy the requirements of data processing equipment. Also, these codes have a correction capability which is always less than their error detection capability. Because of this, the most frequently used method of correction in data transmission consists of coding the data and performing error detection at the destination station. The latter returns a short Positive Acknowledgement (ACK) to the transmitting station if it does not detect any error in the message and a Negative Acknowledgement (NAK) if it does (Figure 2.5).

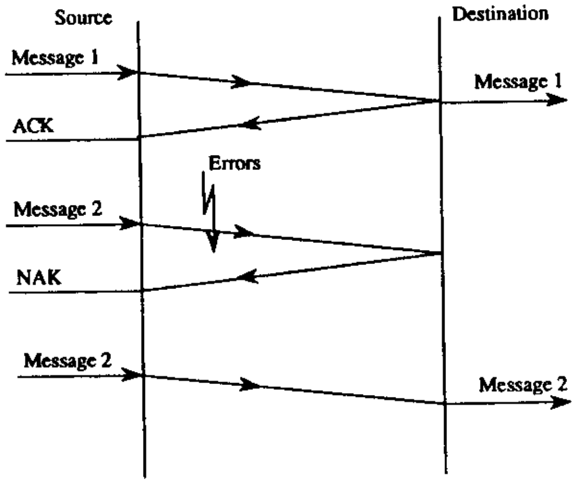

Fig. 2.5 Error correction by retransmission

When the transmitting station receives a positive acknowledgement, it can continue with transmission of the following message. Otherwise, if it receives a negative acknowledgement, it deduces that the message that it has just sent has been subject to errors and it repeats the transmission. At the receiver, the messages are clearly delivered to the user only if no error is detected.
The system of correction by retransmission is simple in principle but its realization poses problems; in particular it is necessary to avoid duplication of message when the acknowledgement messages themselves suffer errors or are lost.

2.2.1 Send and Wait

The simplest procedure **Send and Wait**, also referred to as **Stop and Go**.
With this procedure, station A sends a frame m_1, to which it appends a CRC error check word, to station B If station B does not detect any errors in frame m_1 , it considers it to be correct and transmits m_1 to the processing equipment to which it is connected. Also, B returns a positive acknowledgement frame (ACK) to A to indicate that transmission has been performed successfully. After reception of ACK, station A can send the following frame m_2.
In the case of transmission errors, station B returns a negative acknowledgement (NAK) to A which causes the latter to retransmit m_2.
The send and wait procedure has the merits of simplicity and robustness and so is widely used. However, it has the drawback of using the channel very inefficiently, particularly in the case of short frames exchanged on a circuit whose propagation time is high.
Data link performance can be considerably improved by introducing.more sophisticated retransmission strategies.

2.2.2 Sliding window protocol

In the previous strategy, data frames were transmitted in one direction only. In most practical situations, there is a need for transmitting data in both directions. One way of achieving full-duplex data transmission would be to have two separate communication channels, and use each one for simplex data traffic (in different directions). If this were done, we would have two separate physical circuits, each with a "forward" channel (for data) and a "reverse" channel (for acknowledgements). In both cases the bandwidth of the reverse channel would be almost entirely wasted. In effect, the user would be paying the cost of two circuits but only using the capacity of one.

A better idea is to use the same circuit for data in both directions. After all, in protocols 2 and 3 it was already being used to transmit frames both ways, and the reverse channel has the same capacity as the forward channel. In this model the data frames from A to B are intermixed with the acknowledgement frames from A to B. By looking at the kind field in the header of an incoming frame, the receiver can tell whether the frame is data or acknowledgement.

In all sliding window protocols, each outbound frame contains a sequence number, ranging from 0 up to some maximum. The maximum is usually 2^n-1 so the sequence number fits nicely in an n-bit field. The stop-and-wait sliding window protocol uses n=1, restricting the sequence numbers to 0 and 1, but more sophisticated versions can use arbitrary n.

The essence of all sliding window (see Figure 2.6) protocols is that at any instant of time, the sender **(P=primary)** maintains a list of consecutive sequence numbers corresponding to frames it is permitted to send. These frames are said to fall within the sending window. Similarly, the receiver **(S=secondary)** also maintains a receiving window corresponding to frames it is permitted to accept. The sender's window and the receiver's window need not have the same lower and upper limits, or even have the same size.

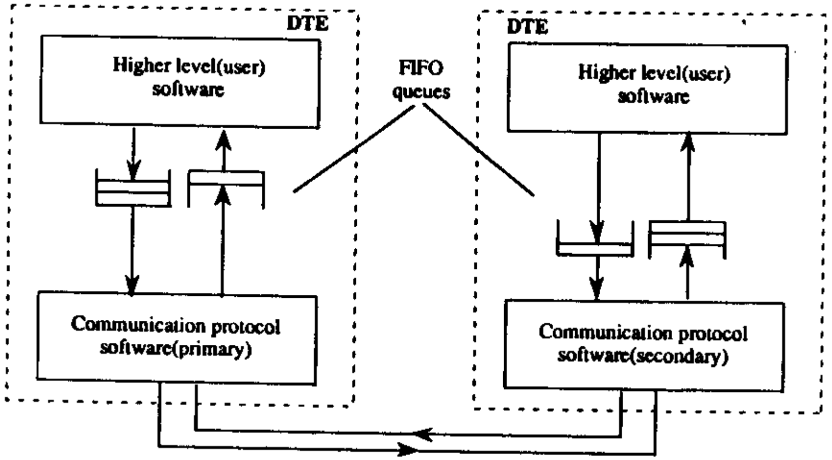

Fig. 2.6 Handling protocol queues

Although these protocols give the data link layer more freedom about the order in which it may send and receive frames, we have most emphatically not dropped the requirement that the protocol must deliver packets to the destination network layer in the same order that they were passed to the data link layer on the sending machine. Nor have we changed the requirement that the physical communication channel is "wire-like" (i.e., it must deliver frames in the order sent).

The sequence numbers within the sender's window represent frames sent but as yet not acknowledged. Whenever a new packet arrives from the network layer, it is given the next highest sequence number, and the upper edge of the window is advanced by one. When an acknowledgement comes in, the lower edge is advanced by one. In this way the window continuosly maintains a list of unacknowledged frames.

Since frames currently within the sender's window may ultimately be lost or damaged in transit, the sender must keep all these frames in its memory for possible retransmission. Thus if the maximum window size is n, the

sender needs n buffers to hold the unacknowledged frames. If the window ever grows to its maximum size, the sending data link layer must forcibly shut off the network layer until another buffer becomes free.

The receiving data link layer's window corresponds to the frames it may accept. Any frame falling outside the window is simply discarded. When a frame whose sequence number is equal to the lower edge of the window is received, it is passed to the network layer, an acknowledgement is generated, and the window is rotated by one. Unlike the sender's window, the receiver's window always remains at its initial size. Note that a window size of 1 means that the data link layer only accepts frames in order, but for larger windows this is not so. The network layer, in contrast, is always fed data in the proper order, regardless of the data link layer's window size.

When an error does occur, two alternative procedures may be followed:
(1) **Selective retransmission**: P detects the out-of-sequence ACK-frame and retransmits just the unacknowledged frame(s).
(2) **Go-back-N**: S detects the receipt of an out-of-sequence I-frame and requests P to retransmit all outstanding unacknowledged I-frames from the last correctly received, and therefore acknowledged, frame.

Although interleaving data and control frames on the same circuit is an improvement over having two separate physical circuits, yet another improvement is possible. When a data frame arrives, instead of immediately sending a separate control frame, the receiver restrains itself and waits until the network layer passes it the next packet. The acknowledgement is attached to the outgoing data frame (using the ack field in the frame header). In effect, the acknowledgement gets a free ride on the next outgoing data frame. The technique of temporarily delaying outgoing acknowledgements so that they can be hooked onto the next outgoing data frame is widely known as **piggybacking**.

2.3 Framing

Error protection implies organizing exchange of information in the form of blocks or frames whose length is defined in accordance with the error rate of the line and the type of application.
The oldest framing method uses a very simple approach by having each frame start with the ASCII character sequence STX and end with the sequence ETX. (STX is Start of TeXt, and ETX is End of TeXt.) In this way, if the destination ever loses track of the frame boundaries, all it has to do is look for STX or ETX characters to figure out where it is (Figure 2.7 a).

Character synchronization is ensured by sending two SYN characters at the start of each frame.

A serious problem occurs with this method when binary data, such as object programs or floating point numbers, are being transmitted. It may easily happen that the characters for STX or ETX occur in the data, which would interfere with the framing. One way to solve this problem is to have the sender's data link layer insert an ASCII DLE (DLE is Data Link Escape) character just before STX and ETX and each "accidental" DLE character in the data. The data link layer on the receiving end removes the DLE before the data are given to the network layer. This technique is called **character stuffing**. Thus a framing DLE STX or DLE ETX can be distinguished from one in the data by the absence or presence of a single DLE. DLEs in the data are always doubled (Figure 2.7 b)
A major disadvantage of using this framing method is that it is closely tied to 8-bit characters in general and the ASCII character code in particular. As networks developed, the disadvantages of embedding the character code in the framing mechanism became more and more obvious so a new technique was developed.

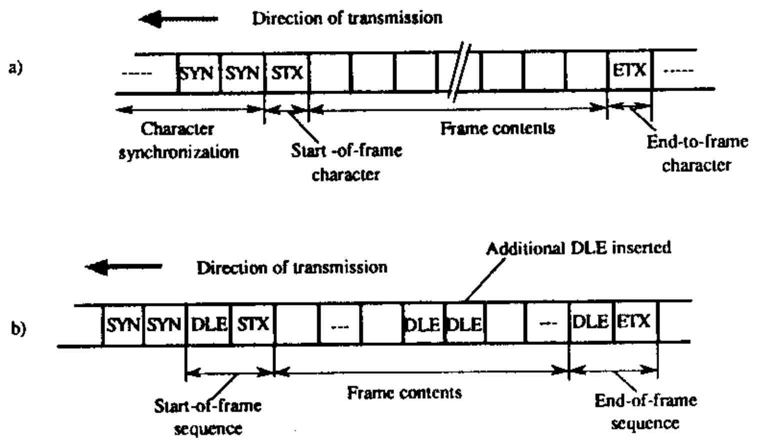

Fig. 2.7 Character oriented framing

The new technique allows data frames to contain an arbitrary number of bits, and allows character codes with an arbitrary number of bits per character. It works like this. Each frame begin and ends with a special bit pattern, namely 01111110 (**Flag**). Whenever the sender's data link layer encounters five consecutive ones in the data, it automatically stuffs a 0 bit into the outgoing bit stream. This bit stuffing is analogous to character stuffing, in which a DLE is stuffed into the outgoing character stream before DLE in the data.

When the receiver sees five consecutive incoming 1 bits, followed by a 0 bit, it automatically destuffs (i.e., removes) the 0 bit. Just as character stuffing is completely transparent to the network layer in both computers, so is bit stuffing. If the user data contains the flag pattern 01111110, it is transmitted as 011111010 but stored in the receiver's memory as 01111110. Figure 2.8 gives an example of bit stuffing.

