TELECOMMUNICATION TERMINALS

In the past, the only function for telecommunication terminals was to provide the user with access to a telecommunications network. Later, many other functions were added to the terminals that were designed to simplify the use of the telecommunications equipment. In the case of the telephone, the old telephone terminals were only providing access to the public switched telephone network (PSTN) while today's telephone terminals provide other services, such as personal directory and answering services. The old text-oriented data terminals that were used to enter the data are now mixed with word processors that were not originally designed for telecommunication applications. By equipping the word processor with a communications interface, however, the differences between telecommunications text terminals and communicating word processors disappear, and we obtain a multifunctional workstation. The old PSTN with analog circuit switches is now replaced by a more complex network using integrated services digital network (ISDN) and asynchronous transfer mode (ATM) switches, thus introducing new services and associated telecommunication terminals. Access to the old PSTN that was once only through copper wires is now complemented with variety of wireless services, opening a new horizon for introducing variety of new terminals. The data services that were using the PSTN through modems are not complemented by long-haul and local packet switched data networks demanding for new terminals to connect to these networks. This article provides an overview of a large class of telecommunication terminals. It defines the meaning of the telecommunication terminal and categorizes them. Based on the terminal definition as an endpoint to offer telecommunication services to end users with wired or wireless interfaces, the article discusses different telecommunication terminals, services, standards, and protocols for wired terminals; it also investigates different wireless terminals. Telecommunication terminals for different wired and wireless access technologies are considered based on their applications in voice, data, and multimedia.

WHAT IS A TELECOMMUNICATION TERMINAL?

Telecommunication is any process that permits the passage from a sender to one or more receivers of information of any nature delivered in any usable form (printed copy, fixed or moving pictures, visible or audible signals, etc.) by means of any electromagnetic system (electrical transmission by wire, radio, optical transmission, guided waves, etc.). The terminal is a point at which information can enter or leave a communication network, or any device capable of sending and/or receiving information over a communication channel.

A telecommunication terminal is a device that can provide telecommunication services used by end users. The term *service* has a very specific meaning in International Telecommunication Union-Telecommunication (ITU-T) standards that is mainly characterized by complete, guaranteed end-to-end compatibility and ITU-T standardized terminals.

Although there is no widely accepted and quoted definition of the term *standard,* it is accepted in the telecommunication industry that standards are required to determine the physical, electrical, and procedural characteristics of communication equipment. The following definition from the 1979 National Policy on Standards for the United States was offered (1): a prescribed set of rules, conditions, or requirements concerning definition of terms; classification of components; specification of materials, performance, or operation; delineation of procedures; or measurement of quantity and quality in describing materials, product, system, services, or practices (2).

Terminal equipment connected by a user shall have a demarcation point between the terminal equipment and the telecommunication facilities used for telecommunications business. The connection method at the demarcation point shall enable easy detachment of the facilities. Table 1 shows some examples of telecommunications terminals that are used domestically or internationally.

Table 1. Telecommunication Terminals Examples

Packet switching terminals
Circuit switching terminals
Telex terminals
Facsimile communications user terminals
Facsimile communication center terminals
Videotex communications user terminals
Videotex communications center terminals
Video conference terminals
Gathering telephone terminals
Four-wire telephone circuit terminals
Nonringing circuit terminals
Signal supervision communications service terminals
Airport radio phone terminals
Convenience radio phone terminals
Teleterminal communications terminals
Maritime mobile telephone terminals
Satellite mobile telephone terminals
Airplane telephone terminals
Marine telephone terminals
Leased circuit terminals
Frame relaying terminals
Cell relaying terminals
Acoustic couplers
Integrated digital telecommunications terminals

Traditional Telecommunication Terminals

There are three fully standardized ITU-T services: telegraphy, telephony, and data. There are four other ITU-T services under the name *telematic* in the process of standardization: teletex, public facsimile, videotex, and message handling service. The goal with all of these services is to ensure high-quality international telecommunications for the end user regardless of the make of the terminal equipment and the type of network to support the service. In the rest of this section we provide a short description of telephony, data, teletex, telex, facsimile, and videotex terminals and their relation to other services.

Telephone Terminals. The invention of the electromagnetic selector in 1896 made telephone service automation possible. Telephone service is now almost completely automated, and a large part of the world can be reached without operator intervention. The transmission structure of the telephone network assisted some other services, such as telegraph and facsimile. ITU-T Recommendation P.342 provides audio performance requirements for telephony terminals in the telephone band (300 Hz to 3400 Hz). The waveform encoding for telephone terminals is specified by Recommendations G.711 for the Pulse Code Modulation (PCM) at both 64 kbit/s and 56 kbit/s and G.726 for adaptive differential pulse code modulation (ADPCM) 32 kbit/s. It gives recommended values or masks for the following parameters: loudness ratings, sensitivity frequency response, harmonic distortion, out-of-band signals, terminal coupling loss, stability loss, and delay. The electrical interface specifications are referred to in Recommendation P.310. ITU-T Recommendation P.310 provides audio performance requirements and associated testing for telephone band (300 Hz to 3400 Hz) digital telephones. This recommendation is only applicable to digital telephones using encoding conforming to Recommendations G.711 (64 kbit/s, PCM) and G.726 (32 kbit/s, ADPCM). The telephone handset has gone through major changes to provide better quality of service (QoS) for users. ITU-T Recommendation P.35 standardizes handset telephones, and P.311 provides preliminary audio performance requirements for wideband audio (7 kHz) handset telephones (3). Requirements and test methods are specified for the major audio transmission parameters affecting wideband audio, including levels, frequency response, noise, distortion, spurious signals, sidetone, echo path, and delay. Wideband audio represents a considerable departure from traditional telephony, offering significantly improved quality. More description of the visual telephone systems described by the H.322 standard is provided later in the section on multimedia terminals.

Data Terminals. Consultative committee on international telephone and telegraphy (CCITT) Recommendation X.25 describes the interface and access protocols for data terminal equipment (DTE) operating in packet mode connected by a data circuit terminating equipment (DCE) to a public data network (PDN). Although the X.25 interface is mainly used by computer systems, display terminals and personal computers are usually connected to X.25 indirectly by a packet assembly/disassembly (PAD) facility which translates the simple terminal protocol to the more complicated X.25 protocols. In reference to the Open system interconnection (OSI) model, X.25 defines three layers: physical layer, data link layer, and packet layer. The service offered is compliant with the OSI network service and permits the transmission of data without loss and duplication. The access speeds and the throughputs may go up to 2 Mbit/s. The procedure to build up a connection or virtual call begins from the terminal (DTE) by transmitting a call request packet, and the called terminal answers with a call accepted packet. A permanent virtual circuit does not require any call buildup, and packets specific to the virtual call are not used by the DTE. In X.25 it is possible to define a maximum of 4000 virtual circuits on one physical layer. X.25 also supports a flow control mechanism. ITU-T Recommendation X.48 specifies the procedures for a DTE operating in accordance with Recommendation X.25 to participate in a multicast data transmission service described in Recommendation X.6. The overriding goal of this Recommendation is to make no changes to the DTE that wishes to participate in this service (4).

