18. INTEGRATED SERVICE DATA NETWORKS

18.1. Introduction

Currently used telephone network is the result of about one hundred year evolution. In the last 20 years major part of analog transmission lines was worldwide replaced by digital lines based on PCM (Pulse Code Modulation) and traditional analog public switch exchanges are also replaced in increasing pace by fully digital exchanges. 8 kHz sampling rate and 8 bit quantization has become a standard for digital transmission within the voice channel (300 Hz - 3.4 kHz), thus requiring 64 kbit/s bitrate. (Note: 64 kbit/s = 64000 bit/s, not 2^{16} = 65536!)

In many aspects, data transmission networks have evolved independetly of the telephone network although they frequently use ordinary telephone voice channels as a medium for low bitrate data transmission. In the last two decades, wide spread use and increased capacities of computers, remote data processing, centralized data bases, etc. resulted in enormous increase of demand for data transmission. Besides the increasing number of users new services and still growing need of higher transmission capacity appear. Because of the above reasons it seemed quite logical in the 80's to develop the digital telephone network to be suitable not only for voice transmission but also for various data transmission services within a single common network. (Note: a kind of data transmission or the PSTN are network services, as opposed to various postal services, e.g. exact time, access to a database etc.) Since the establisment of the subscribers' network is rather expensive, the most economical solution is to transmitt the speech and data signals via the same cables which -in majority of cases- are already at disposal. Fortunately, majority of existing subscribers' lines shorter than 5 to10 km can be used without any special problem up to about 200 kbit/s so that the use of data transmission services upon the current telephone network is attractive, provided the exchanges are suitable for data transmission switching, too. Traditional exchanges, of course, are unsuitable, the up to date digital exchanges, however are designed with respect to this demand.

This development strategy is known as Integrated Service Data Network (ISDN). Main driving power is the rapidly increasing demand for data transmission of subscribers with high traffic -offices, enterprices, banks, etc.- and the fast progress of digital solutions able to satisfy them, moreover the new technologies are inducing new de3mands, as well. While in the developped countries the telephone demand is growing just moderately, demands for data transmission are growing pretty rapidly. One can expect that in the future primary role of the telephone network capable of data transmission will change to opposite, i.e. to the role of a data network giving also possibility to make a call. PCM transmission systems were applied first for the extension of trunk lines used in point-to-point connections of analog exchanges. Using time division multiplex digital channel between analog exchanges proved to be economical even if analog-to-digital and digital-to-analog conversion has to be performed at the trunk ends. Between digital exchanges, conversions are of course not necessery so that not only the additional quantization distortion can be eliminated but digital signals can also be switched and transferred via the exchange.

Basic precondition of a data transmission service is the full, 'from custumer to customer' digitalization of the network, i.e. analog transmission lines and/or exchanges should not be in the signal path. Since telecommunication equipments are designed to work over some decades, the replacement of analog exchanges is a circumstancial process and the ISDN can be installed only step by step. In the beginning, true ISDN services will be provided just within small areas,, for some subsribers of great traffic demand which can later be expanded to extent that anybody requiring such a service could reach it.

As long as the data transmission based on one ore some 64 kbit/s channels is adequate, the above approach seems to be correct. However, even today's demands for high-speed (10 to 100 Mbit/s) data transmission and various image retrieving services (tv-transmission, image databases, etc.) are urging development of a so called Broadband ISDN (B-ISDN) which is technologicaly based on extra high-capacity fibre optic systems which become cheaper year by year. Thus it is very probable that besides the Narrowband ISDN (N-ISDN), as the ISDN based on 64 kbit/s rate is called, B-ISDN will rapidly expand. Even recently the designers and manufacturers already focus on B-ISDN, hence a enormous evolution can be expected in this field.

Of course, great number of designers as well as manufacturers is contributing to the establishment of world-wide networks so the standardization of interfaces has become one of the most important clues to the development. This does not simply mean that the system blocks manufactured at different plants have to be connected are able to communicate with each other but implies also possibility of simple replacement or exchange of older equipments by the newest elements.

In further part of this chapter the narrowband ISDN concept, some of its interfaces and their operation and network synchronization problems are discussed.

