Cable and Wireless Networks

Theory and Practice

Mário Marques da Silva

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Contents

Summary

This book presents a comprehensive approach to networking, cable and wireless communications, and networking security. It describes the most important state-of-the-art fundamentals and system details in the field, as well as many key aspects concerning the development and understanding of current and emergent services.

Three of the author's earlier books, *Transmission Techniques for Emergent Multicast and Broadcast Systems*, *Transmission Techniques for 4G systems*, and *MIMO Processing for 4G and Beyond: Fundamentals and Evolution*, focused on the transition from 3G into 4G and 5G cellular systems, including the fundamentals of multi-input and multi-output (MIMO) systems, and therefore, they spanned a wide range of topics. Another book by the author, *Multimedia Communications and Networking*, focused on networking.

In this book, the author gathers in a single volume his point of view on current and emergent cable and wireless network services and technologies. Different bibliographic sources cover each one of these topics independently, without establishing the natural relationships between the topics. The advantage of the present work is twofold: on the one hand, it allows the reader to learn quickly, thereby helping the reader to master the topics covered, providing a deeper understanding of their interconnection; on the other hand, it collects in a single source the latest developments in the area, which are generally only within reach of an active researcher, such as the author, with a committed research career of several years and regular participation in conferences and international projects.

Each chapter illustrates the theory of cable and wireless communications with relevant examples, contains hands-on exercises suitable for readers with a BSc degree or an MSc degree in computer science or electrical engineering, and ends with review questions. This approach makes the book well suited for higher education students in courses such as networking, telecommunications, mobile communications, and network security. Finally, the book serves as a good reference book for academic, institutional, or industrial professionals with technical responsibilities in planning, design and development of networks, telecommunications and security systems, and mobile communications, as well as for Cisco CCNA and CCNP exam preparation.

Laboratorial Introductory Notes

The lab exercises included in this book focus on three tools: the Emona Telecoms Trainer 101 (ETT-101), for which a dual-channel 20 MHz oscilloscope is required; the free network analyzer, Wireshark; and the Cisco Packet Tracer network simulator.

Emona ETT-101 consists of a telecommunications modeling system that brings block diagrams to life, with real hardware modules and real electrical signals, which are employed in this book to demonstrate the theory about telecommunications. As alternatives to ETT-101, two other pieces of laboratorial equipment can be used: the Emona TIMS 301-C Telecommunications Teaching System and the Emona net*TIMS Telecommunications Teaching System. Emona TIMS 301-C corresponds to ETT-101 with extended capabilities. Emona net*TIMS allows implementation of the same experiments that TIMS 301-C does, but these can be built and controlled remotely by students across a LAN or the Internet (multiple students can do their lab work at any time and from any location in the world). Appendix VI lists experiments that can be implemented with Emona TIMS (both 301-C and net*TIMS), indicating the chapters that discuss each experiment. The free network analyzer, Wireshark, is used to demonstrate the theory on networking, namely signaling, message formats, and network procedures. The Cisco Packet Tracer simulator is used to build networks, to configure them, and to simulate their responses. Some chapters focus on telecommunications, and therefore ETT-101 is used extensively. Other chapters focus on networking, and discuss the utilization of network Wireshark and Packet Tracer. Since the ETT-101 laboratory manual already describes many experiments, the lab exercises presented in chapters on telecommunications simply refer to the different ETT-101 experiments. In this case, the student should refer to the descriptions existing in the ETT-101 laboratory manual, namely Volume 1—*Experiments in Modern Analog and Digital Telecommunications*; Volume 2—*Further Experiments in Modern Analog & Digital Telecommunications*; and Volume 3—*Advanced Experiments in Modern Analog & Digital Telecommunications.*

Author

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From 2005 to 2008, he was with the NATO Air Command Control and Management Agency in Brussels, Belgium, where he managed the deployable communications of the new Air Command and Control System Program. He has been involved in multiple networking and telecommunications projects. His research interests include networking and mobile communications, namely Internet protocol (IP) technologies and network security, block transmission techniques, interference cancellation, MIMO systems, and software-defined radio. He is also a Cisco Certified Network Associate (CCNA) instructor.

He is the author of four books published by CRC Press, *Multimedia Communications and Networking*, *Transmission Techniques for Emergent Multicast and Broadcast Systems*, *Transmission Techniques for 4G Systems*, and *MIMO Processing for 4G and Beyond: Fundamentals and Evolution*. He has authored dozens of journal and conference papers, is a member of IEEE and AFCEA, and has been a reviewer for a number of international scientific IEEE journals and conferences. He has also chaired many conference sessions and has been serving in the organizing committee of relevant EURASIP and IEEE conferences.

1 Introduction to Data Communications and Networking

LEARNING OBJECTIVES

- Describe the fundamentals of communications.
- Identify the key components of networks and communication systems.
- Describe different types of networks and communication systems.
- Identify the differences between a local area network (LAN), a metropolitan area network (MAN), and a wide area network (WAN).
- Identify the different types of media and traffic.
- Define the convergence and the collaborative age of the network applications.

1.1 FUNDAMENTALS OF COMMUNICATIONS

Communication systems are used to enable the exchange of data between two or more entities (humans or machines). As can be seen from Figure 1.1, data consists of a representation of information source, whose transformation is performed by a source encoder. An example of a source encoder is a thermometer, which converts temperatures (information source) into voltages (data). A telephone can also be viewed as a source encoder, which converts the analog voice (information source) into a voltage (data), before being transmitted along the telephone network (transmission medium). In case the information source is analog and the transmission medium is digital, a CODEC (COder and DECoder) is employed to perform digitization. A VOCODER (VOice CODER) is a codec specific for voice, whose functionality consists of converting analog voice into digital at the transmitter side, and the reciprocal at the receiver side.

The emitter of data consists of an entity responsible for the insertion of data into the communication system and for the conversion of data into signals. Note that signals are transmitted, rather than data. Signals consist of an adaptation* of data, such that their transmission is facilitated in accordance with the used transmission medium. Similarly, the receiver is responsible for converting the received signals into data.

The received signals correspond to the transmitted signals subject to attenuation and distortion, and added with noise and interferences. These channel impairments originate that the received signal differs from that transmitted. In the case of analog signals, the resulting signal levels do not exactly translate the original information source. In the case of digital signals, the channel impairments originate corrupted bits. In both cases, the referred channel impairments originate a degradation of the signal-to-noise plus interference ratio (SNIR).† A common performance indicator

* Signals can be, for example, a set of predefined voltages that represent bits used in transmission.

† In linear units, the SNIR is mathematically given by $SNIR = S/(N + I)$, where S stands for the power of signal, N expresses the power of noise, and I denotes the power of interferences. For the sake of simplicity, the SNIR is normally only referred to as SNR (signal-to-noise ratio), but where the interference is also taken into account (in this case N stands for the power of noise and interferences). Furthermore, both SNIR (or SNR) are normally expressed in logarithmic units as $SNIR_{dB} = 10\log_{10}(S/[N + I])$.

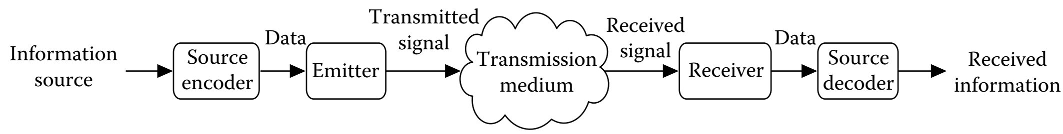

FIGURE 1.1 Generic block diagram of a communication system.

of digital communication systems is the bit error rate (BER). This corresponds to the number of corrupted bits divided by the total number of transmitted bits over a certain time period.

A common definition associated with information is knowledge. It consists of a person's ability to have access to the right information, at the right time. The conversion between information and knowledge can be automatically performed using information systems, whereas information can be captured by sensors and distributed using communication systems.

1.1.1 Analog and Digital Signals

Analog signals present a continuous amplitude variation over time. An example of an analog signal is voice. Contrarily, digital signals present amplitude discontinuities (e.g., voltages or light pulses). An example of digital data includes the bits* generated in a workstation. The text is another example of digital data. Figure 1.2 depicts examples of analog and digital signals.

