Chapter 18
Multimedia Networks

18.1 Network Types:

Multimedia communication involves either person to person (PC to PC) or person to system (PC server or TV set top box to e–library). Multimedia involves text, images, audio and video (clips and movies). To transfer multimedia files, we need to consider 5 basic types of networks:

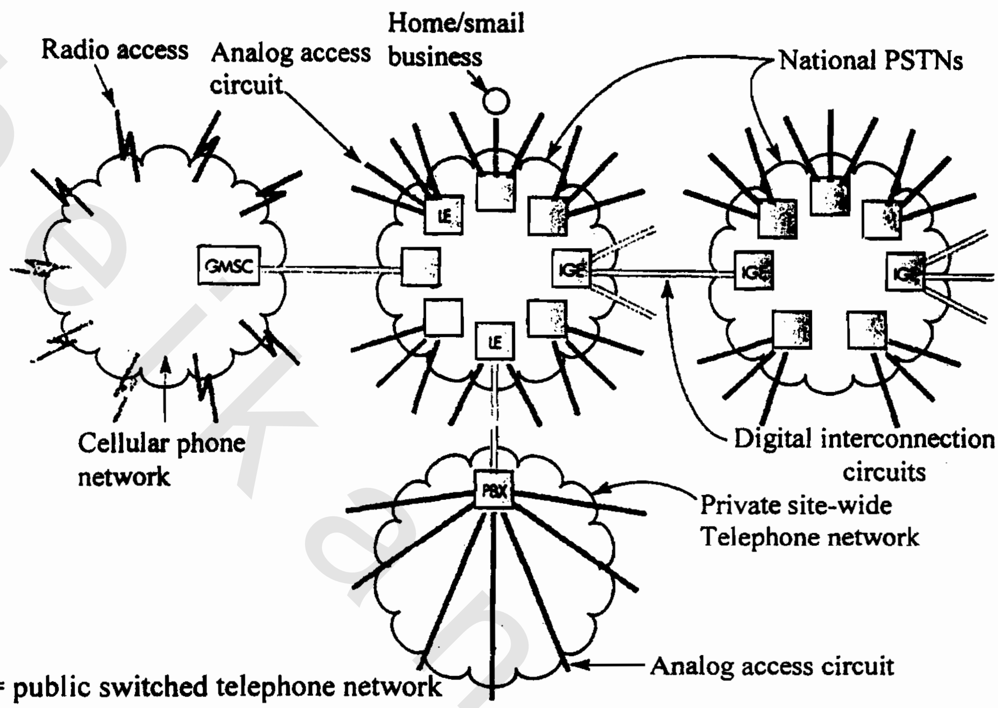
1. Telephone networks both stationary and mobile (cellular).
2. Internet.
3. Broadband TV networks.
4. Integrated services digital network (ISDN).
5. Broadband multi service networks such as DSL (digital subscriber line).

18.2 Telephone Networks:

Plain old telephone service (POTS) is known as public switched telephone networks (PSTN). Switching here means interconnectivity upon request between any two subscribers both nationally and internationally. Telephones located in the home or in small businesses are connected to their nearest local exchange. Telephones in large businesses are interconnected in private branch exchange (PBX), which provides a free switched service between any two telephones connected to it. The PBX is then connected to the nearest local (public) exchange.

Thus, the PBX is concentrated to the PSTN. The switches used in a cellular phone network are known as mobile switching centers (MSCs). These like a PBX are also connected to a switching office in a PSTN, which enables both sets of subscribers to make connections. International calls are routed and switched by international gateway exchange (IGES) (Fig. 18.1).

Telephone networks in circuit mode entail that for each call a separate circuit is set up through the network for the duration of the call. Access circuits link telephone hand sets to a PSTN to carry the two way analog signals associated with a call. Analog circuits can still be used for digital signals using modems (Fig. 18.2). Hence by using a pair of modems one at each subscriber access point a PSTN can be used to provide a switched digital service. High speed (HS) modems (in excess of 1.5 Mbps) are also available for multimedia applications (Fig. 18.3).



PSTN = public switched telephone network
GMSC = gateway mobile switching center
IGE = international gateway exchange
LE = local exchange/end office
PBX = private branch exchange

Fig. (18.1) Schematic telephone network

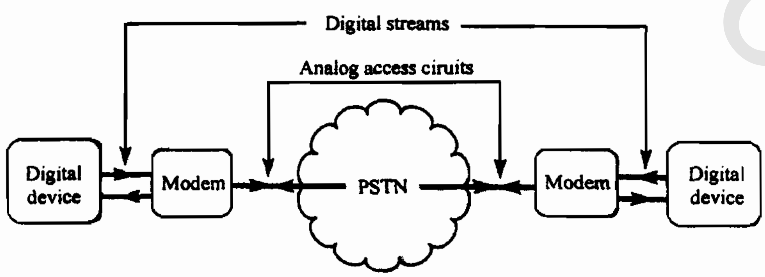


Fig. (18.2) Digital transmission in analog network

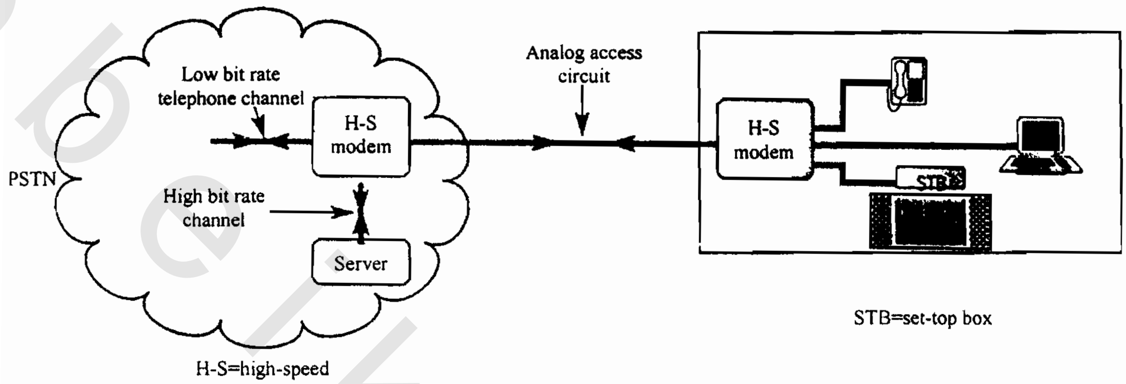


Fig. (18.3) HS modems

18.3 Internet:

Access to the internet is usually provided through an intermediate service provider ISP network for users that use the internet intermittently. The user devices are connected to the ISP through a PSTN with modems (dial up connection) or through an ISDN or even DSL for higher bit rate.

A single site may comprise a local area network (LAN). For an enterprise with network comprising multiple sites, the sites are connected using an intersite backbone network. Such network is called intranet. Different types of networks are all connected to the internet backbone network through gateways responsible for routing and relaying all messages to and from the connected network. Such gateways are called routers (Fig. 18.4).

All data networks operate in packet mode. A packet is a container for blocks of data (text – audio – video) with varying time intervals between each block. At its head, is the address of the intended receiving PC and is used to route the packet through the network. Telephony may also be supported by internet protocols (IP). This is called voice over IP. This is done by digitizing the speech using special software or sound card and a microphone. Also, PC videoconferencing may be supported using a camera, video card and digitizing hardware and software. This way packet mode networks and the internet in particular can support not only general data communication applications but also a range of other multimedia communication applications involving speech, audio, and video.

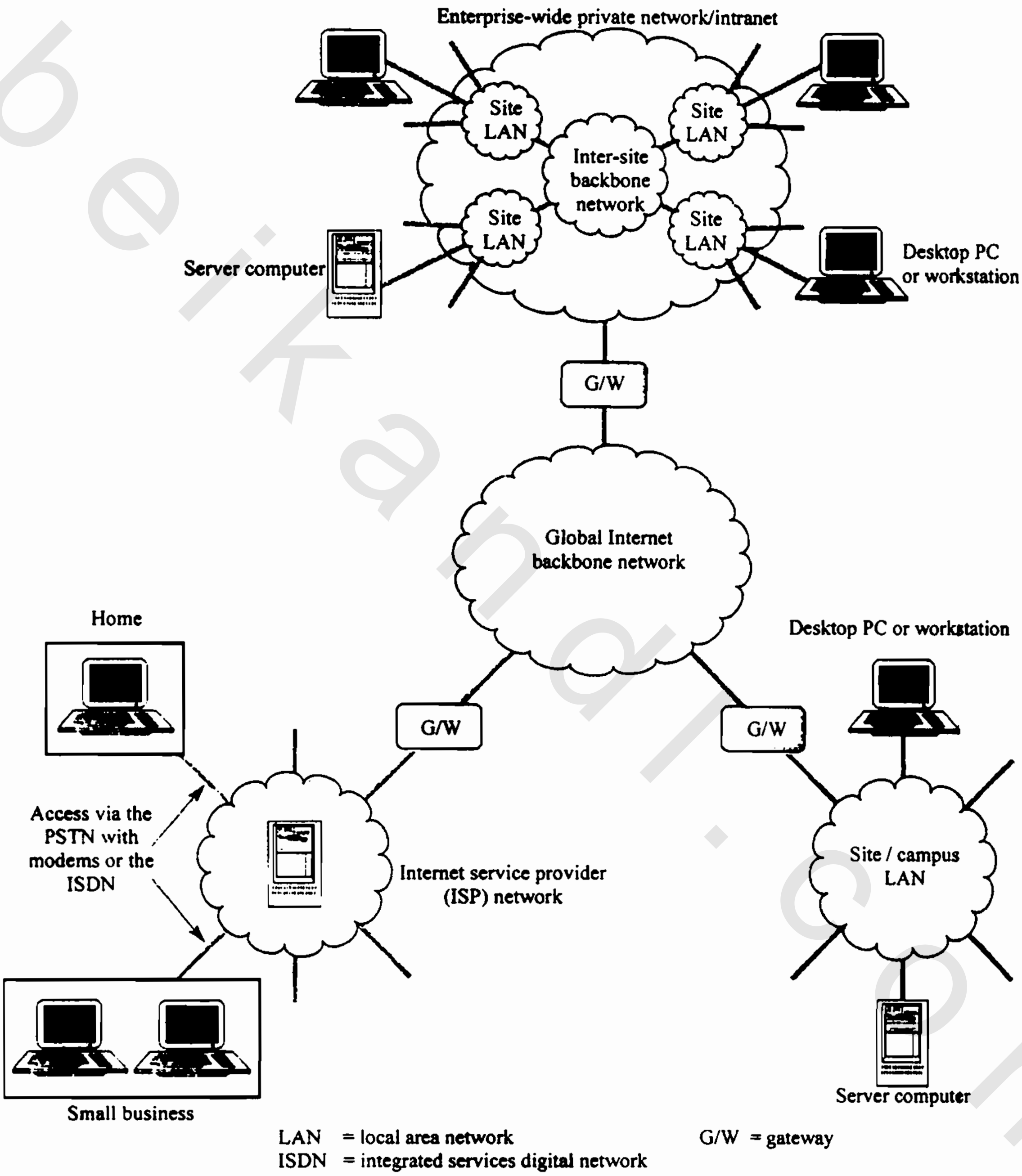


Fig. (18.4) Internet backbone

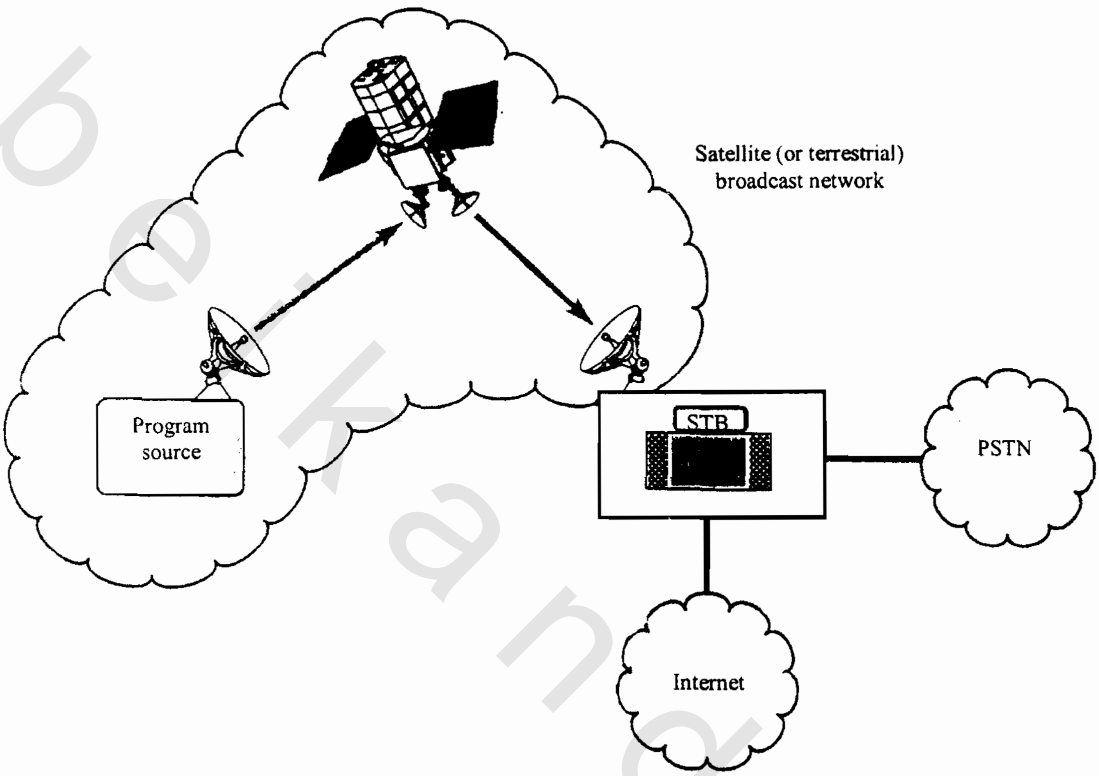


Fig. (18.5) Interactive satellite TV systems

18.4 Broadcast TV Networks:

Satellite network use primarily digital TV services (Chapter 19). Occasionally, a low bit rate return channel is used for interactive purposes through a set top box (STB). Typically, the STB is used to connect the subscriber to a PSTN and the high bit rate channel to connect the subscriber to the internet. Hence, in addition to providing basic broadcast radio and TV services, the high speed PSTN modem is integrated into the STB to provide the subscriber with an interaction channel, so enhancing the range of services these networks can support. This is the basis of interactive TV (Fig. 18.5). Cable networks may also be used for interactive multimedia networks. (Fig. 18.6).