18.2. The ISDN concept

Fig. 18.1. presents the connection of already existing and independently operating networks to the ISDN. The so called Interworking Functions denoted on Fig. as I indicate that ISDN is in tight connection with the formerly formed networks. ISDN integrates all the functions of them and provides further services to subsribers connected to it. The transmission paths among callers and called parties, however, has to be fully digital. If only one analog line or an analog exchange is inserted into the transmission path services based on 64 kbit/s transmission are obviously not accessible.

TE1 is a Terminal Equipment satisfying the ISDN standards and it is directly connected to standardized interface S. This is a bus-type interface to which several equipment can be connected in parallel. The bus architecture is such that speech, data, image, etc. services can be connected to a single connection and used simultaneously to a certain extent.

Terminal Equipment TE2 is a traditional or simpler subscriber equipment (e.g. traditional fax machine) which has to be also connectable to the network. For this purpose, Terminal Adaptor TA is used.

Further significant feature of the ISDN is that the signallings between the subsriber and the network or between two network nodes are transmitted as messages. In the analog and in the PCM systems, Channel Associated Signalling (CAS) ordered to each channel has become the solution to that. Opposite to that, Common Channel Signalling (CCS) is name of the solution used in ISDN. CCS allocates a single 64 kbit/s channel for the transmission of signalling of several connections. These signallings make possible the call set-up and also contribute to the maintenance of the whole network.

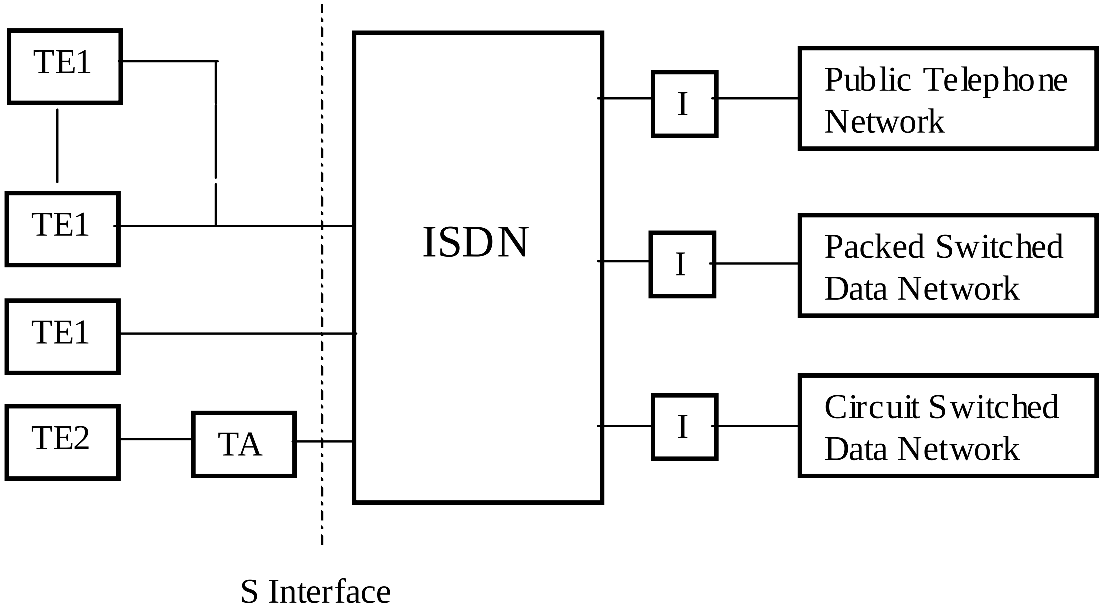

Fig. 18.1. ISDN Connection to Subscribers and to Former Networks

ISDN can be widely used, besides traditional services (telephone, telex, fax, etc.) a great variety of new services can be implemented within a unified system by ISDN. Let us give some examples:

- electronic mail, message handling, electronic mailbox,
- videophone, videoconferencing,
- data transmission (data and file transfer between computer centers and terminals, information services, access to databases, virtual local area network development, etc.),
- mobile communication (car, ship, handy phone set, etc.),
- tranmission of still pictures,
- public radiopaging systems,
- remote control, remote sensing.