Digital signals present several advantages (relating to analog) such as the following:

- Error control is possible in digital signals: corrupted bits can be detected and/or corrected.
- Because they present only two discrete values, the consequences of channel impairments can be more easily detected and avoided (as compared to analog signals).
- Digital signals can be regenerated, almost eliminating the effects of channel impairments. Contrarily, the amplification process of analog signals results in the amplification of signals, noise, and interferences, keeping the SNR relationship unchanged.†
- The digital components are normally less expensive than the analog ones.
- Digital signals facilitate cryptography and multiplexing.
- Digital signals can be used to transport different sources of information (voice, data, multimedia, etc.) in a transparent manner.

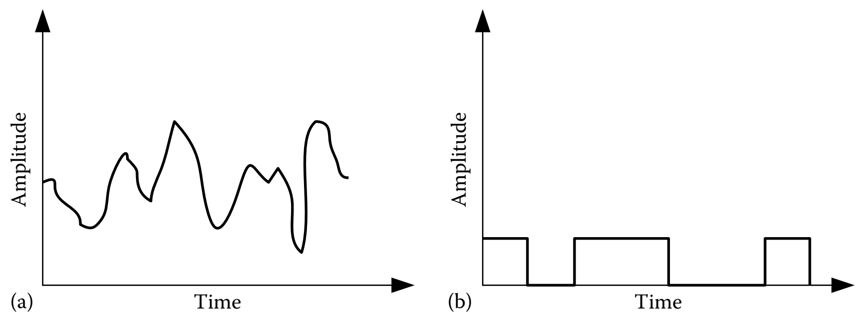

FIGURE 1.2 Example of (a) analog and (b) digital signals.

* With logic states 0 or 1.

† In fact, the amplification process results even in a degradation of the SNR, as it adds the amplifier's internal noise to the signal at its input. This subject is detailed in Chapter 3.

However, digital signals present an important disadvantage:

- For the same information source, the bandwidth required to accommodate a digital signal is typically higher than the analog counterpart.* This results in a higher level of attenuation and distortion.

1.1.2 Modulator and Demodulator

As can be seen from Figure 1.3, when the source (e.g., a computer) generates a digital stream of data and the transmission medium is analog, a MODEM (MOdulator and DEModulator) is employed to perform the required conversion. The modulator converts digital data into analog signals, whereas the demodulator (at the receiver) converts analog signals into digital data. An example of an analog transmission medium is radio transmission, whose signals consist of electromagnetic waves (present a continuous variation in time).

A modem (e.g., asynchronous digital subscriber line [ADSL] or cable modem) is responsible for modulating a carrier wave with bits, using a certain modulation scheme.† The reverse of this operation is performed at the receiver side. Moreover, a modem allows sending a signal modulated around a certain carrier frequency, which can be another reason for using such a device.

In case the data is digital and the transmission medium is also digital, a modem is normally not employed, as the conversion between digital and analog does not need to be performed. In this case, a line encoder/decoder (sometimes also referred to as a *digital modem*, nevertheless not accurately) is employed. This device adapts the original digital data to the digital transmission medium,‡ adapting parameters such as levels and pulse duration. Note that, using such a digital encoder, the signals are transmitted in the baseband.§

The output of a line encoder consists of a digital signal, as it comprises discrete voltages that encode the source logic states. Consequently, it can be stated that the line encoder is employed when the transmission medium is digital. On the other hand, the output of a modulator consists of an analog signal, as it modulates a carrier that is an analog signal.

In the case of high data rate, the required bandwidth necessary to accommodate such a signal is also high.¶ In this scenario, the medium may originate a high level of attenuation or distortion at limit frequency components of the signal. In such a case, it can be a good choice to use a modem that allows the modulation of the signal around a certain carrier frequency. The carrier frequency can be carefully selected such that the channel impairments in the frequencies around it (corresponding to the signal bandwidth) do not seriously degrade the SNR.

The reader should refer to Chapter 6 for a detailed description of the modulation schemes used in modems, as well as for the description of digital encoding techniques.

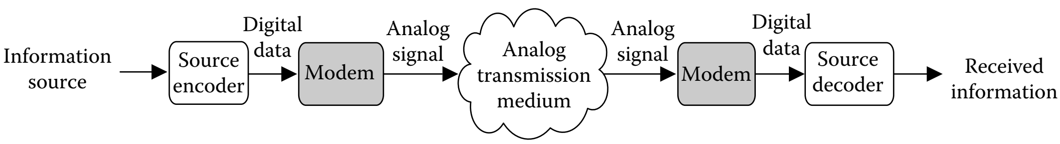

FIGURE 1.3 Generic communication system incorporating a modem.

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* As an example, analog voice is transmitted in a 3.4 kHz bandwidth, whereas the digital pulse code modulation (PCM) requires a bandwidth of 32 kHz (64 kbps).

† Using amplitude, frequency, or phase shift keying. Advanced modems make use of a combination of these elementary modulation schemes.

‡ Using line codes such as return to zero, nonreturn to zero, and Manchester (as detailed in Chapter 6).

§ Instead of carrier modulated (bandpass), as performed by a modem.

¶ According to the Nyquist theorem, as detailed in Chapter 3.

1.1.3 Transmission Mediums

Transmission mediums can be classified as cable or wireless. The examples of cable transmission mediums include twisted pair cables, coaxial cables, multimode or single mode optical fiber cables, and so on.

In the past, LANs were made of coaxial cables. These cables were also used as a transmission medium for medium- and long-range analog communications. Although coaxial cables were replaced by twisted pair cables in LANs, the massification of cable television enabled their reuse.

As a result of telephone cables, twisted pairs are still the dominant transmission medium in houses and offices. These cables are often reused for data. With the improvement in isolators and copper quality, as well as with the development of shielding, the twisted pair has become widely used for providing high-speed data communications, in addition to the initial use for analog telephony.

Currently, multimode optical fibers have been increasingly installed at homes, allowing reaching throughputs of the order of several gigabits per second (Gbps). Moreover, single mode optical fibers are the most used transmission medium in transport networks. A transport network consists of the backbone (core) network, used for transferring large amount of data among different main nodes. These main nodes are then connected to secondary nodes and finally connected to customer nodes.

A radio or wireless communication system is composed of a transmitter and a receiver, using antennas to convert electric signals into electromagnetic waves and vice versa. These electromagnetic waves are propagated over air. Note that wireless transmission mediums can be either guided or unguided. In the former case, directional antennas are used at both the transmitter and the receiver sides, such that electromagnetic waves propagate directly from the transmit into the receive antenna.

The reader should refer to Chapter 4 for a detailed description of cable transmission mediums, while Chapter 5 introduces the wireless transmission mediums.

1.1.4 Synchronous and Asynchronous Communication Systems

Synchronous and asynchronous communications refer to the ability or inability to have information about the start and end of bit instants.*

Using asynchronous communications, the receiver does not achieve perfect time synchronization with the transmitter, and the communication accepts some level of fluctuation. Consequently, start and stop bits are normally included in a frame† in order to periodically achieve bit synchronization of the receiver with the transmitter. Note that between the start and the stop bit, the receiver of an asynchronous communication suffers from a certain amount of time shift. The referred periodic synchronization using start and stop bits is normally included as part of the functionalities implemented by a modem, when establishing a communication in asynchronous mode of operation. Normally, asynchronous communications do not accommodate high-speed data rates. They are typically used for random (not continuous) exchange of data (at low rate).

On the other hand, synchronous communications consider a receiver that is bit synchronized with the transmitter. This bit synchronization can be achieved using one of the following methods:

- By sending a clock signal multiplexed with the data or using a parallel dedicated circuit
- When the transmitted signal presents a high zero crossing rate, such that the receiver can extract the start and end of bit instants from the received signal

Synchronous communications are normally employed in high-speed lines, and for the transmission of large blocks of data. An example of a synchronous communication system is the synchronous digital hierarchy (SDH) networks, used for the transport of large amounts of data in a backbone.

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* Nevertheless, frame synchronization is required in either case.

† A group of exchanged bits.

