Original article
UDC: 621.398:621.316:61:681.324

Telecommunication Requirements in Telemedicine

Irini RELJIN[1]
Branimir RELJIN[2]

[1] PTT COLLEGE, BELGRADE, YUGOSLAVIA
[2] FACULTY OF ELECTRICAL ENGINEERING, UNIVERSITY OF BELGRADE, BELGRADE, YUGOSLAVIA

The paper describes basic telecommunications requirements for telemedicine purposes. Telemedicine, being multimedial in nature, requires the same resources as other multimedial services. That means transferring of large images or even videos may require very large bandwidth and high rates. Fortunately, most of the telemedicine applications are rather simple and may be fulfilled in nowadays communications networks. Having in mind demands in such serious applications, techniques guarantying quality of service, such as the ATM seem as promising solution.

KEY WORDS: *Telemedicine; Telecommunications; High-speed networks*

INTRODUCTION

Telemedicine is generally defined as the use of telecommunications and computer technologies together with medical expertise to facilitate remote health care delivery. Although there is a variety of possible applications, the purpose of telemedicine is to enable health care providers to exercise their expertise at the location of patients or other collaborating care providers using a combination of data, audio, video and externally-acquired images through the networking environment. Telemedicine applications, being multimedia in nature (text, data, audio, video), may require a sophisticated and expensive telecommunications infrastructure. Fortunately, some solutions require only a basic infrastructure to provide health-care services of good quality to remote areas. Different services/signals have different technical requirements; for example in Figure 1 the application bandwidth requirements in bits per second (bps) for different services are depicted.

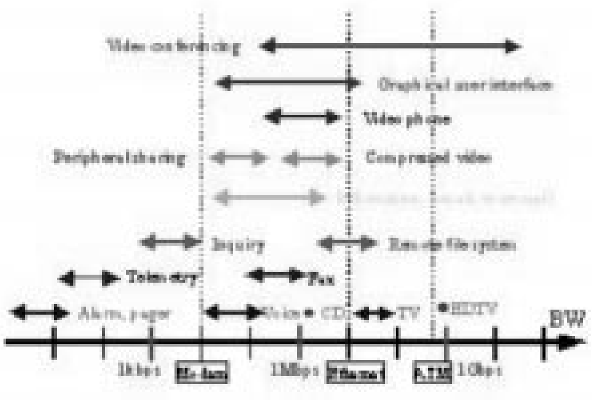

Figure 1. Application bandwidth requirements for different services

Address correspondence to:
Irini Reljin, PTT College, Zdravka Čelara 16, 11000 Belgrade, Yugoslavia

Accepted for publication: 20. 04. 2001.

Telemedicine applications can be categorized as requiring either low, medium or high bandwidth transmission. The range of network choices in telemedicine includes basic telephony, digital land lines, cellular/wireless, broadband networks such as the broadband ISDN (BISDN - Broadband Integrated Services Digital Network) with the asynchronous transfer mode (ATM), as well as satellite networks. Note that when considering telemedicine and telecommunications technologies, it is important to evaluate not only capabilities and the cost/performance trade-off, but also a general technical development.

TELECOMMUNICATION TECHNOLOGIES IN TELEMEDICINE

Typically, telecommunication technologies can be delivered on a variety of transmission media. Although standard telephone lines can support several telemedicine applications, higher bandwidth technologies are necessary for many applications. In the 1999 survey of telemedicine programs, the most common telecommunication technologies were ISDN and the T-carrier (T-1 or fractional T-1, in the USA), or E-carrier (in Europe), and the plain old telephone service (POTS). Today some new technologies, such as Digital Subscriber Line (DSL) and Asynchronous Transfer Mode (ATM), find place in telemedicine.