In majority of the above examples 64 kbit/s rate or multiple of it is neccessary to offer a really attractive, really useful service to the subscriber. E.g., supposing that daily data of a stores chain make 1 Mbyte, a file of these data can be transmitted within about 2 minutes by 64 kbit/s rate while it would engage the line for more than 108 minutes if using 1200 bit/s modems and an analog channel.

Standards are of special importance in every technical activity and this is especially true for the area of international telecommunication. For every essential parameter of ISDN connection and operation, CCITT is releasing recommendations regarded as standards. These are based partly on recomendations accepted for telephone and PCM systems and partly on those accepted for data transmission. Two standard forms of how the interface S is handling the incoming subscriber lines exist:

- S_0: Basic Rate Access (BRA)
- S_0: Primary Rate Access (PRA)

In basic rate access two channels 64 kbit/s each are assigned at S_0 and both called B channel. Besides the two B channels, a 16 kbit/s signalling channel, denoted as D and providing for the call set-up and for signalling during the call. For that, this subsciber line is called also the 2B+D channel.

18.3. Basic Rate Access (BRA)

To operate with basic rate access, 2B+D channel has to be extended up to the subscriber. Transmissions using BRA must work also on existing two-wire lines, replacement of the current cable networking is obviously unthinkable. Functional units required for that are shown on Fig. 18.2. (The units are not neccessarily individual equipments since more functions can be implemented within one equipment.) Standardized interfaces are also shown on the figure.