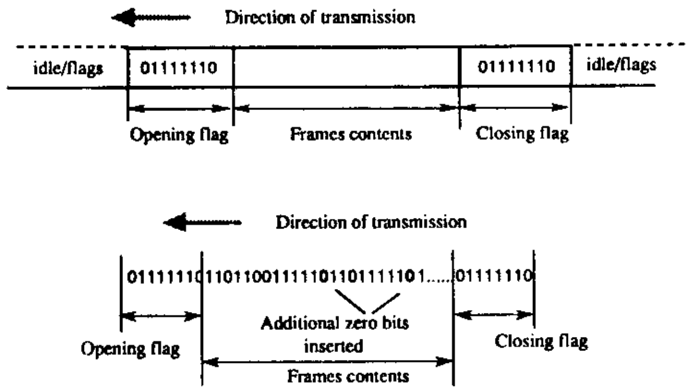

Fig. 2.8 Bit oriented framing

With bit stuffing, the boundary between two frames can be unambiguously recognized by the flag pattern. Thus if the receiver loses track of where it is, all it has to do is scan the input for flag sequences, since they can only occur at frame boundaries and never within the data.

2.3.1 Polynomial codes

The most common type of error that arises when data are transmitted over lines from the switched telephone network are those caused by a burst of electrical interference due, for example, to noise impulses caused by the switching elements within the exchanges. These, in turn, can cause a string or burst of consecutive bits in a frame to be corrupted; hence, this type of error is known as an error burst.
Parity, or its derivative block sum check, does not provide a reliable detection scheme against error bursts. In such cases, the most common alternative is based on the use of polynomial codes. Polynomial codes are used with frame (or block) transmission schemes; that is, a single set of check digits are generated (computed) for each frame transmitted. These digits, which are based on the actual contents of the frame, are appended by the transmitter at the tail of the

frame. The receiver then performs a computation on the complete frame and check digits and, if no errors have been induced, a known result should always be obtained; if a different answer is found, this indicates an error.

The number of check digits per frame varies to suit the worst-case type of transmission errors anticipated, although 16 and 32 bits are the most common. The computed check digits are referred to as the frame check sequence (**FCS**) or the cyclic redundancy check (CRC).
The underlying mathematical theory of polynomial codes is beyond the scope of this note, and the working principle only is shortly discussed.
Essentially, the complete frame contents **M**, together with an appended set of zeros equal in number to the number of FCS digits to be generated (which is equivalent to multiplying the message by 2^n where n is the number of FCS digits) are divided modulo 2 by a second binary number G, the generator polynomial, containing one more digit than the FCS. The resulting remainder R, is then the FCS. This is transmitted at the tail of the information digits. Similarly, on receipt, the received bit stream including the FCS digits is again divided by the same generator polynomial - that is, $(M \times 2^n + R)/G$ - and, if no errors are present, the resulting remainder is all zeros. If an error is present, however, the remainder is non-zero.

The choice of generator polynomial is important since it determines the type of error that are detected. For example, an error pattern that is identical, or has a factor identical, to the generator polynomial will generate the same check bits as the correct transmission, and hence will be undetectable. A polynomial which is prime, in the modulo 2 sense, is therefore normally chosen.

An important characteristic of the polynomial code method is that all bursts of errors with fewer terms than the generator polynomial are detected. As an example, the generator polynomial defined by the CCITT for use on the switched telephone network is:

$$x^{16} + x^{12} + x^5 + x^0$$

In binary form, this is equivalent to 10001000000100001, the power in x being represented as a 1 or 0. In this case, 16 zeros would be appended to the frame contents prior to the generation of the generation of the FCS. The latter would then be the 16-bit remainder. The generator polynomial selected will, therefore, detect any one error burst not exceeding 16 bits in length, in addition to all odd errors, any 2-bit errors and any two error bursts not exceeding two bits.
As this type of error-detection method is now very prevalent, especially with the more recent bit-oriented

transmission control schemes, all of the integrated circuits available for use with this scheme support this type of error-detection method. Although the requirement to perform multiple division operations to compute the FCS digits may seem difficult to implement at a fast transmitter bit rate in practice, since the arithmetic is all performed modulo 2, it is possible to implement an FCS generator checker very readily in hardware using just a combination of shift registers and exclusive-OR gates (modulo 2 adders). A typical arrangement for the example shown earlier is in Figure 2.9.

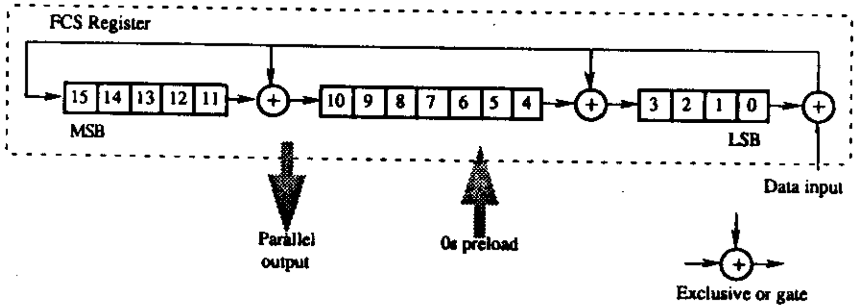

Fig. 2.9 FCS implementation

When the polynomial code method is employed, the sender and receiver must agree upon a generator polynomial G in advance. Three polynomials have become international standards:

CRC-12	$= x^{12} + x^{11} + x^3 + x^2 + x^1 + 1$
CRC-16	$= x^{16} + x^{15} + x^2 + 1$
CRC-CCITT	$= x^{16} + x^{12} + x^5 + 1$

All three contain x + 1 as a prime factor. CRC-12 is used when the character length is 6 bits. The other two are used for 8-bit characters. A 16-bit checksum, such as CRC-16 or CRC-CCITT, catches all single and double errors, all errors with an odd number of bits, all burst errors of length 16 or less, 99.997% of 17-bit error bursts, and 99.998% of 18-bit and longer bursts.

2.4 Flow Control

Another important design issue that occurs in the data link layer (and higher layers as well) is what to do with a sender that systematically wants to transmit frames faster than the receiver can accept them.
This situation can easily occur when the sender is running on a fast (or lightly loaded) computer and the receiver is running on a slow (or heavily loaded) machine. The sender

keeps pumping the frames out at a high rate until the receiver is completely swamped. Even if the transmission is error-free, at a certain point the receiver will simply not be able to handle the frames as they arrive, and will start to lose some. Clearly something has to be done to prevent this situation.

The usual solution is to introduce flow control to force the sender into sending no faster than the receiver can handle the traffic. This mechanism generally requires some kind of a feedback mechanism, so the sender can be made aware of whether or not the receiver is able to keep up.

Various flow control schemes are known, but most of them use the same basic principle. The protocol contains well-defined rules about when a sender may transmit the next frame. These rules generally prohibit frames from being sent until the receiver has granted permission, either implicitly or explicitly.

3. CHARACTER ORIENTED PROCEDURES

Character-oriented procedures are based on the use of certain characters of a specific alphabet to menage the link They were the first to be used in teleprocessing and are still widely used since they have the great merit of simplicity. Character oriented procedures have the disadvantage, however, of not being very robust and, in particular, of not using lines efficiently. This is leading to their progressive replacement by synchronous bit-oriented procedures which are a more recent concept.
The first character-oriented procedures have given rise to numerous incompatible variants. This situation developed with the introduction by IBM in 1965 of the Binary Synchronous Communication (BSC) procedure for synchronous transmission .Important work was then undertaken to standardize a character-oriented procedure, firstly with the appearance of the ANSI X3.28 standard in the United States and ECMA-16 in Europe, then the International Standard ISO 1745 titled Basic Mode Control Procedures for Data Communication Systems .These standards are very similar to BSC, but provide equally for asynchronous as well as synchronous transmission.
In order to impose the minimum constraints on data processing equipment in respect of the type of character which can be transmitted, the link can operate in two distinct modes which are the command mode and the text mode. In command mode, all characters are interpreted as such by the procedure. In text mode, only a reduced set of control characters can take part in control of the line and the other characters are transmitted in a transparent manner. For applications which require totally transparent transmission, use of the DLE character enables all limitations on the type of character transmitted as text to be avoided
Asynchronous line procedures do not need a special mechanism to acquire character synchronization, since the latter is available directly at the physical level. With synchronous transmission, the line procedure must reconstruct character synchronization itself and this is realized with the help of a character which is dedicated to this purpose. In the case of the ISO 1745 procedure, which can equally use synchronous or asynchronous transmission, character synchronization is always ensured by a special character whatever the mode of transmission.
In order to present a specific example of character-oriented line procedures, the BSC procedure will now be described.

3.1 BSC procedure

The BSC procedure is a synchronous character-oriented line procedure which can be operated with one of the three following alphabets: SBT, EBCDIC and ASCII. This procedure permits transmission on a special multipoint hierarchical line operated in two-way alternate mode, with control by polling and selection. It also permits point-to-point transmission on special lines or the switched network with control by polling or contention. Exchanges are performed with a send and wait protocol, without anticipation of acknowledgements.
Transmission can be performed in **transparent mode**, that is without limitation of code, by using the DLE character. In this mode, all control characters are preceded by DLE, and when the DLE character appears in the text it is doubled. Changing to the transparent mode is initiated by the two characters DLE SOH or DLE STX. After entering the transparent mode, all single characters other than DLE are interpreted as text by the receiver. If the latter encounters a DLE character in the transparent text, it examines the following character to determine its significance. If the following character is a command, it is interpreted as such, which enables the end of the transparent text to be identified by the DLE ETX pair. If the DLE character is followed by a second DLE character, the receiver deduces that the initial text contained a DLE character, and it suppresses one of the two DLE characters to re-establish the initial sequence.

4. SYNCHRONOUS BIT-ORIENTED PROCEDURES

Synchronous bit-oriented procedures organize information in the form of frames divided into fields which contain the different types of information to be transmitted such as commands, addresses and text. The nature of the information transmitted is, therefore, no longer tied to a particular alphabet as in the previous procedures. It depends here on the position of the information in the frame.

The most used bit-oriented line procedures are the IBM SDLC (Synchronous Data Link Control) procedure and the ISO HDLC (High-level Data Link Control) procedure together with the LAP (Link Access Procedure) procedures which are both used with the X.25 Network Protocol These different procedures are very similar and are all derived from SDLC.
The discussion of bit-oriented protocols that follows is intended as a general overview, and will target basically HDLC protocol.
All the bit-oriented protocols use the frame structure shown in Fig. 2.10

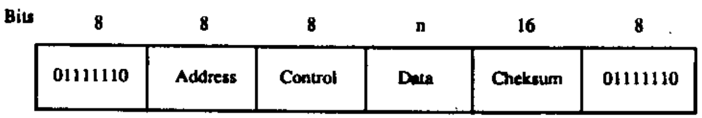

Fig. 2.10 Bit-oriented frame format

The **Address field** is primarily of importance on multidrop lines, where it is used to identify one of the terminals. For point-to-point lines, it is sometimes used to distinguish commands from responses.
The **Control field** is used for sequence numbers, acknowledgements, and other purposes, as discussed below.
The **Data field** may contain arbitrary information. It may be arbitrarily long, although the efficiency of the checksum falls off with increasing frame length due to the greater probability of multiple burst errors.
The **Checksum field** is the well-known cyclic redundancy code, using CRC-CCITT as the generator polynomial.
The frame is delimited with another flag sequence (01111110). On idle point-to-point lines, flag sequences are transmitted continuously, just as SYN characters are usually transmitted during idle periods when BISYNC is used.