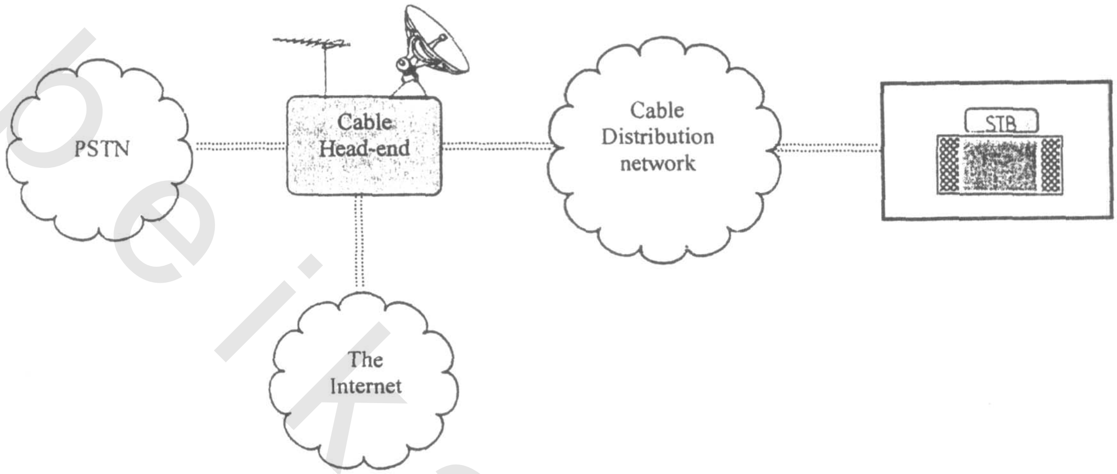


Fig. (18.6) Broadcast cable network

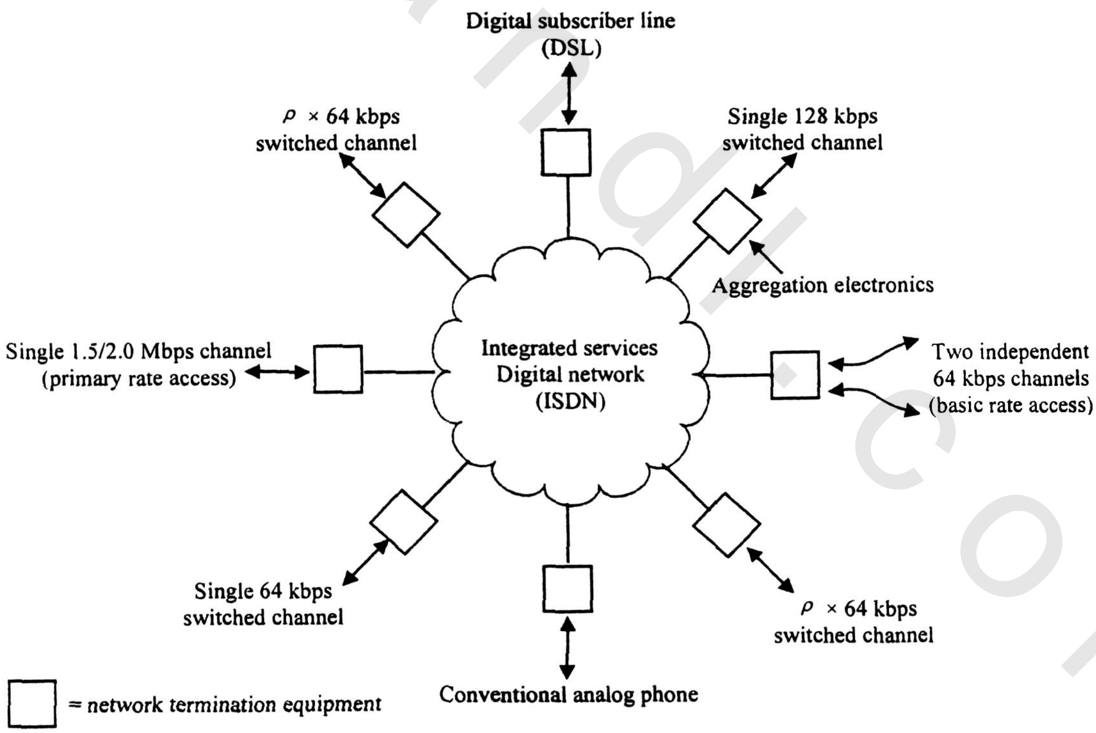


Fig. (18.7) ISDN networks

18.5 ISDN:

The ISDN network incorporates both telephone calls and data calls simultaneously. The subscriber telephone may be either digital or analog. In either case, the handset or the electronics in the network termination equipment makes the digital mode of operation of the network transparent to the user. The bit stream is normally at a bit rate of 64kbps.

The basic digital subscriber line (DSL) of the ISDN is called basic rate access (BRA) which supports two 64kbps channels. The two channels may be independently used for separate calls or may be combined through an aggregation electronics into one 128kbps channel.

Higher bit rate channel of 1.5 or 2Mbps may be achieved and is called primary rate access (PRA). A nonflexible way of obtaining a switched *p*X64kbps services is provided, where *p* = 1, 2….30 (Fig. 18.7).

18.6 Broadband Multiservice Networks:

The term broadband is used to indicate that the circuits involved use bit rates in excess of 2Mbps (30x64kbps) provided by ISDN. As such, they were designed to be an enhanced ISDN or broadband integrated services digital networks (B – ISDN), whereas conventional ISDN is called narrowband ISDN (N – ISDN). Multiservice networks imply that the network must support multiple services. Different multimedia applications require different bit rates depending on the type of media involved.

The switching and transmission methods in these networks must be more flexible than those in PSTN or ISDN networks which were connected with a single type of service. Digital information is integrated together into a binary stream which is divided into multiple fixed size packets or calls.

Since different multimedia applications generate all streams of different rates, this mode of operation means that the rate of transfer of calls through the network also varies. This mode of transmission is called asynchronous transfer mode (ATM) networks. There are ATM local area networks and ATM metropolitan area networks on a city level (Fig. 18.8).

18.7 Fax Communication:

This type of communication involves the use of a pair of fax machines. To send a document, the caller keys in the telephone number of the recipient and a circuit is set up through the network in the same way as for a telephone call. The sending machine starts to scan and digitize each page of the document. Both fax machines have an integral modem within them, and as each page is scanned its digitized image is simultaneously transmitted over the network. A printed version is produced at the receiving side (Fig. 18.11).

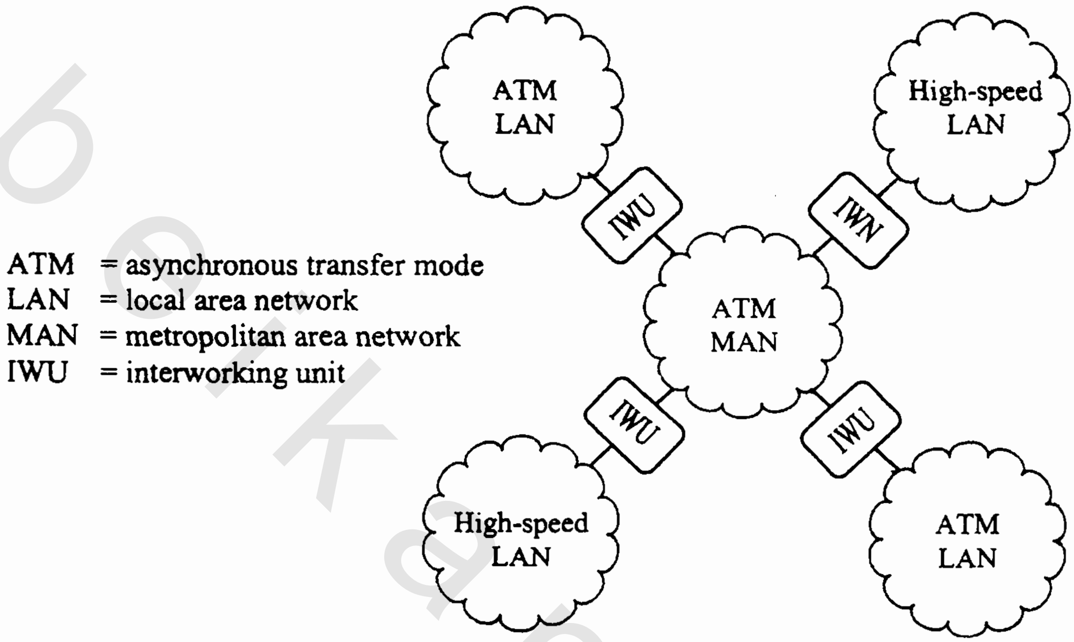


Fig. (18.8) ATM networks

It is also possible to use a PC instead of a fax machine to send an electronic version of the document that is stored within the PC's memory. This is known as PC fax. The digital image is sent in the same way as the digitized scanned document. This requires a telephone interface and appropriate software. The terminal can be either a fax machine or another PC. With the use of a LAN interface it is possible to send the digitized document as invoices and bulletins.

18.8 E-mail:

The user terminal is a PC. Access to the internet is through PSTN / ISDN / DSL and an intermediate service provider (ISP) network or site campus networks. Associated with each network is one or more server (called e mail server) (Fig. 18.12). In the mail box, each user can create or deposit mail or read mail from it. Both e-mail servers and the internal gateway operate using the standard internet communication protocol, using the sender and recipient addresses. A copy of the mail can be sent to multiple recipients through a relevant facility. In text only mail, only strings of ASCII characters are sent. In related applications, chat services seem quite popular. Also, shared white board involves cooperative work comprising text and images.