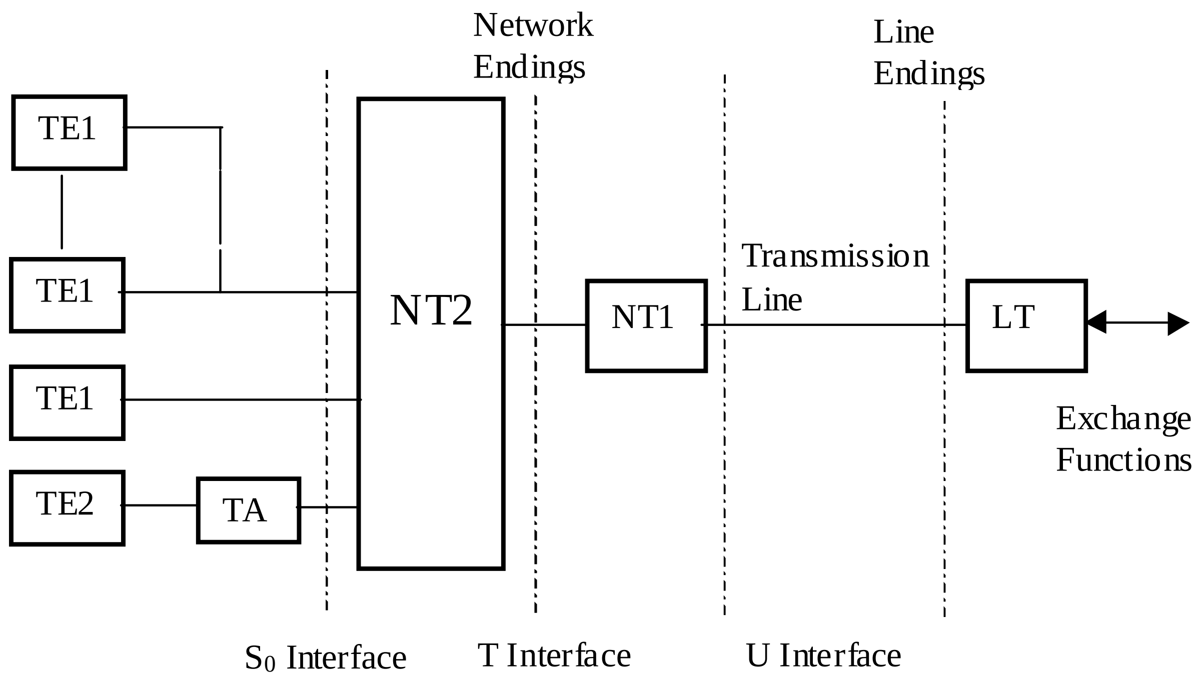

Fig. 18.2. Functional Units of an ISDN Interface

Unit U is the interface between the Line Termination (LT) and Network Termination (NT) through which the digital signal are passed depending on the cable type. Inspite of the poor quality of twisted-pair cables used in the current two-wire analog telephone network, transmission can be realized even on them. Since in this case the cable is used in duplex mode, up-to-date adaptive filters have to be used to eliminate echoes caused by the reflections. The transmission rate is 160 kbit/s since besides the 2B+D channel (2*64+16 = 144 kbit/s) used for information transfer and signalling, an overhead of 16 kbit/s is required for synchronization and maintenance functions.

Units LT and NT1 hence provide for tasks corresponding to actual transmission and in functional decomposition belong to the so-called physical layer.

Unit NT2 is reponsible for the tasks of layers 2. and 3. called the data link and network layer, respectively. furthermore it handles the physical layer at the interface S_0. Functional units NT1 and NT2 may be even drawn together and interface T in between them acts as the limit between the public network and the subscribers' system. E.g., NT2 can be an exchange or a Local Area Network (LAN), carrying the local traffic and offers a connection to the public network. In the absence of interface T, interface S acts as the limit.

Interface S_0 offers possibility of connecting eight Terminal Equipments (TE) at maximum. Maximum distances are limited by the cable attenuation and propagation delay. Depending on cable, the length of a multiple-access passive bus can be about 100 to 200 m (2 μs delay at maximum) and -to keep off erorrs due to mismatches- cable between the bus and theTE shall be not longer than 10 m. In point-to-point connection 6 dB is the specified maximum of attenuation corresponding to a distane of about 1 km. The signal transmission rate is 192 kbit/s since besides 144 kbit/s, another 48 kbit/s is used for synchronization, maintenance and multiple-access control.

Multiple access on the bus S_0 means the statistical multiplexing of the signal sources. Transmission rate of simultaneously transferable information is limited by the 2B channel to 128 kbit/s. An important question is how many equipments may be connected to the bus and how big their traffic without significant delay or with an acceptable loss and given maximum delay can be. These questions are, in general, treated by queueing theory. In the following the simplest case is presented.

18.3.1. Access Delay

Suppose that terminal equipments generate at random λ demands on average within a unit of time. These can be either calls of different length or packets. If λ is constant and probability that a demand arrives within time interval Δt is $\lambda\Delta\, t1$ then arriving can be modelled by Poisson process. In such a case, probability that k demands arrive within time T is given just by the Poisson distribution:

$$P(k) = –\exp(-\lambda T), \quad \text{where } k = 0, 1, 2, \text{ and } P(k) = 1 \qquad (18.1)$$

Thus the expected number of demands arriving within time T is

$$M(k) = k\, P(k) = \lambda T \;\; –\exp(-\lambda T) = \lambda T, \qquad (18.2)$$

which can be directly seen from the above conditions.

In the case of a Poisson process, distribution of the time between two consecutive arrivals

$$F(\tau) = 1 - \exp(-\lambda\tau) \qquad (18.3)$$

is called exponential distribution. Probability density function of this distribution is

$$f(t) = \frac{dF(t)}{dt} = l \cdot e^{-lt} \qquad (18.4)$$

and the expected value of time between two arrivals is

$$M(t) = \int_0^\infty t\, l \cdot e^{-lt}\, dt = \frac{1}{l} \;. \qquad (18.5)$$

E.g if 10 packets arrive within 1 s, then the expected value of time between two packets is 1/10 s which corresponds to natural expectation.

If $l > m$, i.e. the number of demands arriving within a unit of time is greater than that what is (could be) serviced then the demands are congested, theoretically without no limit. If the usage $r = l/m < 1$ then all demands can be serviced but because of their random arrival, a waiting queue is forming, as well. Let $P(n)$ be the probability of n waiting demands. To determine this probability let us see the state diagram shown in Fig. 18.3. where the n-th state represents n wainting demands.