1.1.5 Simplex and Duplex Communications

A simplex communication consists of a communication between two or more entities where the signals flow only in a single direction. In this case, one entity only acts as a transmitter and the other(s) as a receiver. This can be seen from Figure 1.4. Note that the transmitter may be transmitting signals to more than one receiver.

When the signals flow in a single direction, but with alternation in time, it is stated that the communication is half-duplex. Therefore, although both entities act simultaneously as a transmitter and as a receiver (at different time instants), instantaneously, each host acts as either a transmitter or a receiver [Stallings 2010]. The half-duplex communication is depicted in Figure 1.5.

Finally, when the communication is simultaneously in both directions, it is in full-duplex mode. In this case, two or more entities act simultaneously as both a transmitter and a receiver. The full-duplex communication is depicted in Figure 1.6. Full-duplex communications normally require two parallel transmission mediums (e.g., two pairs of wires): one for transmission and another for reception.

FIGURE 1.4 Simplex communication.

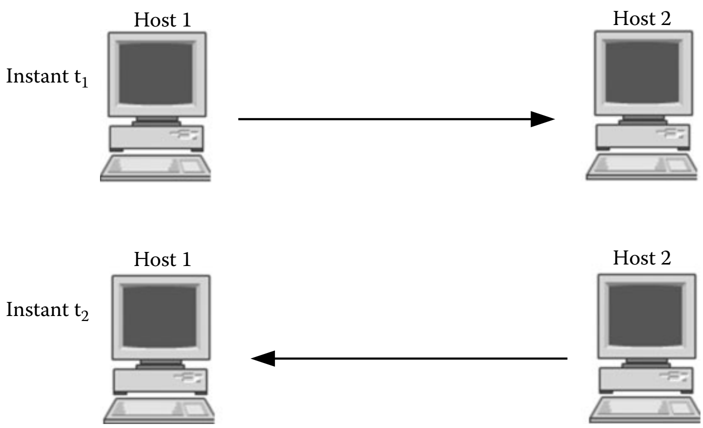

FIGURE 1.5 Half-duplex communication.

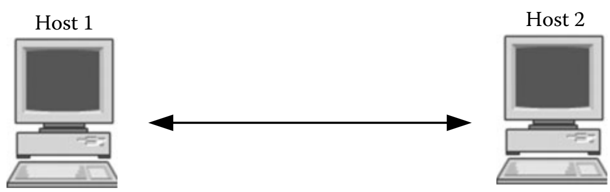

FIGURE 1.6 Full-duplex communication.

1.1.6 Communications and Networks

A point-to-point communication establishes a direct connection (link) between two adjacent end stations, between two adjacent network nodes (e.g., routers), or between an end station and an adjacent node.

A network can be viewed as a concatenation of point-to-point communications, composed of several nodes and end stations, where each node is responsible for switching the data, such that an end-to-end connection is established between two end stations. The examples of point-to-point communications and of a network are depicted in Figure 1.7. An end-to-end network connection consists of a concatenation of several point-to-point links, where each of these links can be implemented using a different transmission medium (e.g., satellite and optical fiber).

A node of a network can be a router or a private automatic branch exchange (PABX). The former device switches packets (packet switching), while the latter is responsible for physically establishing permanent connections, such that a phone call between two end entities is possible (circuit switching). This subject is detailed in Section 1.2.

Depending on the number of destination stations of data involved in a communication, this can be classified as unicast, multicast, or broadcast. Unicast stands for a communication whose destination is a single station. In case the destination of data is all the network stations, the communication is referred to as *broadcast*. Very often broadcast communications are established in a single direction (i.e., there is no feedback from the receiver into the transmitter). Finally, when the destination of the data is more than a single station, but less than all network stations, the communication is referred to as *multicast*.

1.1.7 Switching Modes

1.1.7.1 Circuit Switching

Circuit switching establishes a permanent physical path between the source and the destination. This switching mode is used in classic telephone networks. Only after startup, is allowed a synchronous

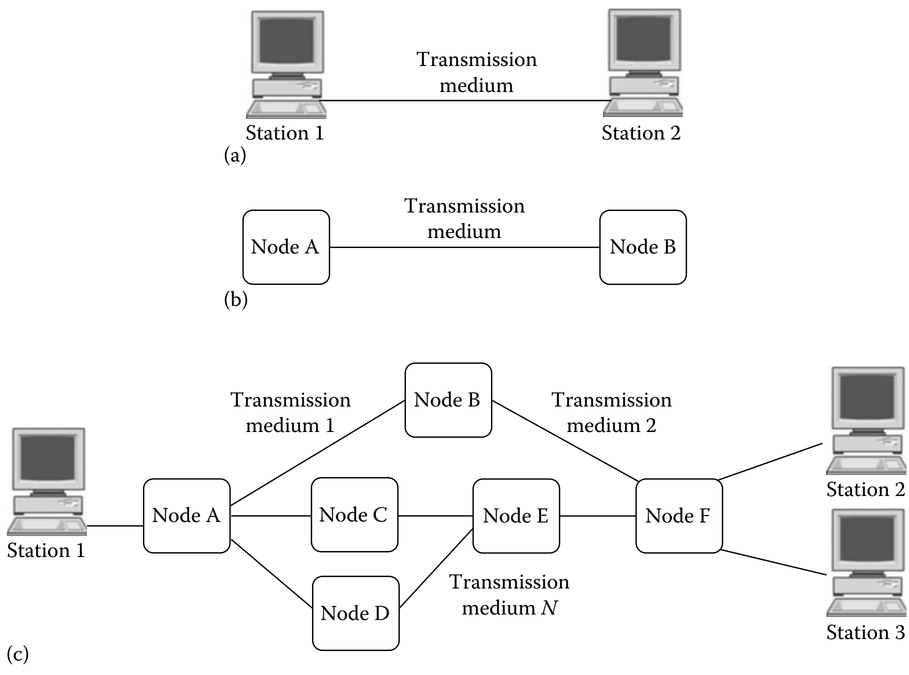

FIGURE 1.7 Examples of (a, b) a point-to-point communication and (c) a network.

exchange of data. This end-to-end path (circuit) is permanently dedicated until the connection ends. The time to establish the connection is long, but a delay is assured only because of the propagation speed of signals. This kind of switching is ideal for delay-sensitive communications, such as voice. If the connection cannot be established because of lack of resources, it is said that the call was blocked, but once established, congestion does not occur. All the bandwidth available is assigned to a certain connection that, for long time periods, may not be used and, in other periods, may not be enough (e.g., if that connection is sending variable data rates). For this reason, it is of high cost. In telephone networks, switching is physically performed by operators using PABX. This consists of a switch whose functionality is typically achieved using space and/or time switching. Space switching consists of establishing a physical shunt between one input and one output. Because digital networks normally incorporate multiplexed data into different time slots* (each telephone connection is transported in a different time slot), there is a need to switch a certain time slot from one physical input into another time slot of another physical output. This is performed by the time and space switching functionality of a digital PABX. An example of a circuit switching (telephony) network is depicted in Figure 1.8.

1.1.7.2 Packet Switching

With the introduction of data services, the notion of packet switching has arrived. Packet switching considers the segmentation of a message into parts, where each part is referred to as a *packet* (with fixed† or variable‡ length). As can be seen from Figure 1.9, a digital message is composed of many bits, while a packet consists of a small number of these bits.

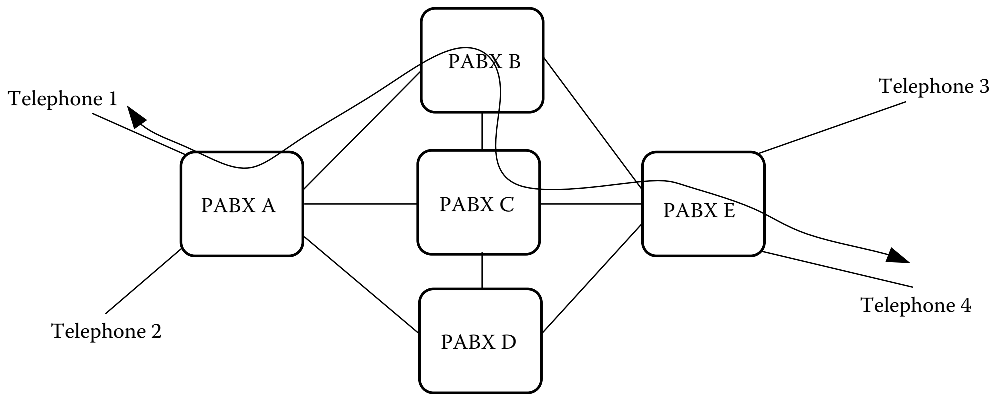

FIGURE 1.8 Example of a circuit switching (telephone) network.