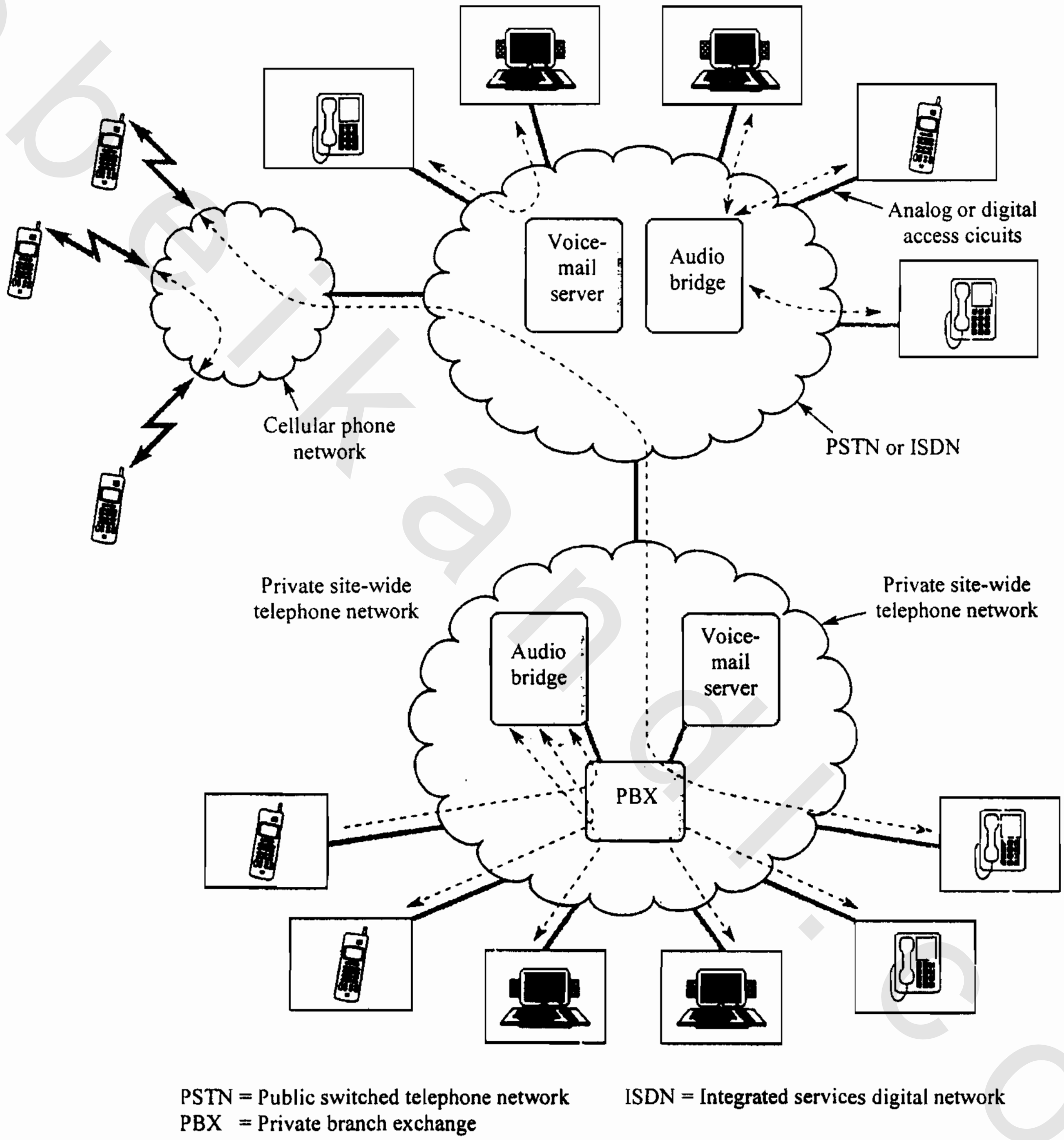


Fig. (18.9) Interpersonal telephony networks

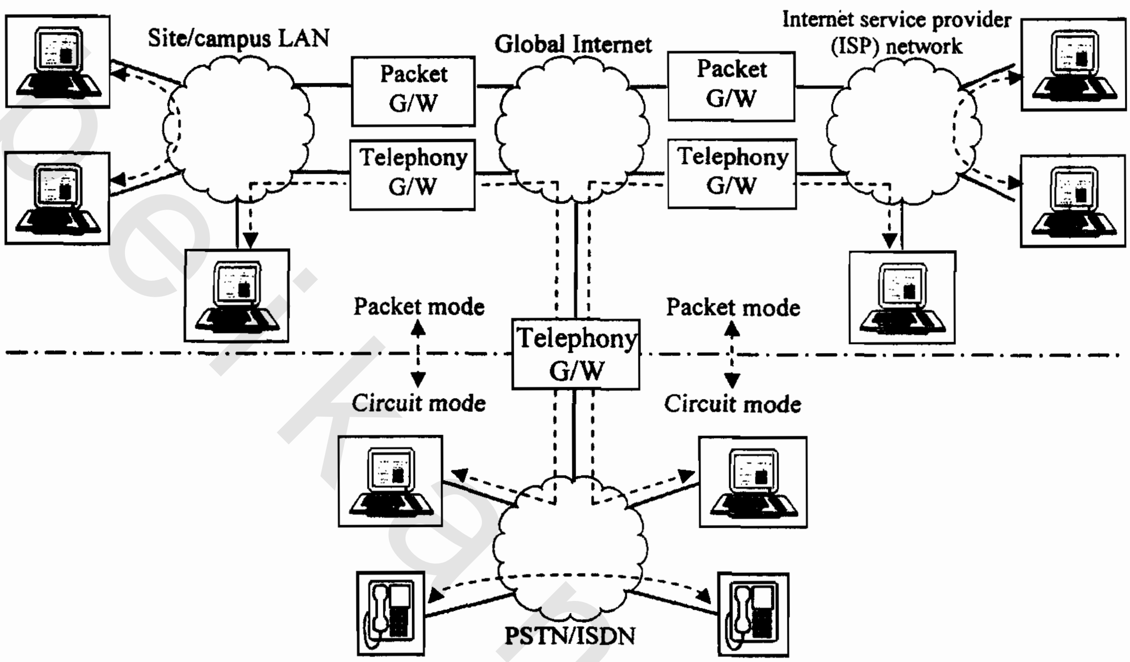


Fig. (18.10) Telephony over the internet

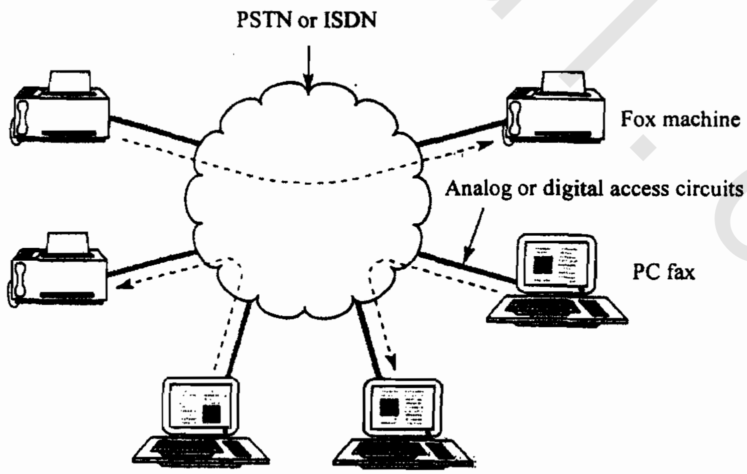


Fig. (18.11) Fax communication

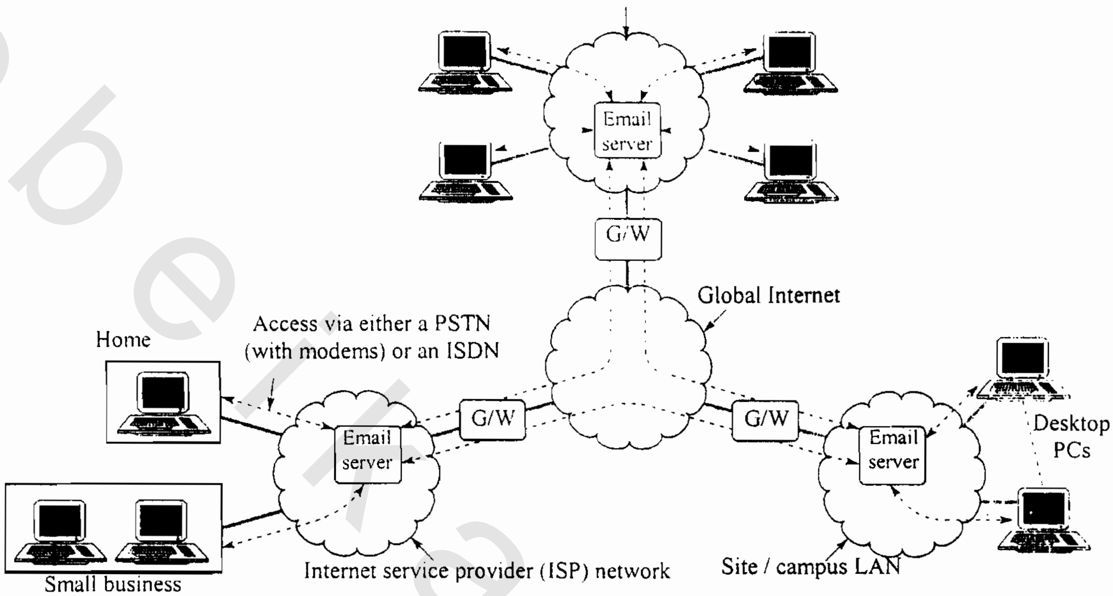


Fig. (18.12) E-mail network

18.9 Videoconference:

One application that uses speech and video integrated through is video telephony using existing network (Fig. 18.13). The terminal PCs incorporate a video camera in addition to the microphone and speaker used for telephony. With a multimedia PC the moving image of the called party is displayed in a window on the PC screen. The network must provide a two way communication channel between the two parties of sufficient bandwidth to support the integrated speech – video generated by each terminal PC.

When several people are involved each in a remote location, we have desk top videoconferencing call. This is used to minimize travel between various locations. In order to support videoconferencing there is a central unit called multipoint control unit (MCU) on a videoconference server (Fig. 18.14). Normally the integrated speech and video information stream from each participant is sent to the MCU which then selects a single information stream to send to each participant. With video activated MCU when the MCU detects a participant speaking, it relays the information stream from that participant to all other participants. In this way, only a single two way communication channel between each location and the MCU is required (Fig. 18.14).

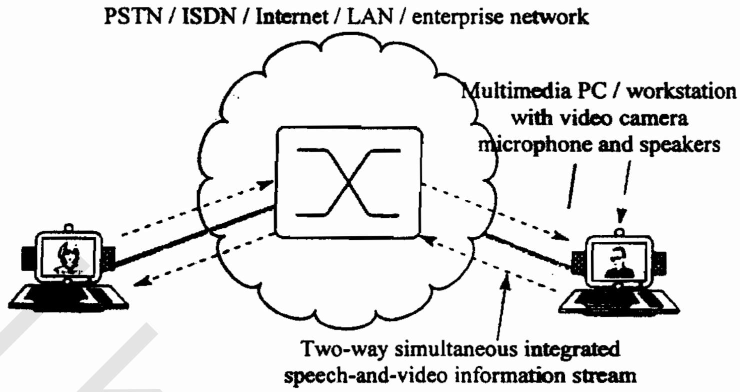


Fig. (18.13) Video telephony

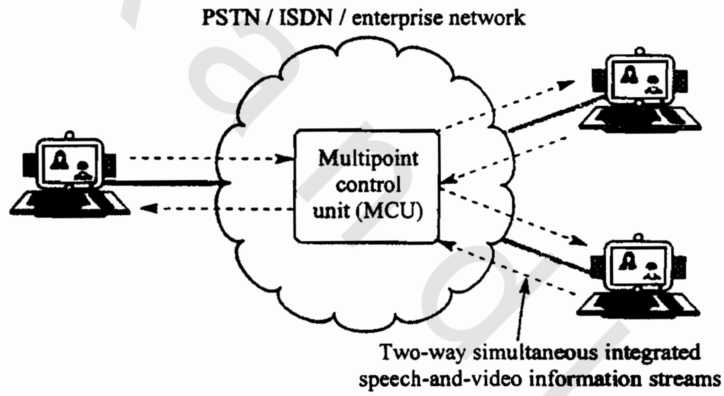


Fig.(18.14) Videoconference using MCU

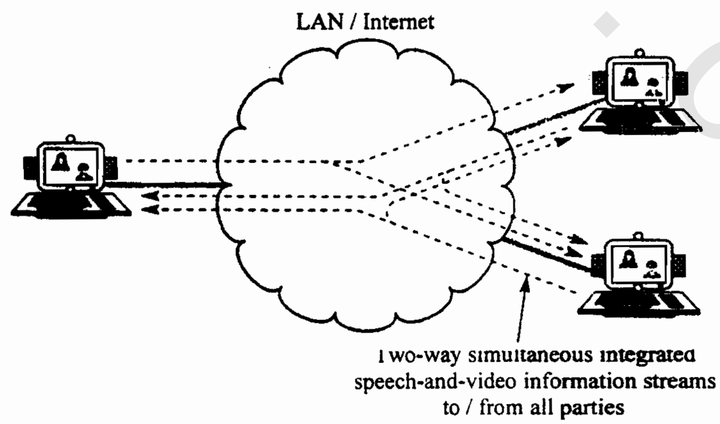


Fig (18.15) Multicasting

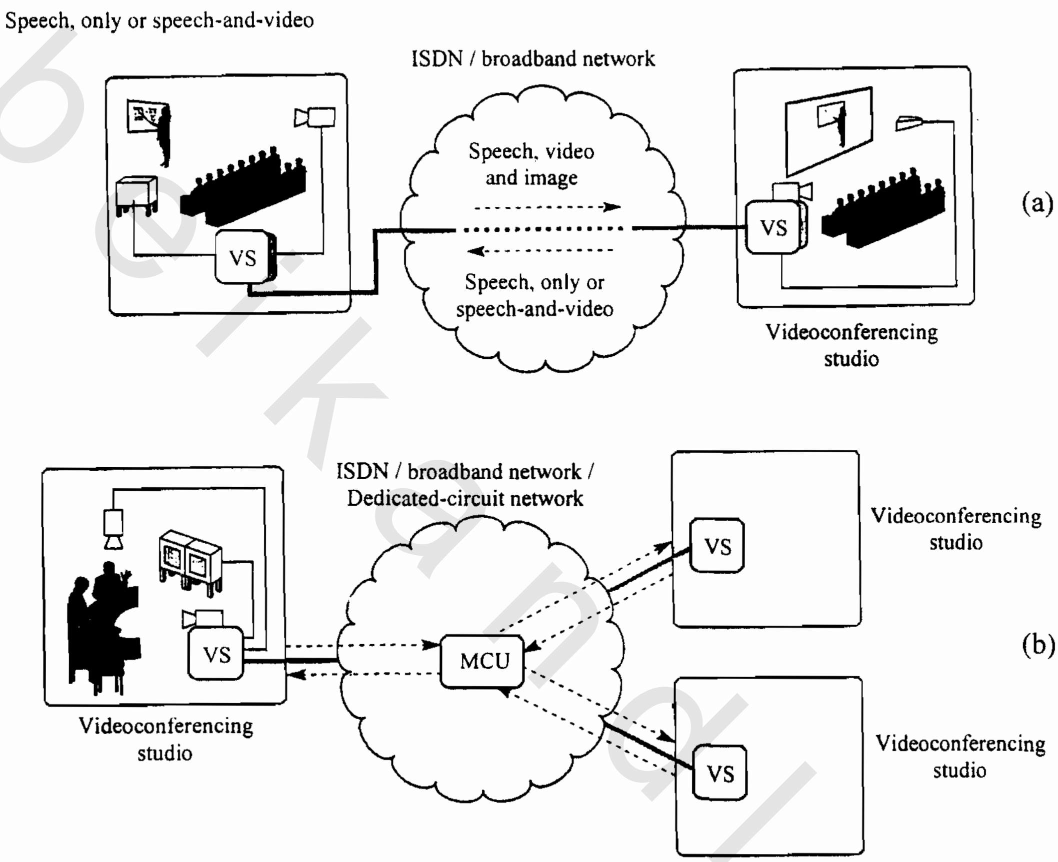


Fig. (18.16) Videoconference
a) one remote class *b) multiparty system using MCU*

Alternatively, LANs and the internet can support multicasting. This means that all transmissions from any of the PCs belonging to a predefined group are received by all the other members of the group. Thus, it is possible to hold a conferencing version without an MCU (Fig. 18.15).

In distance training or distance education, a person at one location is communicating with a group of people at another location. Typically, the information stream transferred from the lecture to the remote class would be integrated speech and video together with electronic material through a document camera. The reverse direction may carry speech (questions) or integrated speech and video for monitoring the remote class (Fig. 18.16a). If the lecture is being relayed to multiple locations an MCU is used at the lecturer's site. Because of the high bandwidth that is involved, the network is either an ISDN that supports multiple 64kbps channels or a broadband multi service network.

Fig. (18.16b) shows a multiparty videoconference system using MCU. In this system, there is a videoconference studio at each location which contains all necessary audio and video equipment such as video cameras, large screen display and audio equipment. The MCU is located at one of the sites. It supports multiple input channels one from each of the other sites and a single output channel. This stream must be broadcast to all other sites.

There are two ways for multipoint conferencing; centralized mode and decentralized mode. The centralized mode is used with circuit switched networks such as PSTN or ISDN. With this mode, a centralized conference server is used. Prior to sending any information, each terminal must first set up a connection to the server. Each terminal then sends its media stream to the server. The server in turn distributes the media stream received to other terminals. (Fig. 18.17).

The decentralized mode is used with packet switched networks such as LANs, intranet, and the internet. In this mode (Fig. 18.18), the output of each terminal is received by all other members of the group. A conference server is not used and it is the responsibility of each terminal to manage the information streams that it receives from the other members. A hybrid system may be used. With data conferencing, the information flow is infrequent. Therefore, a conference server is usually used. In the case of audio conferencing, an audio bridge is used which sets up the connection paths. With video and multimedia conference, a multipoint central unit (MCU) is used.

Because of the volume and rate of information being normally used the centralized mode of operation is used. The MCU consists of two parts, the first is the multipoint controller (MC) which establishes the connections to all the participants. The second part is the multiprocessor (MP) which distributes the information streams generated during the conference. It also carries out the mixing of various media streams into an integrated stream (voice activated switching). When using an MCU, a call is scheduled as for an audio bridge and once the conference starts, each participant may dial in the call and the MCU may dial out. In the voice activated switching mode, the face of the participant is displayed in a window on the screen of the participant's terminal and in a second window in the face of the remote participant who is currently talking. When another participant starts to talk, the face of the new speaker replaces the face of the current remote participant.

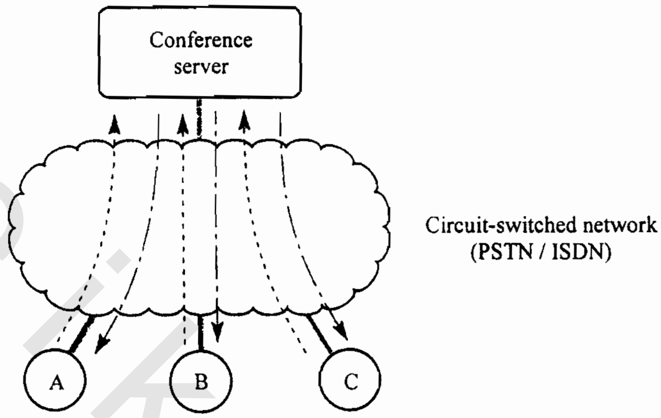


Fig. (18.17). Centralized multipoint conference

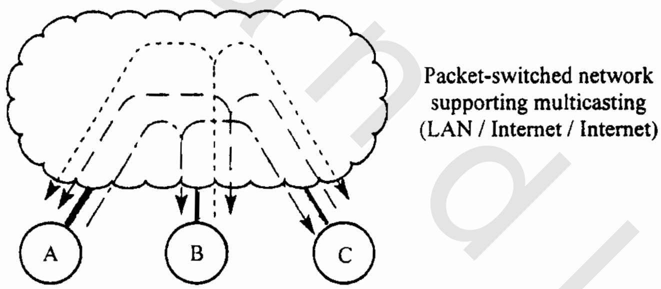


Fig. (18.18) Decentralized multipoint conference

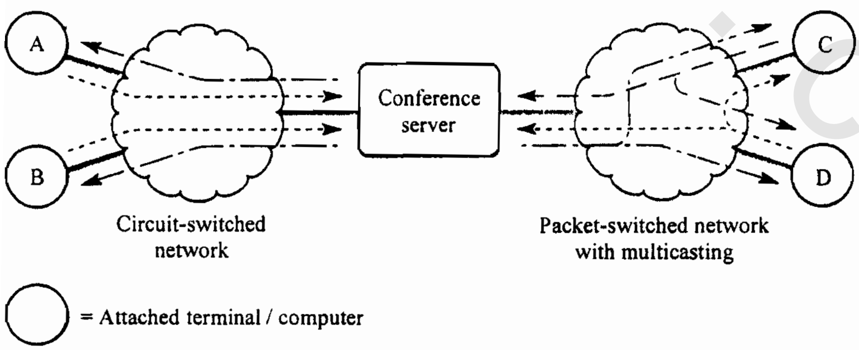


Fig (18.19) Hybrid multipoint conference

In the event of two or more participants starting to talk at the same time, the MCU normally selects the person who speaks the loudest. In the continuous presence mode, the remote window is divided into a number of smaller windows, each displaying the faces of the last set of participants who spoke or are currently speaking. With both modes, the speech from all participants is normally mixed into a single stream, and hence, each participant can always hear what is said by all the other participants.

18.10 The Internet:

The internet is used to support a range of interactive applications with the World Wide Web (WWW) server. This comprises a linked set of multimedia servers which are geographically distributed, thus forming a virtual world wide library (Fig. 18.20).

Multimedia mail may be exchanged among the web participants. Each document comprises a linked set of pages. The linkages of pages are called hyperlinks. These are pointers or references either to other pages of the same document or to any other document within the total web (Fig. 18.21). The optional linkage points within documents are called anchors. Documents comprising only text are created using hypertext, while those comprising multimedia information are created using hypermedia. Each document has a unique address called uniform resource locator (URL), which identifies both the location of the server on the internet, where the first page (home page) of the document is stored and also the file reference on that server. The hyperlinks on this and other pages have similar URLs associated with them. The physical location of a page is transparent to the user and can be located anywhere on the web. The hypertext Mark up Language (HTML) is used for writing client software (browser) to explore the total content of the web, i.e., the contents of the linked information on all the web servers. Once a desired document has been located, the user simply clicks on an anchor point within a page of a document to activate or deactivate the linkage information stored at that point. With a hypertext document, the anchor is usually an underlined word or phrase while with a hypermedia document it is normally an icon. In teleshopping and telebanking, the server must provide additional transaction processing support, such as ordering, purchasing, security and authentication.