Teletex and Telex. Teletex is an international service that enables subscribers to exchange correspondence between two teletex terminals. These terminals operate on the basis of the transfer of data from the memory of the transmitting terminal, via a telecommunication network, to the memory of the receiving terminal. Teletex terminals use the teletex code (CCITT Recommendation T.61), which is similar to the CCITT IA5 8-bit code. The latter is also referred to unofficially by the name American standard code for information interchange (ASCII). Teletex uses communication protocols that conforms to the international organization for standardization (ISO) model, in which the higher-protocol layers are independent of the lower-protocol layers. In a number of countries teletex is being promoted as the successor to telex. While the functionality of the two devices are similar, they are not interoperable and a converter is required. One of the differences in functionality is that telex transfers the message content without placing any format on the text, while teletex does impose certain constraints on the format. The number of available letters, digits, and symbols in teletex is also much larger than in telex. Transmitting diagrams with teletex is possible, while telex does not support diagrams. The mixed mode terminals that support both text and diagrams have been specified by the CCITT. These terminals operate at 2400 bit/s (5).

Facsimile. A facsimile terminal consists of an optical scanner for digitizing images on paper, a printer for printing incoming facsimile messages, and a telephone for making the connection. The optical scanner generally does not offer the same quality of resolution as stand-alone scanners. Some printers on facsimile machines are thermal, which means they require a special kind of paper, while more modern facsimile machines use standard papers. Several standards for the facsimile terminals have been defined by CCITT. Currently, the Group 3 and Group 4 terminals are being used on a very large scale worldwide. The Group 3 facsimile terminals operate over the PSTN. The Group 3 protocol specifies CCITT T.4 data compression and a maximum transmission rate of 9600 Bd. There are two levels of resolution for Group 3 facsimile terminals: 203 by 98 and 203 by 196. The Group 4 terminals have been designed for use on data networks or ISDN. In this group three classes of terminals have been defined. Class I terminals can send/receive facsimile encoded documents. Class II, in addition to Class I, services can also receive teletex encoded documents. Class III is a superset terminal that can transmit teletex documents besides the Class II terminal services. This class can support documents with text and diagrams or mixed mode documents. Group 4 terminals operates on 64 kbit/s channels. CCITT has also specified a store-and-forward facsimile switching service called comfax. This service is intended to provide interoperability between different types of facsimile terminals. Comfax also accepts the input from text terminals and converts the text to facsimile form for delivery to facsimile terminals (6,7).

Videotex. The videotex service is an interactive service that allows users of videotex terminals to communicate with databases servers via telecommunications network. The majority of videotex terminals are connected via PSTN with a special videotex modem made for sending at 75 bit/s and receiving at 1200 bit/s; both text and diagrams from the database can be reproduced on a screen. With a suitable screen, reproduction can be in color. ITU-T Recommendation F.301 describes a high-speed terminal access to videotex services for use on the PSTN. ITU-T Recommendation F.301 provides not only for faster throughput but also access to new capabilities that are not feasible at lower speeds. In spite of the efforts of CCITT, videotex systems have been started as national systems, with international compatibility. In France, the government subsided videotex by automating telephone books using videotex, and terminals were supplied free to subscribers (8,9).

Telecommunication Terminal Classification

In terms of the services, telecommunication terminals can be categorized as

Voice terminal equipment
Data terminal equipment
Visual terminal equipment
Multimedia terminal equipment

The telecommunication terminal equipment can be further categorized in terms of the access interface by which they access the telecommunication network:

Wired terminal equipment
Wireless terminal equipment

An access interface is the physical connection between the user and the telecommunication network that allows the user to request and obtain services. Most residences, for example, have a single-line telephone and, accordingly, a single connection to the local central office (CO). This single local loop can comprise two logical channels, one for user-network signals and one for user data (voice and tones).

As the number of simultaneous users increases at a customer location, the requirement for the number of physical resources to handle those users also increases. A second local loop, for example, can provide a second telephone line, while multiple trunk circuits can provide multiple lines between a customer's private branch exchange (PBX) and the CO. Access to other networks and/or network services (such as a packet or telex network) can be provided by bringing additional lines to the customer's premises. It is not uncommon for a business location to have many individual lines connecting it to the CO for such services as telephony, fax, a point-of-sale terminal, and remote security.

WIRED TELECOMMUNICATION TERMINALS, SERVICES, STANDARDS, AND PROTOCOLS

In this section terminals using wired networks are presented. At first we introduce terminals for the ISDN and ATM services and then we introduce multimedia terminals for the local area networks (LAN) and plain old telephone system (POTS).

The ISDN and ATM are networks capable of providing a variety of services to telecommunication terminals. In the sections related to ATM and ISDN, an overview of these networks and a description of the services available to the terminals are presented. Multimedia for LANs and POTS are application-related topics. Material presented in these sections describes the ITU-T standards for multimedia terminals that are connected to LANs and POTS.

ISDN Terminals and Functional Devices

ISDN is a fully digital communications technology implemented throughout the infrastructure of the existing worldwide telephone network. ISDN uses a standard phone line (a copper wire pair) in a home or office and converts it from a single analog circuit into multiple high-speed digital circuits capable of transmitting audio, still images, motion video, and text data simultaneously.

Several different devices may be present in the connection between consumer premises equipment (CPE) and the network to which the CPE is attached. Consider the relatively simple example of a customer's connection to the telephone network. All of the subscriber's telephones are connected with inside wiring to a junction box in the customer's building; the local loop provides the physical connection between the junction box and the local exchange (LE). As far as the customer is concerned, the CPE is communicating directly with the exchange; the junction box is transparent.

The ISDN standards define several different types of devices. Each device type has certain functions and responsibilities but may not represent an actual physical piece of equipment. For that reason, the standards call them functional devices. Since the ISDN recommendations describe several functional device types, there are several device-to-device interfaces, each requiring a communications protocol. Each of these functional device interfaces is called a reference point.

The following paragraphs describe the different functional devices and reference points, which are shown in Fig. 1.

ISDN Functional Devices. The network device that provides ISDN services is the LE. ISDN protocols are implemented in the LE, which is also the network side of the ISDN local loop. Other LE responsibilities include maintenance, physical interface operation, timing, and provision of requested user services. Some ISDN exchange manufacturers further break down the functions of the LE into two subgroups called local termination (LT) and exchange termination (ET). The LT handles those functions associated with the termination of the local loop, while the ET handles switching functions. Network termination type 1 (NT1), or local loop terminator equipment, represents the termination of the physical connection between the customer site and the LE. The NT1's responsibilities include line performance monitoring, timing, physical

