multimedia communications

1.1 Introduction

Within the context of this book, multimedia communications embraces a range of applications and networking infrastructures. The term "**multimedia**" is used to indicate that the information/data being transferred over the network may be composed of one or more of the following media types:

- **text:** this includes both unformatted text, comprising strings of characters from a limited character set, and formatted text strings as used for the structuring, access, and presentation of electronic documents;
- **images:** these include computer-generated images, comprising lines, curves, and circles, and digitized images of documents and pictures;
- **audio:** this includes both low-fidelity speech, as used in telephony, and high-fidelity stereophonic music as used with compact discs;
- **video:** this includes short sequences of moving images (also known as video clips) and complete movies/films.

The applications may involve either person-to-person communications or person-to-system communications. In general, two people communicate with each other through suitable **terminal equipment (TE)** while a person interacts with a system using either a **multimedia personal computer (PC)** or **workstation.** Typically, these are located either in the home or on a desktop in an office and the system is a **server** containing a collection of files or documents each comprising digitized text, images, audio, and video information either singly or integrated together in some way. Alternatively, the server may contain a library of digitized movies/videos and the user interacts with the server by means of a suitable selection device that is connected to the **set-top box (STB)** associated with a television.

In practice, there are a number of different types of network that are used to provide the networking infrastructure. These include not only networks that were designed from the outset to provide multimedia communication services but also networks that were designed initially to provide just a single type of service and it is as a result of advances in various technologies that these can now provide a range of other services. For example, **public switched telephone networks** (**PSTNs**) – also known as **general switched telephone networks** (**GSTNs**) – were designed initially to provide a basic switched telephone service but, as a result of advances in digital signal processing hardware and associated software, they now provide a range of more advanced services involving text, images, and video. Similarly, data networks that were designed initially to support basic data applications such as electronic mail and file transfers, now support a much richer set of applications that involve images, audio, and video.

In this chapter we shall present an overview of, firstly, how the different media types are represented, secondly, the different types of network that are used to provide multimedia communication services, and thirdly, a selection of the applications that these networks support. Finally, we describe the meaning of a range of terms that are associated with multimedia communications.

1.2 Multimedia information representation

Applications involving text and images comprise **blocks** of digital data. In the case of text, for example, a typical unit is a block of characters with each character represented by a fixed number of **binary digits** (**bits**) known as a **codeword**. Similarly, a digitized image comprises a two-dimensional block of what are called **picture elements** with each element represented by a fixed number of bits. Also, since a typical application involving text and images comprises a short request for some information – a file, for example – and the file contents being returned, the duration of the overall transaction is relatively short.

In applications involving audio and video, however, the audio and video signals vary continuously with time as the amplitude of the speech, audio, or video signal varies. This type of signal is known as an **analog signal** and, typically, the duration of applications that involve audio and/or video can be relatively long. A typical telephone conversation, for example, can last for several minutes while a movie (comprising audio and video) can last for a number of hours.

In applications that involve just a single type of media, the basic form of representation of the particular media type is often used. Similarly, in applications that involve either text-and-images or audio-and-video their basic form is often used since the two media types in these applications have the same form of representation. However, in applications that involve the different media types integrated together in some way, it becomes necessary to represent all four media types in a digital form. In the case of text and images, this is their standard form of representation. For audio and video, however, because their basic forms of representation are analog signals, these must be converted into a corresponding digital form before they can be integrated with the two other media types.

As we shall describe in the next chapter, the digitization of an audio signal produces a digital signal which, because the amplitude of the signal varies continuously with time, is of a relatively high bit rate. This is measured in **bits per second** (**bps**) and, in the case of a speech signal, for example, a typical bit rate is 64 kbps. Moreover, because applications involving audio can be of a long duration, this bit rate must be sustained over an equally long time period. The same applies to the digitization of a video signal except that much higher bit rates and longer time durations are involved. In general, however, as we shall expand upon in the next section, the communication networks that are used to support applications that involve audio and video cannot support the very high bit rates that are required for representing these media types in a digital form. As a result, a technique known as **compression** is first applied to the digitized signals in order to reduce the resulting bit rate to a level which the various networks can support. Compression is also applied to text and images in order to reduce the time delay between a request being made for some information and the information becoming available on, say, the screen of a computer. We shall describe a selection of the compression algorithms that are used with text and images in Chapter 3 and those used with audio and video in Chapter 4.

1.3 Multimedia networks

There are five basic types of communication network that are used to provide multimedia communication services:

- telephone networks,
- data networks,
- broadcast television networks,
- integrated services digital networks,
- broadband multiservice networks.

As the names imply, the first three network types were initially designed to provide just a single type of service: telephony, data communications, and broadcast television respectively. The last two network types, however, were designed from the outset to provide multiple services. We shall describe the essential features of each type of network separately and, in the case of the first three network types, the technological developments that have enabled them to provide additional services.

1.3.1 Telephone networks

Public switched telephone networks have been in existence for many years and have gone through many changes during this time. They were designed to provide a basic switched telephone service which, with the advent of the other network types, has become known as a **plain old telephone service** or **POTS**. The term "switched" is used to indicate that a subscriber can make a call to any other telephone that is connected to the total network. Initially, such networks spanned just a single country but later the telephone networks of different countries were interconnected so that they now provide an international switched service. The main components of the network are shown in diagrammatic form in Figure 1.1(a).

As we can see, telephones located in the home or in a small business are connected directly to their nearest **local exchange/end office**. Those located in a medium or large office/site are connected to a private switching office known as a **private branch exchange** or **PBX**. The PBX provides a (free) switched service between any two telephones that are connected to it. In addition, the PBX is connected to its nearest local (public) exchange which enables the telephones that are connected to the PBX also to make calls through a PSTN. More recently, **cellular phone networks** have been introduced which provide a similar service to mobile subscribers by means of handsets that are linked to the cellular phone network infrastructure by radio. The switches used in a cellular phone network are known as **mobile switching centers (MSCs)** and these, like a PBX, are also connected to a switching office in a PSTN which enables both sets of subscribers to make calls to one another. Finally, international calls are routed to and switched by **international gateway exchanges (IGEs)**.

As we indicated earlier, a speech signal is an analog signal since it varies continuously with time according to the amplitude and frequency variations of the sound resulting from the speech. A microphone is used to convert this into an analog electrical signal. Because of this, telephone networks operate in what is called a **circuit mode** which ~~means that~~, for each call, a separate circuit is set up through the network – of the necessary capacity – for the duration of the call. The **access circuits** that link the telephone handsets to a PSTN or PBX were designed, therefore, to carry the two-way analog signals associated with a call. Hence, although within a PSTN all the switches and the transmission circuits that interconnect them now operate in a **digital mode**,

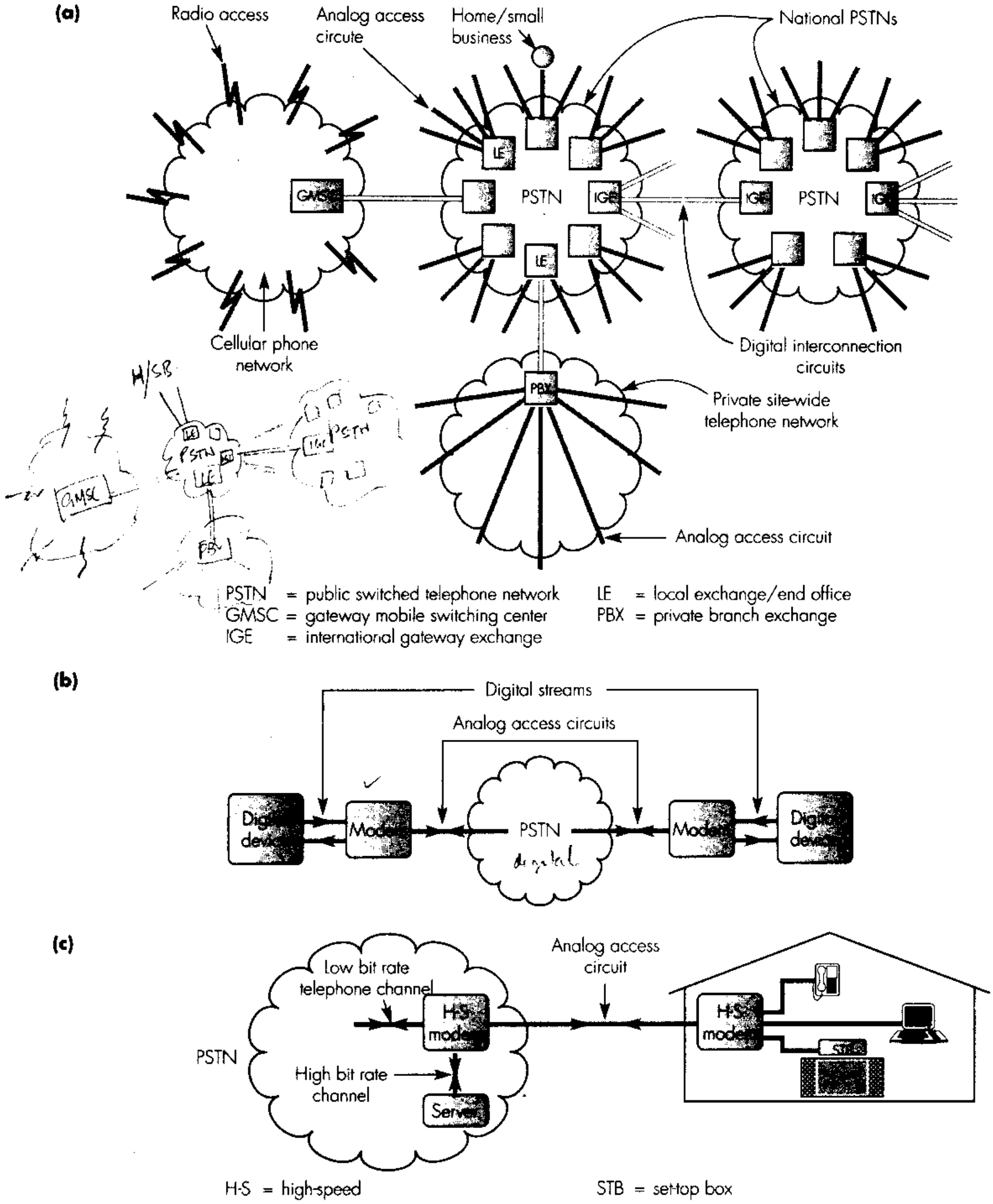


Figure 1.1 Telephone networks: (a) network components; (b) digital transmission using modems; (c) multiple services via an H-S modem.

to carry a digital signal – a stream of binary 1s and 0s – over the analog access circuits requires a device known as a **modem**. The general scheme is shown in Figure 1.1(b).

Essentially, at the sending side, the modem converts the digital signal output by the source digital device into an analog signal that is compatible with a normal speech signal. This is routed through the network in the same way as a speech signal and, at the receiving side, the modem converts the analog signal back again into its digital form before relaying this to the destination digital device. Modems also have the necessary circuits to set up and terminate a call. Hence by using a pair of modems – one at each subscriber access point – a PSTN can also be used to provide a switched digital service. The early modems supported only a very low bit rate service of 300 bps but, as a result of advances in digital signal processing circuits, modems are now available that support bit rates of up to 56 kbps. As we shall expand upon in Chapter 4, this is sufficient to support, not only applications that comprise text and images integrated together, but also services that comprise speech and low-resolution video.

In addition, continuing advances in digital signal processing techniques mean that modems are now available for use with the same access circuits that provide a high bit rate channel which is in addition to the speech channel used for telephony. Typically, the bit rate of this second channel is such that it can support high-resolution audio and video and hence they are used to provide access to servers that support a range of entertainment-related applications. The general scheme is shown in Figure 1.1(c) and, as we shall see in Chapter 4, such applications require bit rates in excess of 1.5 Mbps. This illustrates the technological advances that have been made in this area since the early modems were introduced in the early 1960s and, as we can deduce from this, a PSTN can now support not only speech applications but also a wide range of other multimedia communication applications.

converts ... analog ... necessary ckts to set up and terminate calls

early modems 300bps

modems are now available for use with the same access ckts to provide high bit rate channel which in addition to speech ch. used for telephony

hi-res video audio access to servers that provide range entertainment related apps

1.3.2 Data networks

Data networks were designed to provide basic data communication services such as **electronic mail (email)** and general file transfers. The user equipment connected to these networks, therefore, is a computer such as a PC, a workstation, or an email/file server. The two most widely deployed networks of this type are the **X.25 network** and the **Internet**. Because of its operational mode, however, the X.25 network is restricted to relatively low bit rate data applications and hence is unsuitable for most multimedia applications.

The Internet is made up of a vast collection of interconnected networks all of which operate using the same set of **communication protocols**. A communication protocol is an agreed set of rules that are adhered to by all communicating parties for the exchange of information. The rules define not only the ~~sequence~~ of messages that are exchanged between the communicating parties but also the syntax of these messages. Hence by using the same set of communication protocols, all the computers that are connected to the Internet can communicate freely with each other irrespective of their type or manufacturer. This is also the origin of the term "**open systems interconnection**". Figure 1.2 shows a selection of the different types of interconnected network.

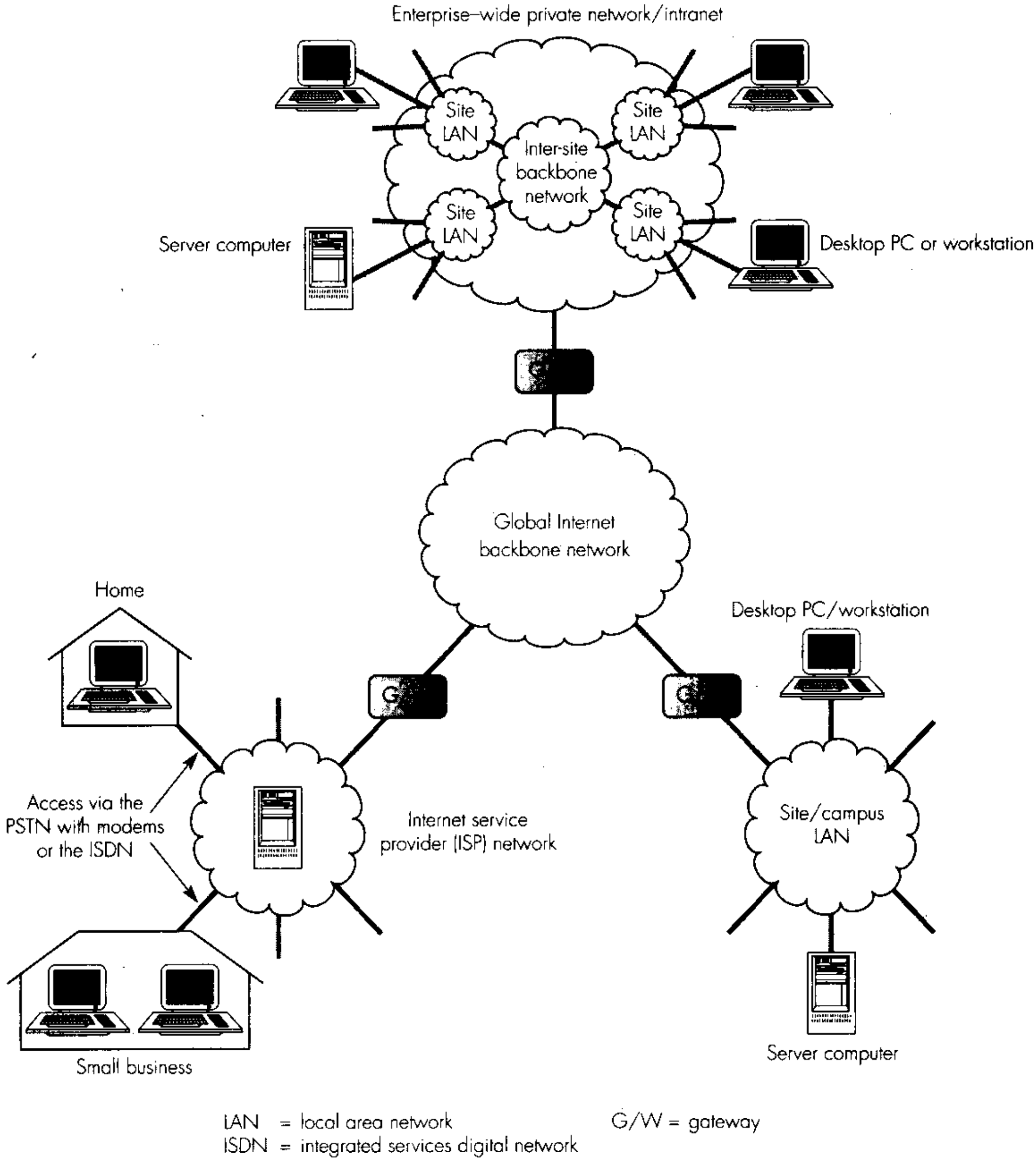


Figure 1.2 A selection of the network types connected to the Internet.

As we can see, in the case of a user at home or in a small business, access to the Internet is through an intermediate **Internet service provider (ISP) network**. Normally, since this type of user wants access to the Internet intermittently, the user devices are connected to the ISP network either through a PSTN with modems or through an **integrated services digital network (ISDN)** which, as we shall explain in Section 1.3.4, provides access at a higher bit rate. Alternatively, business users obtain access either through a **site/campus network** if the business comprises only a single site or, if it comprises multiple sites, through an **enterprise-wide private network**. The same approach is used by most colleges and universities. In the case of a single site/campus, the network is known as a (private) **local area network** or **LAN**. For an enterprise-wide network comprising multiple sites the sites are interconnected together using an **inter-site backbone network** to provide a set of enterprise-wide communication services. In addition, providing the communication protocols used by all the computers connected to the network are the same as those defined for use with the Internet, then all the users also have access to the range of services provided by the Internet. The enterprise network is then known as an **intranet** since all internal services are provided using the same set of communication protocols as those defined for the Internet. The different types of network are all connected to the **Internet backbone network** through an interworking unit called a **gateway** which, because it is responsible for routing and relaying all messages to and from the connected network, is also known as a **router**.