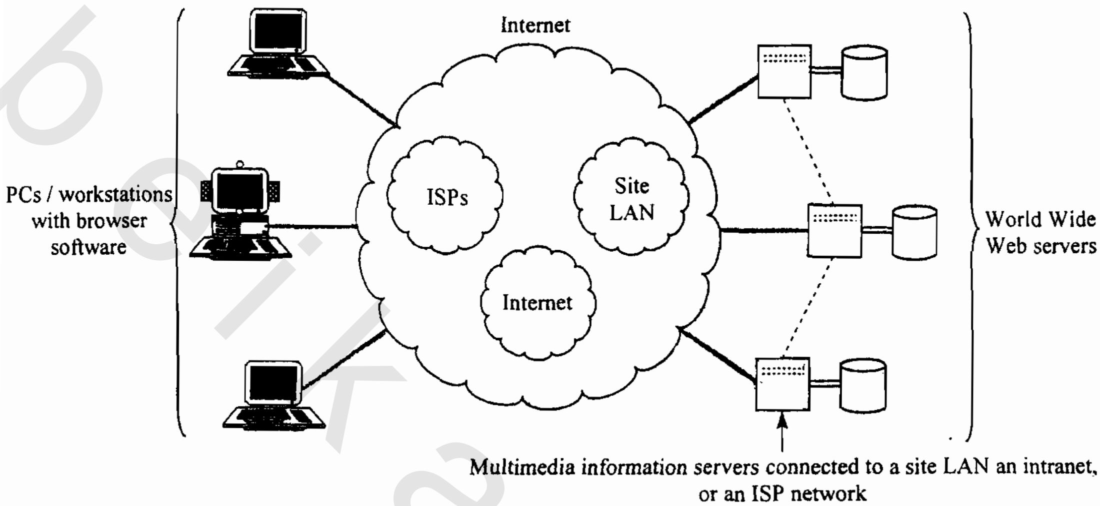


Fig. (18.20) WWW system

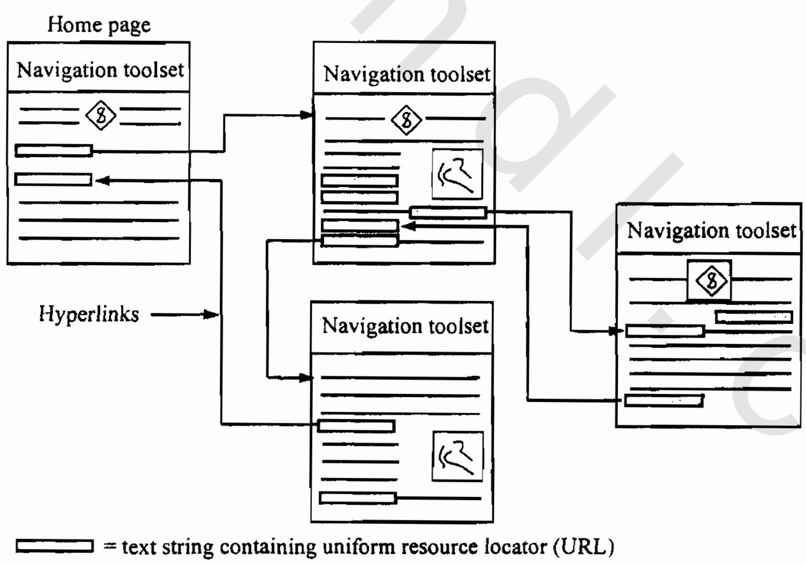


Fig. (18.21) Hypertext linkages

18.11 Interactive TV:

Broadcast TV networks include cable, satellite and terrestrial networks. These networks broadcast analog and digital TV and radio programs. A set top box (STB) is a modem that provides a low bit rate connection to the PSTN and a high bit rate connection to the internet. The subscriber may gain access to all services provided through the PSTN and the internet. In addition to mobile links the user may have a variety of return channels for voting, game participation etc. (Fig. 18.22).

Another application is movie / video on demand (VOD). The video and audio associated with entertainment applications must be of a much higher quality or resolution. A digitized movie / video requires a minimum channel bit rate of 1.5Mbps. The network used to support this type of application must be either a PSTN with a high bit rate modem, cable network, DSL, or fiberoptic network. A video server stores a collection of digitized movies. The subscriber terminal comprises a conventional TV with a selection device for interaction purposes (Fig. 18.22). the user interactions are relayed to the server through a STB, which also contains the high bit rate modem. By means of a suitable menu, the subscriber is able to browse through the set of movies / videos available and initiate the showing of a selected movie. This is called movie on demand (MOD) or video on demand (VOD). The user may control the showing just as in a video cassette recorder (VCR). This means that the server must be capable of playing out simultaneously a large number of video streams equal to the number of the server subscribers watching a certain movie. Thus, the server must support not only the transmission of different movies but also multiple copies of each movie (Fig. 18.23a,24).

In an alternative mode, requests for a particular movie are not played out immediately, but instead are queued until the start of the next play out time of that movie. This is called near movie on demand (NMOD) (Fig. 18.24b). In this way, the viewer is unable to control the play out of the movie. In a small enterprise or campus, a video library may entail a jukebox containing several CDs or a video server connecting several terminals in a LAN.

18.12 Network Types:

The information flow associated with different applications can be either continuous or block mode. In continuous mode, the information stream is generated by the source continuously in a time dependent way. The continuous media is passed directly to the destination as it is generated, and the information stream is played out directly as it is received. This mode of operation is called streaming a real time media. In this case, the bit rate of the channel must be compatible with the bit rate of the source media which is usually audio and or video.

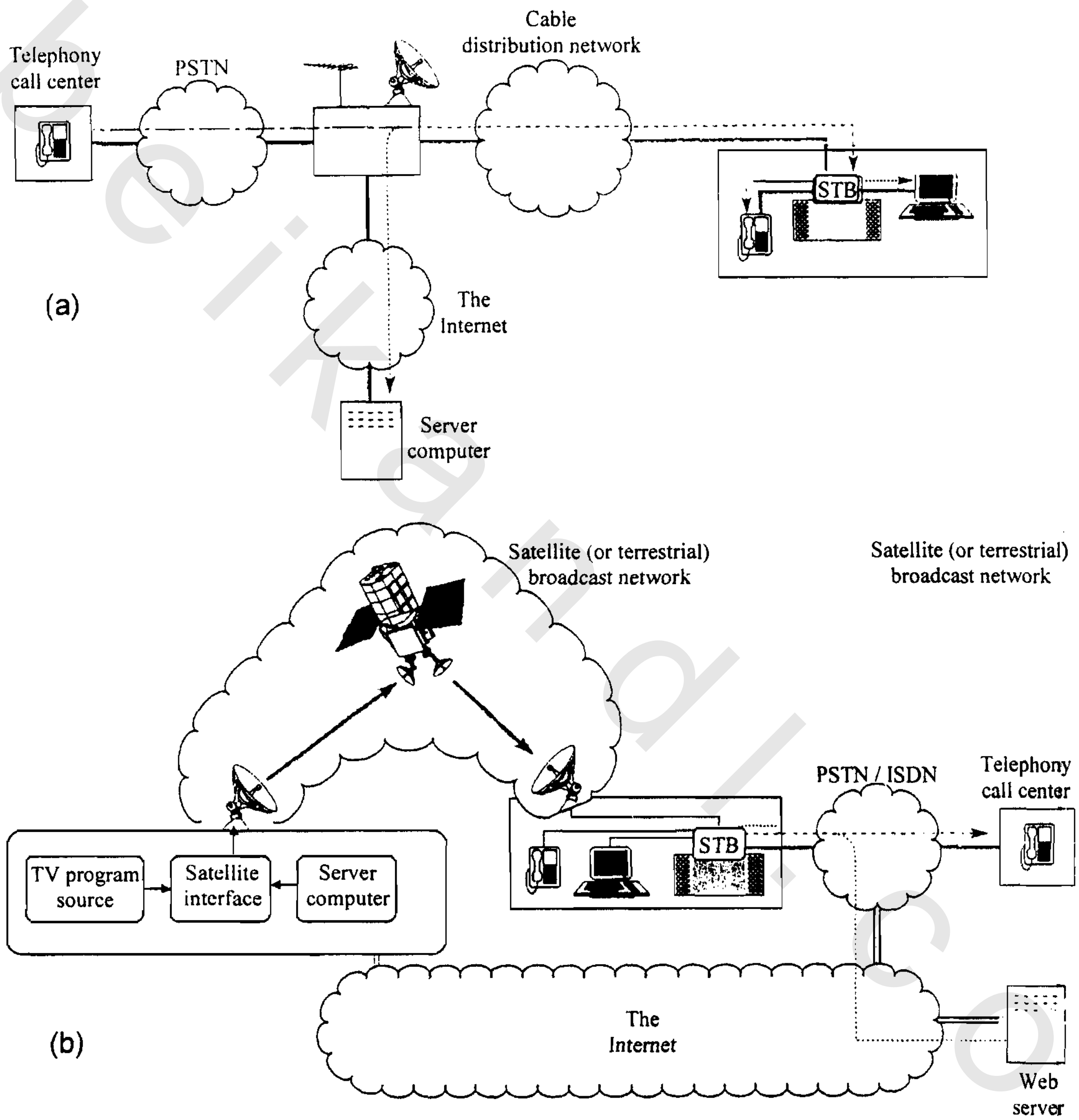


Fig. (18.22) Interactive TV
a) cable distribution network
b) satellite / terrestrial network

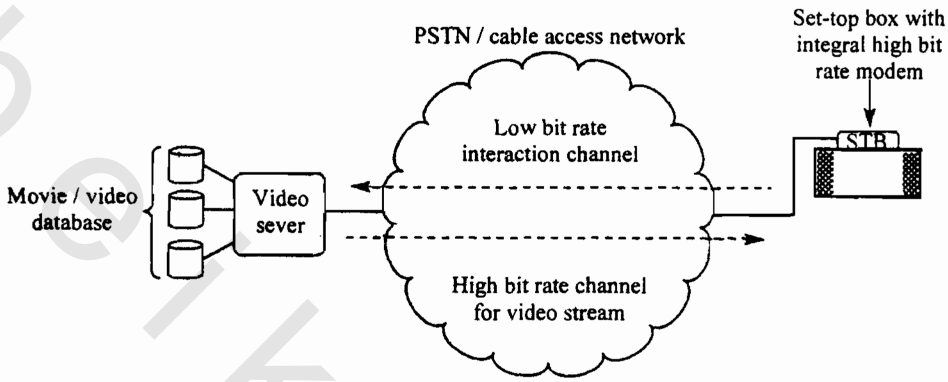


Fig. (18.23) Interaction with a video server

The source information stream may either have a constant bit rate (CBR) or a variable bit rate (VBR). With audio, the digitized audio stream is generated at a constant bit rate which is determined by the frequency of sampling of the audio waveform and the number of bits used to digitize each sample. In the case of video after compression (Chapter 17), the amount of information associated with each frame varies. The information stream associated with compressed video has variable bit rate.

In the case of block mode media, the source information comprises a single block of information that is created in a time independent way. A block of text may represent an e-mail or computer program or a two dimensional matrix of pixel values of an image. The block mode media is stored at the source in a file. When requested, the block of information is transferred across the network to the destination where it is stored and displayed. This is called downloading. Hence, the block mode media need not use a constant bit rate and must be such that when a block is requested, the delay between the request being made and the contents of the block being output at the destination is within an acceptable time interval called round trip delay (RTD) which should be a few seconds.

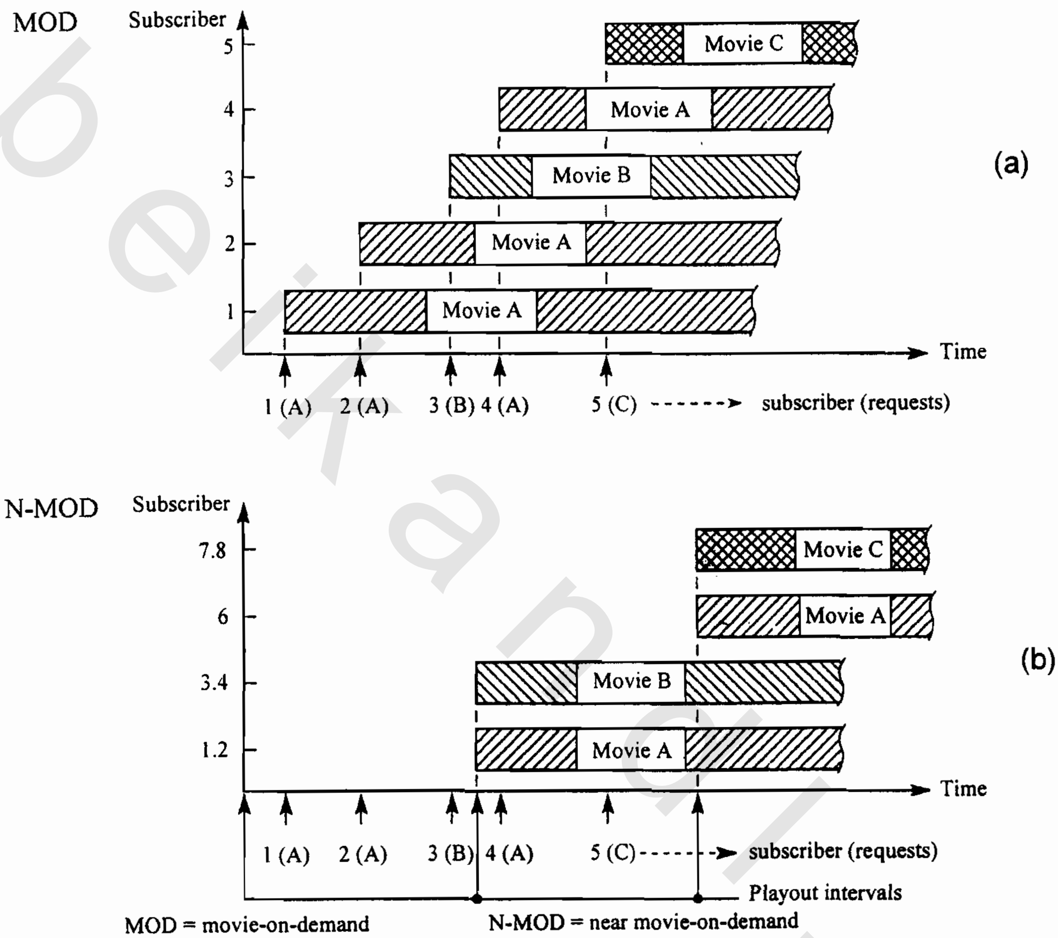


Fig. (18.24) VOD

a) unqueued MOD b) near movie or demand (NMOD)