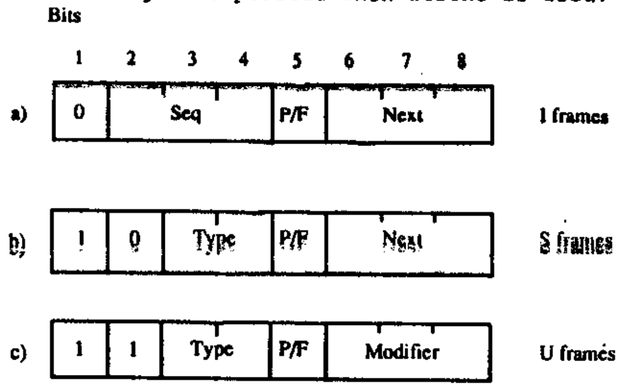

Fig. 2.11 Control field

The minimum frame contains three fields and totals 32 bits, excluding the flags on either end.
There are three types of frame: **Information (I)**, **Supervisory (S), and Unnumbered (U)**; the contents of the Control field for these three kinds are shown in Fig. 2.11.
Concerning (I) frames, the protocols uses a sliding window, with a 3-bit sequence number. Up to seven unacknowledged frames may be outstanding at any instant. The Seq field in Fig. 2.10 (a) is the frame sequence number. The Next field is a piggybacked acknowledgement. However, all the protocols adhere to the convention that instead of piggybacking the number of the last frame received correctly, they use the number of the first frame not received (i.e., the next frame expected). The choice of using the last frame received or the next frame expected is arbitrary; it does not matter which convention is used, provided that it is used consistently of course.
The P/F bit stands for Poll/Final. It is used when a computer (or concentrator) is polling a group of terminals. When used as P, the computer is inviting the terminal to send data. All the frames sent by the terminal, except the final one, have the P/F bit set to P. The final one is set to F.
In some of the protocols, the P/F bit is used to force the other machine to send a Supervisory frame immediately rather than waiting for reverse traffic onto which to piggyback the window information. The bit also has some minor uses in connection with the Unnumbered frames.

The various kinds of Supervisory frames are distinguished by the Type field.
Type 0 is an acknowledgement frame (officially called RECEIVE READY) used to indicate the next frame expected. This frame is used when there is no reverse traffic to use for piggybacking.
Type 1 is a negative acknowledgement frame (officially called REJECT). It is used to indicate that a transmission error has been detected. The Next field indicates the first frame in sequence not received correctly (i.e., the frame to be retransmitted). The sender is required to retransmit all outstanding frames starting at Next.
Type 2 is RECEIVE NOT READY. It acknowledges all frames up to but not including Next, just as RECEIVE READY, but it tells the sender to stop sending. RECEIVE NOT READY is intended to signal certain temporary problems with the receiver, such as a shortage of buffers, and not as an alternative to the sliding window flow control. When the condition has been repaired, the receiver sends a RECEIVE READY, REJECT, or certain control frames.
Type 3 is the SELECTIVE REJECT. It calls for retransmission of only the frame specified. In this sense it is like our protocol 6 rather than 5 and is therefore most useful when the sender's window size is half the sequence space size, or less. Thus if a receiver wishes to buffer out of sequence

frames for potential future use, it can force the retransmission of any specific frame using Selective Reject. HDLC and ADCCP allow this frame type, but SDLC and LAPB do not allow it (i.e., there is no Selective Reject), and type 3 frames are undefined.

The third class of frame is the Unnumbered frame. It is used for control purposes. The various bit-oriented protocols differ considerably here, in contrast with the other two kinds, where they are nearly identical. Five bits are available to indicate the frame type, but not all 32 possibilities are used.
All the protocols provide a command, DISC (DISConnect), that allows a machine to announce that it is going down (e.g., for preventive maintenance). They also have a command that allows a machine that has just come back on line to announce its presence and force all the sequence numbers back to zero. This command is called SNRM (Set Normal Response Mode); it is an **unbalanced** (i.e., asymmetric) mode in which one end of the line is the master and the other the slave. SNRM dates from a time when data communication meant a dumb terminal talking to a computer, which clearly is asymmetric. To make the protocol more suitable when the two partners are equals, HDLC and LAPB have an additional command, SABM (Set Asynchronous **Balanced Mode**), which resets the line and declares both parties to be equals. They also have commands SABME and SNRME, which are the same as SABM and SNRM, respectively, except that they enable an extended frame format that uses 7-bit sequence numbers instead of 3-bit sequence numbers.
A third command provided by all the protocols is FRMR (FRaMe Reject), used to indicated that a frame with a correct checksum but impossible semantics arrived. Examples of impossible semantics are a type 3 Supervisory frame in LAPB, a frame shorter than 32 bits, an illegal control frame, and an acknowledgement of a frame that was outside the window, etc. FRMR frames contain a 24-bit data field telling what was wrong with the frame. The data include the control field of the bad frame, the window parameters, and a collection of bits used to signal specific errors.
Control frames may be lost or damaged, just like data frames, so they must be acknowledged too. A special control frame is provided for this purpose, called UA (Unnumbered Acknowledgement). Since only one control frame may be outstanding, there is never any ambiguity about which control frame is being acknowledged.
The remaining control frames deal with initialization, polling, and status reporting. There is also a control frame that may contain arbitrary information, UI (Unnumbered Information). These data are not passed to the network layer, but are for the receiving data link layer itself.

It is important to discriminate between the services provided by the link layer and the means to actually

implement the services at link layer using L-PDU; in Figure 2.12 is shown an example using HDLC frames.

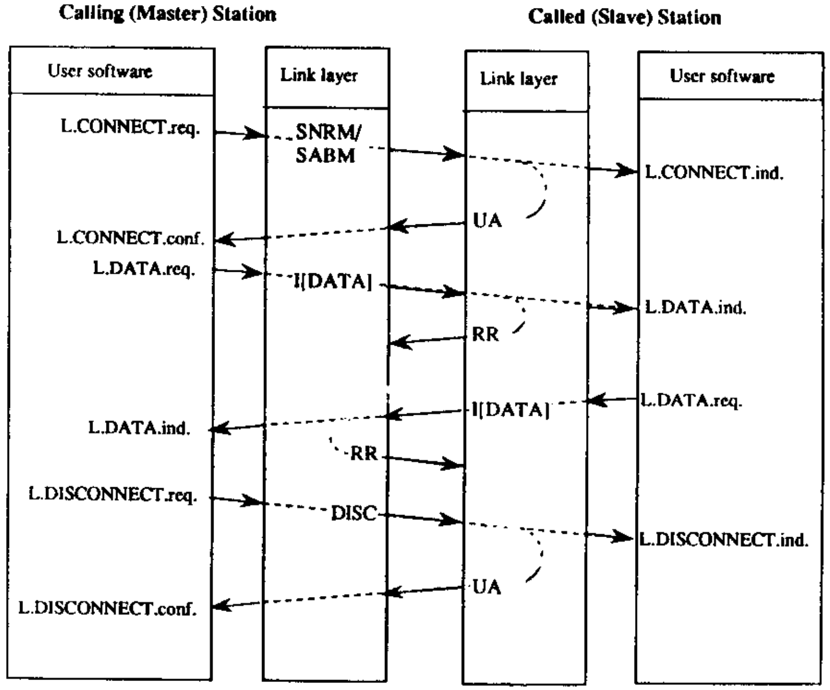

Fig. 2.12 Using L-PDU

In Figure 2.13 a concise summary of the link layer interactions within the OSI reference model is presented.

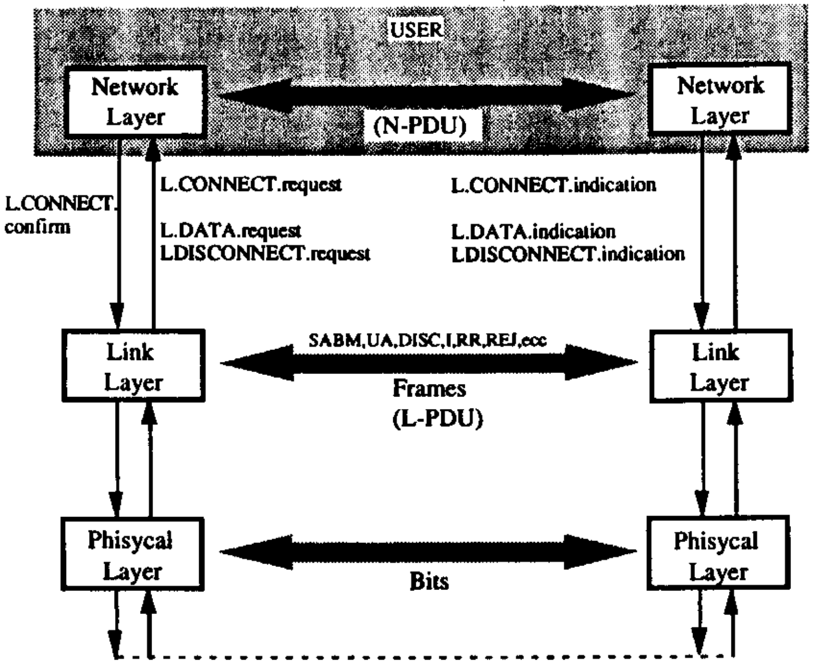

Fig. 2.13 Link layer summary

DATA NETWORKS

1. PRELIMINARY CONSIDERATIONS

Direct communication between stations is provided by data links which ensure error-free transmission of frames. When the number of stations to be connected becomes large, the traffic can no longer be expedited on a single channel and it is then necessary to provide the communication system with a network structure which contains a number of nodes interconnected by data links (Figure 3.1). Once the data exchanged by the stations has to pass through one or more nodes, the design and operation of the communication system rises a series of new problems compared with those arising in a simple data link.

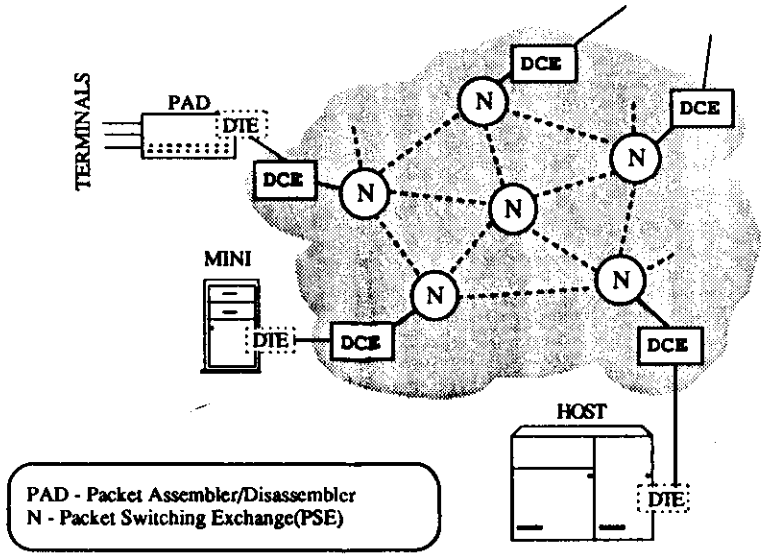

Fig. 3.1 PSDN configuration

There are two main types of PDN: **Packet-Switched Public Data Networks (PSPDN)** and **Circuit-Switched Public Data Networks (CSPDN)**, and different standards have been defined for each type.
Each connection established through a circuit-switched network results in a specific physical communication channel being set up through the network from the calling to the called subscriber equipment, which is then exclusively used by the two subscribers for the duration of the call. This feature which seems the most straightforward solution for data communications, is indeed the main weak point of the cicuit-switched approach.