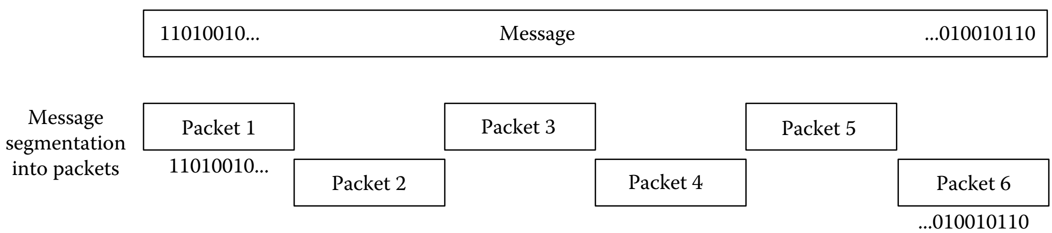

FIGURE 1.9 The segmentation of a message into packets.

* This is normally referred to as *time division multiplexing*.
† For example, asynchronous transfer mode (ATM).
‡ For example, multiprotocol label switching or Internet protocol (IP).

Packets are forwarded and switched independently through the nodes of a network, between the source and the destination. Each packet transports enough information to allow its routing (end destination address included in a header).

While the nodes of a circuit switching network establish a permanent shunt between one input and one output, because packet switching considers a number of bits grouped into a packet, the nodes of a packet switching network only switch data for the duration of a packet transmission. The following packet that uses the same input or output of a node may belong to a different end-to-end connection. This is depicted in Figure 1.10. Consequently, packet switching networks make much better usage of the network resources (nodes) than circuit switching. Note that a node of a packet switching network is typically a router.

Each node of the network is able to store packets, in case it is not possible to send it because of temporary congestion. In this case, the time for message transmission is not guaranteed, but this value is kept within reasonable limits, especially if quality of service (QoS) is offered. Packet switching is of lower costs than circuit switching, and is ideal for data transmission, because it allows a better management of the resources available (a statistical multiplexing is performed). Moreover, with packet switching, we need not assign all of the available resources (i.e., bandwidth) to a certain user who, for long periods, does not make use of them, the network resources being shared among several users, as a function of the resources available and of the users' need. The network resources are made available as a function of each user's need and as a function of the instantaneous network traffic.

There are different packet switching protocols, such as ATM, IP, frame relay, and X.25. The IP version 4 (IPv4) does not introduce the concept of QoS, because it does not include priority rules to avoid delays or jitter (e.g., for voice). Moreover, it does not avoid loss of data for certain types of services (e.g., for pure data communication), and it does not allow the assignment of higher bandwidth to certain services, relating to other (e.g., multimedia vs. voice). On the other hand, ATM and IP version 6 (IPv6) have mechanisms to improve the QoS.

1.1.8 Connection Modes

Depending on the end-to-end service provided, the connection modes through networks can be of two types: connectionless and connection oriented. These modes are used in any of the layers of a network architecture, such as in the Open System Interconnection reference model, or in the transmission control protocol/IP (TCP/IP) stack.

1.1.8.1 Connection-Oriented Service

In order to provide a connection-oriented service, there is a need to previously establish a connection before data is exchanged, and to terminate it after data exchange. The connection is established

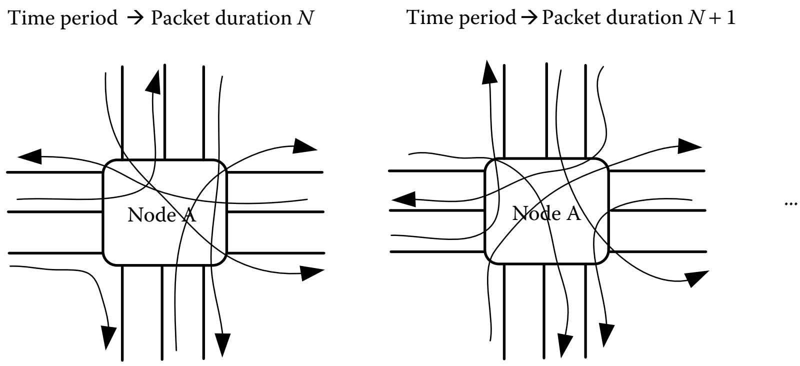

FIGURE 1.10 Switching of packets in different instants.

between entities, incorporating the negotiation of the QoS and cost parameters of the service being provided. The communication is bidirectional, and the data is delivered with reliability. Moreover, in order to prevent a faster transmitter to overload a slower receiver, flow control is employed (to prevent overflow situations). An example of a connection-oriented service is the telephone network, where a connection is previously established before voice exchange. In the telephone network, taking, as a reference, two words transmitted one after the other, we do not experience an inversion of the correct sequence of these words (e.g., receiving the second word before the first one). The TCP of the TCP/IP stack is an example of a connection-oriented protocol.

A connection-oriented service is always confirmed,* as the transmitter has information about whether or not the data reached the receiver free of errors, correcting the situation in case of errors. This can be performed using positive confirmation, such as the positive acknowledgment with retransmission (PAR) procedure, or using negative confirmation, such as the negative acknowledgment (NAK).

In the PAR case, when the transmitter sends a block of data, it initiates a chronometer and expects for the correct reception of an acknowledgment (ACK) message from the receiver within a certain time frame. In case the ACK message is not received in time, the transmitter assumes that the message was received corrupted and performs the retransmission of the block of data. In case the ACK message is received, the transmitter proceeds with the remaining transmission of data. The advantage of this procedure is that the ACK message sent by the receiver to the transmitter allows two confirmations: (1) the data was properly received (error control) and (2) the receiver is ready to receive more data (flow control).

In the case of the NAK, the receiver only sends a message in case the data is received with errors; otherwise, the receiver does not send any feedback to the transmitter. The advantage is the lower amount of data exchanged. The disadvantage is that in the PAR case, flow control is performed together with error control, whereas in the NAK situation, only error control is performed.

The reader should refer to Chapter 2 for a detailed description of the service primitives used in connection-oriented services.

1.1.8.2 Connectionless

The connectionless mode does not perform the previous establishment of the connection, before data is exchanged. Therefore, data is directly sent, without prior connection establishment.

As the connection-oriented mode requires a handshaking between the transmitter and the receiver,† this introduces delays in signals. Consequently, for services that are delay sensitive, the connectionless mode is normally employed. The connectionless mode is also utilized in scenarios where the experienced error probability is reduced (such as in the transmission of bits in an optical fiber).

Depending on whether the service is confirmed or not, data reliability may or may not be assured. Even though if data reliability is not assured, such functionality can be provided by an upper layer of a multilayer network architecture. In such a scenario, there is no need to execute the same functionalities twice.

The connectionless mode can provide two different types of services:

- Confirmed service
- Nonconfirmed service

In the case of the nonconfirmed service, the transmitter does not have any feedback about whether or not the data reached the receiver free of errors. Contrarily, in the case of the confirmed service, although a connection establishment is not required before the data is exchanged (as in the case of

* On the other hand, the connectionless service can be confirmed or nonconfirmed.

† For example, implementing data retransmission, in order to assure data reliability.

the connection-oriented service), the transmitter has feedback from the receiver about whether or not the data reached the receiver free of errors. The reader should refer to the description of the confirmation methods used in confirmed services presented for the connection-oriented service, namely the PAR and NAK.

As an example, Internet telephony (IP telephony) is normally supported by the nonconfirmed service, specifically, by the user datagram protocol (UDP), which is connectionless. However, in IP telephony, the reordering of packets is performed by the application layer.* Another example of a nonconfirmed connectionless mode is the IPv4 protocol, which does not provide reliability to the delivered datagrams and which does not require the previous establishment of the connection before data is sent. In case such reliability is required, the TCP is utilized as an upper layer (instead of the UDP). The serial line IP is an example of a data link layer protocol that is nonconfirmed and connectionless.