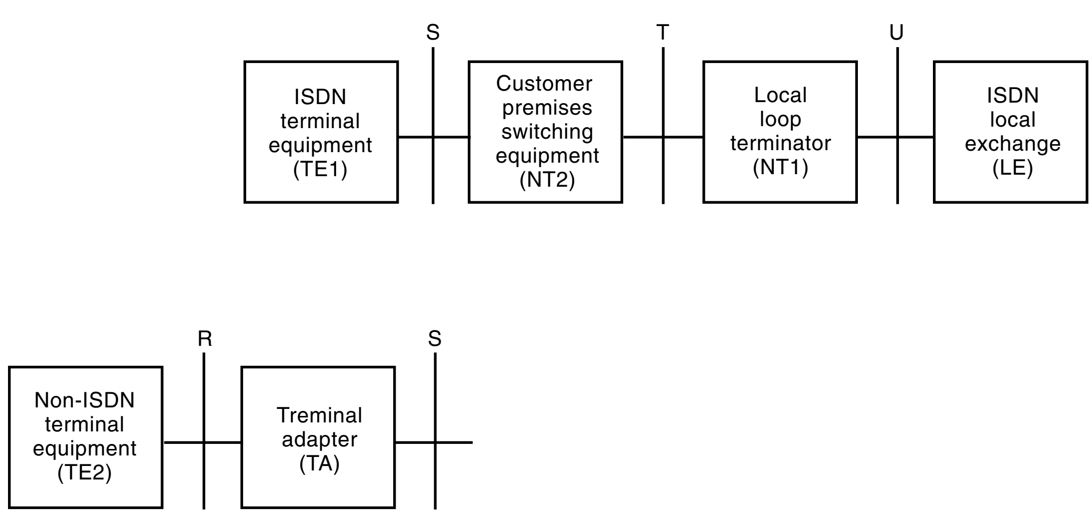

Figure 1. ISDN functional devices and interface access.

signaling protocol conversion, electrical conversion, and power transfer.

Network termination type 2 (NT2) equipment includes those devices providing customer site switching, multiplexing, and/or concentration. This includes PBXs, multiplexers, host computers, terminal controllers, and other CPE for voice and data switching. NT2s distribute ISDN services to other devices that are attached to them. In this role, the NT2 might perform some protocol conversion functions as well as distribution functions. One of the primary distribution functions is network signaling on behalf of the attached terminals. The NT2 is responsible for all signaling to the network. As an example, a PBX might terminate an analog telephone and allow access to an ISDN primary rate interface (PRI) with 1.544 Mbit/s for a connection to other subscribers. In this case the PBX is providing protocol conversion from the analog voice to the ISDN digital voice and is collecting the dialed digits from the telephone and creating a signaling message for the LE.

Terminal equipment (TE) refers to end-user devices, such as an analog or digital telephone terminals, X.25 data terminal equipment, ISDN workstation, or integrated voice/data terminal (IVDT). IVDT configurations include computer terminals with telephones; telephones with built-in display screens and the ability to interface with computers; and, in some instances, systems consisting of a cluster controller and multiple handset/screen keyboard stations, each cable connected to the controller. Some IVDTs are PBX-proprietary devices, while other are independent stand-alone devices. IVDTs generally consist of a dual tone multifrequency (DTMF) telephone handset or speakerphone, an integral or a detachable CRT, a keyboard, and a powerful microprocessor.

Terminal equipment type 1 (TE1) includes those devices that utilize the ISDN protocols and support ISDN services, such as an ISDN telephone or workstation. Terminal equipment type 2 (TE2) includes non-ISDN-compatible devices, such as the analog telephones in use on today's telephone network.

A terminal adapter (TA) allows a non-ISDN device (TE2) to communicate with the network. TAs have particular importance in today's ISDN marketplace; nearly every device in use in today's data and telecommunications environment is TE2. TAs allow analog telephones, X.25 DTEs, PCs, and other non-ISDN devices to use the network by providing any necessary protocol conversion.

ISDN Reference Points. The ISDN reference points define the communication protocols between the different ISDN functional devices. The importance of the different reference points is that different protocols may be used at each reference point. Four protocol reference points are commonly defined for ISDN, called R, S, T, and U.

The R reference point is between non-ISDN terminal equipment (TE2) and a TA. The TA will allow the TE2 to appear to the network as an ISDN device, just like a modem allows a terminal or PC to communicate over today's telephone network. There are no specific standards for the R reference point; the TA manufacturer will determine and specify how the TE2 and TA communicate with each other. Examples of TA-to-TE2 communication include EIA 232-E, V.35, and the industry standard architecture (ISA) bus.

The S reference point is between ISDN user equipment (i.e., TE1 or TA) and network termination equipment (NT2 or NT1). The T reference point is between customer site switching equipment (NT2) and the local loop termination (NT1). ISDN recommendations from the ITU-T, the primary international standards body for ISDN, specifically address protocols for the S and T reference points. In the absence of the NT2, the user-network interface is usually called the S/T reference point. Since the ITU-T considers the physical NT1 device to be owned by the network administration; therefore, the ITU-T views the S or T reference points as the user-network boundary and does not address U reference point (for more information, see Refs. 10 and 11).

ITU-T Study Group 11 (SG11) has responsibility for digital networks, including ISDN. Among other things, it is responsible for writing the I-series recommendations defining ISDN and specifying appropriate services and protocols.

Other SGs participate in the ISDN standards process by virtue of the recommendations that they prepare, which often overlap. Recommendations for public data networks (X series), ISDN and telephone network switching and signaling (Q series), and addressing and numbering plans (E series), for example, can all pertain to ISDN. For this reason, several I-series recommendations are also assigned Q, X, or other series' recommendation numbers. Support of X.25 terminals on an ISDN, for example, is described in Recommendations I.462 and X.31; these two standards are identical. Similarly, ISDN signaling procedures on the D channel are listed as both Recommendations I.451 and Q.931. Supporting other non-ISDN terminals is also considered in ISDN recommendations such as X.31. This Recommendation defines services available and signaling procedures operated at the S/T reference point of an ISDN for subscribing packet mode terminal equipment and terminal adapter functionality to support existing X.25 terminals at the R reference point of the ISDN.

An important aspect for the success of ISDN will be the availability of user applications (in particular, applications based on personal computers) making use of ISDN. ITU-T Recommendation T.200 is the introduction to the multipart specification that defines a standard programming communication interface (PCI) allowing applications to access and manage the services provided by an ISDN. It provides mechanisms to support most protocols used for communication between ISDN applications. The basic services part provides a technical overview and defines the basic functions supported by the ISDN-PCI. It defines the PCI architecture and includes a detailed definition of the PCI messages and parameters used for administration and connection control. In the user plane protocols management architecture part, the general aspects for the management of and access to the user plane protocol supported by the PCI are explained. The three layers of the PCI's user plane protocols are represented in three parts, with detailed procedures, messages, and parameters used to access the ISDN services. One of the important features of this recommendation is providing information for binding PCI to different operating systems, such as Windows and Unix.

The other useful standard that is applicable for interoperability between ISDN and wireless networks is ITU-T Recommendation E.173. This recommendation covers the routing plan for interconnection between public land mobile network (PLMN) and PSTN/ISDN and between PLMN and PDN. It provides guidelines for routing when interconnecting PLMN and fixed terminal networks, and it indicates some basic rules for routing of calls between the networks. Routing between networks must adhere to regulatory conditions that exist within a given country or administration. Recommendation E.173 covers terrestrial-based PLMNs.