As a matter of fact, if data communications where only of point-to-point type, with massive transfer of data between teleprocessing equipments (e.g. File Transfer), the establishment of a temporary physical communication channel would undoubtedly prove as the most simple as well as efficient approach: following the set-up phase, raw transmission of the bulk data stream could be performed without any intermediate relay functionality. Vis-a-vis point-to-point file transfer connections represent just a minor, if not negligible, share of the usual data communication business; more and more frequent is the usage of PDNs as basic infrastructure for Public Services, that is sophisticated processing applications where multi-point capabilities are mandatory (e.g connection to a unique Service Centre), and above all, human interactions (i.e.transmission pauses) are unavoidable. Since network resources are fully subscriber dedicated, they cannot be shared between users when these are inactive, so leading to overdesign of the network and definitely to very inefficient usage.

In contrast, no physical connections are established through the network with a packet-switched approach. Instead, all data to be transmitted are first assembled into one or more message units called packets by the source DTE, which include both the source and destination DTE network addresses. These are then passed bit serially by the source DTE to its local packet-switching exchange (PSE). On receipt of a packet, therefore, the PSE forwards the packet on the appropriate link at the maximum available bit rate.
As no physical connection is established, but just a virtual channel, with a packet-switching network, it is possible for two communicating subscribers (DTEs) to operate at different data rates, since the rate at which data are passed at the two interfaces to the network is separately regulated by each subscriber equipment. This means that a fast DTE, connected to its node by means of a high speed DCE, can host many virtual connections, each one dedicated to many remote subscribers In conclusion beyond the efficient usage of network resources, the packet-switching approach is far superior from a technological viewpoint, allowing a total software approach in the design of networks, resulting in a much higher degree of system flexibility.
The ability of X.25 packet-switching protocol to allow various PSPDN businesses, international connections, and private packet networks to coexist is particularly significant, and is the reason why the protocol set has evolved so quickly to become a worldwide standard.
In the following just PSPDNs will be considered.

2. FUNCTIONS AND SERVICES

From the user viewpoint, the network must permit an exchange of messages between subscribers. At this level, which corresponds to the transport layer of the OSI model, the

size of messages must be determined only by the needs of the users and it must not, therefore, depend on constraints associated with transport of the messages. In general terms, the transport layer manages the end-to-end transport of the messages and provides independence from higher-level layers with respect to all problems of this type. Transport is realized in the form of packets which are routed between two stations connected to a communication network. This allows an intermediate layer to be defined between the transport and data link layers. This intermediate layer is called the **network layer** and its main role is to establish a path between the two nodes that connect the network users. The most important functions provided by the network layer are the segmentation of messages into packets whose size is adapted to the characteristics of the network, together with addressing and routing of the packets and network flow control. The network layer thus makes the transport layer independent of the problems of routing and switching.

With a PSPDN, two types of service are normally supported: **datagram and virtual call circuit.** The difference between the two types of service can be explained by the analogy between exchanging messages by means of letters and by means of a telephone call. In the first case, the letter containing each message is treated as a self-contained entity by the postal authorities and its delivery is independent of any other letters. In the case of a telephone call, however, a communication path is first established through the network and the subsequent message exchange takes place.

The datagram service is analogous to sending a message by means of a letter, since each packet entering the network is treated as a separate, self-contained entity with no relationship to other packets. Each packet is simply received and forwarded in the way just outlined, and hence the datagram service is primarily used for the transfer of short, single-packet messages.

If a message contains multiple packets, the virtual call service is normally selected. This is analogous to sending a message by means of a telephone call, since when using this service, prior to sending any information (data packets) associated with a call, the source DTE sends a special call request packet to its local PSE containing, in addition to the required destination DTE network address, a reference number called the logical channel identifier (LCI). This is noted by the PSE and the packet is then forwarded through the network as before. At the destination PSE, a second LCI is assigned to the call request packet before it is forwarded on the outgoing link to the required destination DTE. Then, assuming the call is accepted, an appropriate response packet is returned to the calling DTE. At this point, a virtual call is said to exist between the two DTEs. The information transfer phase is then entered and all subsequent data packets relating to this call are assigned the same reference numbers on each interface link to the network. In this way, both the source and destination DTEs can readily distinguish between packets arriving on the same link but relating to different calls. Hence, packets belonging to the same call can be passed to the user (the transport layer) in the same sequence as they were entered.

It should be noted that although a virtual circuit may appear to the user to be similar to a connection established through a circuit-switched network, a virtual circuit, as the name implies, is purely conceptual. Moreover, since the PSPDN is able to apply additional error and flow control procedures at the packet level as well as those used at the link level, the class of service supported by a virtual circuit is very high; that is, the probability or all the packets relating to a particular call being delivered free of errors and in the correct sequence without duplicates is very high.

Normally, a virtual circuit is cleared and the appropriate logical channel identifiers released after all data relating to a call have been exchanged. However, it is possible for the virtual circuit to be left permanently established, so that a user who requires to communicate with another user very frequently does not have to set up a new virtual circuit for each individual call. This is then known as a permanent virtual circuit and, although the user must pay for this facility, the cost of each call is based only on the quantity of data transferred.

2.1 The OSI network service primitives

International Standard 8348 defines the network service by specifying the primitives that apply at the boundary between the network layer and transport layer. The user services provided by the network (packet) layer are shown in the time sequence diagram of Figure 3.2

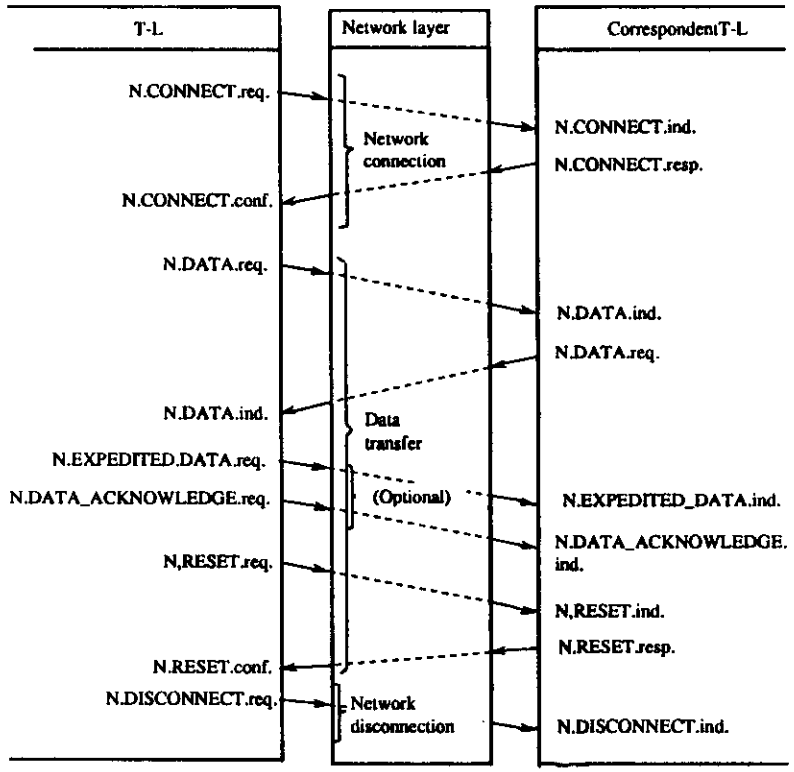

Fig. 3.2 Network layer service primitive

They can be grouped into four categories, for **establishing, releasing, using, resetting** connections, respectively. Most of the primitives have parameters.

The N.CONNECT request primitive is used to set up a connection. It specifies the network address to connect to and the caller's network address. It also contains two Boolean variables used to request optional services.
The first one is used to permit the caller to request acknowledgement of each packet sent; if the network layer does not provide acknowledgement, the variable is set to FALSE when delivered to the destination in the N.CONNECT indication primitive. If the network layer does provide acknowledgements, but the destination does not want to use them, then it sets the variable to FALSE in its N.CONNECT response. Only if both transport entities and the network service provider want to use them are they used. This feature is an example of option negotiation.

The second logical flag, if accepted by all three parties, permits the use of EXPEDITED_DATA; essentially packets may violate the normal queue ordering and skip to the head of the queue. A typical example of EXPEDITED_DATA is a user at a terminal hitting the DEL key to interrupt a running program. The DEL packet will go as N.EXPEDITED_DATA.

The caller may include some user data in the connection request. The called may inspect these data before deciding whether to accept or reject the request. Connection requests are accepted with the N.CONNECT response primitive and rejected with the N.DISCONNECT request primitive. When a request is rejected, the reason field allows the called to tell why it was not willing to accept the connection and whether the condition is permanent or transient. The network layer itself may also reject attempts to establish connections, for example, if the quality of service desired is not available (permanent condition) or the network is currently overloaded (transient condition).

The remaining N.CONNECT primitives and the N.DISCONNECT primitives are straightforward, and need little further comment. After a connection has been established, either party can transmit data using N.DATA request. When these packets arrive, an N.DATA indication primitive is invoked on the receiving end. Expedited data uses primitives analogous to those for regular data.

If acknowledgements have been agreed upon, when a packet has been received, the recipient is expected to issue an N.DATA-ACKNOWLEDGE request. This primitive contains no sequence number, so the party sending the original data must simply count acknowledgements. If the quality of service is low, and data and acknowledgements can be lost, this scheme is not very satisfactory. On the other hand, it is not the task of the network layer to provide an error-free service; that job is done by the transport layer. Network layer acknowledgements are merely an attempt to improve the quality of service, not make it perfect.

The N.RESET primitives are used to report catastrophes, such as crashes of either transport entity or the network service provider itself. After an N.RESET has been original empty state. Data present in the queues at the time of the N.RESET are lost. Again here, it is the job of the transport layer to recover from N.RESETs.

3. NETWORK LAYER DESIGN ISSUES

Messages sent by users can vary widely in size according to the type of application and the nature of the message. A reply to a request can be very short and even reduce to one bit in the case of an unqualified acknowledgement. In contrast, some messages can be very long, for example when

they correspond to a file transfer. On the other hand, effective utilization of a network requires the transmitted packets to be neither too long nor too short. Each packet contains the text and a header which includes, for example, the destination address, an indication of the packet type and an error check word. If the packets are too short, the relative size of the text with respect to the header is small and the lines and node buffers are poorly used. If, on the other hand, the packets are too long, the header becomes negligible with respect to the text, which is favourable in terms of line utilization. However, for a given bit error rate, the probability that a packet is in error and must be retransmitted becomes greater as the packet becomes longer.

Segmentation and reassembly of messages permits the line utilization rate to be optimized and consequently the cost of communication to be reduced. This procedure also reduces the system response time, that is the message transfer time in the network. While passing from node to node, the packets suffer a delay which is proportional to their length, since error checking requires that they can be retransmitted only after having been completely received; this means that they must be temporarily stored at each node.