All data networks operate in what is called a **packet mode**. Essentially, a **packet** is a container for a block of data and, at its head, is the address of the intended recipient computer which is used to route the packet through the network. This mode of operation was chosen since the format of the data associated with data applications is normally in the form of discrete blocks of text or binary data with varying time intervals between each block. More recently, however, multimedia PCs have become available that support a range of other applications. For example, with the addition of a microphone and a pair of speakers – together with a sound card and associated software to digitize the speech – PCs are now used to support telephony and other speech-related applications. Similarly, with the addition of a video camera and associated hardware and software, a range of other applications involving video can be supported. Also, since their introduction, higher bit rate transmission circuits and routing nodes have become available and, as we shall expand upon in Chapters 3 and 4, more efficient algorithms to represent speech, audio, and video in a digital form. Collectively, therefore, this means that packet-mode networks – and the Internet in particular – now support not only general data communication applications but also a range of other multimedia communication applications involving speech, audio, and video.

1.3.3 Broadcast television networks

Broadcast television networks were designed to support the diffusion of analog television (and radio) programs throughout wide geographical areas. In the case of a large town or city, the broadcast medium is normally a **cable**

distribution network while for larger areas, a **satellite network** or sometimes a **terrestrial broadcast network** is used. Since their introduction, digital television services have become available with these networks which, together with a low bit rate return channel for interaction purposes, provide a range of additional services such as games playing and home shopping. The general architecture of a cable distribution network and a satellite/terrestrial broadcast network are shown in Figure 1.3(a) and (b) respectively.

As we can see in Figure 1.3(a), the set-top box attached to the cable distribution network provides not only control of the television channels that are received but also access to other services. For example, when a **cable modem**

backbone n/w

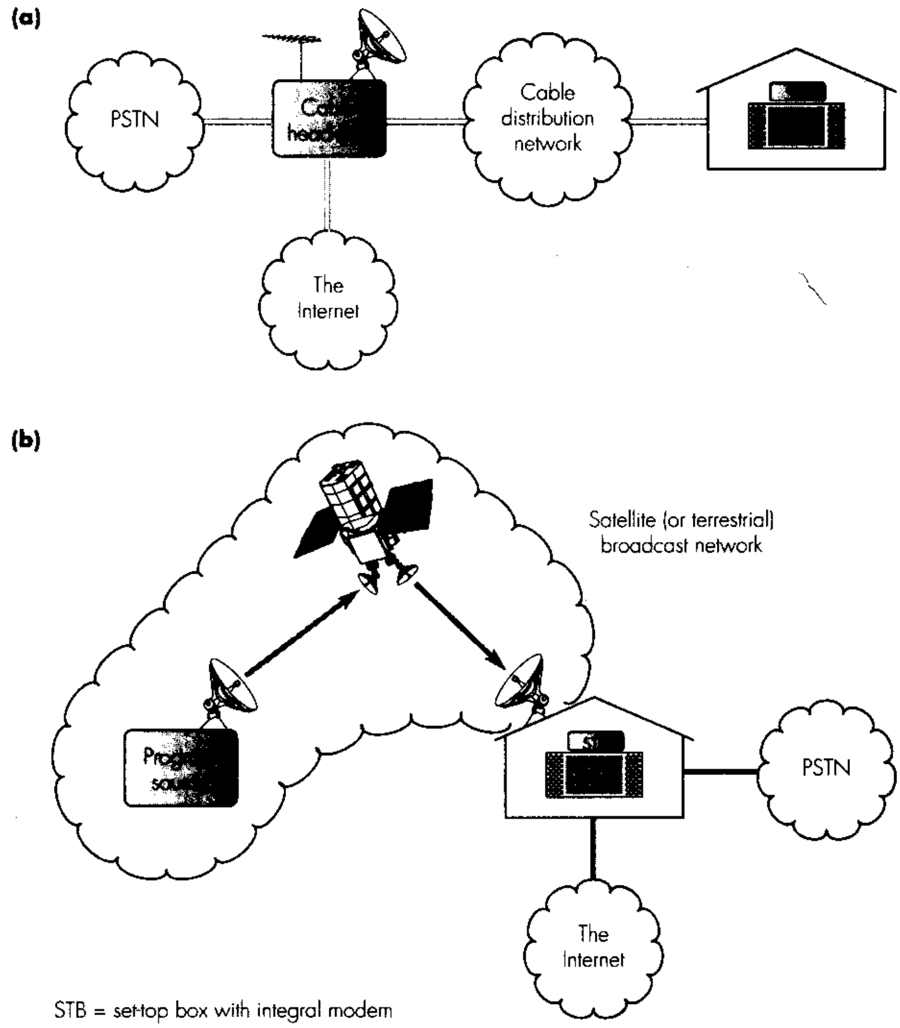


Figure 1.3 Broadcast television networks: (a) cable networks; (b) satellite/terrestrial broadcast networks.

is integrated into the STB this provides both a low bit rate channel and a high bit rate channel from the subscriber back to the **cable head-end**. Typically, the low bit rate channel 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 television services, cable distribution networks also provide access to the range of multimedia communication services that are available with both a PSTN and the Internet. Similarly, as we can see in Figure 1.3(b), in the case of satellite and terrestrial broadcast networks, when a high-speed PSTN modem is integrated into the STB this provides the subscriber with an interaction channel so enhancing the range of services these networks support. This is the origin of the term "**interactive television**".

1.3.4 Integrated services digital networks

Integrated services digital networks started to be deployed in the early 1980s and were originally designed to provide PSTN users with the capability of having additional services. This was achieved firstly, by converting the access circuits that connect user equipment to the network – a telephone for example – into an all-digital form and secondly, by providing two separate communication channels over these circuits. These allow users either to have two different telephone calls in progress simultaneously or two different calls such as a telephone call and a data call. With an ISDN, therefore, the access circuit is known as a **digital subscriber line** (**DSL**).

The subscriber telephone can be either a digital phone or a conventional analog one. In the case of a digital phone, the electronics that are needed to convert the analog voice and call setup signals into a digital form are integrated into the phone handset. With an analog phone, the same electronics are located in the network termination equipment so making the digital mode of operation of the network transparent to the subscriber phone.

As we shall describe in Section 2.5.1, the digitization of a telephone-quality analog speech signal produces a constant bit rate binary stream – normally referred to as a **bitstream** – of 64 kbps. Hence, the basic DSL of the ISDN – known as the **basic rate access** or **BRA** – supports two 64 kbps channels. These can either be used independently (as they were intended) or as a single combined 128 kbps channel. Because of the design of an ISDN, however, since the two channels were intended for two different calls, this requires two separate circuits to be set up through the switching network independently. Hence to synchronize the two separate 64 kbps bitstreams into a single 128 kbps stream requires an additional box of electronics to perform, what is known as, the **aggregation** function.

In addition, a single higher bit rate channel of either 1.5 or 2 Mbps is supported. This is known as the **primary rate access** or **PRA**. Also, a more flexible way of obtaining a switched 128 kbps service has been introduced by many network operators. Indeed, the service provided has been enhanced and now supports a single switched channel of $p \times 64$ kbps where $p = 1, 2, 3 \ldots 30$. The various services provided are summarized in Figure 1.4 and, as we can deduce

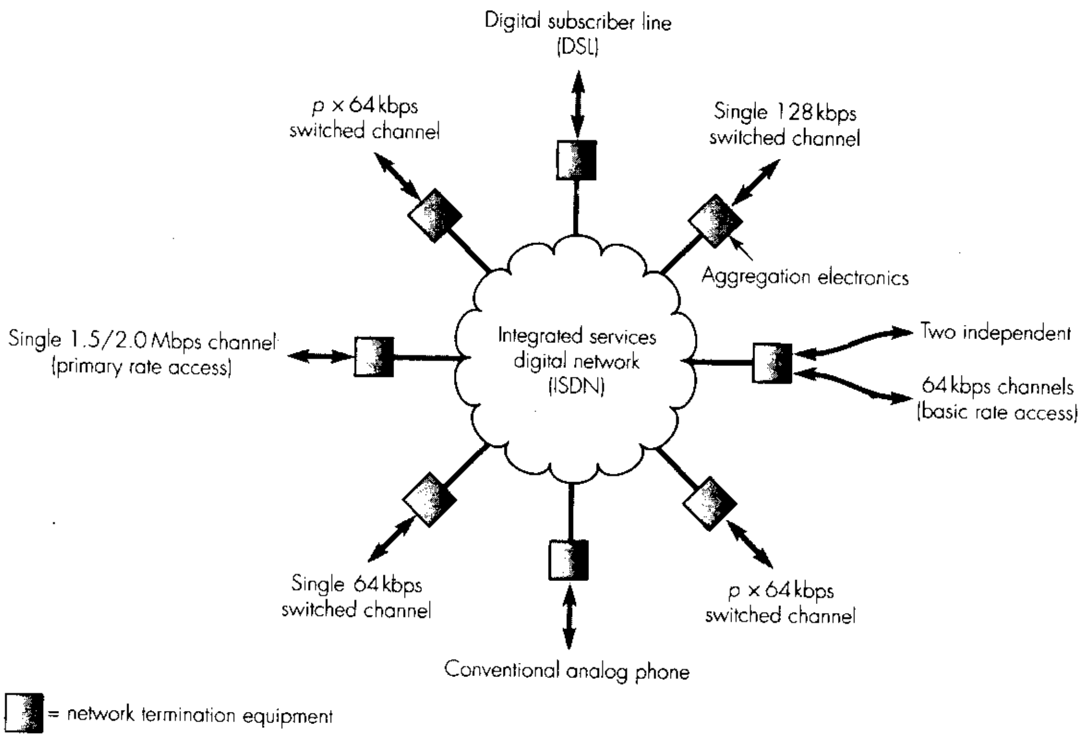


Figure 1.4 Alternative services provided by an ISDN.

from this, an ISDN can support a range of multimedia applications. It should be noted, however, that because of the relatively high cost of digitizing the access circuits, in general the cost of the services associated with an ISDN are higher than the equivalent service provided by a PSTN.

1.3.5 Broadband multiservice networks

Broadband multiservice networks were designed in the mid-1980s for use as public switched networks to support a wide range of multimedia communication applications. The term "**broadband**" was used to indicate that the circuits associated with a call could have bit rates in excess of the maximum bit rate of 2 Mbps – 30 × 64 kbps – provided by an ISDN. As such, they were designed to be an enhanced ISDN and hence were called **broadband integrated services digital networks** or **B-ISDN**. Also, for the same reason, an ISDN is sometimes referred to as **narrowband ISDN** or **N-ISDN**.

At the time the B-ISDN was first conceived, the technology associated with the digitization of a video signal was such that, in general, an ISDN could not support services that included video. Since that time, however, considerable advances have been made in the field of compression with the effect that not only can an ISDN now support multimedia communication applications that include video, but also so can the other three types of network that we have described. The combined effect, therefore, has been to slow down considerably the deployment of B-ISDN. However, a number of the basic design features associated with the B-ISDN have been used as the basis of other broadband multiservice networks.

For example, by definition, multiservice networks implies that the network must support multiple services. In practice, however, as we shall expand upon in the next section, different multimedia applications require different bit rates, the rate being determined by the types of media that are involved. Hence the switching and transmission methods that are used within these networks must be more flexible than those used in networks such as a PSTN or ISDN which were initially designed to provide a single type of service. To achieve this flexibility, all the different media types associated with a particular multimedia application are first converted in the source equipment into a digital form. These are then integrated together and the resulting binary stream is divided into multiple fixed-sized packets known as **cells**. In practice, this type of information stream provides a much more flexible way of both transmitting and switching the multimedia information associated with the different types of application.

For example, in terms of transmission, the cells relating to the different applications can be integrated together more flexibly. Also, the use of fixed-sized cells means that the switching of cells can be carried out much faster than if variable-length packets were used. Since the different multimedia applications generate cell streams of different rates, this mode of operation means that the rate of transfer of cells through the network also varies and hence this mode of transmission is known as the **asynchronous transfer mode** or **ATM**. Broadband multiservice networks, therefore, are also known as **ATM networks** or sometimes **cell-switching networks**. For example, there are **ATM local area networks (ATM LANs)** that span a single site and **ATM metropolitan area networks (ATM MANs)** that span a large town or city. An example of a broadband multiservice network is shown in Figure 1.5 and, as we can see, the ATM MAN is being used as a high-speed backbone network to interconnect a number of LANs distributed around a large town or city. Note also that although two of the LANs are ATM LANs, the other two are simply higher-speed versions of older data-only LANs. This is typical of ATM networks in general which must often interwork with older (legacy) networks.

1.4 Multimedia applications

There are many and varied applications that involve multiple media types. In general, however, they can be placed into one of three categories:

- interpersonal communications,
- interactive applications over the Internet,
- entertainment applications.

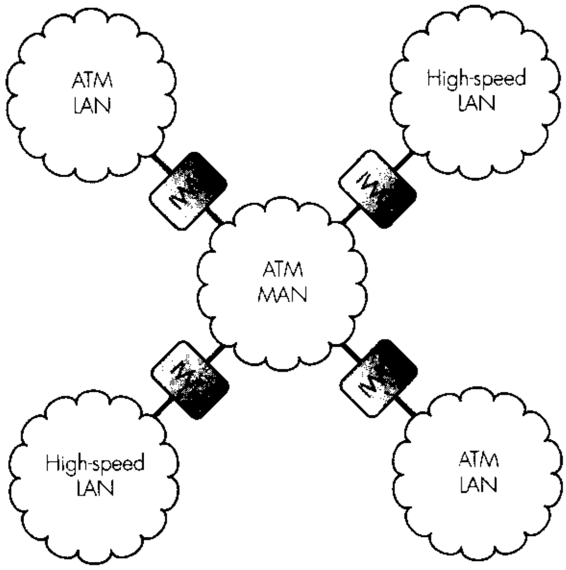


Figure 1.5 Example of an ATM broadband multiservice network.

We shall discuss some examples from each category in the following sections. As we described in Section 1.1, however, in many instances the networks that are used to support these applications were initially designed to provide a service that involved just a single type of medium and it is as a result of advances in various related technologies that they are now used to support multimedia applications. Hence, in addition to these new services, the same networks are still used to support the basic application for which they were designed. Indeed, in some instances, the particular application supported has been enhanced. Thus, although from an applications perspective multimedia communications implies that two or more media types are involved, we shall also include selected examples of applications that these networks were designed to support even though only a single type of medium is involved.

1.4.1 Interpersonal communications

Interpersonal communications may involve speech, image, text, or video. In some cases just a single type of medium is involved while in others two or more media types are integrated together. We shall discuss some examples from each category separately.

Speech only

Traditionally, interpersonal communications involving speech – telephony – have been provided using telephones that are connected either to a public switched telephone network (PSTN/ISDN/cellular phone network) or a PBX. The general scheme is shown in Figure 1.6.

Alternatively, by using a multimedia PC equipped with a microphone and speakers, the user can take part in telephone calls through the PC. This

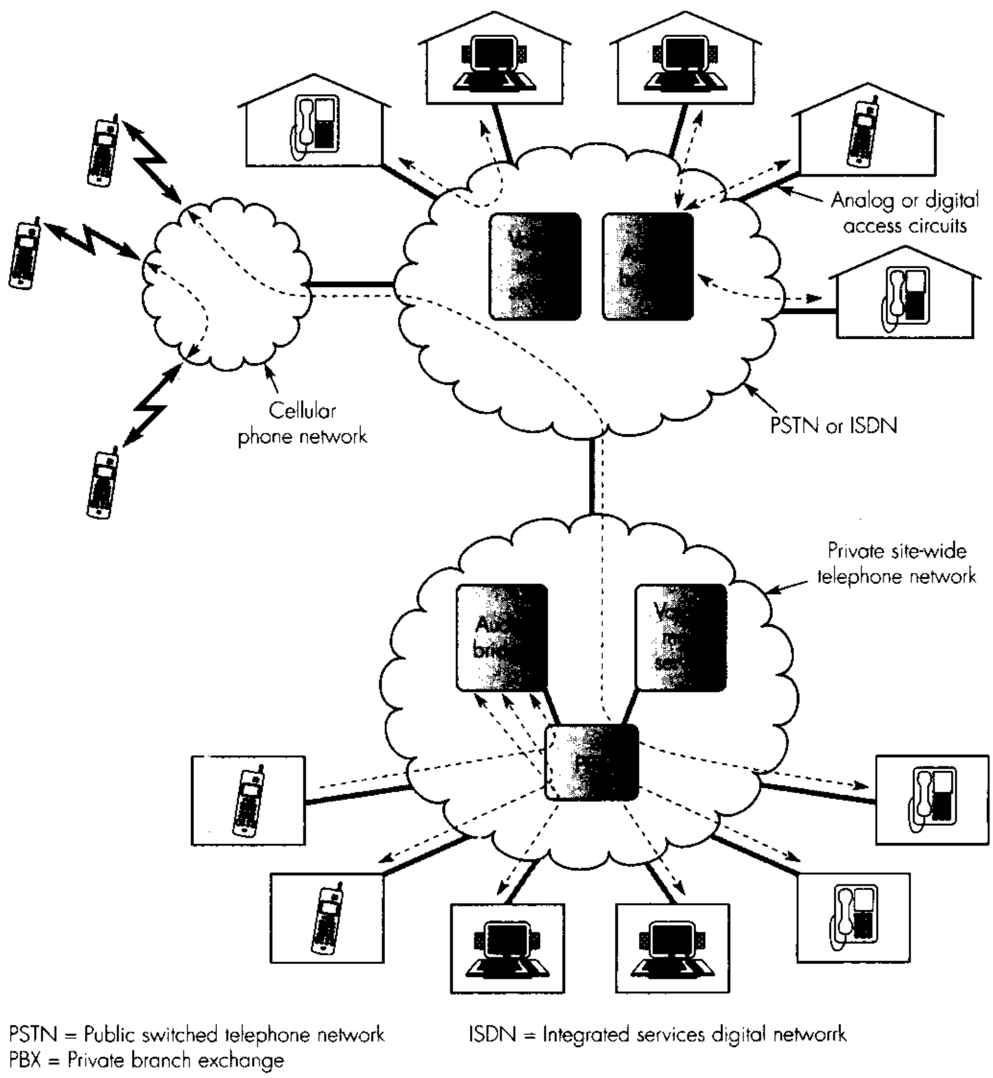


Figure 1.6 Speech-only interpersonal communications: public and private switched telephone networks.

requires a telephone interface card and associated software and is known as **computer telephony integration** or **CTI**. The added advantages of using a PC instead of a conventional telephone are many. For example, the user can create his or her own private directory of numbers and initiate a call simply by selecting the desired number on the PC screen. More generally, providing the access circuit to the network has sufficient capacity – normally referred to as the circuit's **bandwidth** – it is possible to integrate telephony with all the other networked services provided by the PC.