H.320 Terminal. The H.320 terminal is a visual phone terminal over ISDN which has been standardized by ITU-T. ITU-T Recommendation H.320 specifies technical requirements for narrowband visual telephone systems and terminal equipment, typically for videoconferencing and videophone services. It describes a generic system configuration consisting of a number of elements that are specified by respective ITU-T recommendations, definition of communication modes and terminal types, call control arrangements, terminal aspects, and interworking requirements. This Recommendation represents the relationship with other relevant H-series recommendations, such as G.723.1, G.729, H.262, H.263, H.310, H.322, H.323, and H.324.

ITU-T provides the Joint Coordination Group (JCG) on Audio-Visual Multimedia Services (AVMMS), in which study groups of ITU-T are working together on AVMMS. SG15 leads the JCG and has issued many recommendations for videotelephony, videoconferencing, and AVMMs not only in narrowband ISDN (N-ISDN) (e.g., Recommendation H.261 on video coding, Recommendation H.320 on audiovisual terminal and system), but also in the ATM-based B-ISDN, LAN, and general switched telephone network (GSTN) environments. This series of recommendations is composed of multimedia multiplexing, communication procedure and protocol, video and audio coding methods, and audiovisual terminals and systems (12).

ATM Terminals and Services

ATM is a cell-based, connection-oriented switching technology that is designed to support a wide variety of services, including ATM, frame relay, SMDS, and circuit emulation. ATM transmits all information using small (53 byte) fixed-length cells over broadband or narrowband transmission facilities. It is asynchronous because the cells carrying user data are not required to be periodic. The asynchronous and multimedia characteristics of ATM are what make it possible for ATM networks to carry both circuit and packet types of traffic simultaneously, with complete transparency to the applications. ATM was designed to provide large amounts of bandwidth economically and on demand. When a user does not need access to a network connection, the bandwidth is available for use by another connection that does need it.

Flexibility of the ATM network allows variety of services to the user. Typical services provided by ATM terminals are as follows:

Constant Bit Rate (CBR). This is any transfer application that contains smooth traffic or for which the end system's response time requirements justify occupying a fully reserved CBR channel. Examples are videoconferencing; interactive audio (e.g., telephony); audio/video distribution (such as television, distance learning, pay-per-view); audio/video retrieval (e.g., video-on-demand, audio library).

Variable Bit Rate (VBR). This is suitable for any application for which the end system can benefit from statistical multiplexing and can tolerate or recover from a potentially small random loss ratio. Real-time VBR can be used by native ATM voice with bandwidth compression and silence suppression. It may also be appropriate for multimedia communications. Non-real-time VBR can be used for data transfer, such as response-time-critical transaction processing applications like airline reservations, banking transactions, and process monitoring, and frame relay interworking

Available Bit Rate (ABR). Any non-time-critical application running over an end system capable of varying its emission rate can exploit ABR service. This category provides economical support to those applications that show vague requirements for throughput and delay and require low cell loss ratio. Examples include LAN interconnection/internetworking services, LAN emulation, critical data transfer (defense information, banking services), supercomputer applications, and data communications like remote procedure call, distributed file services, and computer process swapping/paging.

Unspecified Bit Rate (UBR). This can provide a suitable solution for less demanding applications. Most data applications, such as background file transfers with minimal service requirements, are very tolerant to delay and cell loss. Examples may include text/data/image transfer, messaging, distribution, retrieval and remote terminal (telecommuting). These services can take advantage of any spare bandwidth (10).

ATM provides two virtual circuit communications services: switched virtual circuits (SVC) and permanent virtual circuits (PVC). SVCs establish short-term connections that require call setup and teardown, while PVCs are similar to dedicated private lines because the connection is set up on a permanent basis.

The following steps summarize the basic functions of an ATM interface:

Packetize or segment the incoming data into fixed-length cells with a header and a payload field.

Address the cell by prefixing it with a logical destination virtual path (VP) address.

Assign different virtual channels (VC) within the virtual path, depending on the type of data.

Multiplex the cells from various sources together onto the outgoing transmission link, mapping the incoming VPI/VCI to the associated outgoing VPI/VCI address.

Demultiplex or "unpack" the cells from various sources, translating them back into their native format, and delivering them to the appropriate device or port. This is handled by the destination ATM switch.

An optimal exploitation framework for the ATM service categories will be achieved with the development of new ATM-aware applications that access the network through an ATM application programming interface (API). This interface, when available on widespread computing platforms, will bypass the bottleneck legacy protocol stacks to provide the application with direct visibility and control of the ATM network resources. Desktop-to-desktop ATM applications will benefit from an ATM API and from other developments, such as signaling capabilities. Supporting an ATM application programming interface in Winsocket 2 is an important issue that facilitates the use of nonexpensive personal computers as ATM terminals.

The major activity for formal ATM standardization is done by ITU-T by investigating proposals from other organizations such as ATM Forum. The ATM Forum was started in October of 1991 by a consortium of four computer and telecommunication vendors. Today's membership comprises network equipment providers, semiconductor manufacturers, service providers, carriers and, most recently, end users. The Forum is not a standards body. The ATM Forum is a consortium of companies that writes specifications to accelerate the definition of ATM technology. These specifications are then passed up to ITU-T (formerly the CCITT) for approval. The ITU-T standard body fully recognizes the ATM Forum as a credible working group.

The main ITU-T recommendation for ATM is Recommendation I.731, which describes the general functional architecture and characteristics of ATM network elements (NE) in terms of specific functional blocks derived from the B-ISDN protocol reference model (PRM) described in Recommendation I.321. The intent of this Recommendation is to enable interoperability between ATM equipment based on the specific requirements described for the functional blocks. A more detailed description of the individual functional elements is given in the companion Recommendation I.732. The I.731 Recommendation provides an overview description of the ATM NE functional blocks in terms of the user plane, layer management, and plane management functions. The physical interfaces required for interoperability between the ATM NEs are defined, with references to the appropriate recommendations that describe these interfaces in detail. This Recommendation also defines the criteria for classification of the ATM equipment types (13).

Since voice communication is very important in ATM and B-ISDN environments, the subworking group Voice and Telephony over ATM (VTOA) has started a draft implementation agreement for VTOA at the desktop that can trigger some other activities, such as explicit call transfer service within the ATM Forum.

Multimedia Terminals over LANs

H.323 Terminal. Recommendation H.323 describes terminals, equipment, and services for multimedia communication over LANs that do not provide a guaranteed quality of service. H.323 terminals and equipment may carry real-time voice, data, and video, or any combination, including videotelephony.

The LAN over which H.323 terminals communicate may be a single segment or ring, or it may be multiple segments with complex topologies. It should be noted that operation of H.323 terminals over multiple LAN segments (including the Internet) may result in variable performance. H.323 terminals may be integrated into personal computers or implemented in stand-alone devices such as videotelephones. Support for voice is mandatory, while data and video are optional, but if supported, the ability to use a specified common mode of operation is required, so that all terminals supporting that media type can interwork. H.323 allows more than one channel of each type to be in use. Other recommendations in the H.323 series include H.225.0 packet and synchronization; H.245 control; H.261 and H.263 video codecs; G.711, G.722, G.728, G.729, and G.723 audio codecs; and the T.120 series of multimedia communications protocols.