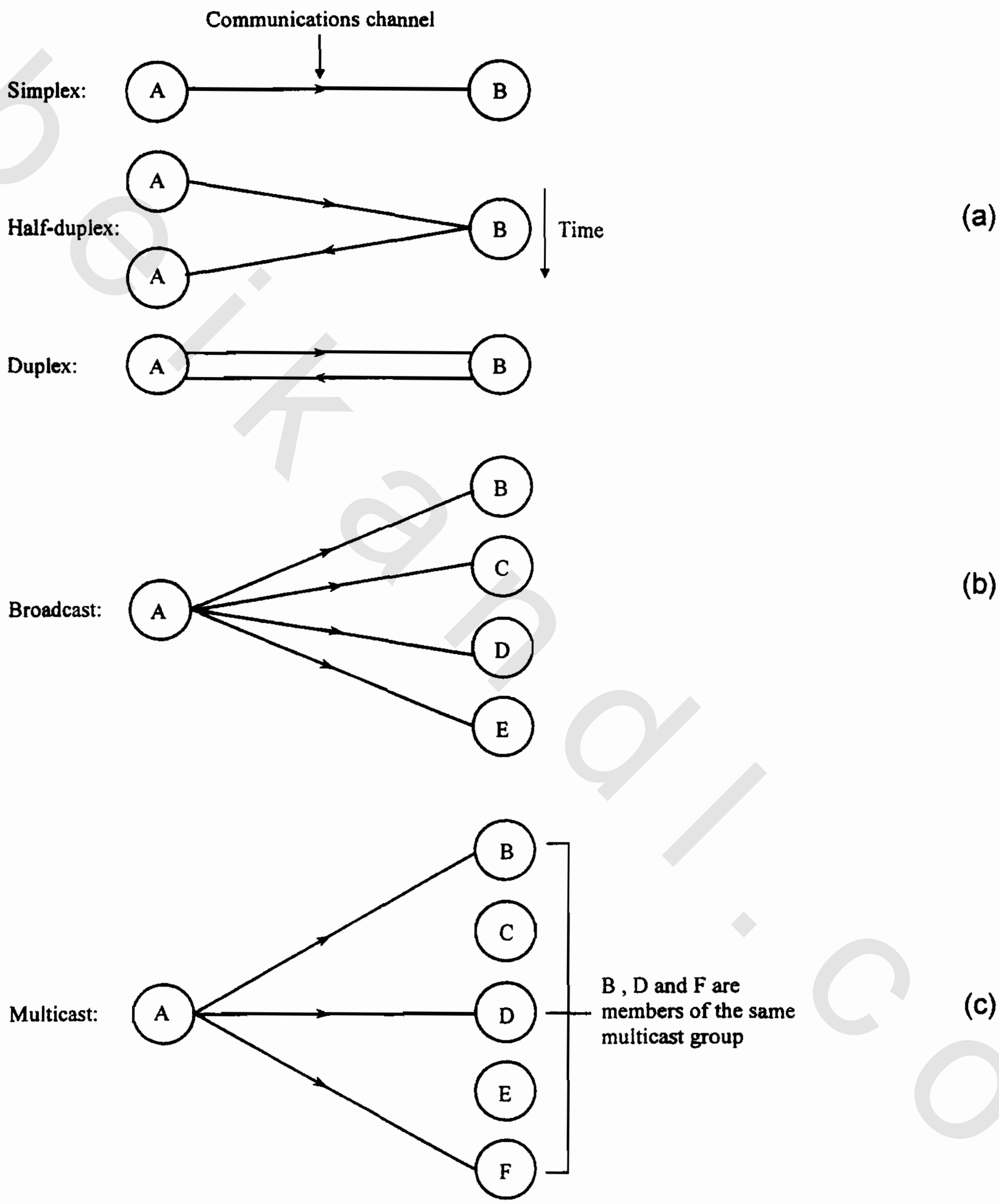


Fig. (18.25) Communication modes
a) unicast b) broadcast c) multicast

In terms of the communication channels provided by the various networks the transfer of the information streams may be one of the following modes (Fig. 18.25).

1. **Simplex**: the information flows in one direction only such as the transmission of satellite remote sensing data to ground station.
2. **Half duplex**: the information flows in both directions but alternately. This is two may alternate mode. This is the case of requesting information from a remote server and the latter responding.
3. **Duplex**: the information flows in both directions simultaneously. It is a two way simultaneous mode, such as in video telephony or videoconference.
4. **Broadcast**: the information output by a single source mode is received by all other modes connected to the same network such as the broadcast of TV programs where all participants receive the same sets of programs.
5. **Multicast**: this is similar to a broadcast except that the information output by the source is selectively received by a specific subset of the nodes connected to the networks, called multicast group. An example is videoconferencing involving a predefined group of terminals connected to a network exchanging integrated speech and video streams or programs on pay networks or pay stations.

In the case of half duplex and duplex communication, the bit rate associated with the flow of information in each direction can be equal if the flows are equal (symmetric flow) or different if the flows are different (asymmetric flow). For example, a video telephone or videoconference involves symmetric duplex communication. However, in an application involving a browser and a web server, a low bit rate channel from the browser to the web server is required for request and control purposes, while a high bit rate is required for the information flow from the server to the subscriber. Hence, this type of application involves an asymmetric half duplex.

There are two types of communication channels associated with the various network types; one that operates in a time dependent mode (circuit mode), also called synchronous channel. The other is time varying mode (packet mode), also called asynchronous channel since it provides a variable bit rate service.

In circuit switched networks, prior to sending any information, the source must first set up a connection through the network. Each subscriber terminal has a unique address or number. The local exchange (switching office) uses this number to establish the connection. The bit rate associated with the connection is fixed. If the destination is free a message is returned to the source indicating that it can now start to transfer information. After the information is received, either the source or destination may terminate the connection. The messages associated with the setting up and cleaning of a connection are called signaling messages. In circuit switched networks (PSTN and ISDN), there is a time delay, while the connection is being established, which is called call / connection setup delay (milliseconds in ISDN and a few seconds in PSTN). For packet mode networks, there are two types, connection oriented (CO) and connectionless (CL). CO networks (packet switched networks) comprise interconnected sets of packet switching exchanges (PSEs). As in circuit switched networks, in packet switched network, each terminal has a unique number or address. Prior to sending the information, a connection is first set up through the network using the addresses of the source and destination terminals. However, the connection that is set up utilizes only a variable portion of the bandwidth of each link, and hence, the connection is known as virtual connection or virtual circuit (VC). To set up a VC, the source terminal sends a call request control packet to its local PSE which contains in addition to the address of the source and destination terminal a virtual circuit identifier (VCI). Each PSE maintains a table that specifies the outgoing link that should be used to reach each network address.

In contrast with a CL network, the establishment of a connection is not required and the two communicating terminals can communicate and exchange information as and when they wish. Each packet must carry the full source and destination addresses in its header in order for each PSE to route the packet into the appropriate outgoing link.

In CL network, the term router is normally used rather than packet switching exchange. In both network types, as each packet is received by a PSE / router on an incoming link it is stored in a memory buffer.

Errors are first checked in the header. If no errors are detected, then the addresses / VCIs carried in the packet header are read to determine the outgoing link that should be used and the packet is placed in a queue ready for forwarding on the selected outgoing link. All packets are transmitted at the maximum link bit rate.

A packet may experience an additional delay, while it is in the output queue for a link waiting to be transmitted. This delay is variable, since it depends on the number of packets to be transmitted. This mode of operation is known as store and forward leading to a mean packet transfer delay and a jitter around this mean. An example of CL packet switched network is the internet. An example of CO packet switched network is the ATM networks.

18.13 Network Performance:

In circuit switched networks, 3 parameters determine the performance of the network:

1. Bit rate.
2. Mean bit error rate.
3. Transmission delay.

The mean bit error rate (BER) of a channel is the probability of a bit being corrupted during its transmission is a definite time. In applications involving financial transactions, no error is tolerated at all. In such application, the source information is divided into blocks, the maximum size of which is determined by the mean BER of the channel. When there is an error, the destination must request the source to send another copy of the missing block. Clearly, this will introduce a delay. To decrease the frequency of retransmission, the block size must be made small. This, however, leads to high overheads, since each block must contain the additional information associated with the retransmission procedure. Therefore the choice of the block size is a compromise between the increased delays resulting from a large block size - and hence retransmission - and the loss of transmission bandwidth resulting from high overheads of using smaller block size.

Transmission delay is determined not only by the bit rate but also by the delays in the terminal interfaces (codec delay) plus the propagation delay of digital signals which is independent of the bit rate.

For packet switched networks, the parameters determining the performance of the network are.

1. Maximum packet size.
2. Mean packet transfer rate.
3. Mean packet error rate.
4. Mean packet transfer delay.
5. Worst case jitter.
6. Transmission delay.

The rate at which packets are transferred is influenced by the bit rate of the interconnecting links because of the variable store and forward delays in each PSE / router. Hence, the rate of transfer of packets is variable.

The mean packet transfer rate is a measure of the average number of packets that are transferred across the network per second and coupled with the packet size being used determines the equivalent mean bit rate of the channel. The mean packet error rate (PER) is the probability of a received packet containing one or more bit errors. It is similar to the block error rate associated with a circuit switched network. Hence, it is related to both the maximum packet size and the worst case BER of the transmission links that interconnect the PSEs / routers in the network.

The mean packet transfer delay is the summation of the mean store and forward delays that a packet experiences in each PSE / router that it encounters along a route and the term jitter is the worst case variation in this delay. The transmission delay is the same for circuit mode or packet mode.

In addition to the network parameters, the application itself has its own parameters. In an application involving image, for example, the parameters may include a minimum image resolution and size, while in an application involving video, the digitization format and refresh rate must be considered. Thus, the application parameters must include:

1. Required bit rate (or mean packet transfer rate).
2. Maximum start up delay.
3. Maximum end to end delay.
4. Maximum delay jitter.
5. Maximum round trip delay.

For applications involving the transfer of a constant bit rate stream, the important parameters are

- Required bit rate (or mean packet transfer rate).
- End to end delay.
- Delay jitter.

For interactive applications, the start up delay defines the amount of time that elapses between an application at the destination. This includes in addition the time needed to establish a network connection plus the delay in setting up the server. The round trip delay involves the delay between a request for some information being made and the start of the information flow. This should be as small as possible (a few seconds) for a meaningful human computer interaction.

Thus, for applications involving the transfer of a constant bit rate stream a circuit switched network appears to be most appropriate. In this case, the set up delay is not important, and the channel provides a constant bit rate service of a known rate.

Conversely, for interactive applications, a CL packet switched networks appear to be most appropriate since there is no network call set up delay and any variations in the packet transfer delay are not important. An example is the transfer of a large file of data from a server connected to the internet to a client PC.

In the case of PSTN or ISDN, a circuit switched mode is used and provides a constant bit rate of 28.8kbps, (PSTN) and 64 /128 kbps (ISDN). Cable modems operate in a packet mode. A typical bit rate of shared channels is 27Mbps. Assuming 270 users, each user would set 100 kbps. In this case, the main parameter of interest is not the bit rate but the time to transmit the complete file. With PSTN and ISDN, this is directly related to the channel bit rate and the size of the file.

To overcome the effect of jitter, a technique called buffering is often used (Fig. 18.26). The effect of jitter is overcome by retaining a defined number of packets in a memory buffer at the destination before the play out of the information bit stream is started. The memory buffer operates using a first in first out (FIFO) scheme and the number of packets retained in the buffer before output starts is determined by the worst case jitter and the bit rates of the information stream. When using a packet switched network for constant bit rate stream (Fig. 18.26a), an additional delay is incurred at the source as the information bit stream is converted into packets. This is known as the packetization delay, and adds to the transmission delay of the channel. Hence, in order to minimize the overall input to output delay, the packet size used for an application is made as small as possible, but of sufficient size to overcome the effect of the worst case jitter.

In the internet packets relating to multimedia applications real time streams are given a higher priority than packets relating to e-mail. Typically, packets containing a real time stream such as audio and video are more sensitive to delay and jitter than packets containing text. Hence, during periods of network congestion, the packets containing real time streams are transmitted first since packets containing video are more sensitive to packet loss than packets containing audio, and hence are given a higher priority.

EX. 18.1

A packet switched network with a worst case jitter of 10ms is to be used for a number of applications, each of which involves a constant bit rate stream. Determine the minimum amount of memory that is required at the destination and a suitable packet size for each of the following input bit rates. Assume that the mean packet transfer rate of the network exceeds the equivalent input bit rate

$i)\, 64 kbps$,

$ii)\, 256 kbps$

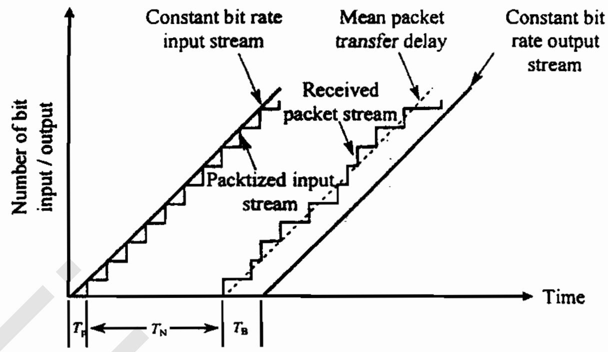
Solution

$i)\, At\ 64kbps \qquad 10ms = 640 bits$

Hence, choose a packet size of say 800bits with FIFO buffer of 1600 bits (2 packets) and start play out of the bit stream after the first packet has been received.

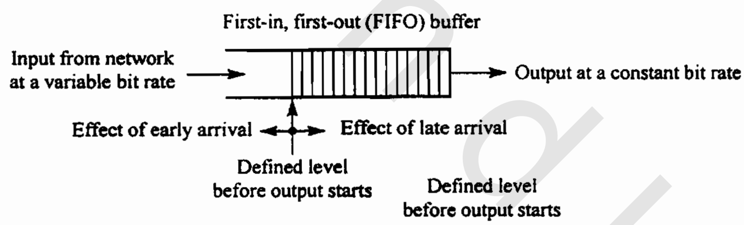
$ii)\, At\ 256kbps \qquad 10ms = 2560 bits$

Hence, choose a packet size of say 2800 bits with FIFO buffer of 5600 bits.



(a)

T_P = packetization delay
T_N = mean network packet transfer delay
= transmission delay + mean store-and-forward delay
T_B = buffering delay at destination (to overcome worst-case jitter)
T_T = total input-to-output delay
= TP + TN + TB
Jitter = variation in store-and-forward delay about the mean



(b)

Fig. (18.26) Transmission for constant bit rate stream over a packet switched network
a) timing schematic. b) FIFO buffering.

18.14 DSL:

Bit rates between $144kbps$ (basic rate) and $1.544/2.048Mbps$ (primary rate) over several kilometers are obtained using baseband transmission in digital subscriber line (DSL). In the case of a basic rate line, this is called ISDN DSL (IDSL), and in the case of a primary rate line, a high speed DSL (HDSL). DSL uses a single pair and an HDSL uses two pairs. The HDSL may use a single pair for bit rates of up to $1.544/2.048Mbps$ for finite wire length.

The basic rate and the primary rate lines of an ISDN are symmetric, i.e., with equal bit rates in both directions. For interaction purposes, usually the information flow is asymmetric, i.e., involving a low bit rate channel from the subscriber for interaction and a high bit rate channel for the down stream direction for the return of the requested information. Asymmetric ratio ranges from10:1 to 100:1. DSL technology has developed based on the twisted pair wires used in common telephony to support high speed interactive services. Two types have evolved, the first known as asymmetric DSL (ADSL) and the second very high speed DSL (VDSL).

In the case of ADSL, the high speed symmetric channel is designed to accommodate the existing analog telephony service. In the case of VDSL, the high speed channel - in addition to operation at a higher bit rate than that of an ADSL - can operate in either an asymmetric or symmetric mode and is designed to coexist with analog telephony as well as basic rate ISDN services. The ADSL was defined to meet the requirements of broadcast quality video on demand, allowing bit rates of up to $8Mbps$ in the down stream channel from the local exchange to the subscriber and $1Mbps$ in the up stream direction.

High speed access to the :Internet requires less limitation on VOD, only $1.5Mbps$ down stream and $384kbps$ for the up stream are satisfactory, The actual bit rates achievable depend on the length and quality of the line. The maximum length of twisted pair cable used in ADSL is around $4\,km$. Longer distances can be achieved using fibers.

The lower frequency band up to $4kHz$ of the bandwidth available with a single twisted pair line is used for analog telephony (POTS). So in order for the two ADSL signals to coexist with the POTS signal on the same line, modulated transmission must be used to take the signals away from the lower frequency band.

In the case of ADSL, the two signals are transmitted in the frequency band from $25kHz-1.1MHz$ (the upper limit may be limited to $500kHz$ called ADSL – Lite.