In addition to telephony, many public and private networks support additional services. Two examples are voice-mail and teleconferencing. **Voice-mail**, for example, is used in the event of the called party being unavailable. A spoken message can then be left in the **voice mailbox** of the called party. This is located in a central repository known as the **voice-mail server**. The message can be read by the owner of the mailbox the next time he or she contacts the server.

Teleconferencing calls involve multiple interconnected telephones/PCs. Each person can hear and talk to all of the others involved in the call. This type of call is known variously as a **conference call** or, since it involves a telephone network, a **teleconferencing call** or sometimes an **audioconferencing call**. It requires a central unit known as an **audio bridge** which provides the necessary support to set up a conference call automatically.

The Internet is also used to support telephony. Initially, because the Internet was designed to support computer-to-computer communications, just (multimedia) PC-to-PC telephony was supported. This was subsequently extended so that a standard telephone could also be used. See Figure 1.7.

In the case of a PC-to-PC telephone call, the standard addresses that are used to identify individual computers connected to the Internet are used in the same way as for a data transfer application. However, because the Internet operates in a packet mode, both PCs must have the necessary hardware and software to convert the speech signal from the microphone into packets on input and back again prior to output to the speakers. Telephony over the Internet is also known, therefore, as **packet voice** or, because the network protocol associated with the Internet is called the Internet protocol (IP), **voice over IP (VoIP)**.

When a PC connected to the Internet needs to make a call to a telephone that is connected to a PSTN/ISDN, because these both operate in a circuit mode, an interworking unit known as a **telephony gateway** must be used. The PC user first sends a request to make a (telephone) call to a preallocated telephony gateway using the latter's Internet address. Then, assuming the user is registered to use this service, the gateway requests from the source PC the telephone number of the called party. On receipt of this, the source gateway initiates a session (call) with the telephony gateway nearest to the called party using the Internet address of the gateway. The called gateway then initiates a call to the recipient telephone using its telephone number and the standard call setup procedure of the PSTN/ISDN. Assuming the called party answers,

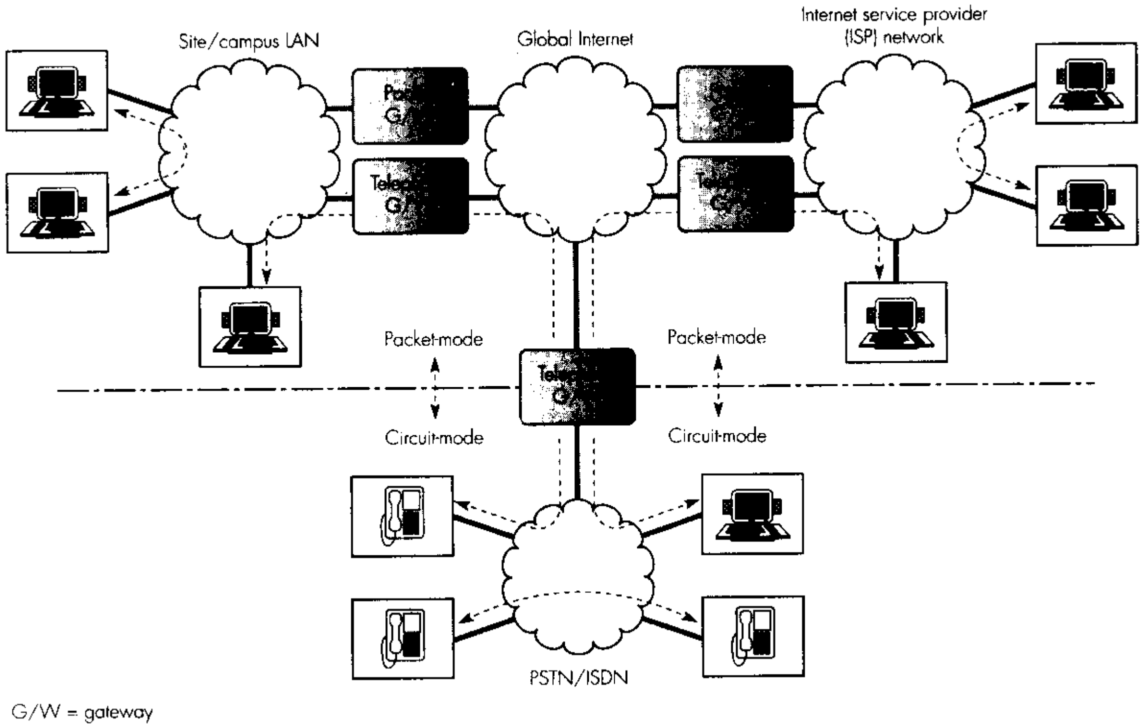


Figure 1.7 Telephony over the Internet.

the called gateway then signals back to the PC user – through the source gateway – that the call can commence. A similar procedure is followed to clear the call on completion.

Image only

An alternative form of interpersonal communications over a PSTN or an ISDN is by the exchange of electronic images of documents. This type of service is known as **facsimile** – or simply **fax** – and is illustrated in Figure 1.8. Normally, this type of communication involves the use of a pair of fax machines, one at each network termination point. To send a document, the caller keys in the (telephone) number of the intended recipient and a circuit is set up through the network in the same way as for a telephone call. The two fax machines communicate with each other to establish operational parameters after which the sending machine starts to scan and digitize each page of the document in turn. Both fax machines have an integral modem within them and, as each page is scanned, its digitized image is simultaneously transmitted over the network and, as this is received at the called side, a printed version of the document is produced. Finally, after the last page of the document has been sent and received, the connection through the network is cleared by the calling machine in the normal way.

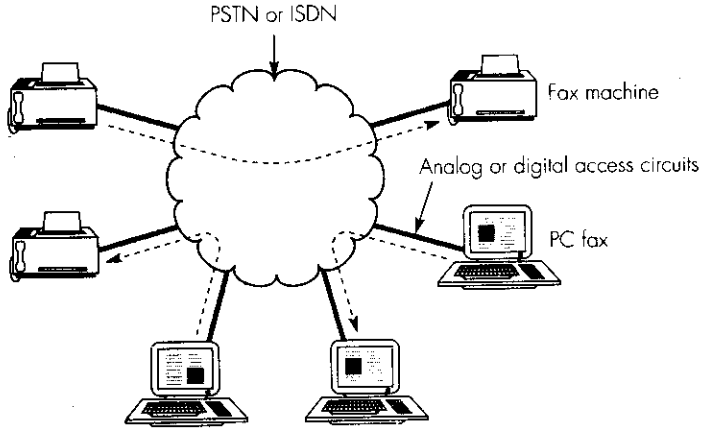


Figure 1.8 Image-only interpersonal communications: facsimile (fax) examples.

It is also possible to use a PC instead of a normal fax machine to send an electronic version of a document that is stored directly within the PC's memory. This mode of operation is known as **PC fax**. Essentially, the digital image of each page of the document is sent in the same way as the scanned image produced by a conventional fax machine. As with telephony, this requires a telephone interface card and associated software. The latter operates in exactly the same way as that in a fax machine and hence the terminal at the called side can be either a fax machine or another similar PC. In addition, with PC fax it is possible to send the digitized document over other network types such as an enterprise network. In this case, a LAN interface card and associated software are used. This mode of operation is particularly useful when working with paper-based documents such as invoices, and so on.

Text only

An example of interpersonal communications involving just text is electronic mail (email). The user terminal is normally a PC or a workstation and, as we described earlier in Section 1.3.2, the most widespread network used is the Internet. Various operational scenarios are shown in Figure 1.9(a).

In the case of a user at home, access to the Internet is through a PSTN/ISDN and an intermediate Internet service provider (ISP) network. Alternatively, business users obtain access either through an enterprise network or a site/campus network. Associated with each network is a set of one or more server computers. Each is known as an **email server** and, collectively, these contain a **mailbox** for each user connected to that network. A user can both create and deposit mail into his or her mailbox and read mail from it. Both the email servers and the internetwork gateway operate using the standard Internet communication protocols.

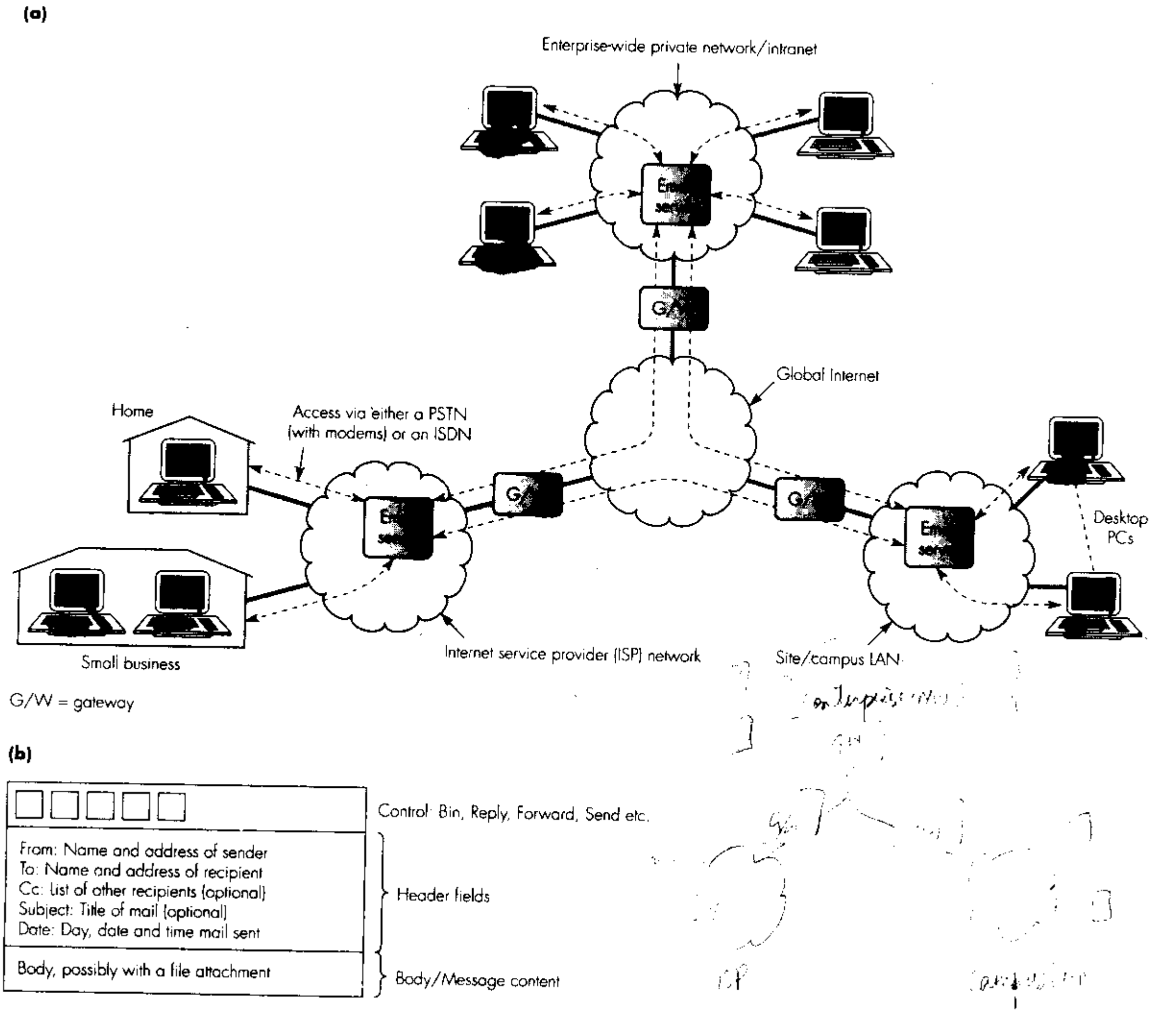


Figure 1.9 Text-only electronic mail: (a) email transfer examples; (b) example email message format.

The format of a typical text-only email message is shown in Figure 1.9(b) and, as we can see, at the head is the unique Internet-wide name of both the sender and recipient of the mail. In addition, a copy of the mail can be sent to multiple recipients each of whom is listed in the cc part of the mail header, the acronym "cc" being the abbreviation for "carbon copy" which was the original means of making (paper) copies of documents. Normally, the contents of text-only mail comprise unformatted text, typically strings of ASCII characters.

Text and images

An example of an application that involves both text and images integrated together is **computer-supported cooperative working** (**CSCW**). The network used is an enterprise network, a LAN, or the Internet and the general scheme is illustrated in Figure 1.10. Typically, a distributed group of people – each in his or her place of work – are all working on the same project. The user terminal is either a PC or a workstation and a window on each person's display is used as a shared workspace. This is known as a **shared whiteboard** and, normally, the display comprises text and images integrated together. The software associated with CSCW comprises a central program – known as the **whiteboard program** – and a linked set of support programs, one in each PC/workstation. The latter is made up of two parts: a change-notification part and an update-control part. Whenever a member of the group updates the contents of his or her whiteboard, the **change-notification** part sends details of the changes to the whiteboard program. This relays the changes to the **update-control** in each of the other PCs/workstations and these in turn proceed to update the contents of their copy of the whiteboard.

Speech and video

An example application that uses speech and video integrated together is **video telephony** which is now supported by all the network types. Figure 1.11(a)shows the general scheme.

In the case of the home, the terminals used are normally dedicated to providing the videophone service, while in an office, a single multimedia

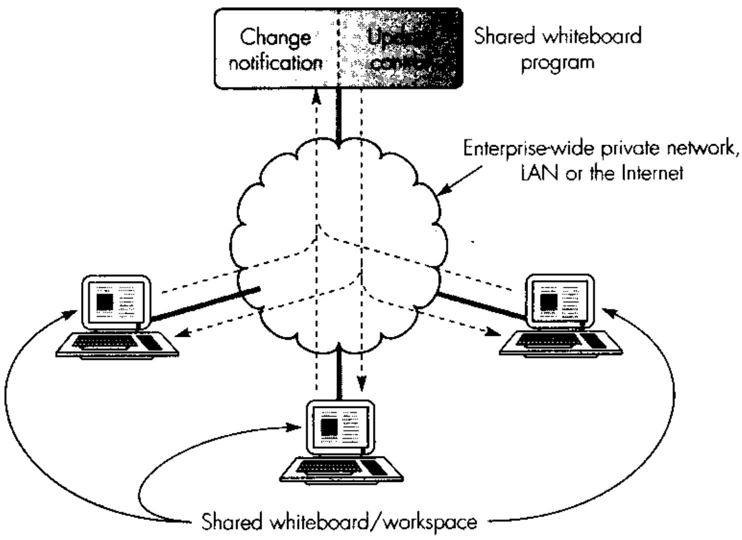


Figure 1.10 Text-and-image computer-supported cooperative working (CSCW).

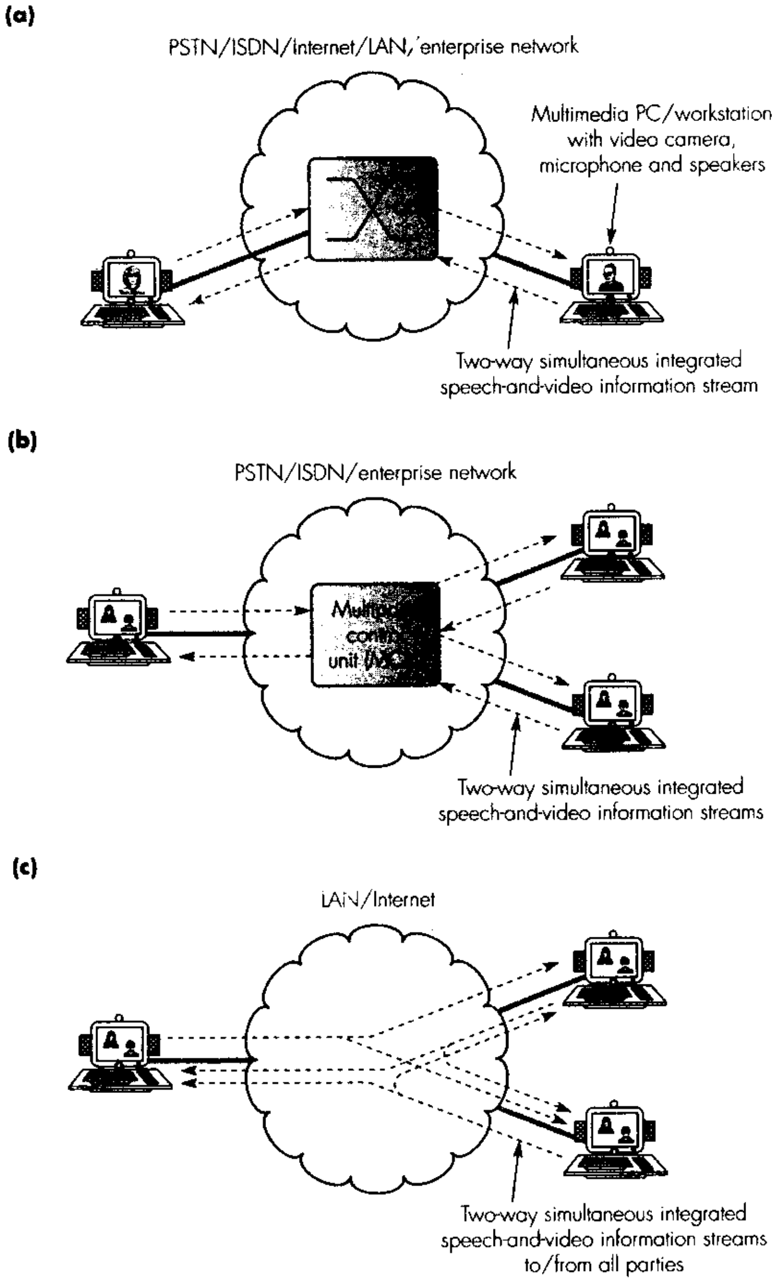


Figure 1.11 Speech-and-video interpersonal communications: (a) two-party video telephone call; (b) videoconferencing using an MCU; (c) videoconferencing using a broadcast network.

PC/workstation is used to provide the videophone service together with a range of other services. In both cases, the terminals/PCs incorporate a video camera in addition to the microphone and speaker used for telephony. With a dedicated terminal, a separate screen is used for the display whilst with a multimedia PC or workstation, the (moving) image of the called party is displayed in a window of the PC/workstation screen. The network must provide a two-way communication channel between the two parties of sufficient bandwidth to support the integrated speech-and-video generated by each terminal/PC.