The reader should refer to Chapter 2 for a detailed description of the service primitives used in confirmed and nonconfirmed connectionless services.

1.1.9 Network Coverage Areas

Packet switching networks may also be classified as a function of the coverage area. Three important areas of coverage exist: LANs, MANs, and WANs.

A LAN consists of a network that covers a reduced area such as a home, office, or small group of buildings (e.g., an airport), using high-speed data rates. A MAN consists of a backbone (transport network) used to interconnect different LANs within a coverage area of a city, a campus, or similar. This backbone is typically implemented using high-speed data rates. Finally, a WAN consists of a transport network (backbone) used to interconnect different LANs and MANs, whose area of coverage typically goes beyond 100 km. While the transmission medium used in a LAN is normally the twisted pair, optical fiber, or wireless, the optical fiber is among the most used transmission medium in a MAN and WAN.

1.1.10 Network Topologies

A network topology is the arrangement of the devices within a network. The topology concept is applicable to a LAN, a MAN, or a WAN. In the case of a LAN, such a topology refers to the way hosts and servers are linked together, while in the MAN and WAN cases, this refers to the way nodes (routers) are linked together. For the sake of simplicity, this description refers to hosts and servers (in the case of LAN) and nodes (in the case of MAN and WAN) just as hosts.†

A bus topology is the topology where all hosts are connected to a common and shared transmission medium. This topology is depicted in Figure 1.11. In this case, the signals are transmitted to all hosts

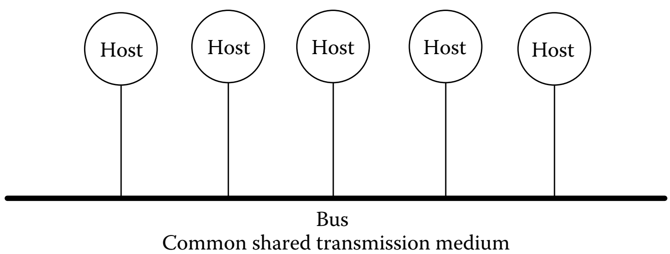

FIGURE 1.11 Bus topology.

* These functions are carried out by the real-time protocol (RTP).

† In fact, both workstations and routers are hosts.

and, because the host's network interface cards (NIC) are permanently listening to the transmitted data, they detect whether or not they are the destination of the data. In case the response is positive, the NIC passes the data to the host; otherwise, the data is discarded [Monica 1998]. This topology presents the advantage that, even though if a host fails, the rest of the network keeps running without problems. The main disadvantage of this topology relies on the high overload of the whole network (including all network hosts) that results from the fact that all data is sent to all network hosts.

In a ring topology, the cabling is common to all the hosts, but the hosts are connected in serial. This topology is depicted in Figure 1.12. Each host acts as a repeater: each host retransmits in a termination, and the data received in the other termination. The main disadvantage of this topology is that if a host fails, the rest of the network is placed out of order. This topology is normally utilized in SDH networks (MAN and WAN), where double rings are normally utilized to improve redundancy. The token ring technology used in LAN is also based on the ring topology.

A star topology includes a central node connected to all other hosts. The central node repeats or switches the data from one host into one or more of the other hosts. Because all data flows through this node, this represents a single point of failure. This topology is depicted in Figure 1.13.

A tree topology is a variation of the star topology. In fact, the tree topology consists of a star topology with several hierarchies. This topology is depicted in Figure 1.14. The central node is responsible for repeating or switching the data to the hosts within each hierarchy. In case the destination of the data received by a certain central node refers to another hierarchy, such central node forwards the data to the corresponding hierarchy central node, which is then responsible for forwarding the data to the destination host.

Finally, in a mesh topology each host is connected to all* or part† of the other hosts in the network. This topology is depicted in Figure 1.15. The advantage of such configuration relies on the existence of many alternative pathways for the data transmission. Even though if one or more paths are interrupted or overloaded, the remaining redundancies represent alternative paths for the data transmission. The drawback of such a topology is that the large amount of cabling is necessary to implement it.

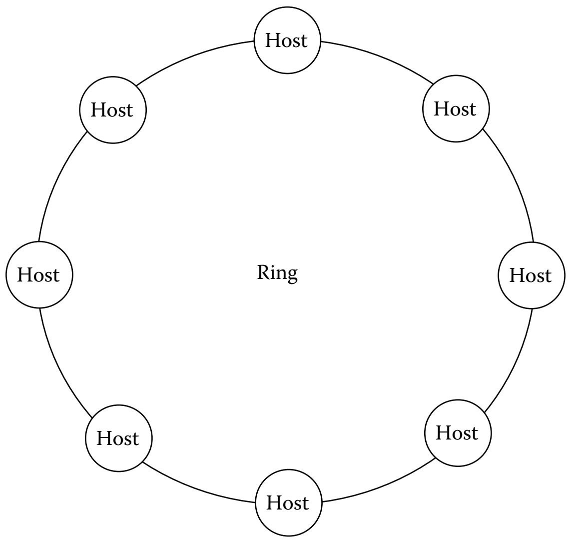

FIGURE 1.12 Ring topology.

* Complete mesh topology.
† Incomplete mesh topology.

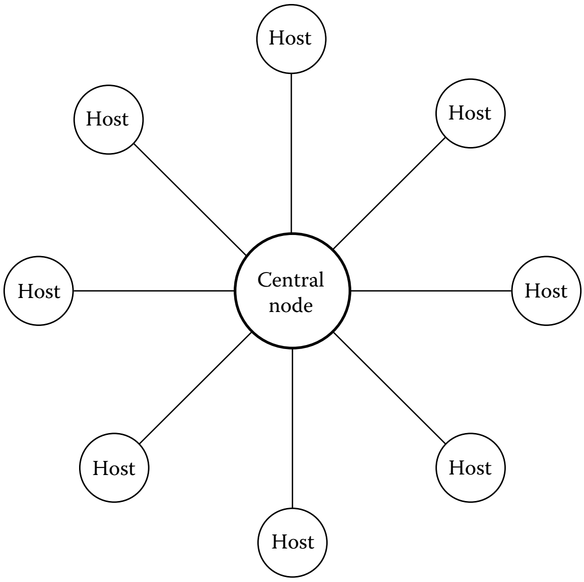

FIGURE 1.13 Star topology.

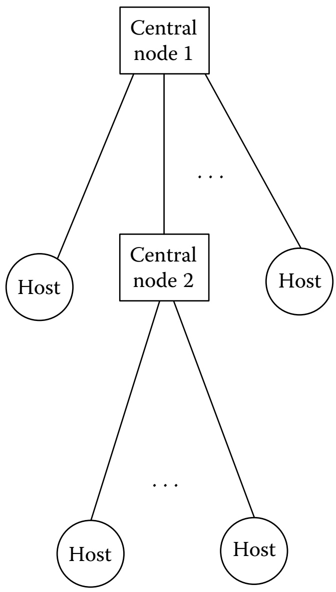

FIGURE 1.14 Tree topology.

It is worth noting that there are two different types of topologies: physical topology and logical topology. The physical topology refers to the real cabling distribution along the network, while the logical topology refers to the way the data is exchanged in the network. A physical star topology with a repeater (a hub)* as a central node presents a medium common and shared by all network hosts. In such a case, the logical topology is the bus topology (common and shared medium).

* A hub/repeater repeats in all other outputs the bits received in one input. In addition, it acts as a regenerator.

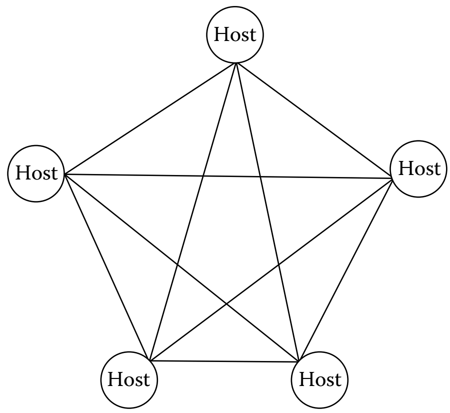

FIGURE 1.15 Mesh topology.