H.323 makes use of the logical channel signaling procedures of Recommendation H.245, in which the content of each logical channel is described when the channel is opened. Procedures are provided for expression of receiver and transmitter capabilities, so transmissions are limited to what receivers can decode, and so that receivers may request a particular desired mode from transmitters. Since the procedures of H.245 are also used by Recommendation H.310 for ATM networks, Recommendation H.324 for GSTN, and V.70, interworking with these systems should not require H.242 to H.245 translation as would be the case for H.320 systems.

H.323 terminals may be used in multipoint configurations and may interwork with H.310 terminals on B-ISDN, H.320 terminals on N-ISDN, H.321 terminals on B-ISDN, H.322 terminals on guaranteed quality of service LANs, H.324 terminals on GSTN and wireless networks, and V.70 terminals on GSTN (14,15).

H.322 Terminal. This terminal, contrary to H.323, guarantees quality of service over LANs. ITU-T Recommendation H.322 covers the technical requirements for narrowband visual telephone services defined in H.200/AV.120-series recommendations, in those situations where the transmission path includes one or more LAN, each of which is configured and managed to provide a guaranteed (QoS) equivalent to that of N-ISDN such that no additional protection or recovery mechanisms beyond those mandated by Recommendation H.320 need be provided in the terminals (16).

Multimedia Terminals over POTS

H.324 Terminal. H.324 is a new standard adopted by ITU that provides a foundation for interoperability and high-quality video, voice, and data-based phone calls. The H.324 standard specifies a common method for video, voice, and data to be shared simultaneously over high-speed modem connections utilizing V.34 operating over GSTN. H.324 is the first standard to specify interoperability over a single analog line. H.324 terminals may carry real-time voice, data, and video, or any combination, including videotelephony. H.324 capability can be used in different videophones, such as a standalone video phone, which has a telephone, a camera mounted on top, and a liquid crystal display (LCD), or TV-based video phone, in which the TV will be used for displaying a videophone image. PC-based video phones can also use the H.324 standard to use videophone by installing compression/decompression software or using some hardware with digital signal processing capability to compress/decompress audio and video signals.

Other recommendations in the H.324 series include the H.223 multiplex, H.245 control, H.263 video codec, and G.723.1 audio codec.

H.324 is interoperable with other terminals, including speech-only terminals and H.320 multimedia telephone terminals over the ISDN. One way to achieve interoperability with an ISDN H.320 terminal is by using an interworking adapter that is located at the interface between ISDN and GSTN signals. It transcodes the H.223 and H.221 multiplexes, and the content of control, audio, and data logical channels between the H.324 and H.320 protocols. To ease communication between H.324 and H.320 terminals via interworking adapters, H.324 terminals that support video shall support the H.261 video codec in the quarter common intermediate format (QCIF) picture format so that the additional delay of video transcoding can be avoided. When this mode is in use, interworking adapters shall insert and remove H.261 and H.263 BCH error correction and error correction framing as appropriate for each terminal type. H.324 terminals shall respond to the H.245 flow control command, so that transmitted H.324 video streams can be matched to the H.320 video bit rate in use by the H.221 multiplex.

The second way to provide interoperability between H.320 and H.324 is by using dual-mode (H.320 and H.324) terminals on the ISDN. This terminal sends H.324 GSTN signals by the use of a "virtual modem," which generates and receives a V.34 analog signal encoded as a G.711 audio bitstream over the ISDN (17,18).

WIRELESS AND MOBILE TERMINALS, SERVICES, AND STANDARDS

On fixed networks the widespread use and popularity of the World Wide Web (WWW) has dramatically increased the volume, richness, and availability of information. Much of this information makes use of video and audio in addition to text and graphics. The growth in mobile communications and portable computing enables multimedia information services on mobile networks. Portable computers are becoming powerful, and personal digital assistants (PDA) now provide pocket-sized computing. However, current wide area wireless networks cannot deliver the bandwidth necessary to adequately support full multimedia applications.

The mobile environment imposes a number of challenges that impair the robustness of client/server operation, which includes adaptation to varying quality of service, robustness in the face of disconnected links, roaming between different operators and network types, movement between different geographical locations, reconfigurable real-time multiparty connections, and personalized information filtering. Mobile terminals have varying capabilities, ranging from a global system for mobile communications (GSM) phone or personal communicator to PDAs with small screens to fully multimedia capable laptops. In this section we first introduce the traditional wireless terminals used for voice, data, and multimedia communications and then we cover evolving broadband wireless terminals.

Traditional Wireless Terminals

Different types of cellular systems employ a variety of multiple access methods. The traditional analog cellular systems, such as those based on the advanced mobile phone service (AMPS) and total access communications system (TACS) standards, use frequency division multiple access (FDMA). FDMA channels are defined by a range of radio frequencies, usually expressed in a number of kilohertz (kHz), out of the radio spectrum.

A common multiple access method employed in digital cellular systems is the time division multiple access (TDMA). TDMA digital standards include North American Digital Cellular (known by its standard number IS-54), GSM, and personal digital cellular (PDC). The latest technology for the voice-oriented networks is the code division multiple access (CDMA), in which channels are identified by specific codes. The IS-95 is the North American standard for the CDMA technology. Data-oriented networks often use a variety of random access methods, such as dynamic slotted-ALOHA and digital sense multiple access (DSMA). In the rest of this section we first describe terminals connected to the voice-oriented digital cellular and PCS services and then we go over the terminals connected to data-oriented wireless networks. We complete this section by addressing multimedia wireless terminals.

Voice-Oriented Terminals

GSM Terminals. ISDN compatibility was one of the goals in designing GSM in terms of the services offered and the control signaling used. However, radio transmission limitations, in terms of bandwidth and cost, do not allow the standard ISDN B-channel bit rate of 64 kbps to be practically achieved.

Using the ITU-T definitions, telecommunication services can be divided into bearer services, teleservices, and supplementary services. The most basic teleservice supported by a GSM terminal is telephony. As with all other communications, speech is digitally encoded and transmitted by the GSM terminal as a digital stream. There is also an emergency service, where the nearest emergency-service provider is notified by dialing three digits (similar to 911).

Speech in a GSM terminal is digitally coded at a rate of 13 kbps, so-called full-rate speech coding. This is quite efficient compared with the standard ISDN rate of 64 kbps. One of the most important Phase 2 additions will be the introduction of a half-rate speech codec operating at around 7 kbps, effectively doubling the capacity of a network.

This 13 kbps digital stream (260 bits every 20 ms) has forward error correction added by a convolutional encoder. The gross bit rate after channel coding is 22.8 kbps (or 456 bits every 20 ms). These 456 bits are divided into eight 57-bit blocks, and the result is interleaved among eight successive time slot bursts for protection against bursty transmission errors. Each time slot burst is 156.25 bits and contains two 57-bit blocks and a 26-bit training sequence used for equalization. A burst is transmitted in 0.577 ms for a total bit rate of 270.8 kbps and is modulated using Gaussian minimum shift keying (GMSK) onto the 200 kHz carrier frequency. The 26-bit training sequence is of a known pattern that is compared with the received pattern in the hope of being able to reconstruct the rest of the original signal. Forward error control and equalization contribute to the robustness of GSM radio signals against interference and multipath fading.