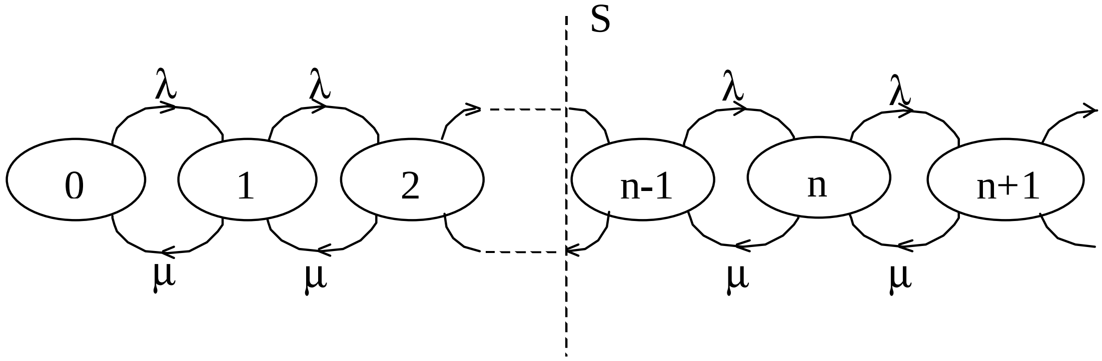

Fig 18.3. State Diagram of Queuing

During a short time Δt, the length of the queue grows by one with probability $\lambda\Delta t$ or it shortens by one with probability $m\Delta t$. (Simultaneous arriving and service of demands and multiple arrivals are neglected here because of short Δt.) If values of λ and μ are taken as constant the system shall get into a state of equilibrium, e.g. at the interface S the probability of .entering and leaving become identical:

$$l\ P(n\text{-}1) = m\ P(n) \tag{18.6}$$

This could be applied for the other states, too, so that

$$P(n) = r^{\mathrm{n}}\ P(0) \tag{18.7}$$

Taking into account all the states

$$\sum_{n=0}^{\infty} P(n) = P(0) \sum_{n=0}^{\infty} r^{n} = P(0) \frac{1}{1-r} = 1 \tag{18.8}$$

so that

$$P(n) = r^{\mathrm{n}}\ (1 - r). \tag{18.9}$$

Expected value of the length of the waiting queue is thus

$$M(n) = \sum_{n=0}^{\infty} n \cdot r^{n} (1-p) = \frac{r}{1-r}\ . \tag{18.10}$$

E.g., if the usage is 80 per cent, i.e. $\rho = 0.8$, then 4 demands will be waiting on average. It can be also well seen from the equations that as the usage approaches 100 per cent, the length of the queue is rapidly growing.

Assuming L byte long messages on average, a 64 kbit/s B channel service rate is

$$m = \frac{64 \cdot 10^{3}}{8 \cdot L}\ \text{message/s} \tag{18.11}$$

and it is the consumer's task then to adapt his demands to this value, i.e. to decide how much of the demands he wants to satify ($\lambda < \mu$) and whether he could tolerate waiting given by above equations. But one must not forget that real conditions greatly differ from those stated in idealized model used in derivation, e.g.. waiting time, influences generation of new demands, etc.

18.4. Primary Rate Access (PRA)

PRA is meant as a primary signal connection of 32·64 = 2048 kbit/s rate since consting of 32 channels, each of 64 kbit/s rate. 30 of them are data channels, one is for signalling and on another one frame synchronization and maintanence information is tranmitted. As previously, the signalling channel carries the information for circuit switching, i.e. for seting up and cancelling the connections. Frame synchronization is required by the TDM, maintenance bits serve for alert and other auxiliary functions.

PRA systems introduced in USA and Japan are different from the above described europen one. They have 24 channels with the total bitrate of 1544 kbit/s and although there are many other differences, the two systems could be connected via appropriate intefacing circuits. The following part deals only with the european system.

Primary frame is built upon a byte-by-byte TDM of channels (see Fig. 18.4.). Repetition of 8 bits in a time slot with 8 kHz results in a 64 kbit/s channel. Each time slot carries a speech signal or another information, e.g. data.