3.1 Routing

One of the main problems which must be solved at the network level is to establish a path between the calling and the called stations. When the network is operated by datagrams, path selection must be made separately for each packet; in contrast, the decision on the route to be followed is taken only at the time of establishment for virtual circuits. In the two cases, the choice of routing algorithm is not easy since it must satisfy a large number of complex requirements. This algorithm should be simple, in order to be implemented easily in the nodes, and it should ensure correct routing of packets for any hazards suffered by the network. The algorithm must thus be capable of giving satisfactory results in spite of variations in traffic and the topology of the network. It must also ensure approximately equal sharing of access rights to the network among the various stations. In most cases,the goal is realization of a network which minimizes the packet transfer time or which maximizes network throughput; the principal objectives are thus minimum transfer time and maximum throughput. In other cases, the main objective of the designer is, for example, to provide absolute communication reliability.
Routing algorithms can be grouped into two major classes: **non-adaptive and adaptive**. Non-adaptive algorithms do not base their routing decisions on measurements or estimates of the current traffic and topology. Instead, the choice of the route to use is computed in advance, off-line,

and down-loaded to the nodes when the network is booted. This procedure is sometimes called **static routing**.

Adaptive algorithms, on the other hand, attempt to change their routing decisions to reflect changes in topology and the current traffic. Three different families of adaptive algorithms exist, differing in the information they use. The global algorithms use information collected from the entire network in an attempt to make optimum decisions. This approach is called **centralised routing**. The local algorithms run separately on each node and only use information available there, such as queue lengths. These are known as **isolated routing algorithms**. Finally, the third class of algorithms uses a mixture of global and local information. They are called **distributed algorithms**.

3.2 Congestion

When too many packets are present in (a part of) the network, performance degrades. This situation is called congestion. When the number of packets dumped into the network by the hosts is within its carrying capacity, they are all delivered (except for a few that are afflicted with transmission errors), and the number delivered is proportional to the number sent. However, as traffic increases too far, the nodes are no longer able to cope, and they begin losing packets.
Packets which cannot be stored in memory must be discarded: this causes their retransmission and thus propagates congestion by generating dummy internal traffic. There is thus an avalanche effect due to the fact that, above a certain limit, the useful transport capacity of the network decreases when the traffic offered increases (Figure 3.3).

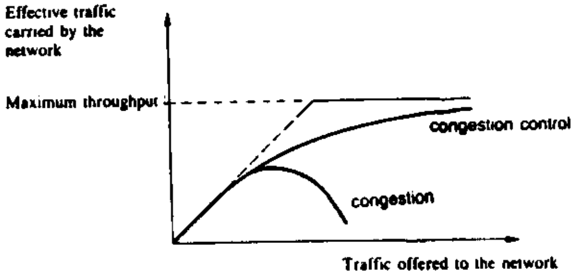

Fig. 3.3 Congestion curve

3.2.1 Control by pre-allocation of resources

In a virtual circuit network, a path is established between the two subscribers at the start of communication by mean of a calling packet. This approach can readily be modified by

requiring the calling packet not only to establish the route but also to reserve, at each node, the resources which will be allocated to the communication being established. In particular, buffers can be assigned in each node and for each active virtual circuit; the size of these corresponds at least to the maximum number of packets in transit, that is the window size. The resource pre-allocation method prevents congestion since the procedure rejects any call for which resources cannot be allocated. The obvious penalty is that resources are very badly used, since they are not shared between users when these are inactive. Pre-allocation of resources thus leads to overdesign of the network.

3.2.2 Using flow control

All congestion control techniques are based on the idea that it is necessary to limit the number of packets in transit in the network in order to prevent network overload. This result can obviously be achieved by end-to-end flow control which limits the maximum number of packets in transit between two users. With such an approach and in a similar manner to that which is used with line procedures, the source marks the packets which it transmits in sequence with a number P(S) defined modulo M. The source cannot send more than W consecutive packets before it receives a packet carrying the number P(R) which acknowledges at least the first of the W packets sent. This sliding window procedure ensures that there are never more than W packets in transit between the source and the destination.

3.2.3 Isarithimic control

With this method, M tokens (credits) flow continuously on the network and a packet can be transferred only if it is in possession of a token. Thus it is impossible for the number of packets in transit on the network to exceed M. The value of M is clearly chosen so as to avoid congestion while ensuring reasonable performance. In practice, each node stores several tokens and sends surplus tokens to neighbouring nodes so as to ensure equal sharing of tokens throughout the network. A packet can enter the network only if the input node has at least one token; if not, the packet is rejected. If the packet is accepted, it seizes the token and returns it on leaving the network.

4. X.25 PROTOCOL

The X:25 protocol was first standardized in 1980. It subsequently became necessary to make modifications to ensure compatibility between this protocol and the OSI model and to specify some points omitted from the standard. This led to the introduction in 1984 of a new version of the X.25 protocol .The 1980 X.25 protocol provided an optional datagram service together with a similar service called **fast select** which is based on transfer of user data at the time of setting up the virtual circuit, with immediate clearing of the circuit at the end of the set-up phase. The 1984 X.25 protocol retained only the fast select service.

The X.25 protocol does not define the internal operation of the network, that is the manner in which the packets are routed between the various PSEs corresponding to the nodes of the network. Practical implementation and performance can thus vary appreciably from one network to another. In contrast, X.25 specifies very precisely the interface between the network and the user in such a way as to permit all X.25 DTEs to be connected to any public network which conforms to the protocol. Recommendation X.25 defines three protocol levels which correspond approximately to the first three layers of the OSI model (Figure 3.4).

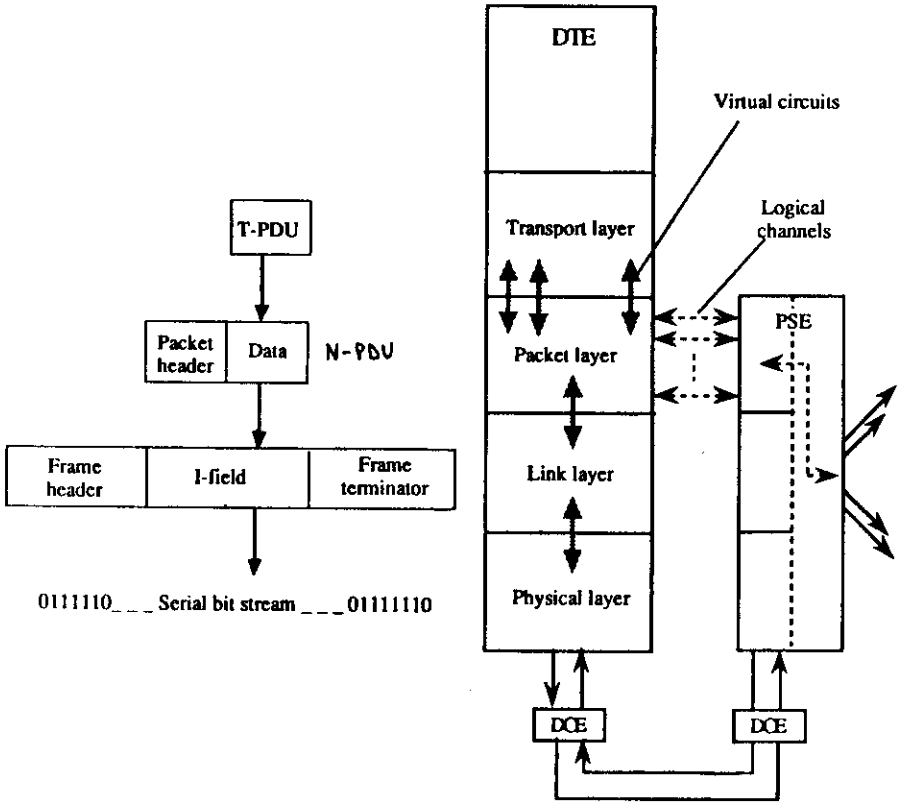

Fig. 3.4 PDU contents

4.1 Physical interface

Level 1 of X.25 specifies the physical, electrical and logical characteristics of the interface between the DTE and the DCE. This level, which corresponds to the transmission of bits by a physical channel, is specified in detail in CCITT Recommendation X.21. This recommendation relates to access to a digital network. As networks of this type are rarely available and the same applies to the corresponding DTE interfaces, access to packet switching X.25 networks is generally achieved by way of the telephone network using modems equipped with a V.24/V.28 interface. In this case, the interface is specified by CCITT Recommendation X.21bis with synchronous transmission (see Figure 3.5).

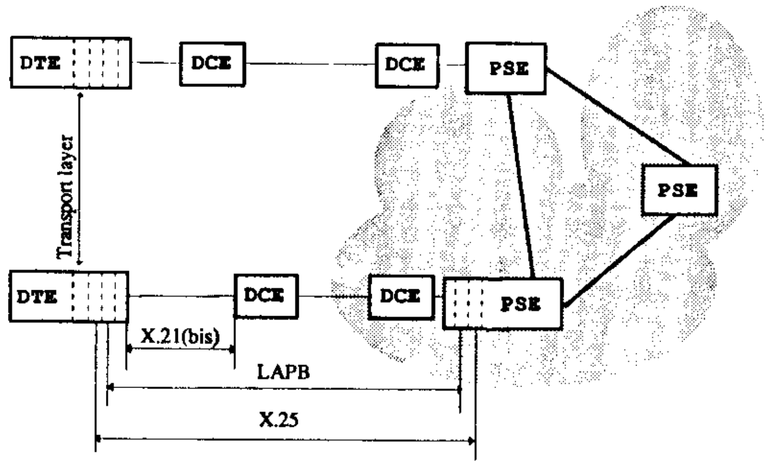

Fig. 3.5 X.25 layers

4.2 Link layer

The LAP and LAPB line procedures form level 2 of the X.25 protocol. These two synchronous procedures are bit oriented and directly derived from the HDLC; they have, in particular, the same frame type and structure. They are normally used for network access with point-to-point operation on a single physical circuit in full duplex.
The LAP and LAPB single link procedures (SLP) provide basic access to public packet switching networks and are generally used as such. However, it is sometimes necessary to connect several parallel circuits to the network in such a way as to prevent system interruption in the case of failure of one of the circuits. To meet this requirement, the CCITT has provided a multi-link procedure (MLP) which permits sharing of packet transmission on the various single link procedures

used between the DTE and the DCE. In this case, the single link procedures can only be of the LAPB type.

The LAP procedure operates in asynchronous response mode (ARM) on an unbalanced line where each station incorporates a primary text transmission function and a secondary text reception function. The LAP procedure is very similar to the HDLC procedure in the asynchronous response mode, but it is incompatible with it. It uses I, RR, RNR and REJ information and supervisory frames with modulo 8 sequence numbering and unnumbered SARM, DISC, CMDR and UA frames. As the configuration is symmetrical, the bidirectional link can be considered to consist of two unidirectional links in opposite directions. This requires the two half links to be established individually by SARM frames which are acknowledged by unnumbered UA frames . Before these operations, the DCE indicates that it is ready establish the link by sending successive flags. Clearing of the link is performed similarly by two DISC command frames.