In (Fig. 18.27), the POTS splitter separates out the POTS and ADSL signals. This is done by means of two filters, a LPF $(0-4kHz)$ that passes only the

POTS signal and a HPF $(25kHz - 1.1MHz)$ that passes only the forward and return ADSL signals. Having separated out the POTS signal, the existing customer wiring can be used to connect telephones to the network termination (NT). In the case of ADSL signals, normally the ADSL modem is located within the NT, and the customer equipment is connected to the NT. This can be PC or TV STB. Alternatively, the ADSL modem can be located within the customer equipment, and the latter is then connected directly to the output of the HPF. In Fig. (18.27b) with an ADSL - Lite installation, passive NT and twisted pair are used. The telephone handsets attached directly to the terminal equipment respond only to low frequency speech signal. The terminal equipment in ADSL - Lite line is PC for internet access. Integrated within the PC, is a HPF and ADSL - Lite modem. To use fast access Internet service, a line termination board containing a HPF and ADSL - Lite modem must be added to the PC. The telecom company connects the customer line to local exchange portion which provides this service. To cut down possible interference with radio signals, a LPF is added to the telephone sets.

The modulation method used with ADSL modems is called discrete multi tone (DMT). The bit stream to be transmitted is divided into fixed length block. The bits in each block are then transmitted using multiple carriers, each of which is separately modulated by one or more bits from the block. The number of carriers is either 256 or 512 (Fig. 18.28). Those carriers that lie in the lower part of the frequency spectrum, $(25-20kHz)$ are reserved for up stream (bit rate 32 – 384 /1000kbps) and the higher frequency band $(250-1100kHz)$ are reserved for down stream (bit rate $640-1500/8000kbps$).

Because with twisted pair wire, the level of attenuation with each carrier may vary, non allocation of bits per carrier is used. Typically, the lower frequency carriers are modulated using multiple bits per carrier (8 QAM), and the higher frequency carriers use fewer bits down to 1 bit (PSK). Because of the high levels of noise present on the lines carried by the various telephony functions, forward error correction (FEC) must be used.

This requires for the bit stream to be transmitted to be segmented into 188 byte blocks, each of which has 16 FEC bytes appended to it. We should note that if we use one carrier we have to increase the bit rate, which introduces more error. By using several carriers we make use of the available spectrum and maintain the same throughput with reduced bit rates for each, hence, decreasing the probability of error.

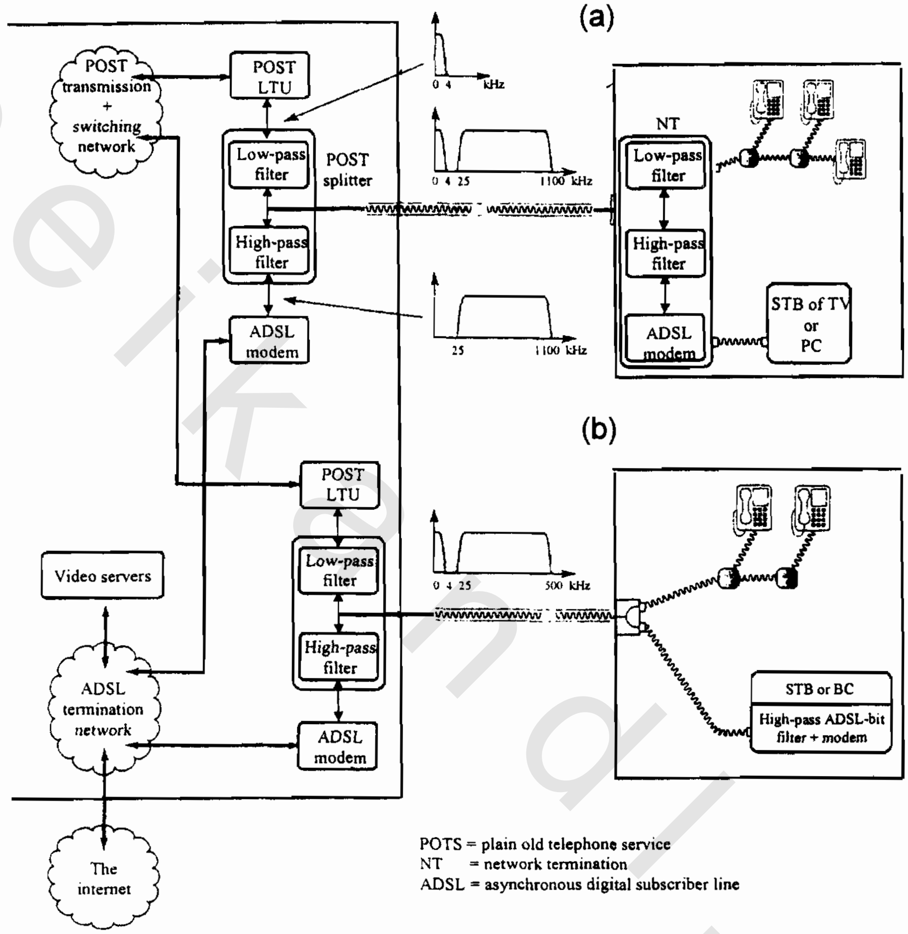
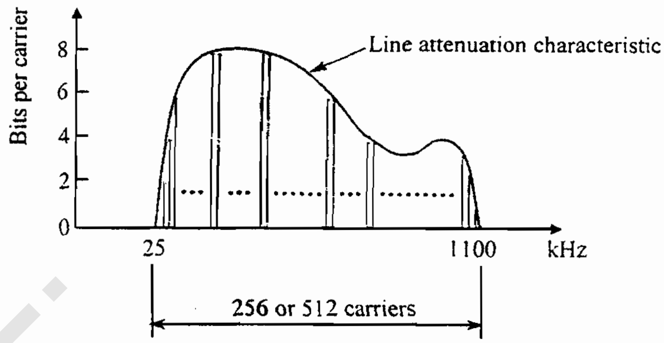


Fig. (18.27) ADSL connections

a) ADSL with active network termination.

b) ADSL – Lite with passive network termination.

We note that the ADSL baseband signal comprises a duplex bit stream of up to 8 Mbps in the down stream and up to 1 Mbps in the return direction. In the case of ADSL – Lite, the bit rates are up to 1.5Mbps down stream and up to 384 kbps up stream.



Upstream : frequency band = 25 – 200 kHz, bit rate = 32 – 384 / 1000 kbps
Downstream : frequency band = 250 – 1100 kHz, bit rate = 640 – 1500 / 8000 kbps

Fig. (18.28) DMT operation

In both cases, since a single twisted pair line is used, a scheme must be employed to enable both bit streams to be transmitted over the line simultaneously. In the case of both ADSL, this is achieved using a technique called frequency division duplex (FDD). With this, the bit stream in each direction is transmitted concurrently using a different portion of the available bandwidth, and hence, set of carriers.

Very high speed digital subscriber line (VDSL) using optical fibers provides high bit rates over existing unshielded twisted pair access lines. Bit rates up to 20Mbps in each direction (or 52Mbps – 1.5Mbps asymmetric configuration) can be achieved.

18.15 Mobile (Cellular) Telephony:

The cellular phone service area is divided into small cells. Each cell has a base station with a tower, which receives and transmits signals to mobile users. All base stations are connected by telephone lines to the mobile telephone switching office (MTSO), which in turn is connected to the telephone control office by phone lines. A subscriber makes a radio call to the cell base station, which sends the signal to the MTSO. If the called party is land based, i.e., has a fixed telephone, the MTSO relays the signal through the telephone control office as a regular telephone call. If the called party is mobile, the MTSO relays the signal to the base station to which the called party belongs. The base station transmits the signal to the called party via the available radio channel in the cell. As the caller moves from one cell to another, the MTSO automatically switches to an available channel in the new cell as long as the call is in progress (Fig. 18.29). Each cellular phone has, a manufacturer's serial number and the phone number assigned by the phone company. These numbers are automatically transmitted to MTSO during the initialization of each call. The MTSO checks the validity of these numbers in a process called authentication. Then, it assigns to the caller two available radio frequencies one for transmission and the other for receiving from the base station. When the call terminates the radio channels are released and made available for another caller.

The MTSO continually monitors the signal strength of a phone call. When the attenuation reaches a certain threshold, the MTSO decides that the caller has moved from one cell to another (handover). The MTSO then searches for a neighboring cell when the signal strength from the caller is stronger, and then automatically switches the caller to this new base station and reassigns two available channels. The switching is so rapid that users do not notice it. The use of different frequencies solved the problem of having a limited number of users in old systems using one frequency over the entire city.

In cellular telephony same frequencies are used in all cells except those immediately adjacent. The transmitted powers are kept sufficiently small so that the signals from one cell do not propagate beyond the immediately adjacent cells. We may increase the number of users by increasing the number of cells as we reduce the cell size and the power levels.

The first generation of cellular systems using analog voice transmission came into operation in the early eighties. The most popular of these systems was advanced mobile phone system (AMPS). It used a $3kHz$ audio signal to frequency modulate a carrier. The frequency deviation is $\pm 12kHz$ and the bandwidth is $30kHz\left(\beta=4\right)$.

The transmission frequency for the base station lies in the range $869-894MHz$, while for the mobile station $824-849MHz$. The spacing between channels is $30kHz$. The number of channels is 832 and the base station coverage radius is $2-25km$. The transmitter output power for a base station is *100W* and *3W* for a mobile station.

In analog systems, the users were assigned exclusively to their own operator and their mobile phones stopped operating when leaving the coverage area of their own mobile communication system.

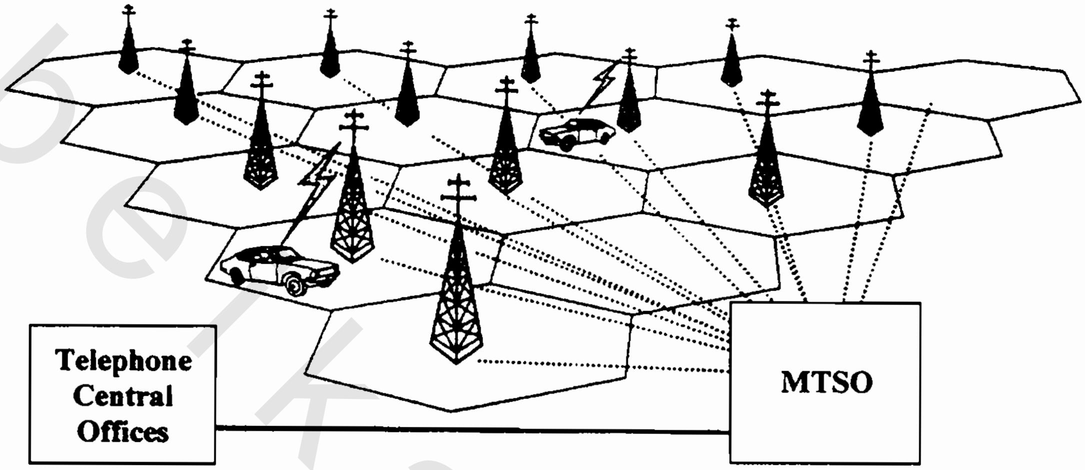


Fig (18.29) Cellular telephone system

As a result of cooperation within the European Union a special working group for mobile telephony Group Special Mobile (GSM) was established to develop the standards for a common mobile cellular telephone system. GSM now stands for Global System for Mobile communication. It is one of the most popular versions of the second generation of the mobile. Two bands each $25MHz$ wide have been assigned to the GSM system. The frequency band between 890 and $915MHz$ is used for the transmission by mobile station, to base stations (uplink), whereas the band between 935 and $960MHz$ is used for the transmission from the base stations to mobile stations (downlink). Thus, a duplex transmission is realized in frequency division duplex (FDD) mode. Both bands are divided into 124 frequency intervals of $200kHz$ each with carrier frequency in their centers. Time along each carrier is divided into 8 slots. Thus, multiple access is realized through assigning the connection a particular carrier frequency or a sequence of them if frequency hopping is performed in a selected time slot. As a result, the GSM can be treated as a system with TDMA / FDMA multiple access scheme.

A physical channel is a sequence of time slots (denoted by the assigned slot numbers which are placed on the selected carrier. Physical channels are arranged in pairs. Each pair consists of one physical channel in each direction (up link and down link). They are marked with the same time slot number and their carriers differ by $45MHz$. A subset of carrier frequencies is assigned to each call. Note the numbering of the time slots is delayed by 3 for down link as compared with up link transmission. Due to that, a mobile station realizing a connection to a base station in the assigned time slot never transmits and receives the signals at the same time. This way an electromagnetic feedback between a mobile station transmitter and receiver is avoided. The TDMA / FDMA combination ensures full utilization of time and frequency domain. None of the carriers and time slots is devoted to a particular exclusive use. The use of SFH / SS technique assures as many subscribers as possible in the allotted bandwidth with minimum interference for low bit rate communication.

CDMA is the dominating method of multiple access in the third generation (3G) mobile communication system such as CDMA 2000 and improved 2G such as IS-95. We have seen that digital TDMA / FDMA (2G) systems require frequency planning in order to make possible use of the same channel frequencies in sufficiently spaced cells. The distance between the cells which use the same carrier frequency is finite and multiple frequency use leads to interference among the mobile stations applying the same carriers in different cells. Multiple reuse factor is the reciprocal of the cell cluster size N .

The application of directive antennas decreases channel interference. The application of disjoint subsets of carrier frequencies used in neighboring cells in TDMA and FDMA systems necessitates rapid switching of the connection from the current base station to the neighboring one at the moment of crossing the border between the cells. This is associated with change of carrier frequency and is called hard handover, which may cause distortion or loss of connection.

One of the dominating distortions occurring in mobile communication systems is the multipath, which results in fading causing distortion. The disadvantage of FDMA / TDMA systems of limited capacity, hard handover and sensitivity to fading are counteracted by CDMA systems, CDMA has an inherent advantage of increased spectral efficiency.

The use of CDMA allows improved frequency reuse. CDMA has an inherent resistance to interference. Although users from adjacent cells will contribute to the total interference level, their contribution will be significantly less than the interference from the same cell users. The result is that frequency reuse efficiency increases by a factor of $4-6$. CDMA may use statistical multiplexing of voice signals since voice activity is about 40%.

In the CDMA access scheme, all users share the same band. The spreading sequence is of much higher rate (chip rate) than the data rate. The sequences of different users do not interfere with each other because they are mutually orthogonal. The called party extracts the signal by correlating the received signal with his assigned code. Mutual orthogonality of spreading sequences is that feature which is crucial for the successful operation of the whole system based on the CDMA scheme.

In the transmitter, the resulting sequence is multiplied by a pair of pseudorandom sequences. Multiplication by pseudorandom sequences perfoms spreading and considerately widens the data signal spectrum. After transforming the spread data sequences into a bipolar form the bipolar pulses are spectrally shaped with filters of transfer faction $H(f)$. Subsequently both in-phase and quadrature components are shifted from the baseband destination frequency range using two modulators with cosine and sine carries. The received signal is shifted back to the baseband using a pair of in-phase and quadrature demodulators, thus extracting the in-phase and quadrature components. Next, both components are filtered by matched filters $H(f)$. The filter outputs are sampled with period T_c and multiplied by a pair of synchronized pseudorandom sequences as those applied in the transmitter. The received samples are then processed resulting in an estimate of the data sequence.

One of the popular schemes for CDMA mobile systems is IS-95. It operates in two bands using FDD in each. The first band class O had been previously occupied by the AMPS. Generally, the $IS-95$ standard allows for mobile stations to operate both in AMPS and $IS-95$ (CDMA) mode. The downlink is realized in the frequency range 824 – 849 MHz and uplink in the range 869-894 MHz . There is 45 MHz difference between both bands. $IS-95$ can be also deployed in the Personal Communications System (PCS) 1800 MHz band called band class1. The frequency ranges are $1930-1990 MHz$ for downlink and 1850-1910 MHz for up link. IS-95 uses a 1.2288 MHz code spreading sequence on top of a variable data rate that ranges from 1200 to 9600 bps. Two pseudorandom sequences of length 2^{15} often denoted as PNI and PNQ are used, separately in the in-phase and quadrature branches. All base station transmitters generate the same PN sequence pair. Transmitting data over I and Q channels simultaneously allows extremely sharp pulse shaping.