The digital TDMA nature of the signal allows several processes intended to improve transmission quality, increase the mobile's battery life, and improve spectrum efficiency. These include discontinuous transmission, frequency hopping, and discontinuous reception when monitoring the paging channel. Another feature used by GSM is power control, which attempts to minimize the radio transmission power of the terminals and thus minimize the amount of co-channel interference generated.

A variety of data services is offered. GSM users can send and receive data, at rates up to 9600 bps, to users on POTS, ISDN, packet switched public data networks (PSPDN), and circuit switched public data networks (CSPDN) using a variety of access methods and protocols, such as X.25 or X.32. Since GSM is a digital network, a modem is not required between the GSM terminal and GSM network, although an audio modem is required inside the GSM network to interwork with POTS.

The mobile terminal in GSM is really two distinct entities. The actual hardware is the mobile equipment, which is anonymous. The subscriber information, which includes a unique identifier called the international mobile subscriber identity (IMSI), is stored in the subscriber identity module (SIM), implemented as a smart card. By inserting the SIM card in any GSM mobile equipment, the user is able to make and receive calls at that terminal and receive other subscribed services. By decoupling subscriber information from a specific terminal, personal mobility is provided to GSM users.

Other data services include Group 3 facsimile, as described in ITU-T Recommendation T.30, which is supported by use of an appropriate fax adapter. A unique feature of GSM terminal, not found in older analog systems, is the support for short message service (SMS). SMS is a bidirectional service for short alphanumeric (up to 160 bytes) messages. Messages are transported in a store-and-forward fashion. For point-to-point SMS, a message can be sent to another subscriber to the service, and an acknowledgment of receipt is provided to the sender. SMS can also be used in a cell-broadcast mode, for sending messages such as traffic updates or news updates. Messages can also be stored in the SIM card for later retrieval.

Supplementary services are provided on top of teleservices or bearer services. In the current (Phase 1) specifications, they include several forms of call forward (such as call forwarding when the mobile terminal is unreachable by the network), and call barring of outgoing or incoming calls (for example, when roaming in another country). Many additional supplementary services will be provided in the Phase 2 specifications, such as caller identification, call waiting, and multiparty conversations.

PCN and GSM both have the ability in their protocols for customers to receive text messages, like a text message pager, but delivery is guaranteed (when the phone is unavailable due to being out of service area, the network will hold the message and deliver it shortly after the phone comes back into range). The message can be sent by a central paging service. SMS is also used for "internal" messages, such as activating a phone for the first time, remote programming of numbers into the directory on the SIM card, and alerting the user of voicemail. It is also used internally within the networks for transfer of call logging information from the switches to the billing centers. If a terminal is stolen, the provider can block the terminal and the SIM card; some terminals will then continue to broadcast their location, and some will display the message "STOLEN HANDSET," which cannot be removed without the operator unblocking the terminal. The European Telecommunications Standards Institute (ETSI) is the organization responsible for the GSM standards (19).

CT2 Terminal. CT2 is the first standard for digital cordless phones, and the range of terminal is very limited. There is no hand-over between base stations, unless the base stations themselves cooperates. Logging onto a base station is the terminal responsibility for receiving the call. Changing to another base station requires re-registration. If the call is unsuccessful, the call is routed onto voicemail. CT2 has the ability to hand over to adjacent base stations. However, in an office situation, a CT2 PBX can be used to support many CT2 phones.

The CT2 terminal uses the 864 MHz to 869 MHz band and provides 40 channels that are 100 kHz apart. The access method is FDMA, and each carrier supports one cell with time division duplex (TDD) for two-way conversation. The channel bit rate is 72 kbit/s. ETSI undertook the task of developing standards for CT2.

DECT Terminal. DECT (Digital European Cordless Telecommunications) is a standard for cordless phones. The DECT terminal is intended to be a far more flexible than the CT2 terminal. In the DECT terminal there are more radio frequency (RF) channels (10 RF carriers $\times$ 12 duplex bearers per carrier = 120 duplex voice channels). DECT uses TDD to support a two-way conversation on the same carrier. It also has a better multimedia performance since 32 kbit/s bearers can be concatenated.

PCN/DCS1800 Terminal. The PCN terminal uses the GSM protocols in higher frequencies, (1800 MHz), therefore, there is a larger frequency range to work with and lower power. Also due to the reduced power levels, batteries should last longer than the equivalent GSM version of the terminal. PCN is also known as DCS1800.

Similar to GSM, PCN uses SIM cards to identify the customer. These are smart cards that hold the user's phone number, subscription details, and a calling directory. Theoretically, even though GSM and PCN phones have different technologies, SIM roaming is possible. In SIM roaming, if users go overseas, they could insert their SIM card and take their phone number with them. ETSI is the organization responsible for the GSM standards.

DCS1900 Terminal. The DCS1900 terminal is one of the many standards proposed for digital mobile phones in the United States. The 900 MHz GSM band and the 1800 MHz DCS1800 band are not available in the United States. The protocol is the same as DCS1800, or GSM.

PHS Terminal. PHS (personal handyphone system) was developed in Japan in mid-1995. The PHS terminal offers similar service to CT2. In Japan, the PHS terminal now supports a protocol called PHS internet access forum standard (PIAFS), which allows 32K data transmission and which means that the PHS terminal could be used like a modem.

JDC Terminal. The Japanese Digital Cellular (JDC) terminal uses a digital protocol unique to Japan that operates in two different frequency bands, 800/900 MHz and 1.5 GHz. This terminal is similar to GSM terminal with similar services.

The channel bit rate is 42 kbit/s. In the first phase of JDC implementation there are three TDMA user channels per frame, and in future implementations there will be six per frame. In the full rate system, there are 224 user data bits per 20 m frame, for a transmitted bit rate of 11.2 kbit/s comprising coded voice bits and error-protection coding.

CDMA Terminals. CDMA is a spread spectrum technique for multiple access. In CDMA, each terminal is assigned a pseudonoise (PN) code to modulate transmitted data. The PN code is a long sequence of ones and zeros similar to the output of a random number generator of a computer. Although the numbers are not really random, by using a specific algorithm they appear to be random. Because the codes are nearly random, there is very little correlation between the different codes. In addition, there is very little correlation between a specific code and any time shift of that same code. Thus, the distinct codes can be transmitted over the same time and the same frequencies and the signals can be decoded at the receiver terminal by correlating the received signal (which is the sum of all transmitted signals) with each PN code.

With growing interest in the integration of voice, data, and imagery traffic in telecommunication networks, CDMA appears increasingly attractive as the wireless access method of choice. For cellular telephony, CDMA is a digital multiple access technique specified by the Telecommunications Industry Association (TIA) as IS-95. In March 1992, the TIA established the TR-45.5 subcommittee, with the charter of developing a spread spectrum digital cellular standard. In July of 1993, the TIA gave its approval of the CDMA IS-95 standard. IS-95 systems divide the radio spectrum into carriers that are 1250 kHz (1.25 MHz) wide. One of the unique aspects of CDMA is that while there are certainly limits to the number of phone calls that can be handled by a carrier, this is not a fixed number. Rather, the capacity of the system will be dependent on a number of different factors, including bandwidth efficiency, number of sectors in each base station antenna, the voice activity factor that determines what percentage of time voice is active, and the interference increase factor (19).