The LAP procedure has the disadvantage of incompatibility with HDLC. Because of this, it is being replaced by the LAPB procedure which is compatible with the HDLC procedure in asynchronous balanced mode (ABM) The DTE and DCE are combined stations in this case, and the link is established by a single initialization frame SABM In a similar manner, the link is cleared by a single disconnection frame DISC. Information transfer is performed using information frames I and supervisory frames RR.. RNR and REJ which are numbered modulo 8 or, optionally, modulo 128.
The multilink procedure makes use of several parallel data links of the LAPB type on which it distributes frame transfer. This mode of operation introduces new problems since frames no longer necessarily arrive sequentially at the destination. More generally, the multilink procedure must implement some functions to control the group of links, for example reset of the group or to put one of the links out of service. To achieve this, the multilink procedure uses frames which have the same general structure as LAPB frames with the exception of an additional two byte command field. This field contains, among others, a 12-bit number which is used to maintain frame sequence.

4.3 Packet layer

Two packet layer protocol entities communicate with each other to implement the user services by exchanging packet protocol data units (N-PDUs). The different N-PDU types used to implement the user services are shown in Table 4.1.

Table 4.1 X.25 packet types.

Type		Service	
From DCE to DTE	From DTE to DCE	Switched virtual circuit	Permanent virtual circuit
Call set-up and clearing of virtual circuits			
INCOMING CALL	CALL REQUEST	x	
CALL CONNECTED	CALL ACCEPTED	x	
CLEAR INDICATION	CLEAR REQUEST	x	
CLEAR CONFIRMATION	CLEAR CONFIRMATION	x	
Data and Interrupts			
DATA from the DCE	DATA from the DTE	x	x
INTERRUPT by the DCE	INTERRUPT by the DTE	x	x
INTERRUPT CONFIRMATION	INTERRUPT CONFIRMATION	x	x
Flows control and reset			
RR from the DCE	RR from the DTE	x	x
RNR from the DCE	RNR from the DTE	x	x
	REJ from the DTE	x	x
RESET INDICATION	RESET REQUEST	x	x
RESET CONFIRMATION	RESET CONFIRMATION	x	x
Restart			
RESTART INDICATION	RESTART REQUEST	x	x
RESTART CONFIRMATION	RESTART CONFIRMATION	x	x
Diagnostic			
DIAGNOSTIC		x	x
Recording			
RECORD CONFIRMATION			
	RECORD REQUEST	x	x

4.4 Virtual call establishment and clearing

A time sequence diagram illustrating the various phases of a virtual call is shown in Figure 3.6. A virtual circuit is established (set up) as a result of the user issuing an N.CONNECT request primitive at a user service access point. The parameter associated with this primitive include the network address of the called DTE and also a limited amount of user data. As can be seen, two alternative procedures may be adopted to set up the network connection.

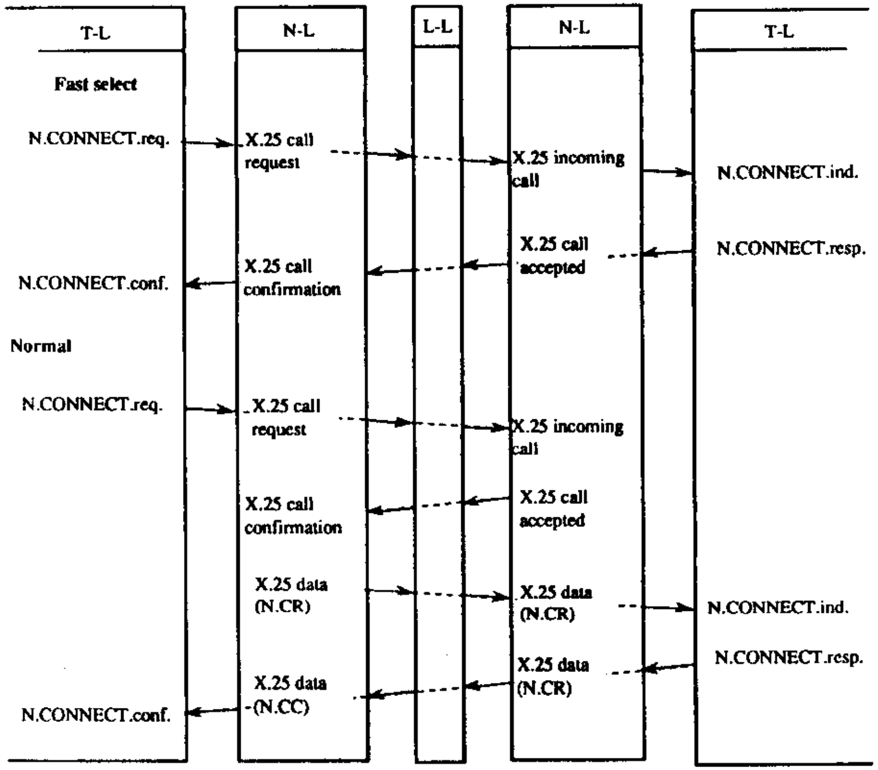

Fig. 3.6 Virtual call establishment

In the second (normal), an X.25 virtual circuit is established on receipt of the N.CONNECT.request primitive and the network connect request (N.CR) is then passed over this circuit using an X.25 data packet. The overheads associated with this method are high, however, and hence the alternative (fast-select) mode has been introduced. As can be seen, the network connect request is mapped directly into an X.25 call-request packet in this method and hence the call set-up overheads are reduced significantly. The reset and disconnect services are mapped in a similar way.

A time sequence diagram illustrating the use of the logical channels is shown in Figure 3.7, assuming the fast-select facility. On receipt of the N.CONNECT. request primitive, the source protocol entity first selects the next free LCI and creates a call-request packet (N-PDU) containing the calling and called DTE addresses and the selected LCI. The

packet is then passed to the link layer for forwarding to its local PSE.

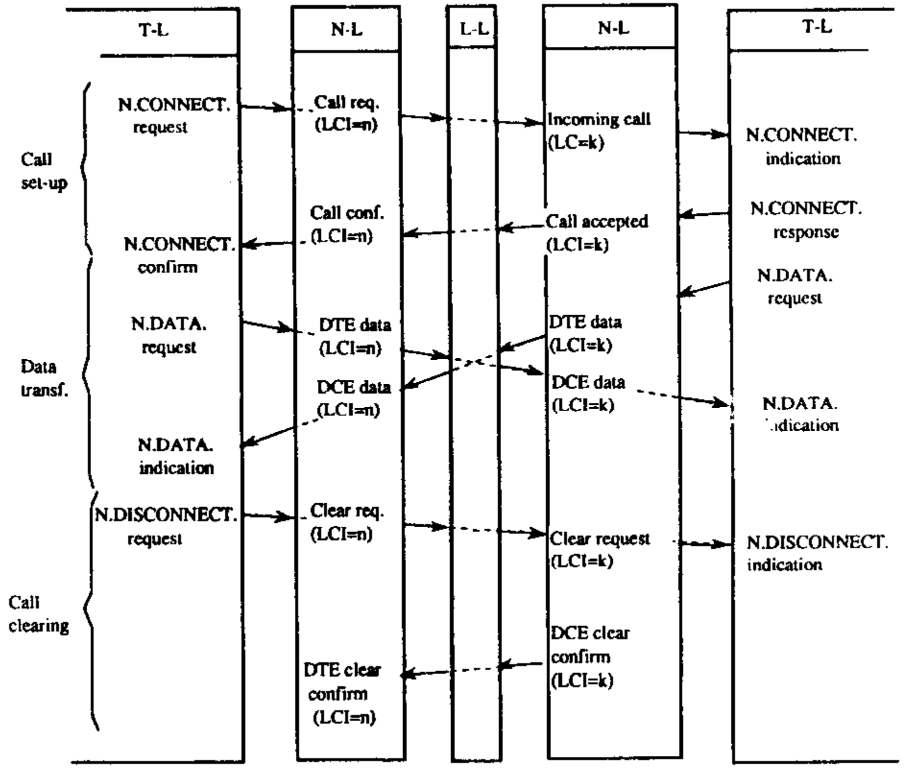

Fig. 3.7 Using Logical Channel Identifiers (LCI)

On receipt of the packet, the local PSE notes the LCI selected and forwards the packet, according to the internal protocol of the network, to the appropriate destination PSE. The latter then selects the next free LCI for use on the link to the called DTE, writes this into the packet and changes the packet type into an incoming-call packet. This is then forwarded to the called DTE where its contents are used by the correspondent packet protocol entity to create an N.CONNECT.indication primitive, which is then passed to the correspondent user.

Assuming that the correspondent user is prepared to accept the call, it responds with an N.CONNECT.response primitive which, in turn, is used by the packet protocol entity to create a call-accepted packet. The latter is assigned the same LCI as the one that was used in the corresponding

incoming-call packet. The call-accepted packet is then forwarded to the called DTE's local PSE and the reserved logical channel on this link then enters the data transfer phase. Similarly, the source PSE, on receipt of the call-accepted packet, inserts the previously reserved LCI for use on this part of the circuit into the packet and sets this logical channel into the data transfer state. It then converts the packet into a call-connected packet and forwards this to the calling DTE. Finally, on receipt of this packet, the calling-packet protocol entity issues an N.CONNECT.confirmation primitive to the user and enters the data transfer state.
If the correspondent user does not wish, or is not able, to accept an incoming call, it responds to the N.CONNECT.indication primitive with an N.DISCONNECT.request primitive. This results in the called packet protocol entity returning a clear-request packet to its local PSE. The latter first releases the previously reserved LCI and then returns a clear-confirmation packet to the called DTE. It then sends a clear-request packet to the source PSE which, in turn, passes the packet to the packet protocol entity in the calling DTE as a clear-indication packet. The DTE first releases the reserved LCI and then passes an N.DISCONNECT.indication primitive to the user. It then returns a clear-confirmation packet to its local PSE to complete the clearance of the virtual circuit. Similarly, either the user or the correspondent user can initiate the clearing of a call at any time by issuing an N.DISCONNECT.request primitive at the corresponding user interface.

4.5 Data transfer

After a virtual call (network logical connection) has been established, both the user and correspondent user may initiate the transfer of data independently of one another by issuing an N.DATA.request primitive at its network interface with the data to be transferred as a parameter. As has been mentioned, the maximum length of each data packet in a packet-switched network is limited, typically to 128 octets of data, to ensure a reliably fast response time. Hence, if a user wishes to transfer a message containing more than this number of octets, the message is first divided into an appropriate number of data packets and each packet sent separately. For the recipient user to know when each message is complete, therefore, each data packet sent through the network contains a single bit in its header known as the more-data bit, which is set whenever further data packets are required to complete a user-level (that is, transport layer) message.

Although the three protocol layers associated with the X.25 protocol set normally have only local significance, there is also a facility provided to allow acknowledgement

information at the packet level to have end-to-end significance. This again is implemented by means of a special bit in each packet header known as the delivery confirmation or D bit. The D bit in the header of a data packet is set to 1 if the source DTE requires an end-to-end confirmation (acknowledgement) of correct receipt by the remote peer packet layer.