Code excited linear prediction (CELP) algorithm is used for voice encoding / compression. Since the voice coder exploits gaps and pauses in speech, the data rate is variable. Whenever the bit rate falls below the peak bit rate of $9600 kbps$ repetition is used to fill the gaps. Thus, multiple access interference is reduced due to this voice activity gating.

18.16 From 2G to 3G:

Although GSM is optimized for circuit switched services such as voice, it offers low rate data services up to $14.4 kbps$. High speed data service up to $115.2 kbps$ are possible with the enhancement of GSM standard toward the

General Packet Radio Service (GPRS) by using a higher number of time slots. GPRS uses the same modulation, frequency and frame structure as GSM. The $IS-95$ renamed cdma 1 was improved up to 64kbps data service instead of $14.4kbps$. Trends toward more capacity for mobile receivers, new multimedia services, new frequencies, and new technologies have led to the third generation (3G). The objectives of the third generation Universal Mobile Telecommunication Technology (UMTS) and cdma 2000 went far beyond the second generation especially with respect to:

1. Wide range of multimedia services (speech, audio, image, video, data) and bit rates (up to $2Mbps$).
2. Lower bit error rate (BER) and higher quality of service.
3. Higher number of users.
4. Variable bandwidth, data rate and power / channel allocation.

From Table (18.1), we see that GSM employed in the $900MHz$ and $1800MHz$ divides the allocated bandwidth into $200kHz$ FDMA subchannels. Then, in each subchannel, up to 8 users share the 8 time slots in TDMA. In the IS-95 up to 64 users share the $1.25MHz$ channel by CDMA. The system is used in the $850MHz$ and $1900MHz$ bands. For UMTS and CDMA , FDD which is known as wideband CDMA employs separate $5MHz$ channels for both the uplink and downlink directions. The end user data rate is $2Mbps$ per carrier.

Table (18.1) Parameters of 2G to 3G mobile radio systems

Parameter	2 G system		3G systems
	GSM	IS-95	IMT-2000/UMTS (WARC 92 [39])
Carrier frequencies	900 MHz 1800 MHz	850 MHz 1900MHz	1900 – 1980 MHz 2010 – 2025 MHz 2110 – 2170 MHz
Peak data rate	64 kbit/s	64 kbit/s	2 Mbit/s
Multiple access	TDMA	CDMA	CDMA
Services	Voice, low rate data	Voice, low rate data	Voice, low rate data

The common feature of the next generation of wireless technology will be the convergence of multimedia services such as speech, audio, video, image, and data. Thus, high speed data will be able to connect to different networks so that the users may switch seamlessly between existing and future standards. The rapid increase in the number of wireless mobile terminal subscribers will exceed the current 1 billion users. Wireless communication systems are not limited to cellular mobile telecommunication systems such as GSM, IS-95, cdma2000 but also includes wireless local area networks (WLANS), e.g., Hiperlan/2, IEEE 802.11a/b, bluetooth and wireless local loops (WLL) as well as broadcast systems, such as digital audio broadcasting (DAB) and digital video broadcasting (DVB).

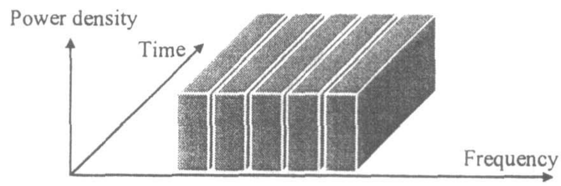
The primary goal of the fourth generation of wireless systems (4G) will not only be the introduction of new technologies to cover the need for higher data rates and new services but also the integration of existing technologies in a common platform. Table (18.2) shows different schemes for some wide area LAN (WLAN) systems used in hotels, train stations, airports and conference rooms where multiple access schemes TDMA or CDMA are employed.

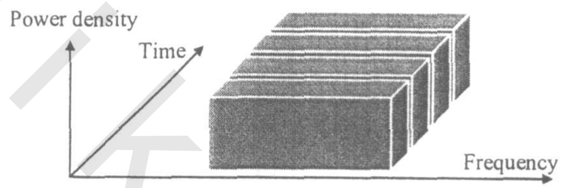
It is noted that FDMA requires low transmission power but requires N_c modulators and demodulators, where N_c is the number of narrow band sub channels. In TDMA, all users employ the same band and are separated by allocating short and distinct time slots. It requires N_c times higher bandwidth.

Table (18.2) WLAN systems

Parameter	Bluetooth	IEEE 802.11b	IEEE 802.11a	HIPERLAN/2
Carrier frequency	2.4 GHz (ISM)	2.4 GHz (ISM)	5 GHz	5 GHz
Peak data rate	1 Mbit/s	5.5 Mbit/s	54 Mbit/s	54 Mbit/s
Multiple access	FH-CDMA	DS-CDMA with carrier sensing	TDMA	TDMA
Services	Ethernet	Ethernet	Ethernet	Ethernet. ATM

In TDMA and FDMA, the channel separation of TDMA and FDMA is based on the orthogonality of signals. Therefore, in a cellular system, the channel interference is only present from the reuse of frequency. On the contrary in CDMA systems, different users transmit at the same time on the same carrier using a wider bandwidth than in a TDMA system. Advantages of the spread spectrum technique are immunity against multi path distortion, simple frequency planning, high flexibility, variable rate transmission and resistance to interference (Fig. 18.30).





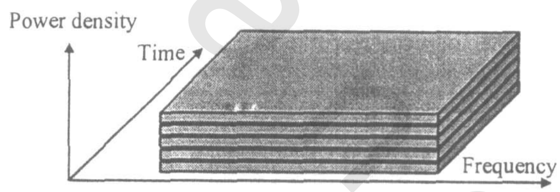


Fig. (18.30) Different multiple access schemes

a) *FDMA (with $N_c = 5$ sub channels).*

b) *TDMA (with 5 time slots).*

c) *CDMA (with 5 spreading codes)*

Table (18.3) Comparison of different multiple access schemes

Multiple access scheme	Advantages	Drawbacks
FDMA	- Low transmit power - Robust to multipath - Easy frequency planning	- Low peak data rate - Loss due to guard bands - Sensitive to narrow band interference
TDMA	- High peak data rate - High multiplexing gain in case of bursty traffic	- High transmit power - Sensitive to multipath - Difficult frequency planning
CDMA	- Low transmit power - Robust to multipath - Easy frequency planning - High scalability - Low delay	- Low peak data rate. - Limited capacity per sector due to multiple access interference

18.17 From 3G to 4G:

The technique multicarrier transmission is recently poised for use as 4G. It is based on the concept of transmitting data simultaneously through a band limited channel without interference between subchannels and without intersymbol interference in time domain through the use of guard time between the transmitted symbols with raised cosine.

The basic principle of multicarrier modulation relies on the transmission of data by dividing a high rate data stream into several low rate substreams. These substreams are modulated on different subcarriers. By using a large number of subcarriers high immunity against multipath dispersion can be provided since the useful symbol duration T_s on each substream will be much longer than the channel time dispersion, since T_s is made low on account of reducing the bit rate within the substream compared to the bit rate of the original data stream. Hence, the effect ISI is minimized.

Since the amount of filters and oscillators needed is considerable for a large number of subcarriers, an efficient digital implementation of a special form of multicarrier modulation - called orthogonal frequency division multiplexing (OFDM) - is used with rectangular pulse shaping and guard time. OFDM can be easily realized by using discrete Fourier transform (DFT). OFDM having densely spaced subcarriers with overlapping spectra of the modulated signals does not require steep BPFs to detect each carrier as it is needed in FDMA schemes. Progress in digital technology has enabled the realization of a DFT for large numbers of subcarriers (several thousands). OFDM has been adopted in ADSL , DAB and DVB standards as well as in WLAN standards (hyperlan /2 and $IEEE\ 802$) (Table 18.4).

The advantages of multicarrier (MC) modulation and the flexibility offered by the spread spectrum technique have led to a new technique called multi carrier spread system (MC – SS). Thus, new multiple access schemes are introduced namely, MC – CDMA and MC – DS – CDMA. Multi carrier modulation and multi carrier spread spectrum are today considered the cornerstone of the 4G high speed wireless multimedia communication systems where spectral efficiency and flexibility are considered be most important criteria for the new era of communication.

Table (18.4) Wireless systems using OFDM

Parameter	DAB	DVB-T	IEEE 802.11a	HIPERLAN/2
Carrier frequency	VHF	VHF and UHF	5 GHz	5 GHz
Bandwidth	1.54 MHz	8 MHz (7 MHz)	20 MHz	20 MHz
Max. data rate	1.7 Mbit/s	31.7 Mbit/s	54 Mbit/s	54 Mbit/s
Number of sub-carriers (FFT size)	192 up to 1536 (256 up to 2048)	1705 and 6817 (2048 and 8196)	52 (64)	52 (64)

18.18 Orthogonal Frequency Division Multiplexing (OFDM):

The principle of multi carrier transmission is to convert a serial high rate data stream on to multiple parallel low rate substreams. Each substream is modulated on another subcarrier. Since the symbol rate on each subcarrier is much less than the initial serial data symbol rate, the effects of delay are minimized reducing ISI. An example of MC modulator with four subchannels $(N_c = 4)$ is shown (Fig. 18.31). A cuboid indicates the 3D time / frequency / power density range of the signal. In OFDM-based mobile communication, the channel may be considered as time invariant during one OFDM symbol. Thus, the OFDM symbol duration should be smaller than the coherence time $(\Delta t_c)$of the channel, which is the time in which the channel is considered time invariant. The subcarrier spacing should be smaller than the coherence bandwidth (Δt_c) of the channel, which is the band over which the signal propagation characteristics are correlated. The channel is frequency selective if the signal bandwidth is greater than the coherence bandwidth Δf_c.

A communication system with MC modulation transmits N_c complex valued source symbols $S_n, n = o..\ N_c - 1$ in parallel on N_c subcarriers. The source symbol duration T_d of the serial data symbols results after serial to parallel conversion in the OFDM symbol duration T_s.

$$T_s = N_c T_d \tag{18 - 1}$$

The principle of OFDM is to modulate the N_c substreams on subcarriers with a spacing of

$$\Delta f_s = \frac{1}{T_s} \tag{18 - 2}$$

in order to achieve orthogonality between the signals on the N_c subcarriers presuming a rectangular pulse shaping. The N_c parallel modulated source symbols $S_n, n = o,..N_c - 1$ are referred to as OFDM symbol. The complex envelope of an OFDM symbol with rectangular pulse shaping has the form

$$x(t) = \frac{1}{N_c} \sum_{n=0}^{N_c - 1} S_n e^{j 2\pi f_n t} \qquad (18 - 3)$$

The N_c subcarrier frequencies are located at

$$f_n = \frac{n}{T_s} \quad n = 0,..N_c - 1 \qquad (18 - 4)$$

The normalized power density of each of OFDM symbols with 16 subcarriers versus the normalized frequency fT_d is shown (Fig. 18.32) as a solid line. The dotted curve illustrates the power density spectrum of the first modulated subcarrier and indicates the construction of the overall power density spectrum as the sum of N_c individual power density spectra each shifted by Δf_s. For large values of N, the power spectrum becomes flatter in the range $-0.5 \le fT_d \le 0.5$ containing the N_c subchannels. As N_c becomes large, the power density spectrum approaches that of single carrier modulation with ideal Nyquist filtering.

A key advantage of using OFDM is that MC modulation can be implemented in the discrete domain by using IDFT or IFFT [(1)] .When sampling the complex envelope $x(t)$ of an OFDM symbol with rate $1/T_d$ the samples are

$$x_v = \frac{1}{N_c} \sum_{n=0}^{N_c - 1} S_n e^{j 2\pi v / N_c} \qquad (18 - 5)$$

The sampled sequence $x_v, v = 0,..N_c - 1$ is the IDFT of the source symbol sequence $S_n \; n = 0,..N_c - 1$.

The block diagram of a MC modulator employing OFDM based on an IDFT and an MC demodulator employing inverse OFDM based on DFT is illustrated in Fig. (18.33).

Thus, the first step in MC modulation is serial to parallel conversion, then modulating by carriers of subbands. This is done by software rather than by oscillators. This is in the frequency domain. Using IDFT, samples in the time domain are obtained, then converting parallel to serial and adding guard intervals. Digital to analog converter changes the digital signal to analog. In the receiver, the reverse process is carried out.

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(1) IDFT = Inverse discrete Fourier transform
IFFF = Inverse fast Fourier

The key parameters of various MC based communication standards are summarized in Tables (18.5),(18.6) and (18.7). It should be noted that multitone operation in DSL is a form of OFDM.

Table (18.5) Broadcasting standards DAB and DVB - T

Parameter	DAB			DVB-T	
Bandwidth	1.5 MHz			8 MHz	
Number of sub-carriers N_c	192 (256 FFT)	384 (512 FFT)	1536 (2k FFT)	1705 (2k FFT)	6817 (8k FFT)
Symbol duration T_s	125 μs	250 μs	1 ms	224 μs	896 μs
Carrier spacing f_S	8 kHz	4 kHz	1 kHz	4.464 kHz	1.116 kHz
Guard time T_g	31 μs	62 μs	246 μs	$T_s/32$, $T_s/16$, $T_s/8$, $T_s/4$	
Modulation T_g	D-QPSK			QPSK, 16-QAM, 64-QAM	
FEC coding	Convolutional with code rate 1/3 up to 3/4			Reed Solomon + convolutional with code rate 1/2 up to 7/8	
Max. data rate	1.7 Mbits			31.7 Mbit/s	

Table (18.6) Wireless local area network WLAN standards

Parameter	IEEE 802.11a, HIPERLAN/2
Bandwidth	20 MHz
Number of sub-carriers N_c	52 (64 FFT)
Symbol duration T_s	4 μs
Carrier spacing f_S	312.5 kHz
Guard time T_g	0.8 μs
Modulation T_g	BPSK, QPSK. 16-QAM, and 64-QAM
FEC coding	Convolutional with code rate 1/2 up to 3/4
Max. dat rate	54 Mbit/s

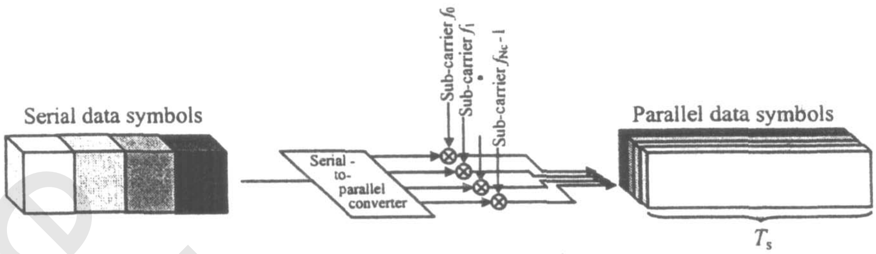

Fig. (18.31) MC Modulation with $N_c = 4$

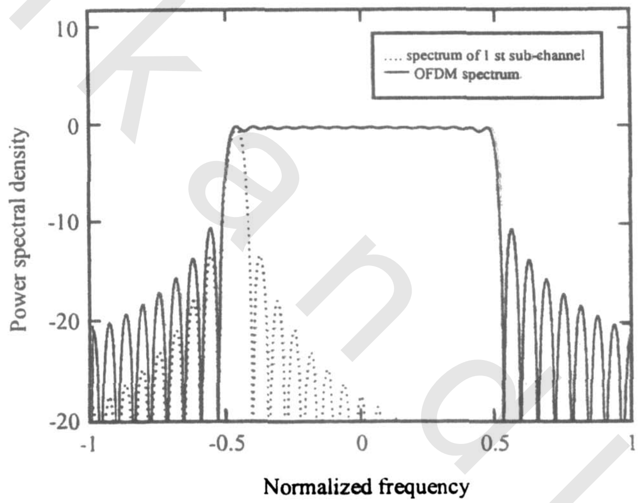

Fig. (18.32) OFDM spectrum with 16 subcarriers

Table (18.7) Wireless local log (WLL) standards

Parameter	Draft IEEE 802.16a, HIPERMAN	
Bandwidth	from 1.5 to 28 MHz	
Number of sub-carries N_c	256 (OFDM mode)	2048 (OFDMA mode)
Symbol time T_s	from 8 to 125 μs (depending on bandwidth)	from 64 to 1024 μs (depending on bandwidth)
Guard time T_g	from 1/32 up to 1/4 of T_s	
Modulation	QPSK, 16-QAM, and 64-QAM	
FEC coding	Reed Solomon + convolutional with code rate 1/2 up to 5/6	
Max. data rate (in a 7 MHz channel)	up to 26 Mbit/s	

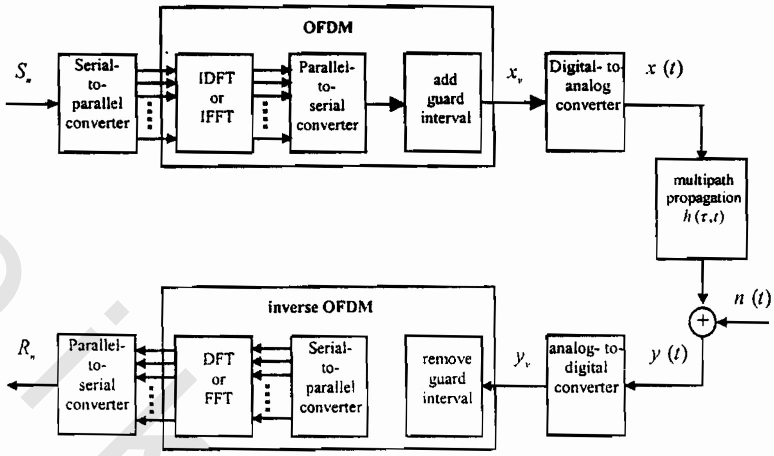

Fig (18.33) MC system employing OFDM

18.19 Multicarrier Spread Spectrum:

Fig. (18.34) shows the principle of DS – CDMA before MC is incorporated. Since 1993, various combinations of MC modulation with the spread spectrum technique as multiple access schemes have been introduced. Two philosophies exist namely MC – CDMA (OFDM – CDMA) and MC – DS – CDMA (Fig. 18.35 and Table 18.8).