The integration of video with speech means that the bandwidth of the access circuits required to support this type of service is higher than that required for speech only. Moreover, as with telephony, a call may involve not just two persons – and hence terminals/PCs – but several people each located in their own office. This type of call is then known as a **desktop videoconferencing call** and is now widely used in large corporations involving multiple geographically distributed sites in order to minimize travel between the various locations. As we indicated earlier, large corporations of this type have an enterprise-wide network to link the sites together and, in order to support videoconferencing, there is a central unit called a **multipoint control unit (MCU)** – or sometimes a **videoconferencing server** – associated with this network. An example is shown in Figure 1.11(b).

In principle, a separate window on the screen of each participant's PC/workstation should be used to display the video image of all the other participants. In practice, however, this would require multiple integrated speech-and-video communication channels, one for each participant, being sent to each of the other participants. Normally, this would require more bandwidth than is available. Hence instead, the integrated speech-and-video information stream from each participant is sent to the MCU which then selects just a single information stream to send to each participant. For example, with a voice-activated MCU, whenever the MCU detects a participant speaking, it relays the information stream from that participant to all the other participants. In this way, only a single two-way communication channel between each location and the MCU is required thereby reducing considerably the communication bandwidth needed.

Alternatively, some networks such as LANs and the Internet support what is called **multicasting**. This means that all transmissions from any of the PCs/workstations belonging to a predefined **multicast group** are received by all the other members of the group. Thus with networks that support multicasting, it is possible to hold a conferencing session without an MCU. The principle is shown in Figure 1.11(c) and, as we can deduce from this, this is only feasible when only a limited number of participants are involved owing to the high load it places on the network.

While the application just described involves only a single person at each location, there are other applications that involve groups of people at one or more of the locations. Two examples are shown in Figure 1.12. In part (a) a person at one location is communicating with a group of people at another location. This is the case, for example, with the transmission of a live lecture or seminar. Typically, the information stream transferred from the lecturer to the (remote) class would be integrated speech-and-video together with electronic copies of transparencies and other documents used in the lecture. In the reverse direction, the information may comprise just speech – for questions – or integrated speech-and-video to enable the lecturer to both see and hear the members of the class at the remote location. In terms of communications requirements, these are similar to those for a two-party videophone call. Alternatively, if the lecture is being relayed to multiple locations, either a separate communications channel is required to each remote site or an MCU

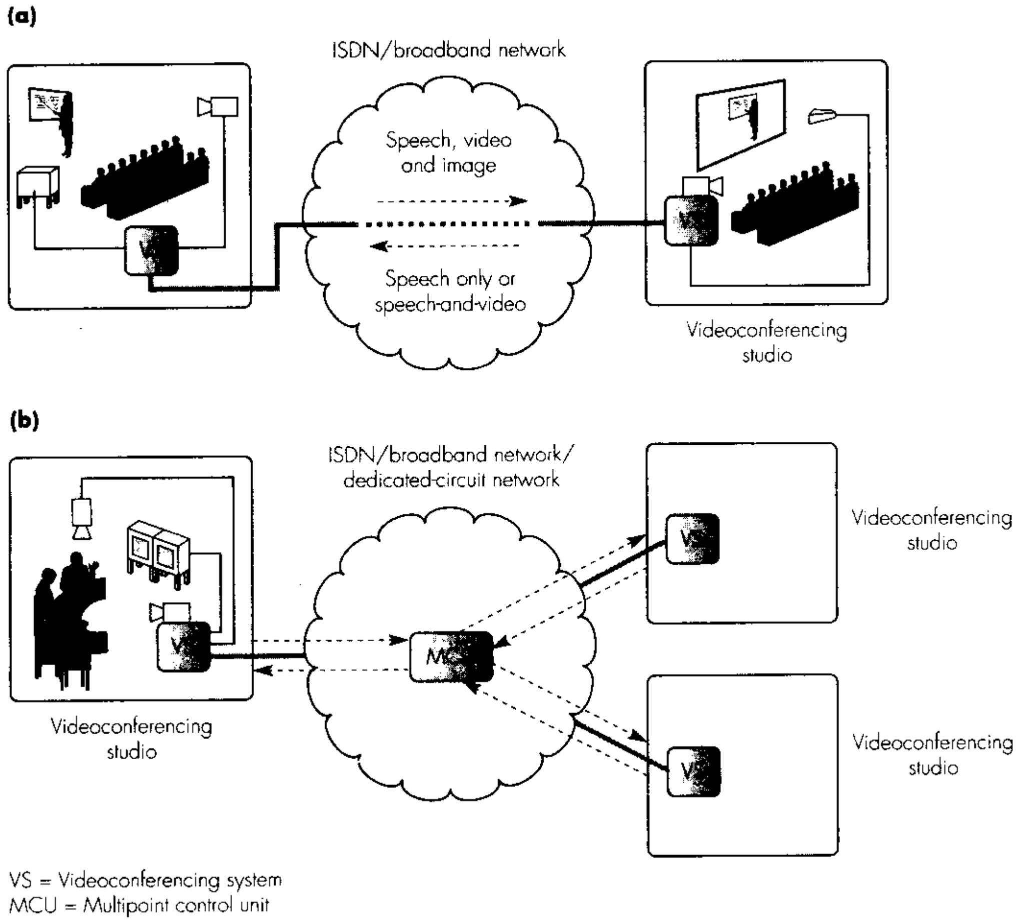


Figure 1.12 Speech-and-video interpersonal communications: (a) remote lecture; (b) multiparty (group) videoconferencing.

is used at the lecturer's site. As we can see, because of the relatively high bandwidth that is involved, the network is either an ISDN that supports multiple 64 kbps channels or a broadband multiservice network if one is available.

In the example in Figure 1.12(b), there is a group of people at each location. This type of application has been in use for many years and was the first example of videoconferencing. Normally, since a group of people are present at each location, specially equipped rooms called **videoconferencing studios** are used which contain all the necessary audio and video equipment. This comprises one or more video cameras, a large-screen display, and associated audio equipment, all of which is connected to a unit called a **videoconferencing system**. A conference can involve just two locations or, more usually, multiple locations as shown in the figure. In the case of the latter, an MCU is normally used to minimize the bandwidth demands on the access circuits to the network. In the figure, the MCU is shown as a central facility within the network and hence only a single two-way communications channel is required for each access circuit of the network. This is the type of arrangement with a telecommunications-provider conference, for example. Alternatively, if a private network is being used, the MCU is normally located at one of the sites. The communication requirements at that site are then more demanding since it must support multiple input channels – one from each of the other sites – and a single output channel, the stream from which must be broadcast to all of the other sites.

Multimedia

The example discussed earlier concerning Internet-based electronic mail – email – assumed the information content of each email message consisted of text only. In addition, however, mail containing other media types such as images, audio, and video are also used. Three examples of electronic mail consisting of media types other than text are voice-mail, video-mail, and multimedia mail.

Voice-mail is similar in principle to that described earlier in relation to telephone networks. With Internet-based voice-mail, however, there is a voice-mail server associated with each network. This is in addition to the email server shown earlier in Figure 1.9(a). The user first enters a voice message addressed to the intended recipient and the local voice-mail server then relays this to the server associated with the intended recipient's network. The stored voice message is then played out the next time the recipient accesses his or her voice-mailbox. The same mode of operation is used for video-mail except in this case the mail message comprises an integrated speech-and-video sequence.

Multimedia mail is an extension of text-only mail inasmuch as the basic content of the mail comprises textual information. With multimedia mail, however, the textual information is annotated with a digitized image, a speech message, or a video message, as shown in Figure 1.13. In the case of speech-and-video, the annotations can be sent either directly to the mailbox

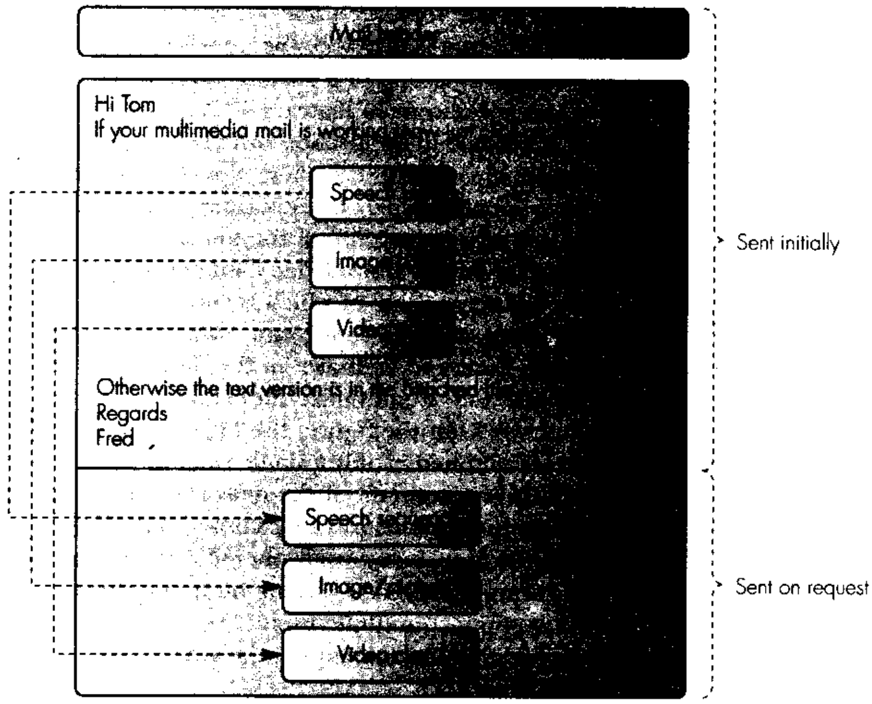


Figure 1.13 Multimedia electronic mail structure.

of the intended recipient together with the original textual message – and hence stored and played out in the normal way – or they may have to be requested specifically by the recipient when the textual message is being read. In this way, the recipient can always receive the basic text-only message but the multimedia annotations can be received only if the terminal being used by the recipient supports voice and/or video.

1.4.2 Interactive applications over the Internet

In addition to a range of interpersonal communication applications, the Internet is also used to support a range of interactive applications, the most widely used being for interactions with a **World Wide Web (WWW)** or simply **Web**, server. This comprises a linked set of multimedia information servers that are geographically distributed around the Internet. The total information stored on all the servers is equivalent to a vast library of documents. The general principle is illustrated in Figure 1.14(a).

Each document comprises a linked set of **pages** and the linkages between the pages are known as **hyperlinks**. These are pointers – also known as **references** – either to other pages of the same document or to any other document within the total Web. In this way, a reader of a document has the

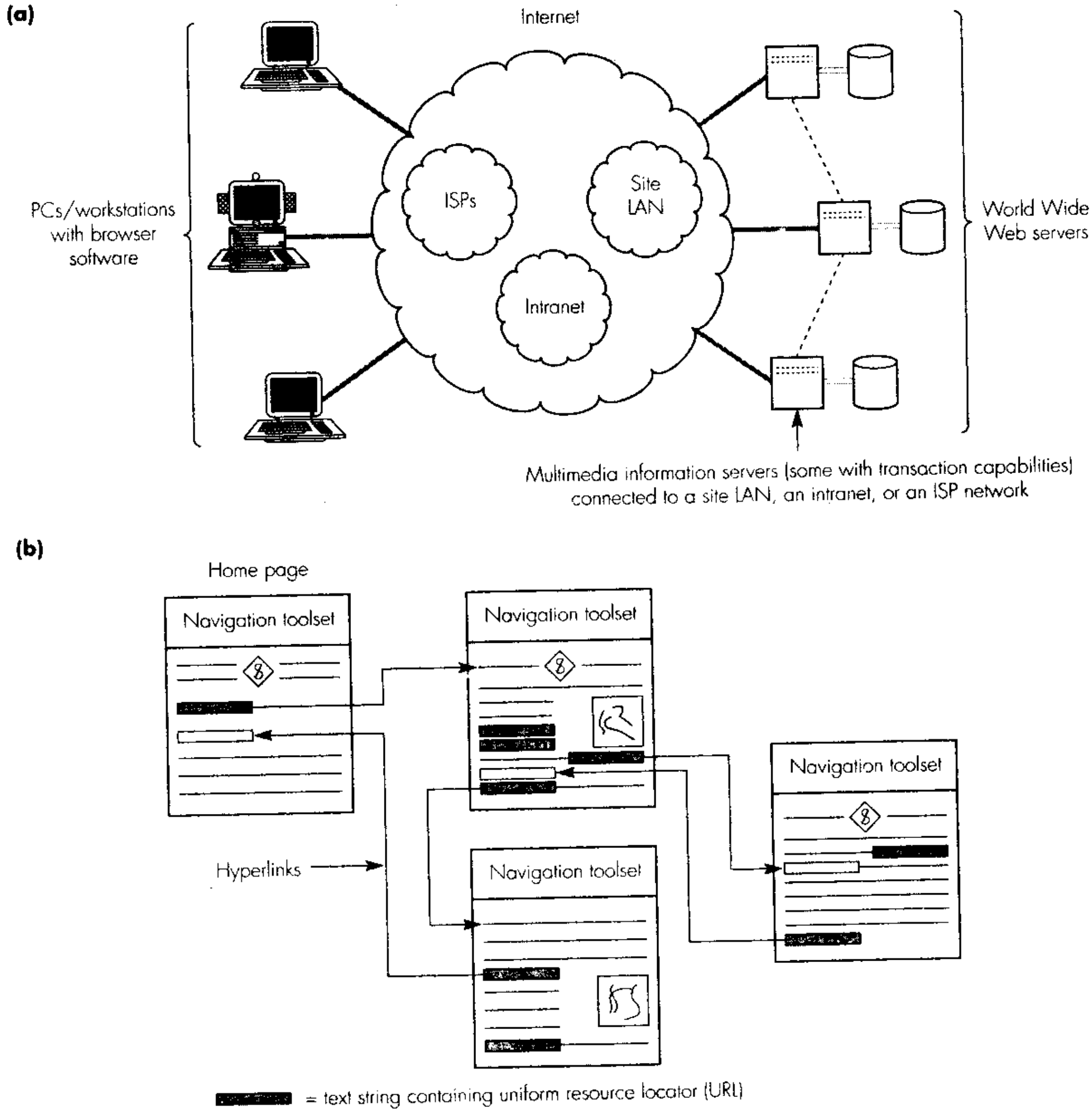


Figure 1.14 Interactions with a World Wide Web server: (a) schematic; (b) hypertext linkages between the pages of a set of documents.

option, at well-defined points throughout the pages that make up a document, to jump either to a different page of the same document or to a different document. Also, to return subsequently to a specific point on a page at a later time. The optional linkage points within documents are defined by the creator of the document and are known as **anchors** since it is to these that the necessary linkage information is attached. Documents comprising only text are created using what is called **hypertext,** while those comprising multimedia information are created using what is known as **hypermedia.** The general structure of this type of document is shown in Figure 1.14(b).

There is no central authority for the introduction of new documents into the Web. Anyone can create a new document at a particular server site – providing the server has been allocated an Internet address – and make hyperlink references from it to any other document on the Web. Each document has a unique address – known as a **uniform resource locator** or **URL** – which identifies both the location of the server on the Internet where the first page of the document is stored and also the file reference on that server. The first page of a document is known as the **home page** and all the hyperlinks on this and the other pages have similar URLs associated with them. As we can deduce from this, the physical location of a page is transparent to the user and, in theory, can be located anywhere on the Web.

A standard format is used for writing documents. This is known as the **Hypertext Markup Language (HTML)** and it is also used for writing client software to explore the total contents of the Web, that is, the contents of the linked information on all the Web servers. The client function is called a **browser** and there are a number of user-friendly browsers available to explore the contents of the Web. These allow a user to create a directory of previously visited servers and to open up a dialog with a particular server at the click of the mouse. Once a desired document has been located, the user simply clicks on an anchor point within a page of the document to activate the linkage information stored at that point. It is also possible to return to the previous anchor at any time. With a hypertext document, the anchor is usually an underlined word or phrase while with a hypermedia document it is normally an icon of an appropriate shape; for example, a loudspeaker for a sound annotation or a video camera for a video clip. It is of course the presence of sound and video annotations that brings a document to life and adds value over a simple printed page.

In some applications the client simply wishes to browse through the information stored at a particular site. Examples include browsing through sales literature, product information, application notes periodicals, newspapers, and so on. In general, there is no charge for accessing this information. However, access to books, journals, and similar documents may be by subscription only.

In applications such as **homeshopping, homebanking,** and so on – more generally known as **teleshopping** and **telebanking** – a client may wish not only to browse through the information at a site but also to initiate an additional **transaction.** Here the server must provide additional transaction processing support for, say, ordering and purchasing. Since this will often involve a financial transaction, more rigorous security procedures are required for access and authentication purposes.

1.4.3 Entertainment applications

Entertainment applications can be one of two types:

- movie/video-on-demand,
- interactive television.

We shall discuss each application separately.

Movie/video-on-demand

This category of application is similar in principle to that described in the previous section except that, in general, the video and audio associated with entertainment applications must be of a much higher quality/resolution since wide-screen televisions and stereophonic sound are often used. As we shall describe in Section 4.3.3, a digitized movie/video – with sound – requires a minimum channel bit rate (bandwidth) of 1.5 Mbps. Hence the network used to support this type of application must be either a PSTN with a high bit rate modem – as we showed earlier in Figure 1.1(c) – or a cable network of the type we showed in Figure 1.3(a). As we saw earlier in Section 1.3.1, in the case of a PSTN, the high bit rate channel provided by the modem is used only over the access circuit and provides additional services to the other switched services that the PSTN supports. The general operational scheme in both cases is shown in Figure 1.15(a).

As we can see, the information stored on the server is a collection of digitized movies/videos. Normally the subscriber terminal comprises a conventional television witl ection device for interaction purposes. The user interactions are relayed to the server through a set-top box 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 type of application is known as **movie-on-demand (MOD)** or sometimes **video-on-demand (VOD)**. In addition to selecting a movie, the subscriber can control the showing of the movie by using similar controls to those used on a conventional video cassette recorder (VCR), that is, pause, fast-forward, and so on.