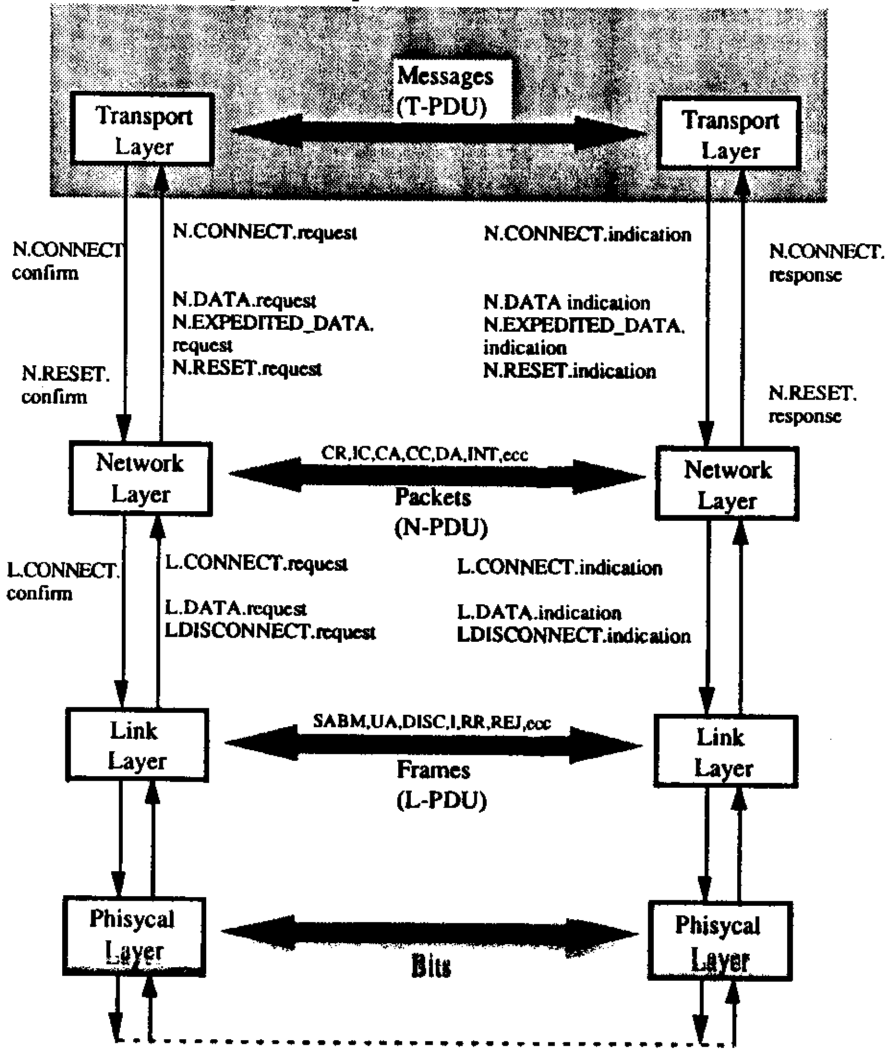

Fig.3.8 Network layer summary

In Figure 3.8 a concise summary of the network layer interactions within the OSI reference model is presented.

4.6 Packet formats

All X.25 packets contain at least three bytes which contain a general format identifier, a logical channel identifier and a packet type identifier (Figure 3.9). According to the packet type, the basic header can include extra bytes which contain additional fields. All packets have a length which is an integral number of bytes.. As for the HDLC procedure, X.25 packets can be subdivided into data packets which contain send and receive numbers, supervisory packets which include receive numbers and unnumbered packets which are used to control the virtual circuit.

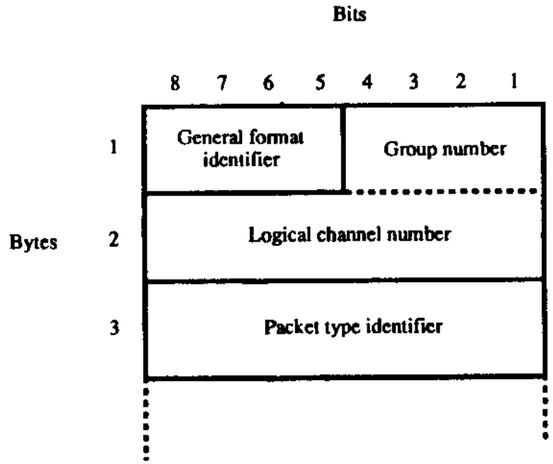

Fig.3.9 Typical packet header

4.7 Option facilities

The X.25 protocol provides a number of optional facilities which can be established at the time of installation or negotiated on establishment of the virtual circuit by means of a service specification field of the CALL REQUEST packet. In the following some of these services are listed.

Incoming calls barred
Outgoing calls barred
One-way logical channel outgoing
One-way logical channel incoming
Non-standard default packet sizes
Non-standard default window sizes
Default throughput classes assignment
Flow control parameter negotiation

Throughput class negotiation
Closed user group
Fast select acceptance
Reverse charging
Local charging barred
Identification of network user
Charging information

4.8 Packet assembly and disassembly service (PAD)

Implementation of the X.25 protocol is relatively complex, so that it is economically justified only for computers, microcomputers and intelligent terminals. Asynchronous terminals are very common and it is, therefore, necessary to provide a facility to connect them to public packet switching X.25 networks. This connection can be realized by means of auxiliary packet assembly and disassembly equipment (PAD) which provides protocol conversions between asynchronous terminals and the X.25 network and whose use is shared by several terminals (see Figure 3.10).

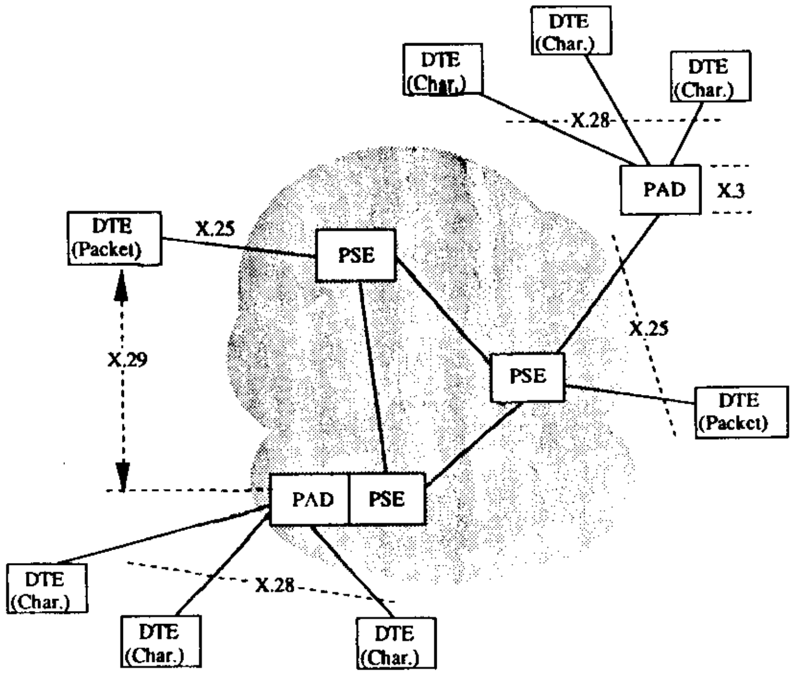

Fig. 3.10 PADs in the PSPDN

The PAD serves essentially to manage the X.25 protocol on behalf of the asynchronous terminals. As the latter can be of various types, PAD operation may be defined by a number of parameters which allow the PAD to be compatible with the terminals. The main functions provided by the PAD are as follows:

- Assembly of terminal characters into packets
- Disassembly of the user data field of packets received from the network into characters for the terminal.
- Establishment and clearing of virtual circuits
- Control of reset and interrupt procedures
- Generation of service signals
- Processing of a break signal from an asynchronous terminal
- Editing in the terminal

The CCITT has specified the PAD functions in Recommendation X.3.

4.8.1 Recommendation X.28

This recommendation specifies the protocol to be used between an asynchronous character-mode terminal and the PAD. It contains the procedures to be followed to:

- access the PAD;
- set the terminal parameters to the required values;
- establish a virtual call to a destination packet-mode DTE;
- control the exchange of user's data between the terminal and the PAD;
- clear an established call.

The asynchronous terminal can be connected to the PAD by a leased line or by the switched network. In the most frequent case, where the connection uses the public telephone network, the circuit can be equipped with V.21 modems up to 300 bit/s. V.22 bis or V.23 modems for rates up to 1200 bit/s or V.23 modems with a 75 bit/s return channel for the terminal-PAD direction and transmission at 1200 bit/s for the PAD-terminal direction. The circuit must be established to conform to Recommendation V.25(bis). If the connection is made by way of a public data network, the interface must conform to Recommendation X.20 of X.20bis.
Commands and data are exchanged between the terminal and the PAD in asynchronous mode using characters which contain seven effective bits and one parity bit and are coded in accordance with International Alphabet No 5.
Once connected to the PAD, the terminal can be operated in Command mode or Text mode under complete control of the PAD.

[illegible] Recommendation X.29

This recommendation specifies the interaction between the PAD and the remote packet-mode DTE. The basic procedures associated with X.29 for call establishment and data transfer are essentially the same as those used in X.25. Additional procedures are also defined in the recommendation, however, that reflect the presence of the PAD between the terminal and remote packet-mode DTE.

5. X.75 PROTOCOL

Up until now, we have assumed that there is a single homogeneous network, with each node using the same protocol in each layer.
Inter-networking is a common requirement for the network layer, either between homogeneous networks (e.g. PSPDN in different countries) or between networks conforming to different standards.
The X.75 protocol is defined by the CCITT X.75 Recommendation to specify the packet-switched signalling system between PSPDNs.
The two entities involved in the signalling system are generally called STE (Signalling TErminal). To identify each of them specifically, the suffixes "X" and "Y" are used and the relevant interface is referred to as X/Y interface.
From a technical viewpoint the X.75 is designed to be a symmetric protocol (in terms of procedures), as no different role is assigned to a STEs on the same X/Y interface (e.g. no functional differentiation is made on the basis of direction of crossing the interface).
Due to the analogy of X.75 with respect to the X.25 protocol (from which it is derived), no further description is strictly necessary, except for minor technical details:

- besides the optional user facilities field, a new field is reserved for the network utilities.
- to minimize the risk of call collision, X.25 protocol fixes different rules to DTE and DCE in the use of the suggested mechanism for assignment LCI; as far as STEs are concerned, no mechanism is specified by X.75.
- some code used by X.25 for clearing/reset cause field is not applicable to X.75.

6. COMMERCIAL PSPDN COMPANIES

The commercial PSPDN companies are typically based in the United States, started in the 1960s as providers of computer time-sharing services
BT Tymnet and SprintNet are the leading pure PSPDNs in the United States, representing over 85% of the market [DATAPRO, 1987]. Also reviewed below is Infonet, a PSPDN with an established international basis.

6.1 BT Tymnet

The Tymnet PSPDN was created in the early 1970s by Tymshare, Inc., a leading computer time-sharing company. It was purchased by McDonnell Douglas, and in 1989 was subsequently sold to British Telecom International. The developers of Tymnet pioneered a number of important commercial developments in packet switching; so that Tymnet can be really considered the first PSPDN in the world.