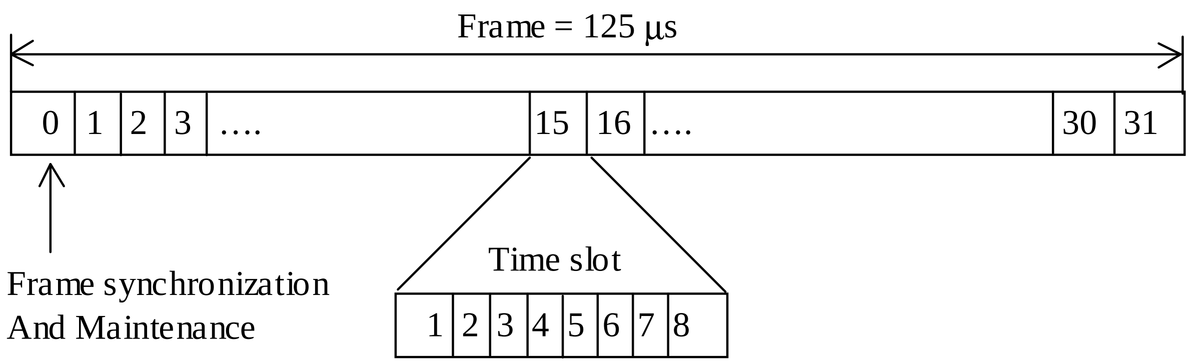

Fig. 18.4. Structure of a Primary Frame

Time slots and bits inside them used to be denoted as IR0 ... IR31 and B1 ... B8, respectively, where B1 is the Most Signigicant Bit (MSB). The maintenance bits are transmitted in slot IR0, whereas bits B2 ... B8 are fixed (00011011) in every second frame and serve as frame synchronization codeword. In other frames value of bit B2 is 1, so that a constant data could not coinceede with the frame sync codeword. This is important for the receiver for to recognize the codeword and to restore the frame synchronization.

IR16 is the signalling channel carrying informations for setting up and cancelling the connections.

18.5. Clock Synchronization, PDH and SDH Systems

In a PCM transmission, clock frequencies must be the same at the transmitter and receiver. Since the clock frequency can be directly recovered from the bitstream in the receiver, it is not neccessary to provide an extra channel for the clock transmission. In the case of a point-to-point connection it can be relatively easily realized. If, however, digital signals arriving from two ore more points have to be multiplexed and their clock frequencies are equal just in nominal sense, bitrate must be controlled by *justification*, i.e. by inserting 'empty bits'.

Clock or bit synchronization may refer to one ore more point-to-point connection or even to a whole network. A network is called *plesiochronous* if its units operate with their own independent clocks, frequencies of which may even differ to a certain specified extent from the nominal value. On the contrary, in synchronous networks, all clock frequencies are identical since a separated synchronization system supplies the clock signals usually using a common master oscillator. The major part of the current telephone network is not a synchronous one but a PDH (Plesiochronous Digital Hierarchy) system capable of effective communication just in point-to-point applications. In case of more complex network topology, branching, time-slot switching, reconfiguration, etc. is becoming difficult and costly. Thus the first digital solutions developped on the ground of plesiochronous systems had to cope with problems of full frequency and phase equalization. In switching systems which manage independent clock signals, phase shifts get cummulated regardless on how accurate incoming clock signals are. When the shift reaches the value that can be handled by the justification buffer , a frame (125 μs) must be either dropped or repeated (called a slip).Thus, if signals incoming to an exchange are not synchronized, their accuracy (which for PDH modules is typically 10 to 50 ppm) must be strongly improve to reduce the slip number to an acceptable level. E.g., if the stability of two digital exchanges operated ,from independent clocks is as good as 10^{-11}, then a slip shall occur only in every 70 days.

In speech transmission, a one-frame slip may cause at most, an audible click which does nor disturb a call too much. This is not the case, however, for rapidly expanding use of data and other, eventually compressed or encrypted information. Since in such cases even seldom slips may have serious consequencies, it is reasonable to gradualy extend synchronous operation in wide and wider area.