A key feature of MOD is that a subscriber can initiate the showing of a movie selected from a large library of movies at any time of the day or night. Hence, as we can deduce from Figure 1.15(b), this means that the server must be capable of playing out simultaneously a large number of video streams equal to the number of subscribers currently watching a movie. This requires the information flow from the server to be extremely high since it must support not just the transmission of a possibly large number of different movies, but also multiple copies of each movie. Technically this is very challenging and costly since, in general, the cost of the server is directly related to the aggregate information flow rate from it.

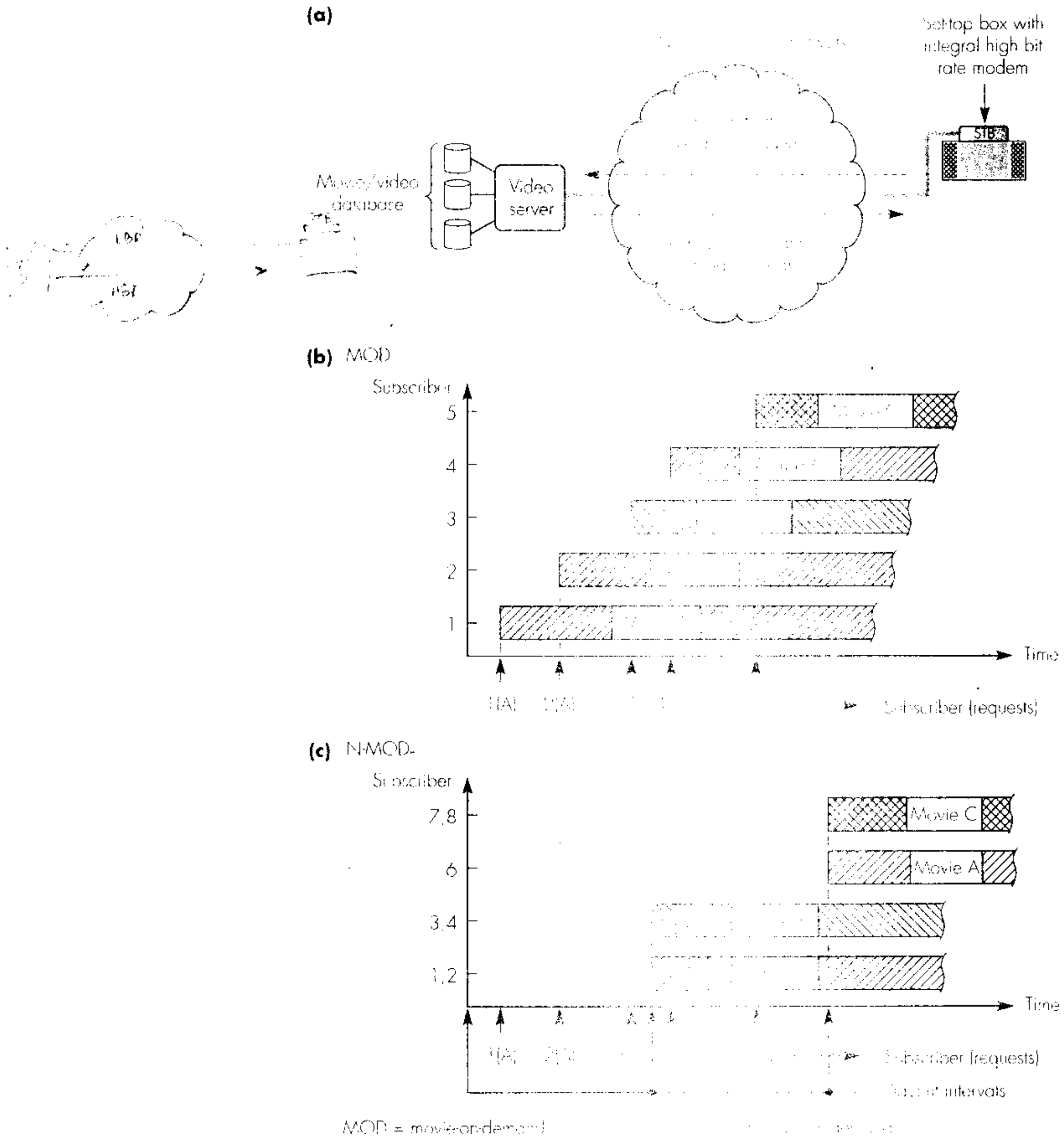


Figure 1.15 Interactions with a video server: (a) networking schematic; (b) movie-on-demand; [illegible] on-demand.

In practice, if the server is supporting a large number of subscribers, then it is common for several subscribers to request the same movie within a relatively short time interval between each request. An alternative mode of operation is also used, therefore, in which requests for a particular movie are not played out immediately but instead are queued until the start of the next playout time of that movie as shown in Figure 1.15(c). In this way, all requests for the same movie which are made during the period up to the next playout time are satisfied simultaneously by the server outputting a single video stream. This mode of operation, is known as **near movie-on-demand** or **N-MOD**. Clearly, however, the viewer is unable to control the playout of the movie.

Similar applications are also used within a business environment, except that the stored information in the server is typically training and general educational material, company news, and so on, and thus the number of stored videos is normally much less as is the number of simultaneous users. This means that the video servers required are less sophisticated than those used in public MOD/N-MOD systems. Also, the stored video streams/programs are often in a different format. The format used is the same as that used with **CD-ROMs** since the received video stream can then be displayed directly on the screen of a multimedia PC or workstation. The communication requirements of the private network, however, are the same as those identified for use with a public network.

Interactive television

Broadcast television networks include cable, satellite, and terrestrial networks. The basic service provided by these networks is, of course, the diffusion of both analog and digital television (and radio) programs. In addition, however, as we saw earlier in Section 1.3.3, the set-top box (STB) associated with these networks also has a modem within it. In the case of a cable network, as we show in Figure 1.16(a), the STB provides both a low bit rate connection to the PSTN and a high bit rate connection to the Internet. Hence by connecting appropriate terminal equipment to the STB – a keyboard, telephone, and so on – the subscriber is able to gain access to all the services provided through the PSTN and the Internet. In addition, through the connection to the PSTN, the subscriber is able to actively respond to the information being broadcast. This is the origin of the term "interactive television" and typical uses of the return channel are for voting, participation in games, home shopping and so on. As we see in Figure 1.16(b), a similar set of services are available through satellite and terrestrial broadcast networks, except that the STB associated with these networks requires a high-speed modem to provide the connections to the PSTN and the Internet.

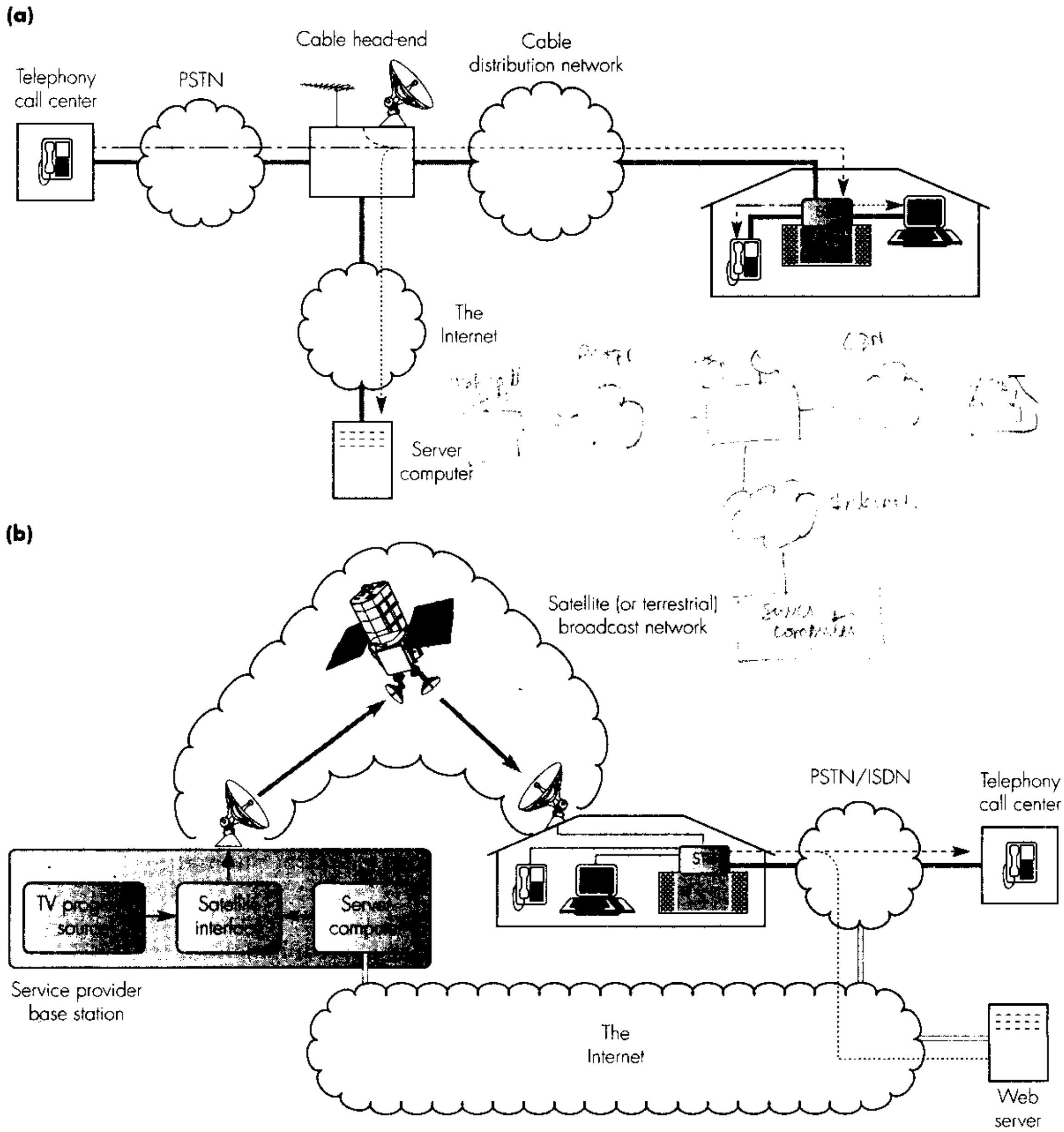


Figure 1.16 Interactive television: (a) cable distribution network; (b) satellite/terrestrial broadcast network.

1.5 Application and networking terminology

Before we leave this chapter it will be helpful if we first review some of the terminology used in relation to the different media types and also the terminology and operational characteristics of the different types of communication channels provided by the various networks we have identified. A selection of the terms that are used are shown in Figure 1.17. We shall describe each term separately as well as its origin and interrelationship with the other terms.

1.5.1 Media types

As we identified in Section 1.4 when we described the different multimedia applications, the information flow associated with the different applications can be either continuous or block-mode. In the case of **continuous media**, this means that the information stream is generated by the source continuously in a time-dependent way. In general, therefore, continuous media is

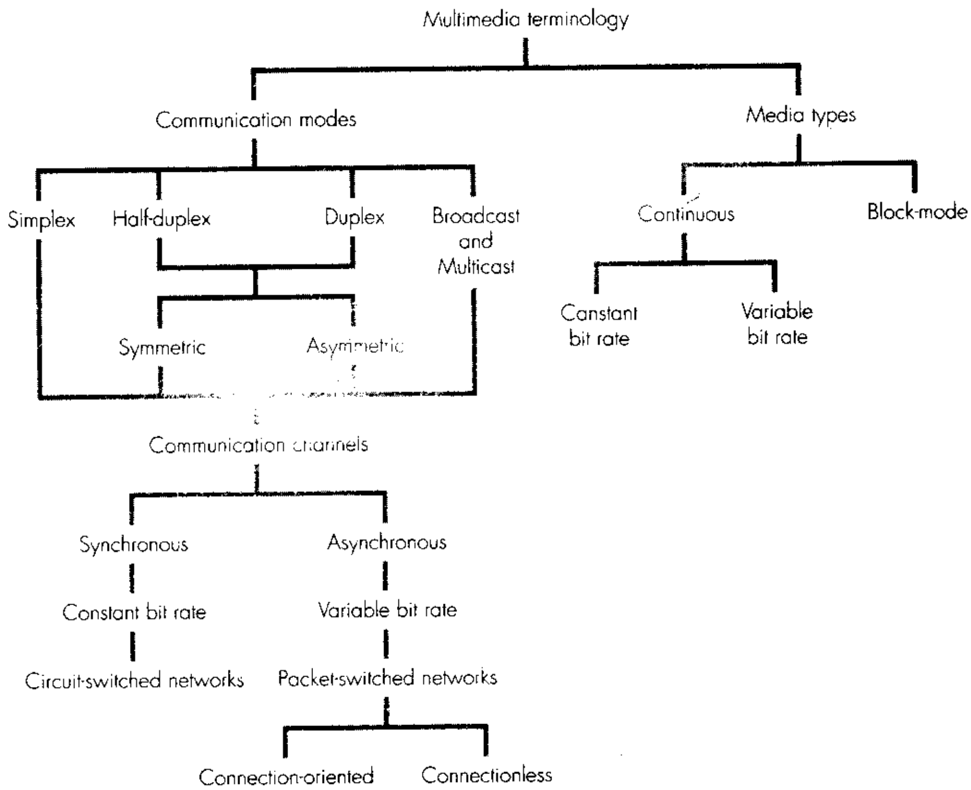


Figure 1.17 A selection of the terms used with multimedia.

passed directly to the destination as it is generated and, at the destination, the information stream is played out directly as it is received. This mode of operation is called **streaming** and, since continuous media is generated in a time-dependent way, it is also known as **real-time media**. With continuous media, therefore, the bit rate of the communications channel that is used must be compatible with the rate the source media is being generated. Two examples of media types that generate continuous streams of information in real time are audio and video.

In terms of the bit rate at which the source information stream is generated, this may be at either a **constant bit rate (CBR)** or a **variable bit rate (VBR)**. With audio, for example, the digitized audio stream is generated at a constant bit rate which is determined by the frequency the audio waveform is sampled and the number of bits that are used to digitize each sample. In the case of video, however, as we shall expand upon in Section 2.6.1, although the individual pictures/frames that make up the video are generated at a constant rate, after compression, the amount of information associated with each frame varies. In general, therefore, the information stream associated with compressed video is generated at fixed time intervals but the resulting bit rate is variable.

In the case of **block-mode media**, the source information comprises a single block of information that is created in a time-independent way. For example, a block of text representing an email or computer program, a two-dimensional matrix of pixel values that represents an image, and so on. Normally, therefore, block-mode media is created in a time-independent way and is often stored at the source in, say, a file. Then, when it is requested, the block of information is transferred across the network to the destination where it is again stored and subsequently output/displayed at a time determined by the requesting application program. This mode of operation is known as **downloading** and, as we can deduce with block-mode media, the bit rate of the communications channel need not be constant but 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. This is known as the **round-trip delay (RTD)** and, for human–computer interactions, ideally, this should be no more than a few seconds.

1.5.2 Communication modes

In terms of the communication channels that are provided by the various network types, as we show in Figure 1.18, the transfer of the information streams associated with an application can take place in one of five modes:

- **simplex:** this means the information associated with the application flows in one direction only. An example is the transmission of photographic images from a deep-space probe at predetermined times since this involves just a unidirectional flow of information from the probe to an earth station;

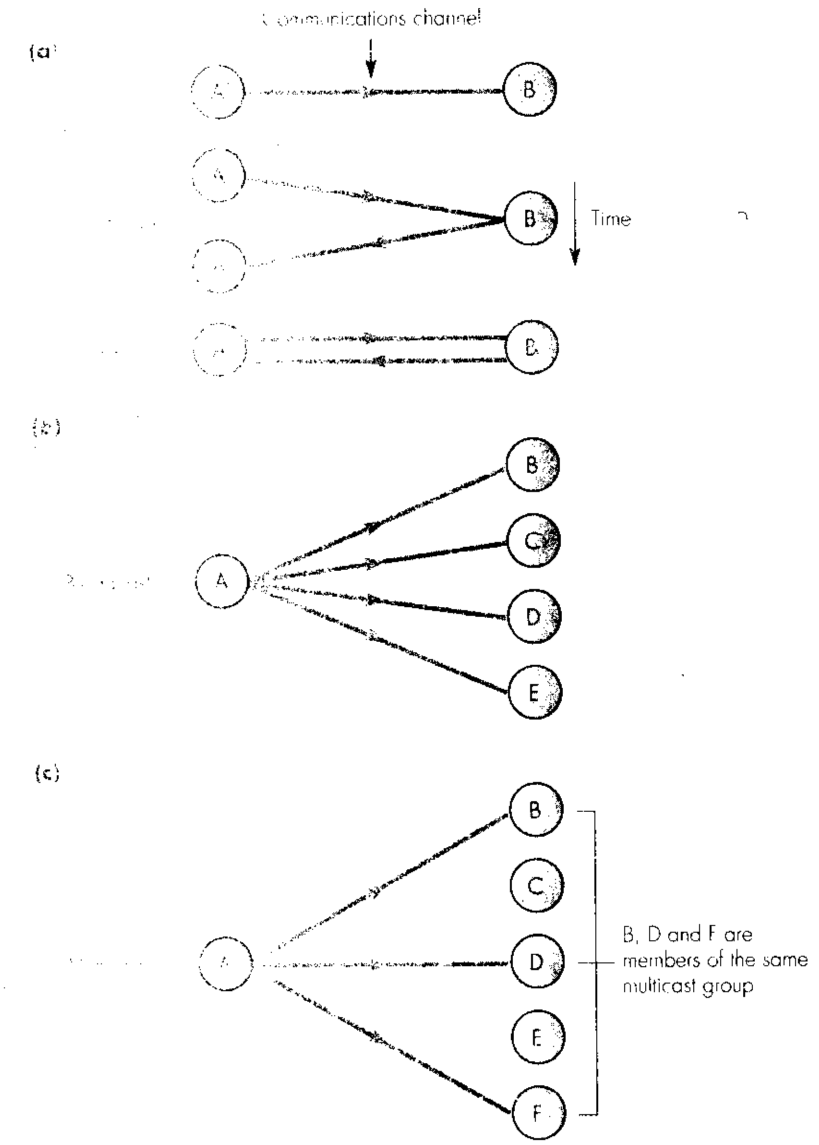


Figure 1.18 Communication modes: (a) unicast; (b) broadcast; (c) multicast.