On the other hand, a physical star topology with a switch* as a central node corresponds, as well, to a logical star topology. Moreover, a logical ring topology corresponds to a physical star topology with a central node that rigidly switches the data to the adjacent host (left or right).

1.1.11 Classification of Media and Traffic

Different media can be split into three groups [Khanvilkar et al. 2005]:

- *Text*: Plaintext, hypertext, ciphered text, and so on.
- *Visuals*: Images, cartography, videos, videoteleconference (VTC), graphs, and so on.
- *Sounds*: Music, speech, other sounds, and so on.

While the text is inherently digital data (mostly represented using a string of 7-bit ASCII characters), the visuals and sounds are typically analog signals, which need to be digitized first, in order to allow its transmission through a digital network, such as an IP-based network (e.g., the Internet or an intranet). As can be seen from Figure 1.16, the multimedia is simply the mixture of different types of media, such as speech, music, images, text, graphs, and videos.

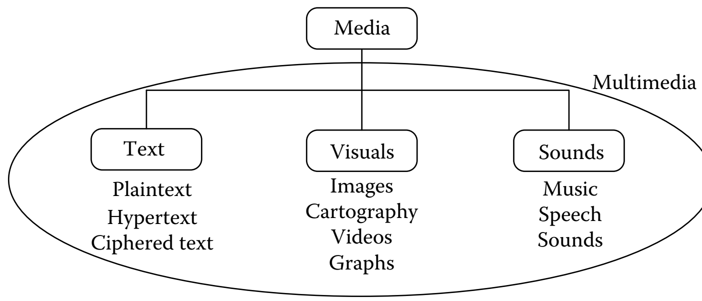

FIGURE 1.16 Basic types of media.

* A switch only switches data to the output where the destination host is located. This is performed based on the address of the destination.

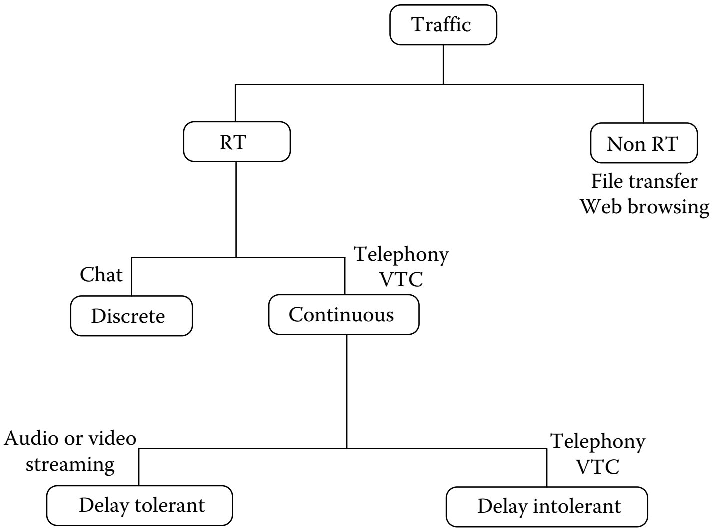

FIGURE 1.17 Classification of traffic.

When media sources are being exchanged through a network, it is generically referred to as *traffic*. As depicted in Figure 1.17, the traffic can be considered as real time (RT) or non-real time (NRT). While RT traffic is delay sensitive, NRT media is not. An example of RT traffic is telephony or VTC, whereas a file transfer or the web browsing can be viewed as NRT traffic.

RT traffic can also be classified as continuous or discrete. Continuous RT traffic consists of a stream of elementary messages with interdependency. An example of continuous RT traffic is telephony, whereas the chat is an example of discrete RT traffic.

Finally, RT continuous traffic can still be classified as delay tolerant or delay intolerant. RT continuous delay-tolerant traffic can accommodate a certain level of delay in signals, without sudden performance degradation. Such tolerance to delays results from the use of a buffer that stores in memory the difference between the received data and the played data. In case the transfer of data is suddenly delayed, the buffer accommodates such delay, and the media presented to the user does not translate such delay introduced by the network. Video streaming is an example of a delay-tolerant media. Contrarily, the performance of delay-intolerant traffic degrades heavily when the data transfer is subject to delays (or variation of delays). An example of RT continuous and delay-intolerant media is telephony or VTC. IP telephony or VTC allows a typical maximum delay of 200 ms, in order to achieve an acceptable performance.

1.2 PRESENT AND THE FUTURE OF TELECOMMUNICATIONS

Current and emergent communication systems tend be IP based and are meant to provide acceptable QoS in terms of speed, BER, end-to-end packet loss, jitter, and delays for different types of traffic.

Many technological achievements have been made in the last few years in the area of communications and others are planned for the future to allow the new and emergent services. However, whereas in the past new technologies pushed new services, nowadays the reality is the opposite: end users want services to be employed on a day-by-day basis, whatever the technology that supports it. Users want to browse over the Internet, get e-mail access or use the chat, establish a VTC, regardless of the technology used (e.g., fixed or mobile communications). Thus, services must be delivered following the concept of *anywhere* and at *anytime*. Figure 1.18 presents the bandwidth requirements for different services.

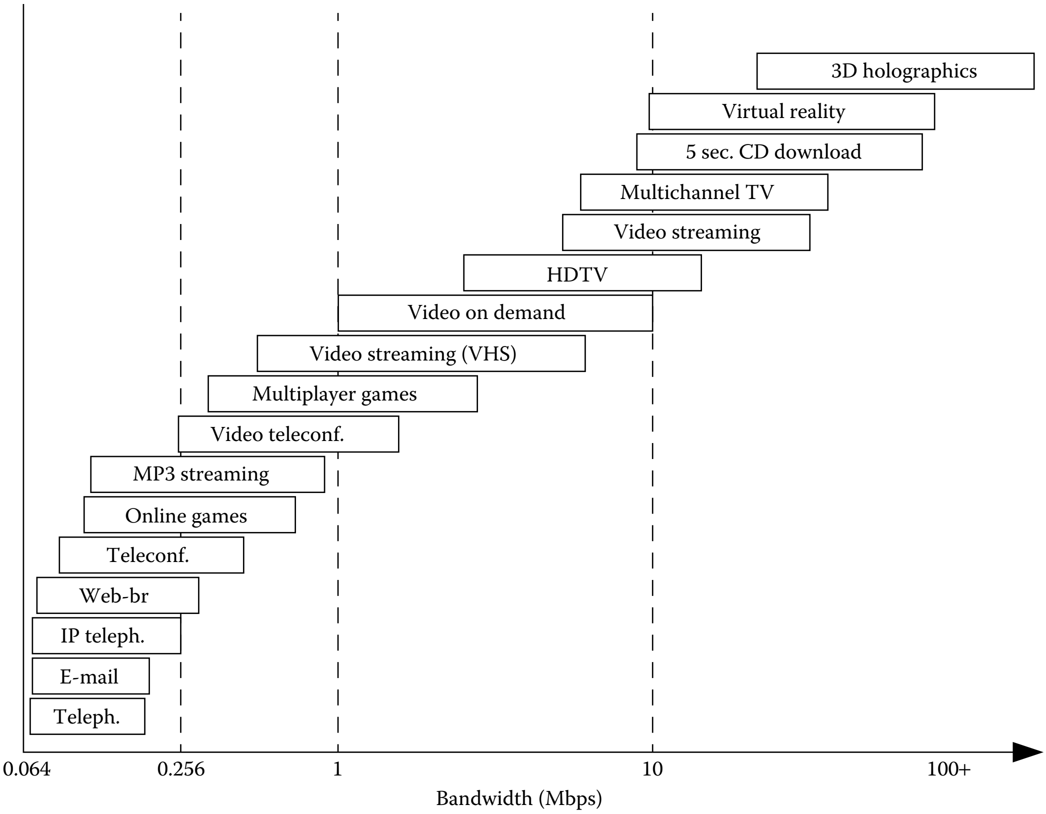

FIGURE 1.18 Bandwidth requirements of the different services.