The network can be accessed in 234 U.S. cities and 30 countries, and consists of over 1000 packet-switched nodes, each of which is interconnected with at least two others. This provides the alternate routing capability that gives packet switching its robustness. Packet assembly and disassembly are done with a proprietary structure, using virtual circuit routing. Tymnet is interconnected with the Canadian Datapac PSPDN and networks in Europe via U.S. international record carriers. Furthermore, Tymnet can be accessed through the domestic and international telex networks.

The packet-switched nodes in the Tymnet PSPDN are actually designed and manufactured by BT Tymnet. Consequently, it is a significant supplier of this equipment to private companies and common carriers. Interconnecting Tymnet nodes are leased lines, satellite links, and private microwave links. The network is used as a public utility, serving over 10,000 public access ports.

Users are basically charged for service in two ways. First, there is a charge per hour of connection time, analogous to telex service. The second charge is for the packets actually sent over the network. Users are charged for blocks of 1000 characters (kilocharacters), monthly bills being usage-sensitive (this means that the more kilocharacters you send, the lower the rate per kilocharacter). Also, there are peak-time and off-time rates to encourage use during nights and nonworking days. The combination of the two charges is typically less than using the PSTN would cost and possibly even the telex network, depending on the device speed. Users can employ Tymnet to reach information services on distant hosts, a capability common to many PSPDNs and E-mail networks. A third-party host provider using Tymnet (such as a financial information service) would pay a fee per user and per access port connected to the PSTN.

Tymnet is a flexible PSPDN in that the network can allow devices of dissimilar speeds and different character formats to communicate with one another by means of PAD services. The general arrangement and operation of Tymnet is very representative of other U.S. PSPDNs and those in different countries as well. Tymnet also provides X.75 gateways into external X.25 PSPDNs and private networks.

6.2 SprintNet

The SprintNet PSPDN was started by BBN, the manufacturer supplying packet switches for the original ARPANET. SprintNet has grown impressively to become one of the most sophisticated packet PSPDNs in the world. The company was acquired by GTE in 1979; subsequently, GTE transferred SprintNet to US Sprint, where it was combined with Uninet

PSPDN. In July 1989, SprintNet announced that it would be merged and fully integrated into US Sprint so that all sales, administration, and research and development (R&D) activities would be conducted as one entity.

SprintNet provides a broad range of data communication services and also is a manufacturer of packet-switching equipment. In 1986, there were 800 host computers and 100,000 terminals connected to the network. The traffic on the network for the year amounted to 16 milion domestic calls and more than a million international calls..To facilitate interconnection with PSPDNs in other countries, SprintNet has built on the X.25 protocol and interfaces using the CCITT Recommendation X.75 on internetworking. The first such interconnection was demonstrated with Datapac of Canada, a service of Telecom Canada, in 1980.
SprintNet nodes can perform PAD services so that most types of terminals, personal computers and modems, with a range of line speeds from 110 baud to 56 kb/s, can be used to access a distant host computer. Host computers are connected to the SprintNet PSPDN through dedicated access facilities, which individually would be a leased line from a local Telecom company and, if appropriate, a long-haul carrier.
One of SprintNet's strong points is the advanced stage of development of SprintMail, their E-mail offering.

6.3. Infonet

The Computer Sciences Corporation (CSC) introduced the Infonet PSPDN in 1968 to support the data communication needs of its time-sharing and software customers. Importantly, Infonet developed into an entity of its own with particular strength as an international backbone between users in the United States and Europe.
In 1988, CSC set a new strategic direction for Infonet by selling pieces of the network enterprise to PTTs. As of 1989, ownership is divided among: France Telecom, DB Telekom, Singapore Telecom, Telecom Australia, PTT Nederland, RTT Belgium, Telefonica of Spain, and Teleinvest of Sweden. CSC retains a minority interest.
Infonet provides packet-switched data communication services in a manner similar to the U.S. domestic leaders, BT Tymnet and SprintNet. Packet-switched nodes employ the X.25 network protocol and are connected primarily by leased digital private lines operating at 56 kb/s. The full range of user terminal devices and host computer systems can be connected to the network, with nodes performing whatever protocol or speed conversion is necessary. Many users employ Infonet to extend their IBM SNA information environments to diverse overseas locations. The access points support a variety of IBM devices that connect through the following protocols: SNA/SDLC, 3270 BSC, and 2780/3780 remote job entry (RJE). Of course, Infonet supports CCITT X.25 and XX.75 and asynchronous telex and teletypewriter attachments.

Infonet provides dedicated access lines for customer host computers. Occasional access to Infonet is primarily via the U.S. domestic telephone network, with points-of-presence in approximately 100 cities. What is more important is that Infonet has a substantial global presence, with direct access in 21 countries via dedicated data lines. This connectivity is increased to a total of nearly 100 countries. Furthermore, there are X.75 bridge connection to PSPDNs in 18 countries of the ensemble. Infonet's international backbone can be reached via the domestic PSTN in a particular country, a dedicated access line, or the domestic PSPDN.

6.4 GE Information Services (GEIS)

GE Information Services (GEIS), the telecommunication and information service organization established by General Electric Company, is a leading provider of PSPDN and associated enhanced services. GEIS is a worldwide operation, having grown rapidly over two decades. The service name of MARK*NET has been adopted for the PSPDN capabilities offered by GEIS.

Domestically, GEIS provides asynchronous dial-up service from 650 U.S. and Canadian cities. Leased line connections would be required for host access, for a variety of protocols including IBM SNA/SDLC and X.25. GE has supplied the bulk of the packet-switching node equipment and can perform a variety of protocol conversion services. Equipment such as data multiplexers and packet nodes can be installed on the customer's premise if required for the particular application.

GE maintains a full-time staff and major processing node in Amsterdam, the Netherlands. This facility has the capability to monitor the European and transatlantic networks. There are leased trunk lines and nodes deployed in some 30 countries, which are also interconnected with 70 PSPDNs worldwide.

Network service are charged in the typical way, with customers paying for connection time and for kilocharacters transmitted or received. Dedicated access lines for hosts and other applications like point-of-sales terminals are charged on a monthly basis.

GEIS has long been a major force in teleprocessing and information services. Before the personal computer was invented, GE was the principal supplier of engineering and scientific computing services. The worldwide extent of the network has become attractive to strategic users needing a well integrated data communication medium. In this context, GEIS is a viable competitor to Infonet.

7 PSPDN OFFERINGS OF THE PTTs

While the first PSPDNs to use packet switching were implemented in the United States, other nations were quick

to adopt the technology. By 1988, PSPDNs were operating in more than 100 countries, many with 100 or more nodes.

7.1. PSPDNs in Japan

At the foundation of Japanese networking is the major telecommunication provider, Nippon Telegraph and Telephone. Since 1981, NTT has offered packet-switching services to business customers, particularly financial institutions. International packet-switched network services are provided by KDD as a part of their established position as a Japanese overseas carrier. Under the trade name VENUS-P, KDD served 14,000 business and individual users in 1988 with packet-switched data transfer at 9600 b/s on international lines to 43 regions of the world. Users can be provided with electronic mailboxes for storage of E-mail message, and access is provided to U.S.-based public databases and information services. Charges are assessed separately for connection and volume of data transmitted. KDD provides full-time access lines to the customer's premises via NTT's network at a variety of speed between 1200 kb/s, for which there are separate charges for initial connection and each month of access.

Other PSPDNs have been established since deregulation opened the market in 1985, primarily through joint ventures of Japanese and U.S. companies; an example is the Fujitsu Enhanced Information and Communication Service (FENICS). Based on packet-switching principles, FENICS provides a wide range of services from basic data communication to more specific applications. In particular, Fujitsu has been focusing on the financial community, where more sophisticated applications are desired As with U.S. PSPDNs, FENICS can be reached by dial-up through the PSTN or via dedicated access lines. The network can provide real-time connections with protocol and speed conversion. Customers may connect host computers to the network and offer information services to other users. In addition, E-mail services are offered through a store-and-forward capability.

7.2 Canada's Datapac

The Canadian PSPDN, Datapac, is operated by Telecom Canada, one of the two terrestrial long-haul carriers in Canada. The Datapac network began service in 1977 and provided the foundation for the development of packet-switching products by Northern Telecom. Although users may access the network with the virtual circuit mode of X.25, packets are actually routed as independent datagrams. The network architecture ensures that packets are all transmitted correctly and arrive at the distant port without loss and in the proper sequence. A user who access the Datapac network will "see" the same end-to-end virtual circuit connection as in SprintNet. Between the nodes are fixed 56 kb/s data circuits maintained by Telecom Canada as part of its long-haul network.

The rates for using Datapac are volume and distance-sensitive; there are additional charges for installation and special features. The access charge is dependent on the service provided, and usage charge are based on the number of packets transmitted.

7.3 PSPDNs in Europe

In Europe, all major Administrations have made a clear commitment to PSPDN, as soon as the trends became clear. Notable PSPDNs are Transpac in France, PSS in United Kingdom, Datex-P in Germany, and last not least, Itapac in Italy.

7.3.1 Transpac

France Telecom claims that Transpac, its domestic PSPDN, is the world's largest and most successful packet-switched data network. The network celebrated its tenth anniversary in 1988. In 1987 alone, Transpac experienced and annual growth rate of 38%. A total of 12,000 dedicated customers use the network, with 60,000 direct accesses as well as 3120 accesses via the PSTN and telex networks. Approximately 50% of the packet traffic on Transpac is generated by the Teletel videotex service .

The Transpac packet-switched architecture uses the virtual circuit mode of CCITT Recommendation X.25. Each network node has at least two permanent data lines connected, which operate at 72 kb/s. This arrangement ensures that therefore is diverse routing for any access point and destination. France Telecom announced in 1988 that access to Transpac is being expanded through dial-up facilities. Traditionally, only the X.25 interface was supported. The new access facilities include dial-up asynchronous access with error correction or with V.32 modems. An international gateway from Transpac using Recommendation X.75 is available through France Telecom. This is in addition to the international data transport facilities provided in France by Infonet .

Expansion and improvements in the capabilities of Transpac are common occurrences. France Telecom is pursuing a growth policy by expanding capabilities in E-mail via this packet-switched network. As an example, the Transpac network is the backbone for EDI market within France.

7.3.2 Itapac

Itapac is the Italian public packet switched data network. Since the beginning, 6 nodes have been installed in cities as Milano, Torino, Firenze, Roma and Napoli; several (at least 1 for region) PADs have been installed throughout the

Italy. A fast increase of the number of components (mainly for the PADs) is expected in the future.

For physical level, the network offers the X.21bis interface. In the future, the X.21 interface will be available as well. Line speeds available are 2400 b/s, 4800 b/s and 9600 b/s (300 b/s and 1200 b/s for X.28 accessing). For Link level, the LAPB procedure is used. The following values of HDLC parameters apply:

maximum number of bits in a I frame = 1080
maximum number of outstanding I frames = 7
maximum number of attempts to complete a transmission = 10
sequence numbering of frames = modulo 8

For Packet layer level, Itapac performs modulo 8 as the sequence numbering of packets and 128 octets as packet size. Both SVC and PVC services are provided in order to allow the multiplexing of logical virtual connections between pairs of DTE connected together via Itapac. The optional user facilities offered by the Italian network are a subset of those specified by CCITT. They include Reverse charging, Closed user Group, Incoming call barred, Outgoing call barred, one-way logical channel incoming, one-way logical channel outgoing, throughput class negotiation and hunt group.