- half-duplex: this means that information flows in both directions but alternately. This mode is also known as **two-way alternate** and an example is a user making a request for some information from a remote server and the latter returning the requested information;

- **duplex:** this means that information flows in both directions simultaneously. It is also known as **two-way simultaneous** and an example is the two-way flow of digitized speech and video associated with a video telephony application;
- **broadcast:** this means that the information output by a single source node is received by all the other nodes – computers, and so on – that are connected to the same network. An example is the broadcast of a television program over a cable network as all the television receivers that are connected to the network receive the same set of programs;
- **Multicast:** this is similar to a broadcast except that the information output by the source is received by only a specific subset of the nodes that are connected to the network. The latter form what is called a **multicast group** and an example application is videoconferencing which involves a predefined group of terminals/computers connected to a network exchanging integrated speech and video streams.

In the case of half-duplex and duplex communications, the bit rate associated with the flow of information in each direction can be either equal or different; if the flows are equal, the information flow is said to be **symmetric** and if the flows are different, **asymmetric**. For example, a video telephone call involves the exchange of an integrated digitized speech and video stream in both directions simultaneously and hence a symmetric duplex communications channel is required. Alternatively, in an application involving a browser (program) and a Web server, a low bit rate channel from the browser to the Web server is required for request and control purposes and a higher bit rate channel from the server to the subscriber for the transfer of, say, the requested file. Hence for this type of application, an asymmetric half-duplex communications channel is sufficient.

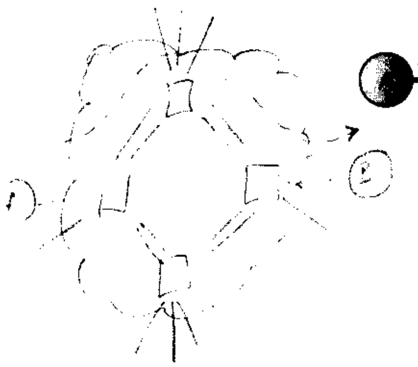
1.5.3 Network types

In the same way that there are two types of information stream associated with the different media types – continuous and block-mode – so there are two types of communications channel associated with the various network types, one that operates in a time-dependent way known as **circuit-mode** and the other in a time-varying way known as **packet-mode**. The first is known as a **synchronous communications channel** since it provides a constant bit rate service at a specified rate. The second is known as an **asynchronous communications channel** since it provides a variable bit rate service, the actual rate being determined by the (variable) transfer rate of packets across the network.

Circuit-mode

A circuit-mode network is shown in Figure 1.19 and, as we can see, it comprises an interconnected set of **switching offices/exchanges** to which the subscriber terminals/computers are connected. This type of network is known as a **circuit-switched network** and, prior to sending any information, the source must first set up a connection through the network. Each subscriber terminal/computer has a unique network-wide number/address associated with it and, to make a call, the source first enters the number/address of the intended communication partner. The local switching office/exchange then uses this to set up a connection through the network to the switching office/exchange to which the destination is connected and, assuming the destination is free and ready to receive a call, a message is returned to the source indicating that it can now start to transfer/exchange information. Finally, after all the information has been transferred/exchanged, either the source or the destination requests for the connection to be cleared. The bit rate associated with the connection is fixed and, in general, is determined by the bit rate that is used over the access circuits that connect the source and destination terminal/computer to the network.

The messages associated with the setting up and clearing of a connection are known as **signaling messages**. As we can deduce from the above, with a circuit-switched network there is a time delay while a connection is being established. This is known as the **call/connection setup delay** and two examples of networks that operate in this way are a PSTN and an ISDN. With



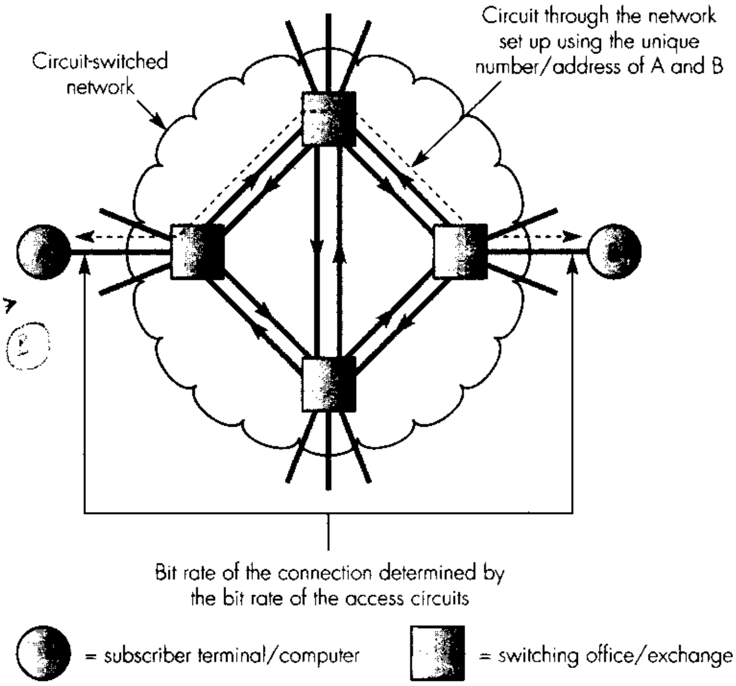


Figure 1.19 Circuit-switched network schematic.

a PSTN, the call setup delay can range from a fraction of a second for a local call through to several seconds for an international call. With an ISDN, however, the delay ranges from tens of milliseconds through to several hundred milliseconds.

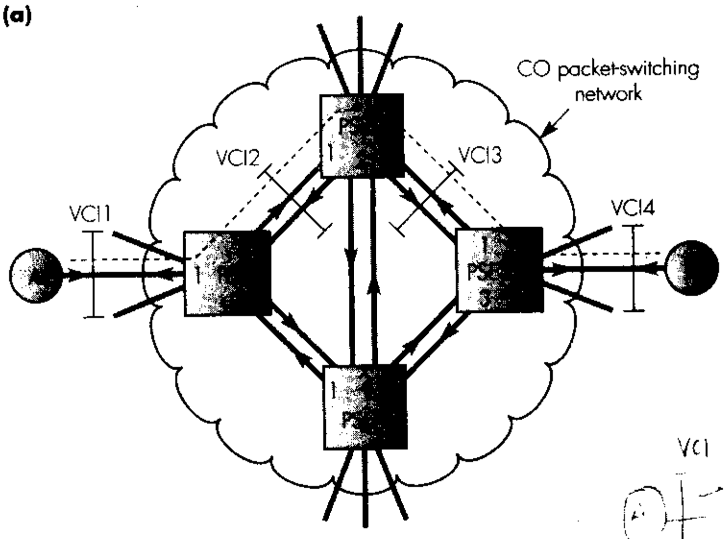
Packet-mode

As we see in Figure 1.20, there are two types of packet-mode network: **connection-oriented (CO)** and **connectionless (CL)**. The principle of operation of a connection-oriented network is shown in Figure 1.20(a) and, as we can see, it comprises an interconnected set of **packet-switching exchanges (PSEs)**. This type of network is known as a **packet-switched network** and, as with a circuit-switched network, each terminal/computer that is connected to the network has a unique network-wide number/address associated with it. With a connection-oriented network, as the name implies, prior to sending any information, a connection is first set up through the network using the addresses of the source and destination terminals. However, in a packet-switched network, the connection/circuit that is set up utilizes only a variable portion of the bandwidth of each link and hence the connection is known as a **virtual connection** or, more usually, a **virtual circuit (VC)**.

To set up a VC, the source terminal/computer sends a *call request* control packet to its local PSE which contains, in addition to the address of the source and destination terminal/computer, a short identifier known as a **virtual circuit identifier (VCI)** Each PSE maintains a table that specifies the outgoing link that should be used to reach each network address and, on receipt of the *call request* packet, the PSE uses the destination address within the packet to determine the outgoing link to be used. The next free identifier (VCI) for this link is then selected and two entries are made in a **routing table**. The first specifies the incoming link/VCI and the corresponding outgoing link/VCI and the second, in order to route packets in the reverse direction, the inverse of these, as we show in the example in the figure. The *call request* packet is then forwarded on the selected outgoing link and the same procedure is followed at each PSE along the route until the destination terminal/computer is reached.

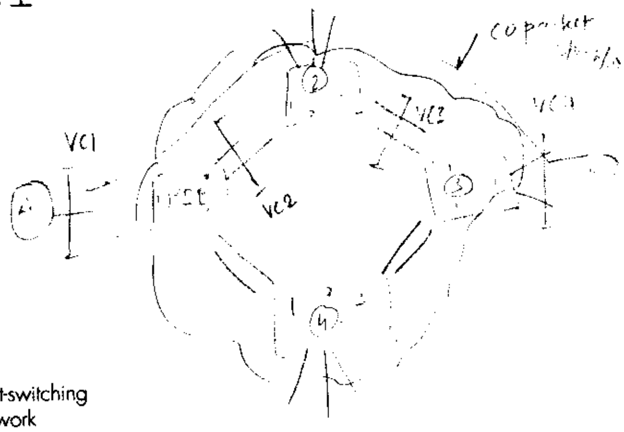
Collectively, the VCIs that are used on the various links form the virtual circuit and, at the destination, assuming the call is accepted, a *call accepted* packet is returned to the source over the same route/virtual circuit. The information transfer phase can then start but, since a VC is now in place, only the VCI is needed in the packet header instead of the full network-wide address. Each PSE first uses the incoming link/VCI to determine the outgoing link/VCI from the routing table. The existing VCI in the packet header is then replaced with that obtained from the routing table and the packet is forwarded on the identified outgoing link. The same procedure is followed to return information in the reverse direction and, when all information has been transferred/exchanged, the VC is cleared and the appropriate VCIs are released by passing a *call clear* packet along the VC.

(a)



PSE1 routing table:

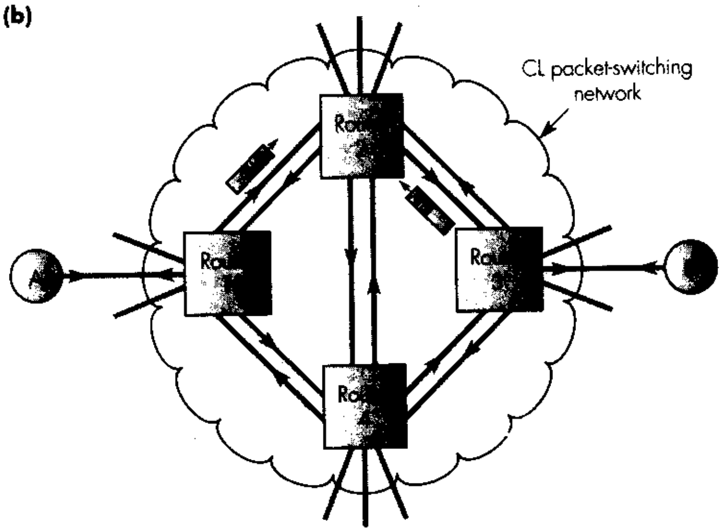
	IN		OUT
PSE1 routing table:	VCI1/Link1	→	VCI2/Link2
	VCI2/Link2	→	VCI1/Link1
PSE2 routing table:	VCI2/Link1	→	VCI3/Link3
	VCI3/Link3	→	VCI2/Link1
PSE3 routing table:	VCI3/Link1	→	VCI4/Link2
	VCI4/Link2	→	VCI3/Link1



CO = connection-oriented
--- = virtual circuit

VCI = virtual circuit identifier
PSE = packet-switching exchange

(b)



CL = connectionless
A, B = full network-wide addresses

= packet
information content
source address
destination address

Figure 1.20 Packet-switching network principles: (a) connection-oriented; (b) connectionless.

In contrast, with a connectionless network, the establishment of a connection is not required and the two communicating terminals/computers can communicate and exchange information as and when they wish. In order to do this, however, as we show in Figure 1.20(b), each packet must carry the full source and destination addresses in its header in order for each PSE to route the packet onto the appropriate outgoing link. In a connectionless network, therefore, 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 its entirety in a memory buffer. A check is then made to determine if any transmission/bit errors are present in the packet header – that is, the signal that is used to represent a binary 0 is corrupted and is interpreted by the receiver as a binary 1 and vice versa – and, if an error is detected, the packet is simply discarded. The service offered by a packet-switched network is said, therefore, to be a **best-effort service**. 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. However, with this mode of operation, it is possible for a sequence of packets to be received on a number of incoming links all of which need forwarding on the same outgoing link. Hence a packet may experience an additional delay while it is in the output queue for a link waiting to be transmitted. Clearly, this delay will be variable since it depends on the number of packets that are currently present in the queue when a new packet arrives for forwarding. This mode of operation is known as (packet) **store-and-forward** and, as we can see, there is a packet store-and-forward delay in each PSE/router. The sum of the store-and-forward delays in each PSE/router contributes to the overall transfer delay of the packet across the network. The mean of this delay is known as the **mean packet transfer delay** and the variation about the mean the **delay variation** or **jitter**.

An example of a packet-switched network that operates in the connectionless mode is the Internet, which we shall describe in some detail in Chapter 9. Two examples of networks that operate in the connection-oriented mode are the international X.25 packet-switching network and ATM networks. As we explained in Section 1.3.2, the X.25 network is used primarily for the transfer of files containing text and binary data between large computers. Because of the packet format that is used, the routing of packets is relatively slow with the effect that the X.25 network is unsuitable for most multimedia applications. In contrast, as we described in Section 1.3.5, ATM networks have been designed from the outset to support all types of multimedia applications. This is achieved by using high bit rate interconnecting links and, once a virtual circuit has been set up, a very small fixed-sized packet of 53 bytes is used to transfer the information associated with the call. Each small packet is known as a **cell** and includes a short 5-byte header which enables cells to be switched at the very high link bit rates that are used. It is for this reason that ATM networks are also known as **fast packet-switching networks** or sometimes **cell-switching networks**. We shall describe the operation of ATM networks in Chapter 10.

1.5.4 Multipoint conferencing

As we described in Section 1.4.1, multipoint conferencing features in many interpersonal applications including audio- and videoconferencing, data sharing, and computer-supported cooperative working. Essentially, these involve the exchange of information between three or more terminals/computers. In practice, because of the different modes of operation of the two network types – circuit-switched and packet-switched – multipoint conferencing is implemented in one of two ways: centralized and decentralized.

The **centralized mode** is used with circuit-switched networks such as a PSTN or an ISDN and, as we show in Figure 1.21(a), with this mode a centralized conference server is used. Prior to sending any information, each terminal/computer to be involved in the conference must first set up a connection to the server. Each terminal/computer then sends its own media stream – comprising, say, audio, video, and data integrated together in some way – to the server using the established connection. The server, in turn, then distributes either the media stream received from a selected terminal/computer or a mix of the media streams received from several terminals/computers back to all the other terminals/computers that are involved in the conference.

The **decentralized mode** is used with packet-switched networks that support multicast communications. Examples include local area networks, intranets, and the Internet. In this mode, as we show in Figure 1.21(b), the output of each terminal/computer is received by all the other members of the conference/multicast group. Hence a conference server is not normally used and instead it is the responsibility of each terminal/computer to manage the information streams that it receives from the other members.

In addition, a third mode known as the **hybrid mode** can be used. This is shown in Figure 1.21(c) and, as we can see, it is used when the various terminals/computers that make up the conference are attached to different network types. In the example shown, the conference comprises four terminals/computers, two attached to a circuit-switched network and two to a packet-switched network that supports multicasting. As in the centralized mode, a conference server is used and the output of each terminal/computer is sent to the server either over individual circuits – terminals A and B – or using multicasting – terminals C and D. However, in this mode, as in the centralized mode, it is the server that determines the output stream(s) to be sent to each terminal.

As we explained in Section 1.4.1, there are four types of conferencing:

- data conferencing: this involves data only and examples include data sharing and computer-supported cooperative working;

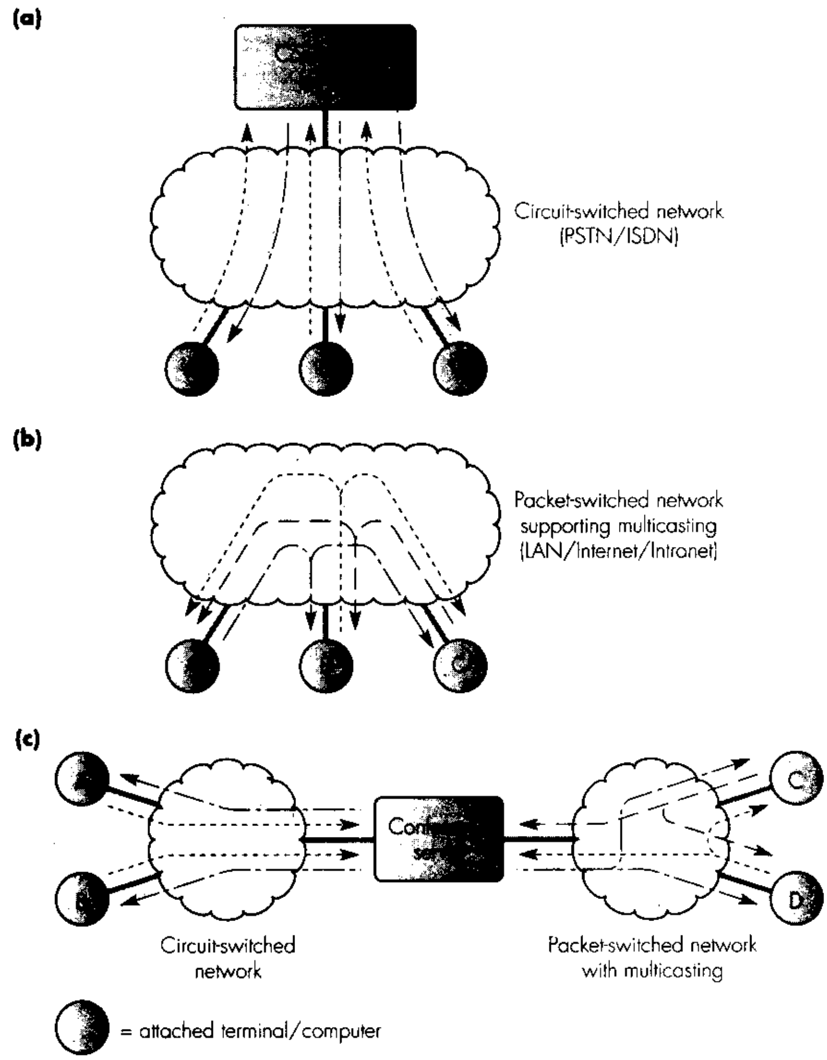


Figure 1.21 Multipoint conferencing modes of operation: (a) centralized; (b) decentralized; (c) hybrid.