1.2.1 Convergence

The main objective of the telecommunications industry is to create conditions to make the convergence a reality [Raj et al. 2010]. The convergence of telecommunications can be viewed in different ways. It can be viewed as the convergence of services, that is, the creation of a network able to support different types of service, such as voice, data (e-mail, web browsing, database access, files transfer, etc.), and multimedia, in an almost transparent way to the user [Raj et al. 2010].

The convergence can also be viewed as the complement between telecommunications, information systems, and multimedia, in a way to achieve a unique objective: make the information available to the user with reliability, speed, efficiency, and at a low price. According to the Gilder law, the speed of telecommunications will increase three times every year in the next 20 years, and according to the Moore law, the speed of microprocessors will duplicate every 18 months.

The convergence can be viewed as the integration of different networks in a single one, in a transparent way to the user. It can also be viewed as the convergence between fixed and mobile concepts [Raj et al. 2010], as the mobile is covering indoor environments (e.g., femtocells* of long-term evolution [LTE]) allowing data and television/multimedia services, traditionally provided by fixed services, whereas fixed telecommunications are giving mobility with the cordless systems, whose example is the Digital European Cordless Telephone standard. There are terminals that are able to

* A femtocell is a cellular base station for use at home or in offices that creates an indoor cell, in locations where cellular coverage is deficient, inexistent, or to provide high-speed data services. It typically interconnects with the service provider via broadband xDSL or cable modem.

operate as cellular phones or as fixed network terminals. New televisions not only receive the TV broadcast but also allow browsing over the Internet.

The convergence is viewed by many people as the convergence of all the convergences, which will lead to a deeply different society, whose results can already be observed nowadays with the use of the following services:

- Telework
- Telemedicine
- Web-TV
- E-Banking
- E-Business
- Remote control over houses, cars, offices, machines, and so on
- VTC

Human lives, organizations, and companies will tend to increase their efficiency, with the new communication means, and with the increase of the available information, as well as with multicontact.

With technological evolutions—increase of user data rates, improved spectral efficiency, better performances (lower BER), increase of network capacity, and decrease of latency (RT communications)—and with the massification of telecommunications as a result of lower prices (as a result of technological evolution and the increase of competition), it is expected that virtual reality and 3D holographic will be a reality in the near future.

1.2.2 Collaborative Age of the Network Applications

While the convergence approach was based on the ability to allow information sharing using a common network infrastructure, the new approach consists of the use of the network as an enabler to allow sharing of knowledge. It consists of the ability to provide the right information to the right person at the right time. For this to be possible, a high level of interactivity made available to each Internet user is required. In parallel, business intelligence is an important platform that allows decision makers to receive the filtered information* required for the decision to be made in a correct moment. The concept of Internet of Things enables the knowledge by making available a large amount of data captured by multiple machines and sensors, and by enabling machine-to-machine communications. Moreover, to enable the sharing of knowledge, there is a need to complement the Internet of Things with the processes and applications. This is required to process the data captured by sensors and machines.†

We observe, nowadays, an explosion of ad hoc applications that allow any Internet user to inject nonstructured information (e.g., Wikipedia) into the Internet world, in parallel with an increase of mobile-cloud and peer-to-peer applications such as Torrent, eMule, and IP telephony. Social networks are currently being used by millions of people that allow the exchange of unmanaged multimedia by groups of people just to share information or by groups interested in the same subject. Note that this multimedia exchange can be text, audio, video, multiplayer games, and so on. This can only be possible with the ability of the IP to support all types of services in parallel with the provision of QoS by the network, that is, with the convergence as a support platform. This is the new paradigm of the modern society: the collaborative age. The collaborative age of the Internet can also be viewed as the transformation of man-to-man communication into man-to-machine and machine-to-machine communication, using several media, and where the source or destination party can be a group instead of a single entity (person or equipment).

Figure 1.19 shows the evolution of the network usage. Initially, this was viewed merely for the data applications. Afterward, as referred to in Section 1.2.1, convergence was an important issue to allow

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* For example, key performance indicators.

† This also presents a relationship with big data analysis.

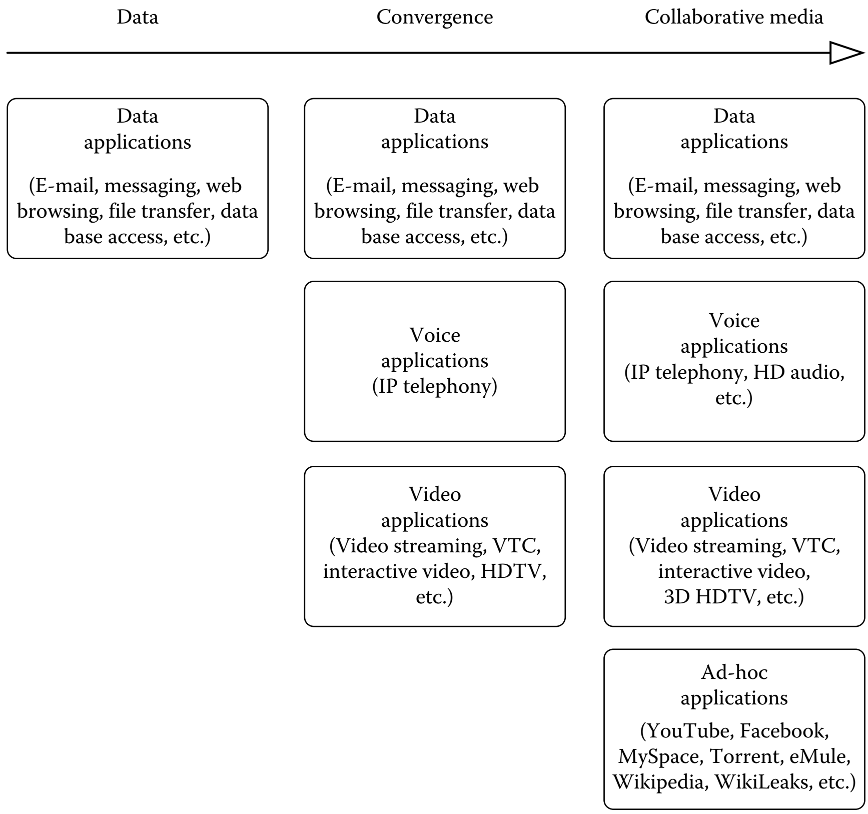

FIGURE 1.19 Evolution of network applications: from data to collaborative tool.

a better usage of the network. An increase in the level of Internet users interactivity made the Internet world a space for deep collaboration between entities, but with a higher level of danger as well.

1.2.3 Transition toward the Collaborative Age

To reach demands of the modern society, in terms of both convergence and collaborative services, several problems need to be solved from the scientific and industrial community. From Section 1.2.2, we may conclude that the convergence can be viewed as an important requirement to support the collaborative services.

Although we observe an enormous demand for convergence, we see that there are still problems that need to be solved. An example is the universal mobile telecommunication system (UMTS), which still treats voice and data in different ways, as data is IP based whereas voice is still circuit switching based. The LTE is the cellular standard that deals with this issue and makes the all-over-IP a reality.

From the point of view of services, the total digitalization of several information sources and the use of efficient encoding and compressing data algorithms are very important. The information sources can be voice, fax, images, music, videoconference, e-mail, web browsing, positioning systems, high-definition television, and pure data transmission (database access, file transfer, etc.). Different services need different transmission rates, different margin of latencies and jitter, different performances, or even fixed or variable transmission rates. Several MPEG protocols for voice or video, those already existent and those that are still in the research and development phase, intend to perform an adaptation of several information sources to the transmission media, allowing a reduction of the number of encoded bits to be transmitted.

Different services present different QoS requirements, namely:

- Voice communications are delay sensitive, but are low sensitive to loss of data, and require low data rate but approximately constant.
- Iterative multimedia communications (e.g., web browsing) are sensitive to loss of data, requiring considerable data rate, with a variable transmission rate, and are moderately delay sensitive.
- Pure data communications (e.g., database access and file transfer) are highly sensitive to loss of data, requiring relatively variable data rate, without sensitivity to delay.