MC – CDMA is based on a serial concatenation of direct sequence (DS) spreading with MC modulation. The high rate DS spread data stream is multicarrier modulated in the way that chips of a spread data symbol are transmitted in parallel and the assigned data symbol is simultaneously transmitted on each subcarrier.

As for DS – CDMA, a user may occupy the total bandwidth for the transmission of a single data symbol. Separation of the user's signal is performed in the code domain. Each data symbol is copied on the substreams before multiplying it with a chip of the spreading code assigned to the specific user. This reflects that an MC – CDMA system performs the spreading in frequency direction and, thus has an additional degree of freedom compared to a DS – CDMA. Mapping of the chips in the frequency direction allows for simple methods of signal detection.

A guard time is needed between adjacent OFDM symbols to prevent ISI as to assure that the symbol duration is longer than the time dispersion of the channel. The number of subcarriers N_c has to be large to guarantee frequency non selective fading on each subchannel. From Fig. (18.35), MC – DS – CDMA modulates substreams on subcarriers with a carrier spacing proportional to the inverse of the chip rate. This will guarantee orthogonality between the specta of the substreams. If the spreading code length is smaller or equal to the number of subcarriers N_c a single data symbol is not spread in the frequency direction. Instead, it is spread in the time direction. Spread spectrum is obtained by modulating N_c time spread data symbols on parallel subcarriers. By using high numbers of subcarriers we may benefit from time diversity.

Frequency diversity can only be exploited if the same information is transmitted on several subcarriers in parallel. Furthermore, higher frequency diversity could be achieved if the carrier spacing is chosen larger than the chip rate.

When $N_c = 1$, the classical DS – CDMA transmission is obtained, whereas without spreading $(P_G = 1)$ the result is pure OFDM.

Table (18.8) Characteristics of MC - SS

Parameter	MC-CDMA	MC-DS-CDMA
Spreading	Frequency direction	Time direction
Sub-carrier spacing	$f_s = \frac{P_G}{N_c\ T_d}$	$f_s \geq \frac{P_G}{N_c\ T_d}$
Specific characteristics	Very efficient for the synchronous downlink by using orthogonal codes	Designed especially for an asynchronous uplink
Applications	Synchronous uplink and downling	Asynchronous uplink and downlink

18.20 Wireless Networks: Bluetooth, WiFi and WiMAX:

Wireless networks can be classified into 3 categories: Wireless local area networks (WLANs), wireless metropolitan area networks (WLANs) and wireless personal area networks (WPANs). Each of these groups is designed to accommodate the specific needs of the networks they provide.

WPANs are short range networks that provide high data transfer over short distances. (WLANs) are medium range networks that reduce data rate to accommodate for larger range, while WMANs have a long range with higher data rate than (WLANs) at the same range but is still less than (WPANs) (Fig. 18.36). WMANs are used to network an area the size of a city within an area of $50km$ radius.

WLANs network a small community of users such as a campus. WPANs are usually used for multimedia communication such as Bluetooth in cell phones which have a range of up to $10m$.

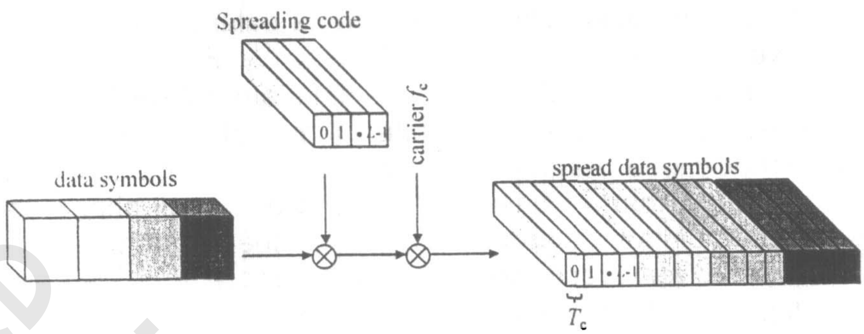

Fig. (18.34) Principle of DS – CDMA

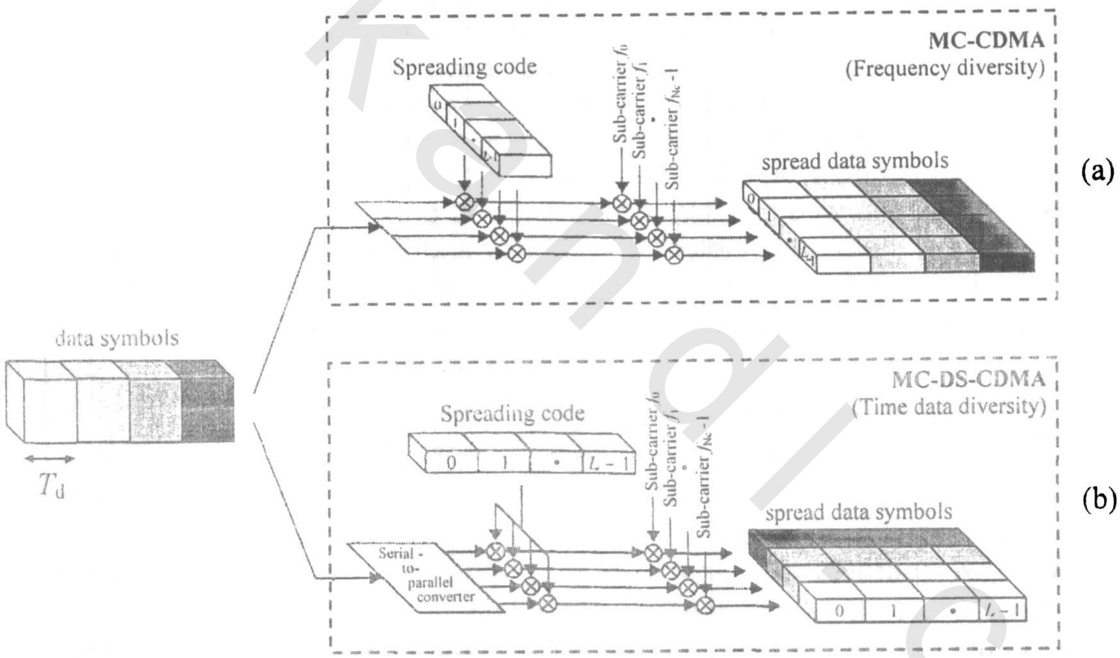

Fig. (18.35) Principle of MC – CDMA and MC – DS - CDMA systems
a) frequency diversity (data symbol spread in frequency)
b) time diversity(data symbol spread in time)

To choose the best networks requires a balance between range, data rate, cost and power consumption. Wireless Fidelity (Wi-Fi) is a name commonly given to IEEE802.11x standard. WiFi is a WLAN standard. WiMAX is a WMAN standard.

Wireless technology can operate in one of two modes *viz* infrastructure mode and adhoc mode. As the name implies, the first mode of operation requires infrastructure in the form of base stations similar to cell phone users. This mode requires access points so that a device can connect to the wireless network. This makes this mode very dependent on the hardware and also very prone to the network collapsing if the base station is damaged.

In the adhoc mode of operation, the devices that form the network work on peer to peer manner, i.e., the routing from one device to the other is done via the other devices in the network, i.e., connections are made spontaneously from the transmitting device to the receiver. This makes the network robust and independent of a single base station. Adhoc networks cost less than infrastructure networks and are faster to set up but they must have a limited number of users.

WPANs are used to provide data transfer over short distances. WPANs are also used in multimedia. Another example is lighting access control, monitoring of patients in hospitals, wireless keyboards and PC peripherals, DVD, VCR, and TV remotes. Among them are Bluetooth, Wi Media and Zig Bee. Bluetooth operates in the $2.4GHz$ band and is capable of transmitting up to $2.178Mbps$ data rate. The Wi Media is used for devices such as TVs and digital cameras of data rates $55Mbps / 480Mbps$. The Zig Bee operates in the $2.4GHz$ band and has low data rate $9.6kbps - 250kbps$ and has a range $70-300m$.

The $WLANs$ have different standards. They operate in $2.4GHz / 5GHz$ and have data rates $11Mbps$ for a range 50m indoors and 250 outdoors, or $54Mbps$ for 18m indoors and 30m outdoors.

WiMAX operates between $2-66GHz$ and has $120Mbps$ data rate with a single wired base station. WiMAX uses OFDM. WiMAX is considered to be the broadband alternative to cable and DSL. It will provide fixed, nomadic, portable and mobile wireless broadband connectivity without the need for direct line of sight. With a base station with capacity of $40Mbps$ / channel for fixed and portable access application, there is enough bandwidth for simultaneously supporting hundreds of business with $T-1(E-1)$ speed connectivity and thousands of residences with DSL speed connectivity. WiMAX technology will be incorporated in notebook computers, allowing for urban areas and cities to become metro zones for portable outdoor broadband wireless access.

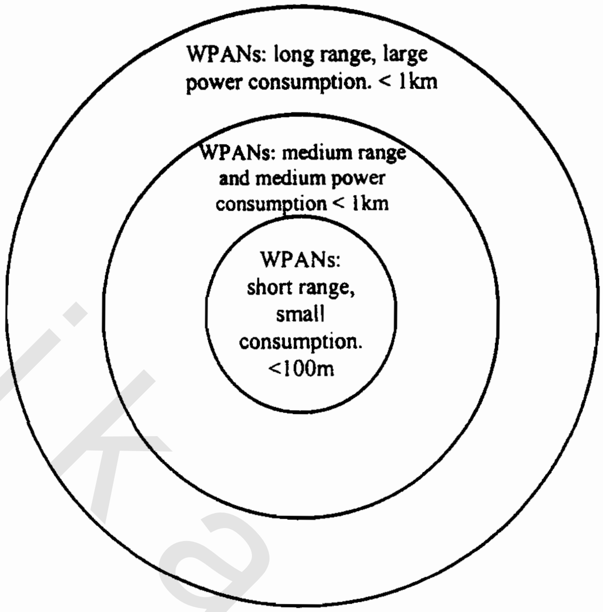

Fig. (18.36) Typical range for wireless networks

The radio channel of a wireless communication is often described as being either line of sight (LOS) or no line of sight (NLOS). In a LOS link, a signal travels over a direct and unobstructed path from the transmitter to the receiver.

In an NLOS link, a signal reaches the receiver through reflections and diffractions. The signals arriving at the receiver consists of components from the direct path, multiple reflected paths, scattered energy and diffracted propagation paths. Thus, signals have different delay spreads, attenuation, polarization, stability relative to the direct path. How a radio system uses these multi path signals to an advantage is the key to providing service in the NLOS conditions. There are several advantages that make NLOS deployment desirable. For instance, strict planning requirements and antenna height restrictions often do not allow the antenna to be positional for LOS. Base stations are also forced to operate in NLOS conditions since antenna height should not be too great or else cause interference between adjacent sites.

18.21 Personal Satellite Communication Systems:

An example of satellite mobile communication systems is INMARSAT for global maritime communication. The systems belonging to this family may also be used in individual communications with a mobile station of the size of a briefcase or laptop. The characteristic feature of currently existing systems is unidirectional or bidirectional voice and / or data communications at a limited quality in very large area. In recent years, a few new satellite systems are operational, among others those known under the names of Iridium, Globalstar and ICO. All of these make use of satellites which are located on low or medium orbits. In case of low orbits, the number of satellites has to be large, the coverage of each of them is small but the size of the satellite and its cost is low as compared with a satellite requiring higher orbits. Additionally, the delay introduced by the way to and from the satellite is relatively small and the power of a mobile station located on the Earth surface can be low allowing for the application of handheld phones. A large number of low orbit satellites having a small coverage implies a high overall system capacity due to the multiple usage of the channel frequencies. The geostationary satellite systems require the smallest number of satellites. However, each is expensive. Due to a high satellite orbit ($37000km$ above the Earth) each satellite covers a large area, thus, the channel reuse is much lower as compared with the low orbit satellite systems. Additional difficulty is a substantial delay introduced due to a long signal propagation to and from a geostationary satellite$(0.5s)$. Such a delay makes the voice transmission unpleasant for the user. A compromise between low orbit and geostationary systems is the system with intermediate circular orbits (ICO).

Problems

1. If BER is 10^{-3} show why information should be divided into blocks. What is the size of each block?
2. If BER is p and the number of bits in a block is N then assuming random errors show that the probability of a block containing a bit error P_B is given by $P_B = 1-(1-p)^N$.
3. What is the expression for P_B if $Np < 1$
4. Derive the maximum block size that should be used over a channel which has $BER = 10^{-4}$ if the probability of a block containing an error and hence discarded is 0.1
5. Determine the propagation delay in
   a) connection through a private network of 2 km and the velocity of propagation is $2 \times 10^8\, m/s$
   b) connection of PSTN of $300km$ and the velocity of propagation is $2 \times 10^8\, m/s$
   c) connection over a satellite channel of $60000km$ and the velocity of propagation is $3 \times 10^8\, m/s$
6. Assuming a file size of $100Mbp$, find the minimum time to transmit the file using the following different Internet access modes.
   a) PSTN and $288kbps$ modem
   b) ISDN at $64kbps$
   c) ISDN at $128kbps$
7. Redo Ex. 18.1 if the input bit rate is $1.5Mbps$. What do you conclude?
8. What happens if the maximum network packet size is 8000 bits in the problem above?
9. A web page of $10MB$ is being retrieved from a web server. Assuming negligible delays within the server and trunk network quantify the time to transfer the page over the following types of access circuits.
   - PSTN with modem at $28.8kbps$
   - ISDN at $1.5Mpbs$
   - High speed modem at $6Mbps$
   - Cable modem at $27Mbps$
10. Verify the OFDM spectrum shown in Fig. (18.34)

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