At the beginning of '70s, roughly parallel to standardization of PDH systems, CCITT has standardized sychronous mode digital concentrators, as well. There were, however, several reasons why they have not been implemented in practice. Temperature dependent changes of the group delay of cable systems, accuracy and realibility of clock generators, stability of synchronization systems, bridging the clock dropout periods, etc. induced serious technical and economical problems for which technology of the '70s had no answer. Opposite to that, PHD was relatively simple, and robust against errors.

Recently the situation has, however, significantly changed and several factors convince about the perspectivity of synchronous networks:

- Instead of point-to-point transmission, meshed network of digital exchanges is spreading in wide and wider area.
- Traditional copper wire cables are step by step replaced by optical systems, having much smoother envelope delay.
- Wide spread use of digital switching resulted in nodes operated by accurate and highly realible clock signal.
- Developments in circuit technology made enable application of highly flexible (long) buffers and complex synchronization systems.
- While the speech is not too sensible to phase slips, major part of the new data type services is unreliable because of this unavoidable feature of PDH systems.

Because of this reasons, switching from plesiochronous to synchronous networks has already started and is under intensive construction. To facilitate this process, CCITT has even standardized a new hierarchy of multiplexing, called the Synchronous Digital Hierarchy (SDH) which brings about following advantages:

- world-wide unified standard, specifing a base bitrate (155.52 Mbit/s) and multiples (4, 16, ...) of it,
- data packets can be retrieved from synchronous frames without their decomposition, simplifying thus the multiplexing, branching and switching equipments,
- network maintenance information, role of which rapidly increases with the growing size and complexity of networks, is included,
- a network can be easily reconfigured without manual operation, by virtual networks can be set up or cancelled with high flexibility,
- using ring networks, rerouting to a reserved path in the of an error can be solved automatically and in economical way.

On the basis of all advantageslisted above it is probable, that the future B-ISDN shall also be founded on SDH as substrating network. This is suitable also for the transmission of signals used in PDH systems, however the main transmission mode will be the ATM (Asynchronous Transfer Mode) which is similar to packet switching but it is based on transmission of cells, i.e. fixed length packets. ATM is accomodated to efficient transmission of burst-type information. This is a obvious requirement for data transmission services, however, efficiency can be increased even in speech transmission since during pauses in speech, the channel can be used by other service(s). Such a method was already used earlier but real statistical advantages are much more significant for the greater channel-number of broadband systems.

A universal network of data, speech, video and other various services, majority of which can not be even foreseen is bringing within reach new horizons in the development of telecommunication and computer science, the driving forces of evoluation of an information society.

Control Questions

1. What is the condition for the subscribers to have an access to ISDN services? List some of these services!
2. Where are the physical and operational boundaries between an ISDN service supplier and a subscriber?
3. What is the condition for a queuing system to reach the state of equilibrium?
4. Why it is impossible for a data channel to disturb the sychronization of the primary frame?
5. Is it economical to transmitt asynchronous data on synchronous network?

Example

Each of N = 80 data terminals is sending every t_0 = 10s on average a message (a packet) of m = 800 bytes on average. What is the expected length of the waiting queue and how much is the average waiting time provided the packets are transmitted via a 64 kbit/s data concentrator?

Solution: $\lambda = N/t_0 = 8$ demands/s and $\mu = 64\cdot10^3/8m = 10$, thus $\rho = \lambda/\mu = 0{,}8$, i.e. $M(n) = 4$. Waiting time t_W corresponds to the time required to service the queue, i.e. $M(t\mathrm{w}) = M(n)/\mu$ = 0,4 s.

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Abbreviations

ATM	Asynchronous Transfer Mode
B-ISDN	Broadband ISDN
BRA	Basic Rate Access
CAS	Channel Associated Signalling
CCITT	Comité Consultatif International Télégraphique et Téléphonique
CCS	Common Channel Signalling
ISDN	Integrated Services Digital Network
LAN	Local Area Network
LT	Line Termination
MSB	Most Significant Bit
NT	Network Termination
PCM	Pulse Code Modulation
PRA	Primary Rate Access
PDH	Plesiochronous Digital Hierarchy
SDH	Synchronous Digital Hierarchy
TE	Terminal Equipment