Jitter is defined as the delay variation through the network. Depending on the application, jitter can be a problem, or jitter issues can be disregarded. For instance, data applications that only deliver their information to the user if the data is completely received (reassembling of data) pay no attention to the jitter issues (e.g., file transfer). This is totally different if voice and video applications are considered; those applications degrade immediately if jitter occurs.

The transmission of data services (e.g., pure data communications and web browsing) through most of the reliable mediums (e.g., optical fiber and twisted pair) usually considers error detection algorithms jointly with automatic repeat request* (ARQ), instead of error correction (e.g., block coding or forward error correction). This happens because these services present very rigid requirements in terms of BER, whereas not very demanding in terms of delay sensitivity (in this case, stopping the transmission and requesting for repetitions are not crucial). Note that the utilization of error correction requires more redundant bits per frame than error detection (amount of additional data beyond the pure information data). This can be seen from Figure 1.20.

Nevertheless, the transmission of data services through a nonreliable medium (e.g., wireless) is normally carried out using error correction, as the number of repetitions would be tremendous, creating much more overhead (and corresponding reduction of performance due to successive repetitions) than the overhead necessary to encode the information data with error correction techniques. A similar principle is applied to services that are delay sensitive (voice), where, to reduce latency, error correction is normally a better choice, instead of error detection.

These are the notions that introduce the QoS concept, implying that each service will impose certain requirements. For the convergence to become a reality, the network should be able to take all these requirements into account.

Taking into account all the previously described factors, one that presents a great contribution to support the new collaborative services is the maximum transmission rate, as it is associated with the user data rate. The factors that limit the use of higher transmission rates are several sources of interference and noise. The effects of noise can be minimized through the use of regenerators, as

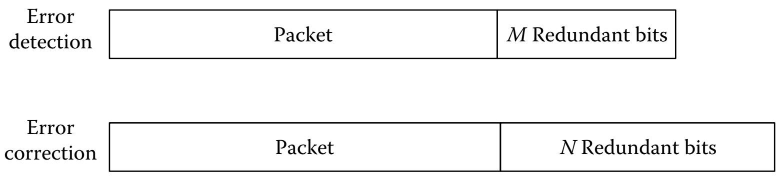

FIGURE 1.20 Types of error control: the error detection and error correction. Their differences in terms of the amount of redundant bits ($N > M$).

* ARQ works associated with error detection. The transmitter sends groups of bits (known as frames), which are subject to an encoding in the transmitter. The decoding process performed in the receiver allows this station to gain knowledge about whether or not there was an error in the propagation of the frame. In the case of error, the receiver requests a repetition of the frame from the transmitter.

well as advanced detection algorithms (e.g., matched filters). Interferences tend to increase with the increase in the used bandwidth (which corresponds to an increase of transmission rates), this being the main limitation of the use of higher data rates.

The challenge facing the today's telecommunications industry is how to continually improve the end-user experience, to offer appealing services through a delivery mechanism that offers improved speed, service attractiveness, and service interaction. In order to deliver the required services to the users with the minimum cost, the technology should allow better and better performances, higher throughputs, improved capacities, and higher spectral efficiencies.

What can be done in order to increase the throughput of a wireless communication system? One can choose a shorter symbol duration T_S. This, however, implies that a larger fraction of the frequency spectrum will be occupied, because the bandwidth required by a system is determined by the baud rate $1/T_S$. Wireless channels are normally characterized by multipath propagation caused by reflections, scattering, and diffraction in the environment. The shorter symbol duration might therefore cause an increased degree of intersymbol interference (ISI) and thus performance loss. As an alternative to the shorter symbol duration, one may choose using a multicarrier approach, multiplexing data into multiple narrow subbands, as adopted by orthogonal frequency division multiplexing (OFDM) [Marques da Silva et al. 2010]. The OFDM technique has been selected for LTE, as opposed to wideband code division multiple access that is the air interface technique that has been selected by European Telecommunications Standard Institute for UMTS. Thus, the problem of ISI can be mitigated. But still, the requirement for increased bandwidth remains, which is crucial with regard to the fact that the frequency spectrum has become a valuable resource. This imposes the need to find schemes able to reach improved spectral efficiencies, such as higher order modulation schemes, the use of multiple antennas at transmitter and at receiver such as multiple input multiple output systems, more efficient error control, and so on [Marques da Silva et al. 2010].

CHAPTER SUMMARY

This chapter provided an introduction to multimedia communications and networking, including the study of most important fundamentals of communications and future trends.

It was described that digital signals allow regeneration, multiplexing, and error control, functionalities not possible when analog signals are employed. Nevertheless, it was viewed that digital signals tend to require a higher bandwidth than the analog counterpart.

It was also viewed that the modem is employed when the transmission medium is analog, whereas the digital encoder (also referred to as the *line encoder*) is employed with digital transmission mediums. Moreover, the modem sends carrier modulated signals, that is, signals modulated around a certain carrier, whereas the digital encoder sends baseband signals, that is, signals modulated around a null frequency.

It was shown that transmission mediums can be cable or wireless. In the latter case, the difference between guided and unguided wireless transmission mediums was described. Among the cable transmission mediums, the optical fiber is the most resistant to interferences, and supports the higher bandwidth. Moreover, single mode optical fibers support higher bandwidths than multimode optical fibers.

We have viewed that synchronous communications allow higher data rates than asynchronous communication systems. Synchronous communications extract the synchronism reference from the received signal, or using an additional transmission pair or channel. Contrarily, asynchronous communication systems need to periodically use start and stop bits for allowing the receiver to determine the bit transition instants.

It was shown that simplex communications send signals only in a single direction, whereas duplex communications allow bidirectional communications. In the case of full duplex, two channels are required to allow simultaneous bidirectional communications.

It was described that a network is composed of a concatenation of point-to-point links, which can be of different types. In this case, intermediate nodes are responsible for linking the required sequence point-to-point links.

We have also viewed that circuit switching uses all assigned resources during the connection, whereas packet switching allows a more efficient use of the network resources. In case of packet switching, this can be of two modes: connection oriented or connectionless. The connection-oriented mode provides flow control and error control, which allows the service being confirmed. It was described that the connectionless mode can provide a confirmed service, or a nonconfirmed service.

The difference between a LAN, a MAN, and a WAN was described. The LAN is used within an office, or a house. A MAN is used to cover typically a city, or a university campus, being used to interconnect different LANs. Finally, a WAN corresponds to a network that typically covers a wide territory, such as a country, being also employed to interconnect different LANs.

The logical topology corresponds to the way data is interchanged, whereas the physical topology corresponds to the way network devices are physically interconnected. Bus, star, ring, tree, and mesh are examples of topologies that can be employed.

It was described that the traffic consists of the exchange of media sources through a network, or through a communication system. Moreover, media can be text, visual, or sounds. In addition, traffic can be RT, or NRT, discrete, or continuous. In the case of RT, and continuous traffic, this can be delay tolerant, or delay intolerant.

REVIEW QUESTIONS

1. What are the advantages of using digital communications relating to analog communications? What are the disadvantages?
2. What are the reasons that may imply the use of a modem?
3. What is the difference between simplex, half-duplex, and full-duplex communication?
4. What is the physical topology used to implement a logical bus? In such a case, what is the central node?
5. What is the difference between unicast, multicast, and broadcast communication?
6. What is the difference between an analog and a digital signal?
7. What is the difference between a LAN, MAN, and WAN?
8. What is the difference between a connectionless and connection-oriented service?
9. What is the difference between a point-to-point communication and a network?
10. What is the difference between a circuit switching and a packet switching network? Give examples of networks based on these two switching types.
11. What are the most important QoS requirements?
12. What is the convergence of telecommunications?
13. What is the difference between physical topology and logical topology?
14. Which types of media do you know?
15. How can the different types of traffic be grouped?
16. What is the collaborative age of the telecommunications?

LAB EXERCISES

1. Using the Emona Telecoms Trainer 101 laboratory equipment, and volume 1 of its laboratory manual, perform experiment 1—Setting up an oscilloscope.
2. Using the Emona Telecoms Trainer 101 laboratory equipment, and volume 1 of its laboratory manual, perform experiment 2—An introduction to Telecoms Trainer 101.