- audioconferencing: this involves audio (speech) only;
- videoconferencing: this involves speech and video synchronized and integrated together;
- multimedia conferencing: this involves speech, video, and data integrated together.

speech/video/data

With data conferencing, the information flow between the various parties is relatively infrequent. Normally, therefore, the conference server is a general-purpose computer with the conference function implemented in software. With the other three types of conferencing, however, the information flows demand the use of special purpose units. In the case of audioconferencing, the unit is called an **audio bridge** and typical units support from six through to 48 conference participants. With video and multimedia conferencing, the unit is called a multipoint control unit (MCU) and, because of the volume and rate of the information being exchanged, normally, the centralized mode of working is used with both network types.

An MCU consists of two parts: the first is known as the **multipoint controller (MC)** part and is concerned with the establishment of connections to each of the conference participants and with the negotiation of an agreed set of operational parameters – screen resolution, refresh rate, and so on. The second part is known as the **multipoint processor (MP)** and is concerned with the distribution of the information streams generated during the conference. The latter include such functions as the mixing of the various media streams into an integrated stream, voice-activated switching and continuous presence.

concerned with dist. of info streams

When using an audio bridge, a call is scheduled for a particular date, time, and duration and everyone who is to take part in the call is assigned a user ID and password. At the appropriate time, all participants call in and, after they have been verified to join the conference, they can hear and speak to the other participants. In a similar way, when using an MCU, a call is scheduled as for an audio bridge and, once the conference starts, each participant can hear, see, and share data with the other participants. With an MCU, however, in addition to the participants calling in – known as the **dial-in mode** – in some instances, the MCU calls the participants – the **dial-out mode** – which, in general, provides better security.

In the **voice-activated switching mode**, the face of the participant is displayed in a window on the screen of the participant's terminal/computer and, in a second window, is 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. 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**, however, the remote window is divided into a number of smaller windows, each of which displays the face of the last set of participants who spoke or who 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.

1.5.5 Network QoS

The operational parameters associated with a communications channel through a network are known as the **network Quality of Service (QoS) parameters** and collectively they determine the suitability of the channel in relation to its use for a particular application. In practice, the QoS parameters associated with a circuit-switched network are different from those associated with a packet-switched network and hence we shall discuss each separately.

Circuit-switched network

The QoS parameters associated with a constant bit rate channel that is set up through a circuit-switched network include:

- the bit rate,
- the mean bit error rate,
- the transmission delay.

The mean **bit error rate (BER)** of a channel is the probability of a bit being corrupted during its transmission across the channel in a defined time interval. Hence, for a constant bit rate channel, this equates to the probability of a bit being corrupted in a defined number of bits. A mean BER of 10^{-3}, therefore, means that, on average, for every 1000 bits that are transmitted, 1 of these bits will be corrupted. In some applications, providing the occurrence of bit errors is relatively infrequent, their presence is acceptable while in other applications it is imperative that no residual bit errors are present in the received information. For example, if the application involves speech, then an occasional bit error will go unnoticed but in an application involving the transfer of, say, financial information, it is essential that the received information contains no errors. Hence with such applications, prior to transmission the source information is normally divided into blocks the maximum size of which is determined by the mean BER of the communications channel.

For example, if the mean BER is 10^{-3}, then the number of bits in a block must be considerably less than 1000 otherwise, on average, every block will contain an error and will be discarded. Normally, however, bit errors occur randomly and hence, even with a block size of, say, 100 bits, blocks may still contain an error but the probability of this occurring is considerably less. In general, if the BER probability is P and the number of bits in a block is N, then, assuming random errors, the probability of a block containing a bit error, P_B, is given by:

$$P_B = 1 - (1-P)^N$$

which approximates to $N \times P$ if $N \times P$ is less than 1.

In practice, most networks – both circuit-switched and packet-switched – provide an **unreliable service** which is also known as a **best-try** or **best-effort** service. This means that any blocks containing bit errors will be discarded either within the network – packet-switched networks – or in the network interface at the destination – both packet-switched and circuit-switched networks. Hence if the application dictates that only error-free blocks are

BER

Example 1.1

Derive the maximum block size that should be used over a channel which has a mean BER probability of 10^{-4} if the probability of a block containing an error – and hence being discarded – is to be 10^{-1}.

Answer:

$$P_B = 1 - (1 - P)^N$$

Hence $0.1 = 1 - (1 - 10^{-4})^N$ and $N = 950$ bits $0.1 = 1 - (1 - 10^{-4})^N$

$N = 950$.

Alternatively, $P_B = N \times P$

Hence $0.1 = N \times 10^{-4}$ and $N = 1000$ bits

$P_B = N \times P$
$0.1 = N \times 10^{-4}$
$N = 1000$ bits

acceptable, it is necessary for the sending terminal/computer to divide the source information into blocks of a defined maximum size and for the destination to detect when a block is missing. When this occurs the destination must request that the source send another copy of the missing block. The service offered is then said to be a **reliable service**. Clearly, this will introduce a delay so the retransmission procedure should be invoked relatively infrequently, which dictates a small block size. This, however, leads to high overheads since each block must contain the additional information that is associated with the retransmission procedure. Normally, therefore, the choice of block size is a compromise between the increased delay resulting from a large block size – and hence retransmissions – and the loss of transmission bandwidth resulting from the high overheads of using a smaller block size.

The **transmission delay** associated with a channel is determined not only by the bit rate that is used but also delays that occur in the terminal/computer network interfaces (known as codec delays), plus the propagation delay of the digital signals as they pass from the source to the destination across the network. This is determined by the physical separation of the two communicating devices and the velocity of propagation of a signal across the transmission medium. In free space, for example, the latter is equal to the speed of light ($3 \times 10^8\,\mathrm{ms}^{-1}$) while it is a fraction of this in physical media, a typical value being $2 \times 10^8\,\mathrm{ms}^{-1}$.

Notice that the propagation delay in each case is independent of the bit rate of the communications channel and, assuming the codec delay remains constant, is the same whether the bit rate is 1 kbps, 1 Mbps, or 1 Gbps.

Packet-switched network

The QoS parameters associated with a packet-switched network include:

- the maximum packet size,
- the mean packet transfer rate,

Example 1.2

Determine the propagation delay associated with the following communication channels:

(i) a connection through a private telephone network of 1 km,

(ii) a connection through a PSTN of 200 km,

(iii) a connection over a satellite channel of 50 000 km.

Assume that the velocity of propagation of a signal in the case of (i) and (ii) is 2×10^8 ms^{-1} and in the case of (iii) 3×10^8 ms^{-1}.

Answer:

Propagation delay T_p = physical separation/velocity of propagation

(i) $T_p = \frac{10^3}{2\times10^8} = 5\times10^{-6}$ s

(ii) $T_p = \frac{200\times10^3}{2\times10^8} = 10^{-3}$ s

(iii) $T_p = \frac{5\times10^7}{3\times10^8} = 1.67\times10^{-1}$ s

- the mean packet error rate,
- the mean packet transfer delay,
- the worst-case jitter,
- the transmission delay.

In a packet-switched network, although the rate at which packets are transferred across the network is influenced strongly by the bit rate of the interconnecting links, because of the variable store-and-forward delays in each PSE/router, the actual rate of transfer of packets across the network is also variable. Hence 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** or **PER** is the probability of a received packet containing one or more bit errors. It is the same, therefore, as the block error rate associated with a circuit-switched network which we derived in the previous section. Hence it is related to both the maximum packet size and the worst-case BER of the transmission links that interconnect the PSEs/routers that make up the network.

We defined the meaning of the term "mean packet transfer delay" in Section 1.5.3 when we described the operation of packet-mode networks. It is the summation of the mean store-and-forward delay 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. As we just explained, the transmission delay is the same whether the network operates in a packet mode or a circuit mode and includes the codec delay in each of the two communicating computers and the signal propagation delay.

1.5.6 Application QoS

The network QoS parameters define what the particular network being used provides rather than what the application requires. The application itself, however, also has QoS parameters associated with it. In an application involving images, for example, the parameters may include a minimum image resolution and size, while in an application involving video, the digitization format and refresh rate may be defined. The application QoS parameters that relate to the network include:

- the required bit rate or mean packet transfer rate,
- the maximum startup delay,
- the maximum end-to-end delay,
- the maximum delay variation/jitter,
- the maximum round-trip delay.

For applications involving the transfer of a constant bit rate stream, the important parameters are the required bit rate/mean packet transfer rate, the end-to-end delay, and, equally important, the delay variation/jitter since this can cause problems in the destination decoder if the rate of arrival of the bitstream is variable. For interactive applications, however, the **startup delay** defines the amount of time that elapses between an application making a request to start a session and the confirmation being received from the application at the destination – a server, for example – that it is prepared to accept the request. Hence this includes, in addition to the time required to establish a network connection – if this is required – the delay introduced in both the source and the destination computers while negotiating that the session can take place. As we saw earlier in Section 1.5.1, the round-trip delay is important since, for human–computer interaction to be successful, the delay between a request for some information being made and the start of the information being received/displayed should be as short as possible and, ideally, should be less than a few seconds.

As we can see from the above, for applications that involve the transfer of a constant bit rate stream, a circuit-switched network would appear to be most appropriate since, firstly, the call setup delay is often not important and secondly, the channel provides a constant bit rate service of a known rate.

Conversely, for interactive applications, a connectionless packet-switched network would appear to be most appropriate since with this there is no network call setup delay and any variations in the packet transfer delay are not important.

An example application that illustrates the benefits of a packet-switched network over a circuit-switched network is the transfer of a large file of data from a server computer connected to the Internet to a client PC/workstation in a home. As we showed earlier in Figures 1.2 and 1.18, access to the Internet from home can be by means of a PSTN (with a modem), an ISDN connection, or a cable modem. In the case of a PSTN and an ISDN, these operate in a circuit-switched mode and provide a constant bit rate channel of in the order of 28.8 kbps (PSTN with modem) and 64/128 kbps (ISDN). In contrast, cable modems operate in a packet mode and, as we shall see later in Section 11.2.1, the modems in each of the homes in a cable region time-share the use of a single high bit rate channel/circuit. A typical bit rate of the shared channel is 27 Mbps and the number of concurrent users of the channel may be several hundred. Hence, assuming 270 concurrent users, each user would get a mean data rate of 100 kbps.

With this type of application, however, the main parameter of interest is not the mean data/bit rate but the time to transmit the complete file. With a PSTN and an ISDN, this is directly related to the channel bit rate and the size of the file. With a cable modem, however, although they time-share the use of the 27 Mbps channel, when they gain access to it, the file transfer takes place at the full rate. Hence assuming the file size is 100 Mbits, the minimum time to transmit the file using the different Internet access modes is:

PSTN and 28.8 kbps modem:	57.8 minutes
ISDN at 64 kbps:	26 minutes
ISDN at 128 kbps:	13 minutes
cable modem at 27 Mbps:	3.7 seconds

In the case of a cable modem, if other transfer requests occur during the time the file is being transmitted, then the completion time of each transfer request will increase as they share the use of the channel. Nevertheless, with this type of application, the probability of multiple users requesting a transfer in this short window of time is relatively low.

In many instances, however, this does not mean that the alternative network types cannot be used. For interactive applications, for example, the call setup delay with an ISDN or an ATM network – and a PSTN for local calls – is very fast and, for many applications, quite acceptable. Similarly, for constant bit rate applications, providing the equivalent mean bit rate provided by the network is greater than the input bit rate and the maximum jitter is less than a defined value, then a packet-switched network can be used. To overcome the effect of jitter a technique known as **buffering** is used, the general principles of which are shown in Figure 1.22.

As we show in the figure, the effect of jitter is overcome by retaining a defined number of packets in a memory buffer at the destination before playout of the information bitstream is started. The memory buffer operates

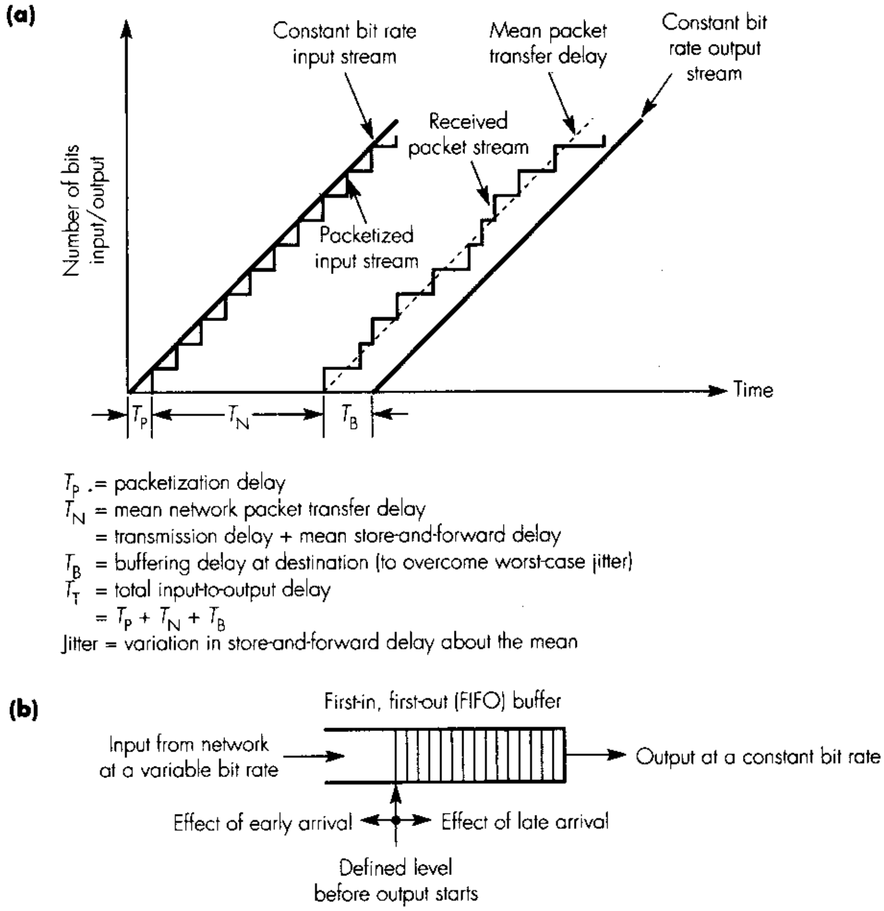


Figure 1.22 Transmission of a constant bit rate stream over a packet-switched network: (a) timing schematic; (b) FIFO buffer operation.

using a first-in, first-out (FIFO) discipline and the number of packets retained in the buffer before output starts is determined by the worst-case jitter and the bit rate of the information stream. However, as we show in part (a) of the figure, when using a packet-switched network for this type of application, an additional delay is incurred at the source as the information bitstream 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 order to simplify the process of determining whether a particular network can meet the QoS requirements of an application, a number of standard application **service classes** have been defined. Associated with each service class is a

Example 1.3

A packet-switched network with a worst-case jitter of 10 ms is to be used for a number of applications each of which involve a constant bit rate information 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. It can be assumed that the mean packet transfer rate of the network exceeds the equivalent input bit rate in each case.

(i) 64 kbps

(ii) 256 kbps

(iii) 1.5 Mbps.

Answer:

(i) At 64 kbps, 10 ms = 640 bits
Hence choose a packet size of, say, 800 bits with a FIFO buffer of 1600 bits – 2 packets – and start playout of the bitstream after the first packet has been received.

(ii) At 256 kbps, 10 ms = 2560 bits
Hence choose a packet size of, say, 2800 bits with a FIFO buffer of 4800 bits.

(iii) At 1.5 Mbps, 10 ms = 15000 bits
Hence choose a packet size of, say, 16000 bits with a FIFO buffer of 32000 bits.

Notice that if the computed packet size exceeds the network maximum packet size, then the equivalent number of packets must be sent before playout starts. For example, if the maximum network packet size was 8000 bits, then for case (iii) above playout would not start until two packets have been received and the FIFO buffer should hold four packets.

specific set of QoS parameters and a network can either meet this set of parameters or not. Also, for networks that support a number of different service classes – the Internet for example – in order to ensure the QoS parameters associated with each class are met, the packets relating to each class are given a different **priority**. It is then possible to treat the packets relating to each class differently.

For example, as we shall see in Chapter 9, in the Internet, packets relating to multimedia applications involving real-time streams are given a higher priority than the packets relating to applications such as email. Typically, packets containing real-time streams such as audio and video are also more sensitive to delay and jitter than the packets containing textual information. Hence during periods of network congestion, the packets containing real-time streams are transmitted first. Packets containing video are more sensitive to packet loss than packets containing audio and hence are given a higher priority.

Summary

In this chapter we have discussed:

- the different types of media that are used in multimedia applications,
- the different types of communication networks that are used to support these applications,
- a selection of the different types of application.