Because in a CDMA system every terminal is a source of interference to every other CDMA terminal, control of mobile terminal power is a critical element of the system design (19).

Data-Oriented Terminals

ARDIS Terminal. The Advanced Radio Data Information Service is a two-way radio service that was first implemented by IBM and Motorola. The service is suitable for two-way data transfer of size less than 10 kbytes. The ARDIS terminal is a laptop radio terminal that uses the 800 MHz band with separate transmit and receive frequencies. The user data rate is 8000 bit/s. The radiated power from terminal is 4 W. The laptop portable terminals access the network using data sense multiple access (DSMA), in which a remote terminal listens to the base station transmitter to determine if a busy-bit is on or off. When the busy-bit is off, the terminal can transmit. When the busy-bit is off, chances are two or more terminals transmit simultaneously, which results in collision and retransmission is required.

Mobitex Terminal. The Mobitex system is a nationwide, interconnected trunked radio network developed by Ericsson and Swedish Telecom. While the Mobitex system was designed to carry both voice and data service, the U.S. and Canadian networks are used to provide data service only.

The Mobitex terminal transmits at 896 MHz to 901 MHz and receives at 935 MHz to 940 MHz. The terminal uses dynamic power setting in the range of 100 mW to 10 W for mobile terminals and 100 mW to 4 W for portable terminals. The transmission rate is 8000 bit/s half duplex in 12.5 kHz channels, which is suitable for file transfer up to 20 kbytes. The packet size is 512 bytes, with 1 to 3 seconds delay. Forward-error correction as well as retransmission are used to ensure the bit-error-rate quality of delivered data packets. The access method is dynamic slotted-ALOHA.

To access the network, the Mobitex terminal finds the base station with the strongest signal and registers with the base station. When the terminal enters an adjacent service area, it automatically reregisters with new base station and the terminal's location information is relayed to a higher level, which provides roaming for the mobile terminal.

CDPD Terminal. CDPD stands for cellular digital packet data, which consists of a laptop computer with a cellular modem transmitting data over the existing cellular telephone network, commonly referred to as AMPS (advanced mobile phone system). CDPD uses the same transmission channels as the cellular telephone network for short transmissions of small data packets. There are two types of CDPD networks: dedicated channel and channel hopping. In a dedicated channel network, certain channels are set aside solely for use by the CDPD network. In a channel hopping network, all the channels are shared by CDPD and the cellular telephone network. A cellular phone call has preemptive priority over any CDPD call. The CDPD terminal vacates the channel within 40 ms of detecting voice activity to transfer the channel to the cellular voice user. The CDPD terminal then moves, or "hops" to another available channel and continues its sessions. If there is no channel available to the CDPD terminals, they are dropped.

Each channel on a CDPD network can be accessed by up to 30 CDPD terminals at a time. The terminals take turns transmitting on the channel stream. If more than one terminal tries to send data at the same time, a "collision" occurs and all users involved try to transmit again a short time later.

The CDPD terminal consists of a personal communications device, such as a laptop computer connected to a cellular data modem, or a point-of-sale terminal. The CDPD terminal may operate as a normal computer, used for downloading files or checking e-mail, or it may serve a specialized purpose, such as credit card verification. The CDPD terminal both sends data to the network and receives data from the network at a rate of 19.2 kbps. (Because of overhead, the actual data rate is approximately 9.6 kbps.) Before transmitting the data, the CDPD terminal packetizes the data and also encrypts them for security (19).

Multimedia Terminals

Portable Wireless Multimedia Terminal. Convenient, ubiquitous access to the network requires a portable input/output (I/O) device. Reading large amounts of text on a computer screen, instead of paper, is inconvenient because of the usual fixed desktop placement of the screen. Thus the electronic viewing device should have the convenience of paper, with the capabilities of a multimedia terminal. This implies that the portable device should weigh less than a pound and a form factor that allows convenient observation (such as a full size 8×11 inch notepad) with a long battery life, while providing color video and audio output and pen and microphone inputs (20).

There continues to be rapid progress in the area of LCD displays, which usually require a backlight to give sufficient contrast (this light dominates the power consumption of most portable terminals). However highly portable terminals provide low-quality displays since they have limited space, resolution, and power available for display. For these reasons the user interface of a portable terminal needs to be designed for monochrome presentations in a very small screen area. Since portable terminals lack the space for keyboards, icon-based interfaces and pen-based input have been considered as an alternative.

Another challenge in designing wireless multimedia terminals is decreasing the delay time for accessing information on the network. This includes all the processing and delays in the portable unit, base station, and gateway, and the delays in the Internet. Latency control is the most important factor in those applications in which there is a tight interactive loop between the user input and the display output. If the delay between two consecutive displays on the monitor is performed in under 30 ms, it will not be detected by the human eye since this is the update rate of the display. Similarly, experience with echoes in long-distance phone connections shows that delays under 30 ms are also not objectionable to the user. Thus a latency specification of less than 30 ms appears to be adequate and should be considered as a design parameter.

From the networking point of view, the network protocols can be divided into three classes: heavyweight, lightweight, and filters. Filters are not strictly protocol implementations; they are mechanisms for extracting data from a network. Heavyweight protocols make guarantees such as in-order delivery and reliability and provide multiplexing/demultiplexing as a standard feature; therefore, they require high processing power, which makes portable terminals heavier and slower. Lightweight protocols, on the other hand, often do not provide strong guarantees and can be specialized. They tend to be fast and can be based on a model other than the stack. They attempt to minimize processing, connection, and flow control overhead, and network contention, such as Xerox remote procedure call (RPC). Filters provide user-level access to the raw network at the device driver level. Thus the lightweight protocols and specialized filters can be useful in designing portable terminals to lower the weight and to increase speed.

The portable wireless multimedia terminals require high performance at low energy. Thus the focus will be on highly integrated complementary metal on semiconductor (CMOS) implementations that have the lowest possible energy consumption while using advanced communication algorithms. The primary challenge of reducing the weight of the terminal is the power consumption of the circuitry to convert the data from the downlink data stream to the form required by the video and graphics displays and audio output. The most demanding function is the decompression and associated frame buffering of the video data, which must be performed in real time, and its subsequent conversion to analog signals to drive the display. One solution for this problem is to move as much of the processing as possible out of the portable unit into servers on the network. This results in a design in which there is no user-accessible computation in the portable terminal, which not only relieves the portable unit of the requirement to support a general-purpose operating systems such as Windows NT or Unix, but also eliminates the need for expensive and power-hungry mass storage devices, costly memory, and, implicitly, a high-speed general-purpose processor subsystem. Also, by not incorporating user-accessible general-purpose processing in the portable unit, there is no need to send error-free computer data over the wireless link. As mentioned, this results in very inefficient transfer when the error rates are 10^{-3} or higher, requiring some form of error correction capability in the wireless protocols. The computation remaining in the mobile terminal is implemented with the lowest possible power consumption by design of a custom chip set (for more information see Refs. 21 and 22).