Media types

The different types of media that are used in multimedia applications are summarized in Figure 1.23. In some applications only a single type of medium is involved while in others multiple media types are used. As we described in Section 1.2, the basic form of representation of both text and images consists of a block of binary codewords. Each codeword consists of a fixed number of binary digits and represents, for example, a single character (text) or a single picture element (image). In contrast, the basic form of

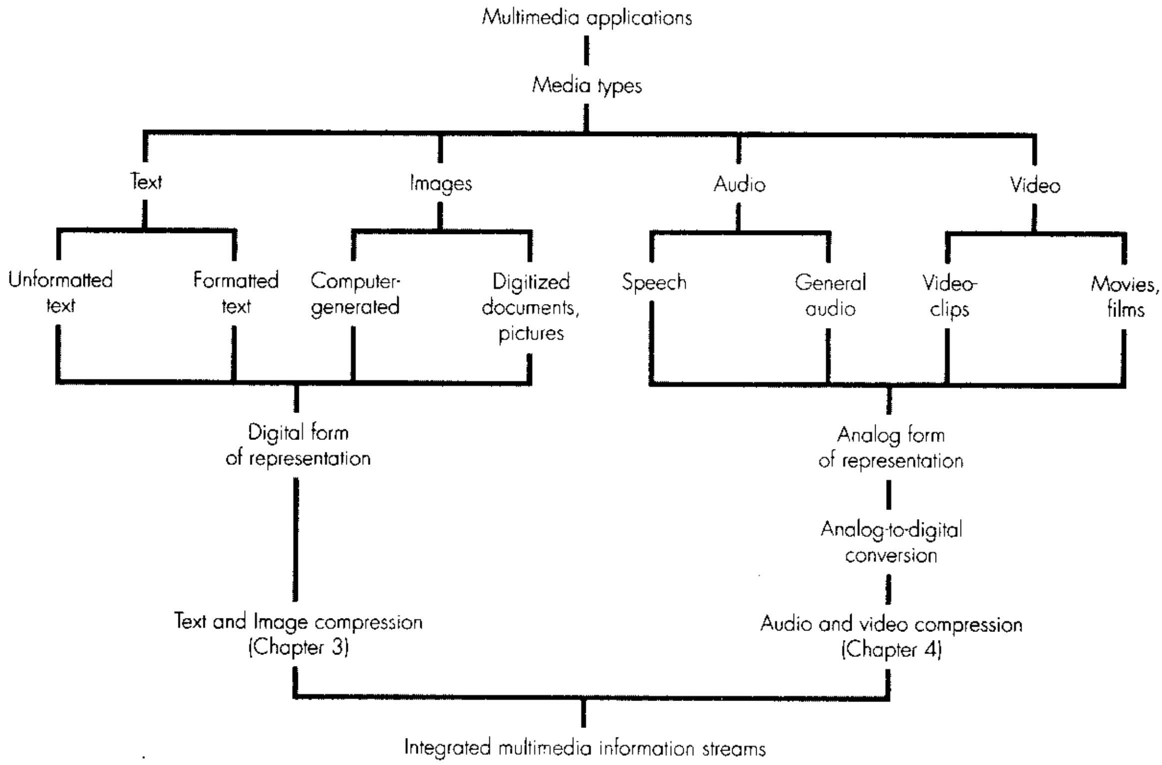


Figure 1.23 Alternative types of media used in multimedia applications.

representation of both audio and most video signals is in the form of an analog signal whose amplitude and frequency varies continuously with time. Hence in applications that involve audio and video integrated with text and/or images, it is necessary first to convert the audio and video signals into a digital form. We shall describe the digital form of representation of all four media types in more detail in the next chapter.

In practice, the bandwidth associated with the basic form of representation of the different media types is greater than the bandwidth of the communication networks that are used to support multimedia applications. Hence in most applications, a technique known as compression is applied to the source information prior to it being transferred over the network. We shall describe a selection of the different compression algorithms that are used with text and images in Chapter 3 and a selection of those used with audio and video in Chapter 4.

Network types

The different types of communication networks that are used to support multimedia applications are summarized in Figure 1.24 and a selection of the applications that each network type supports are shown in Figure 1.25.

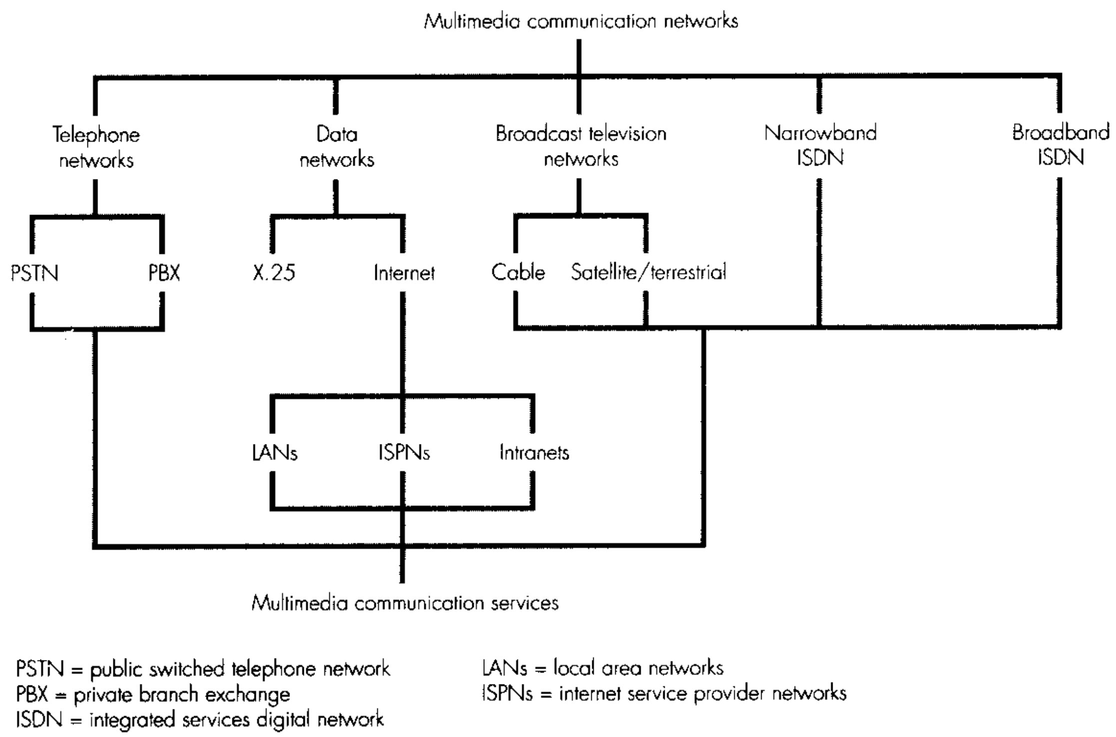


Figure 1.24 Multimedia communication networks.

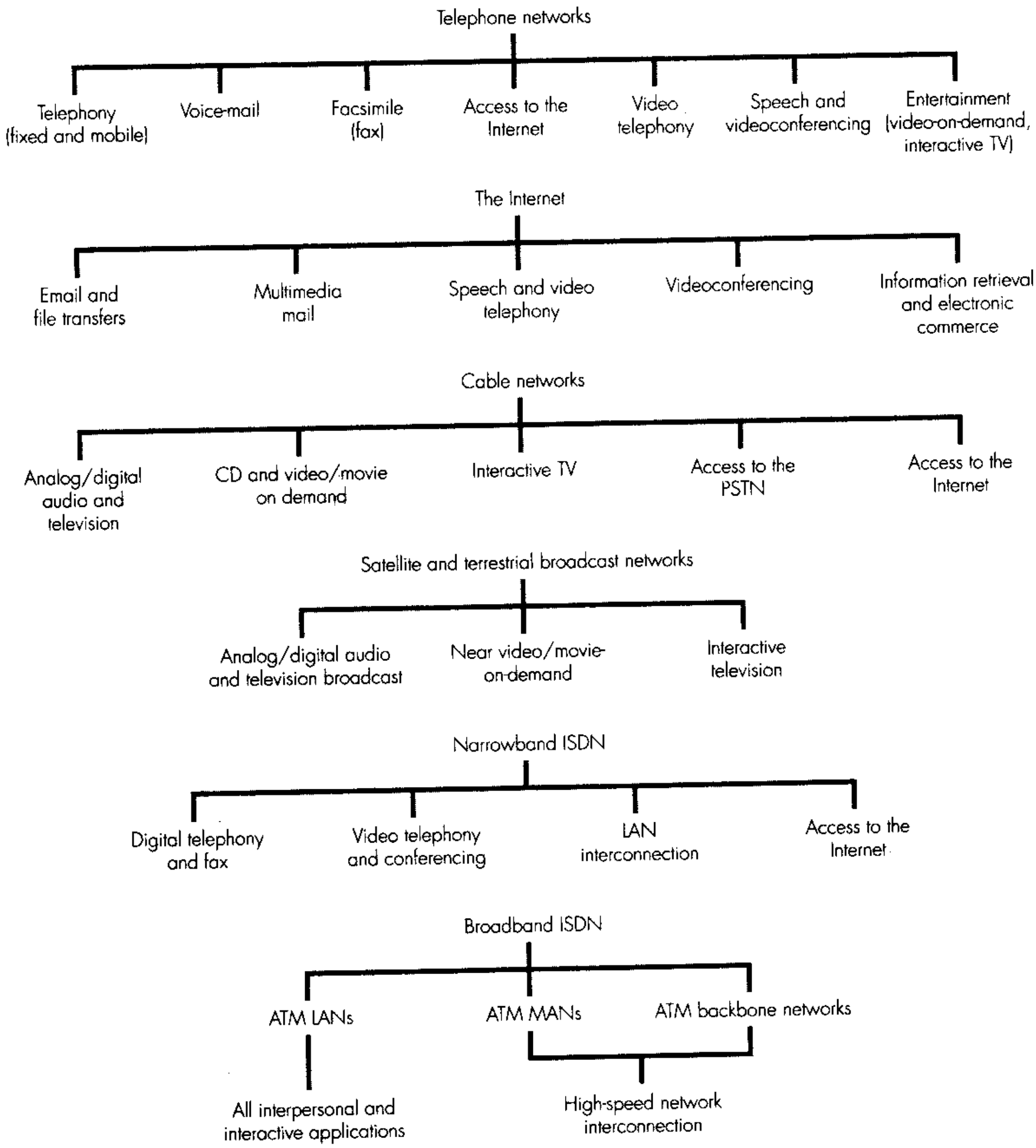


Figure 1.25 Multimedia communication networks and their services.

In the case of telephone networks, these were designed initially to provide a switched telephone service both nationally/internationally (a PSTN) and at a single site (a PBX). Also, with the advent of the first modems, they provided a low bit rate switched data service. Later, as a result of advances in modem technology and related compression algorithms, the transfer of digitized documents (facsimile) was supported. These developments have continued and the PSTN now supports a wide range of applications that involve all four media types either singly or integrated together in some way.

Similarly, with data networks – and in particular the Internet. Although designed initially to support applications that involve only text and images, the advances in compression algorithms means that the Internet now supports a similar range of multimedia applications. The applications associated with broadcast television networks have been enhanced by the same advances in modem technology and compression algorithms.

The last two network types – narrowband and broadband ISDN – were both designed from the outset to provide an enhanced set of services to those provided by a PSTN. A narrowband ISDN, for example, has a range of digital access alternatives which, collectively, support a wide range of multimedia applications. Broadband ISDN was designed from the start to support multimedia applications that involve all four media types. In practice, however, because of the wider set of (multimedia) applications that the other networks can now support, broadband ISDN is not as yet widely deployed. Nevertheless, the basic transmission and switching technology associated with it – known as the asynchronous transfer mode (ATM) and cell switching respectively – has been used in high-speed LANs and for high-speed network interconnection.

We shall discuss in some detail the operation of these various network types in Chapters 7–11 and the communication protocols and other application-related issues that are associated with them in Chapters 12–14.

Multimedia applications

We have chosen to place all multimedia applications into one of three categories:

- interpersonal communications,
- interactive applications over the Internet,
- entertainment applications.

There are a number of applications in each category each of which involves various media types. A selection of the applications, together with the type of media involved, are summarized in Table 1.1. Note that each type of application can be supported by a number of different network types. For example, video telephony is supported by a PSTN, the Internet, and both narrowband and broadband ISDN.

Table 1.1 Multimedia applications summary.

Category	Media	Application descriptions
Interpersonal communications	Speech	Telephony, voice-mail, teleconferencing
	Image	Facsimile
	Text	Electronic mail
	Text and images	Computer-supported cooperative working (CSCW)
	Speech and video	Video telephony, video mail, videoconferencing
	Text, image audio and video	Multimedia electronic mail, multiparty video games etc.
Interactive applications over the Internet	Text, image, audio and video	Information retrieval (news, weather, books magazines, video games, product literature etc.)
		Electronic commerce
Entertainment services	Audio and video	Audio/CD-on-demand
		Movie/video-on-demand
		Near movie/video-on-demand
		Analog and digital television broadcasts
		Interactive television

Exercises

Section 1.2

1.1 (a) State the basic form of representation of:
- (i) text,
- (ii) an image,
- (iii) audio,
- (iv) video.

State the form of representation that is used when all are integrated together and give your reason.

1.2 State the meaning of the term "bits per second" in relation to digitized audio and video. What is the meaning of the term "compression" and why is compression used?

Section 1.3

1.3 With the aid of a diagram, explain the meaning of the following terms relating to a switched telephone network:
- (i) POTS,
- (ii) local exchange/end office,
- (iii) PBX,
- (iv) mobile switching center,
- (v) international gateway exchange.

1.4 Explain why a pair of modems is required to transmit a digital signal over a PSTN. With the aid of a diagram, show the location of the two modems when two digital devices communicate over a PSTN and the types of signal – analog or digital – that are used over each part of the access circuit.

1.5 Show in the form of a diagram how a high-speed modem provides a range of additional services to basic telephony and data services. State a typical access bit rate that is used.

1.6 With the aid of a diagram, describe how the following gain access to the Internet:
(i) a user at home or in a small business,
(ii) a distributed community of users around a single site or campus,
(iii) a distributed set of users all attatched to the same enterprise-wide private network. Why is the latter sometimes called an intranet?

1.7 Explain why most data networks operate in a packet mode. Hence explain why services involving audio and video are supported.

1.8 State the aim of all broadcast television networks.
With the aid of diagrams, explain how additional services are provided with
(i) a cable distribution network and
(ii) a satellite/terrestrial broadcast network.

1.9 With aid of a diagram, explain the meaning of the following terms relating to ISDN:
(i) digital subscriber line,
(ii) basic rate access,
(iii) aggregation,
(iv) primary rate access,
(v) $p \times 64$.

1.10 Explain the meaning of the term "broadband" in relation to a B-ISDN and why the deployment of such networks has been delayed.

1.11 Explain the meaning of the following terms relating to a B-ISDN:
(i) a cell,
(ii) ATM,
(iii) cell switching.

Section 1.4

1.12 In relation to speech-only interpersonal communications involving both public (PSTN/ISDN) and private (PBX) networks, with the aid of a diagram explain how voice-mail and teleconferencing are supported. Include in your descriptions the role of a voice-mail server and an audio bridge.

1.13 With the aid of a diagram, explain the function of a telephony gateway in relation to Internet telephony. Hence state the origin of the term "voice over IP" (VoIP).

1.14 Describe the principal operation of a fax machine and why modems are required. What is the meaning of the term "PC fax"?

1.15 Show in the form of a diagram the networks and essential items of equipment that are used to send an email message from a PC user at home to
(i) a PC attached to a site/campus LAN,
(ii) a PC attached to a site/campus LAN,
(iii) a PC attached to an enterprise-wide private network/intranet.

1.16 With the aid of a diagram, explain the basic principles behind computer-supported cooperative working (CSCW). Include in your explanation the role of the following:
(i) shared whiteboard,
(ii) whiteboard program,
(iii) change-notification,
(iv) update control.

1.17 Use a schematic diagram to illustrate each of the following:
(i) a two-party video phone call,
(ii) a videoconferencing call using an MCU,
(iii) a videoconferencing call using a broadcast network. Show on your diagrams the bandwith requirements of the access circuits in each case.

1.18 Explain the role of an MCU in relation to a videoconferencing session involving multiple geographically distributed videoconferencing studios.
Quantify the bandwidth implications of locating the MCU at one of the sites.

1.19 With multimedia electronic mail, when a message in a number of different formats is sent –

for example, speech-with-video, speech-only and text – normally the first version in output first. Why is this?

1.20 Use a schematic diagram to explain the meaning of the following terms relating to interactive applications over the inernet:
(i) a Web server,
(ii) a browser,
(iii) the World Wide Web.

1.21 In relation to a document stored on a Web server, explain the meaning of the following terms:
(i) Web page,
(ii) home page,
(iii) hyperlink,
(iv) URL,
(v) HTML.

1.22 With the aid of a diagram, explain the principle of operation of movie/video-on-demand. Identify the bandwidth requirements associated with this type of application.

1.23 What is the limiting factor associated with movie/video-on-demand? Explain how this effect is reduced with near movie-on-demand.

1.24 Explain the meaning of the term "interactive television" and, with the aid of diagrams, how this is using either a cable distribution network or a satellite/terrestrial broadcast network. Explain how Internet access is obtained with each network type.

Section 1.5

1.25 What is the meaning of the following terms relating to the different types of media used in multimedia communications:
(i) block-mode media,
(ii) continuous media,
(iii) streaming,
(iv) constant bit rate,
(v) variable bit rate?

1.26 With the aid of diagrams, explain the meaning of the following operational modes of a communication channel:
(i) simplex,
(ii) half-duplex,
(iii) duplex,
(iv) broadcast
(v) multicast,
(vi) asymmetric and symmetric.

1.27 With the aid of a diagram explain the principle of operation of a circuit-mode network. Include in your explanation the need for signaling (messages) and the overheads of the connection setup delay associated with this.

1.28 With the aid of diagrams, explain the principle of operation of a CO packet-mode network. Include in your explanation the need for a virtual connection/circuit, a virtual circuit identifier, and a routing table.

1.29 In relation to a CL packet-mode network, explain the meaning of the following terms:
(i) best effort service,
(ii) store-and-forward display,
(iii) mean packet transfer delay,
(iv) jitter.

1.30 With the aid of diagrams, explain the following operational modes of multipoint conferencing:
(i) centralized,
(ii) decentralized,
(iii) hybrid.

1.31 Identify the application domain and explain the operation of the following types of conference server:
(i) general purpose,
(ii) audio bridge,
(iii) multipoint control unit,

1.32 In relation to conference servers, explain the meaning of the following terms:
(i) multipoint controller,
(ii) multipoint processor,
(iii) dial-in mode,

(iv) dial-out mode,
(v) voice activated switching,
(vi) continuous presence.

1.33 Identify and explain the meaning of the key QoS parameters associated with the following network types:
(i) circuit-switched,
(ii) packet-switched.

1.34 Define the BER probability of a transmission line/channel. How does this influence the maximum block size to be used with the line/channel?

1.35 Define the transmission delay of a transmission line/channel. First identify the individual sources of delay that contribute to this.

1.36 Identify and explain the meaning of the key parameters associated with application QoS.

1.37 With the aid of a diagram explain the meaning of the following terms relating to packet-switched network:
(i) packetization delay,
(ii) mean packet transfer delay,
(iii) jitter.

Hence describe how the effects on a constant bit rate stream of packetization delay and jitter can be overcome by buffering.

1.38 A Web page of 10 Mbytes 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 circuit:
(i) a PSTN modem operating at 28.8 kbps,
(ii) an aggregated basic rate access line of 128 kbps,
(iii) a primary rate ISDN access line of 1.5 Mbps,
(iv) a high-speed modem operating at 6 Mbps,
(v) a cable modem operating at 27 Mbps.

1.39 Discuss the term "application service classes". Include in your dicussion how packets belonging to different classes are treated within the network.