Another factor in wireless portable multimedia terminal is the antenna system for transmission of information at increasingly higher bit rates per hertz, which requires practical methods like adaptive equalization, diversity at the receiver and transmitter, and several combining techniques from the reception of multiple antenna systems (23).

Multimedia Telephone Terminals over Mobile Radio. It is expected that multimedia telephone terminals will also be used on mobile radio networks. Rate matching between wireless terminals and GSTN terminals can be achieved by the use of the H.245 flow control command. Wireless operation is for further study.

PDA. A PDA is a terminal that consists of a backlit LCD, a few navigational keys, a stylus, and a wireless connectivity device. To enter data, either the on-screen keyboard or the on-screen graphics-based writing pad can be used; the latter responds to a form of shorthand and comes with a stick-on guide for easy reference.

Usually several PIM application icons reside within the display. Standard PIM applications are an expense tracker and an e-mail application. Transferring data through a serial connection from the PDA is also possible. The PDA usually is used by field engineers to access information remotely.

Broadband Wireless Terminals

Wireless ATM. The key element in the wireless ATM terminal is represented by a wireless ATM adaptation layer (AAL), capable of coping with the increased latency and poor reliability of wireless links. Terminal mobility is supported using both wireless terminal features and B-ISDN signaling for end-to-end connections between a mobile and a fixed ATM terminal (24,25).

In the long term, this development will offer the technological base for a seamless multimedia multimode personal communication system, which will have a number of programmable AAL interfaces capable of supporting ATM applications over different wireless and wireline physical layers and allow universal roaming.

The ATM Forum Wireless ATM Working Group is standardizing mobility support within an ATM network and a radio access layer for ATM-based wireless access. There are some activities in the European Community (EC) for implementing and demonstrating wireless ATM, among which are Advanced Communications Technologies and Services' (ACTS) ATM wireless access communication system (AWACS), system for advanced mobile broadband applications (SAMBA), and wireless ATM network demonstrator (WAND) are important. AWACS will operate in the 19 GHz band with user bit rates up to 34 Mbit/s and radio transmission ranges between 50 m and 100 m. The project will support limited, slow-speed mobility in line with expected use of high data services, but with improved portability. The key objectives for SAMBA are to demonstrate mobile applications of bit rates up to 34 Mbit/s; design and realize transparent ATM connections via radio transmission for mobile applications; and specify and implement medium access, handover, and radio resource management for a cellular system. The Magic WAND project covers the whole range of functionality from basic (wireless) data transmission to shared multimedia applications. The primary goal of the project is to demonstrate that wireless access to ATM, capable of providing real multimedia services to mobile users, is technically feasible. The project partners have chosen to use the 5 GHz frequency band for the demonstrator and perform studies on higher bit rate operation (>50 Mb/s) in the 17 GHz frequency band. In this project mobile terminals based on the portable computers in combination with wireless access points (AP) will be used.

FPLMTS/IMT2000 Terminals. FPLMTS/IMT2000 are third-generation mobile communications systems. They are considered by ITU-T and will follow their second-generation counterparts such as digital cellular phones. FPLMTS/IMT2000 terminals provide services ranging from basic wide area paging, to voice telephony (probably the prime requirement of the personal terminal), to digital data services, and to audio and visual communications. The actual services obtained by a user depend on the user's terminal capabilities and subscribed set of services and the service set provided by the relevant network operator. The FPLMTS/IMT2000 terminals have high transmission speeds (below 2 Mbps and capable of transmitting simple moving pictures), service accessibility from anywhere in the world, and high quality in the mixed networks.

The user of the personal terminal will be able to take it anywhere world wide and have access to at least a minimum set of services comprising voice telephony, a selection of data services, access to universal personal telecommunications (UPT), and an indication of other services available. FPLMTS cover the application areas presently provided by separate systems such as cellular, cordless, telepoint, mobile data, and paging.

In FPMTLS multiple calls can be connected at the terminal. FPLMTS mobility is available only within FPLMTS, while the UPT can be enjoyed across several telecommunication platforms. UPT is a telecommunication service that enables users to access various services through personal mobility on any fixed or mobile terminal. In FPLMTS an identical procedure between mobile terminals and networks is applied to both UPT and FPLMTS mobility. Thus, multiple users can simultaneously communicate using their ID numbers at FPLMTS mobile terminals. For terminal mobility and FPLMTS user mobility, a standardized international mobile user identity (IMUI) is required. At present it is considered that Recommendation E.212, "Identification Plan for Land Mobile Stations," can be applied to IUMI. FPLMTS provides user identity module (UIM) portability, which enables an FPLMTS user identity to be physically separate from the FPLMTS terminal. In addition, FPLMTS provide standard terminal portability, enabling standard terminal equipment to be connected to FPLMTS mobile terminals (19,26).

UMTS Terminal. UMTS is a third-generation mobile communication system currently being developed in Europe. UMTS related activities are lead by research conducted within the UMTS Forum program and standardization activities within the ETSI. UMTS terminals should support existing mobile services and fixed telecommunications services up to 2 Mbit/s and unique mobile services, such as navigation, vehicle location, and road traffic information services, which will become increasingly important in a pan-European market. UMTS terminals also can be used anywhere, in the home, the office, and in the public environment, both in rural areas and city centers. These terminals are available in different sizes, from mobile terminals such as a low-cost pocket telephone (to be used by almost anyone anywhere) to sophisticated terminals that provide advanced video and data services.

It is expected that UMTS and IMT-2000 will be compatible so as to provide global roaming, but it is too early to say whether this goal will eventually be achieved (27,28).

ABBREVIATIONS

ACTS	Advanced Communications Technologies and Services
AMPS	Advanced Mobile Phone System
ATM	Asynchronous Transfer Mode
B-ISDN	Broadband Integrated Services Digital Network
CDMA	Code Division Multiple Access
DAMPS	Digital Advanced Mobile Phone System
DCS-1800	Digital Cellular System 1800
DECT	Digital European Cordless Telephone
EC	European Community
ETSI	European Telecommunications Standards Institute
EU	European Union
FPLMTS	Future Public Land Mobile Telecommunication System
GSM	Global System for Mobile Communications
IMT2000	International Mobile Telecommunications 2000
IN	Intelligent Network
IP	Internet Protocol
IS-95	Interim Standard-95
ISDN	Integrated Services Digital Network
ITU	International Telecommunication Union
JDC	Japanese Digital Cellular Standard
LAN	Local Area Network
MAC	Medium Access Control
NMT	Nordic Mobile Telephone
PACS	Personal Analog Cellular System
PCS-1900	Personal Cellular System
PDC	Personal Digital Cellular
PDN	Packet Data Network
PHS	Personal Handy Phone System, a wireless telecommunication system used in Japan
RACE	Research into Advanced Communication in Europe
SGSN	Specific General Packet Radio Service Node
TACS	Total Access Communication System
TDMA	Time Division Multiple Access
UMS	UMTS Mobility Server
UMTS	Universal Mobile Telecommunications System
WARC	World Administrative Radio Conference
W-CDMA	Wideband Code Division Multiple Access

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