A. Basic telephony

It is the oldest technology used in communications. It can be delivered via a copper wire, fiber optic cable, radio (in the HF, VHF, or UHF band), microwave radio, or satellite. A typical analog copper line is used in most homes and offices for telephone and facsimile machine (fax); this technology is known as the POTS. The massive introduction of the computers in the last decade provokes the necessity of high-speed communications between different users. For this purpose relatively low-cost and robust modulator-demodulator units (modems) are used. The most common

rates for the data transmission are 28.8 kbps, 33.6 kbps while since 1998 the advanced technology has been permitting the data transmission rates of 56 kbps (although the actual rates are generally around 46 kbps). This transmission rate is suitable for audio conferencing, store-and-forward communication, Internet, and low bandwidth videophone conferencing: by applying H.324 compression algorithm videoconferencing over POTS is possible with video resolution FCIF 352x288 color pixels (7 frames per second) or QCIF 176x144 color pixels (15 frames per second).
The connection between the users can be point-to-point or point-to-multipoint, as depicted in Figure 2. The point-to point connection is either uni-directional or bi-directional and point-to-multipoint connection is uni-directional.

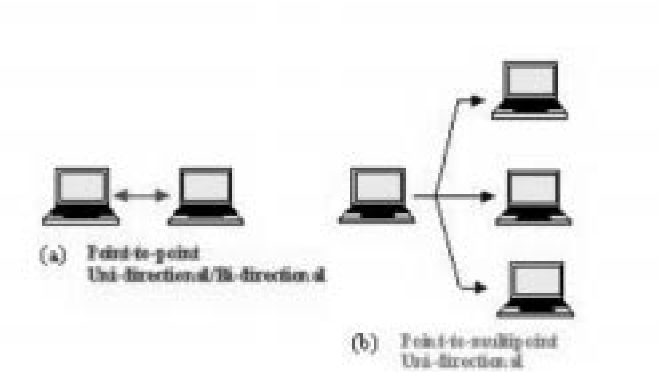

Figure 2. Different connections between users: (a) point-to-point, (b) point-to-multipoint

B. Integrated Services Digital Network (ISDN)

ISDN is a telecommunication technology that allows the transport of voice and data on-demand. Basic Rate Interface (BRI) defines an ISDN digital communications line consisting of three independent channels: two bearer (or B) channels, each at 64 kbps, and one data (or D) channel at 16 kbps, so it can carry information at nearly five times that is the fastest rate achievable using analog modems over POTS. The B channels are used for carrying the information in digital mode, whether computer data, digitized voice, or motion video. With appropriate equipment these B channels can be linked together to provide an aggregate 128 kbps data channel. Primary Rate Interface (PRI) defines an ISDN bearing 30 B channels. The ISDN is a modular standard allowing users to configure their installation to their bandwidth needs (by using inverse multiplexors different N x 128 kbps levels can be reached), and to access a wide range of additional services supplied either by the telecommunications network operator or by the third parties. Multimedia telemedicine conference systems can be delivered over ISDN networks. The main advantage of ISDN is that it is generally dial-up (like a phone). Although in some areas narrowband ISDN (N-ISDN) is not much more expensive than an analog phone, this technology is not yet available in all countries, particularly, in those of the developing world. The N-ISDN has been derived for the purpose of CBR (Constant Bit Rate) telecommunication traffic, which had been the first type of digital traffic ever produced - coding by the fixed code words. In the mean time the variable length coding with different techniques of data compression has been induced.

C. Digital Subscriber Line (DSL)

DSL is a relatively new technology that uses regular telephone lines to transmit a high volume of data at a very high speed. Namely, the telephone uses only the low-frequency part of the frequency available on these copper lines. The DSL gets more from them by splitting the spectrum of a line - using the higher frequencies for data, the lower for voice and fax. This procedure is illustrated in Figure 3, for the so-called asymmetric DSL (ADSL), probably a major DSL transmission technology. Upstream is the data stream from the customer to a network while the downstream is the data stream in the opposite direction.
Although DSL technology has been known at least for a decade it started generating serious interest only recently, due to the explosive growth of the Internet. Currently, the local Internet access is primarily available through dial-up telephone modems at speed up to 56 kbps. Soon, the customers had begun to demand even faster speed and the ability to connect direct to the Internet. As a result, industry standards for a particular type of DSL (known as a xDSL; the 'x' defines the method or protocol used to transmit data) were developed. Thanks to the new techniques of digital signal processing and the advancement of computer integration, it is possible nowadays to manufacture DSL systems. By applying DSL we can use voice-quality telephone lines for supporting video signals, because the maximum transfer speed can reach even 52 Mbps, depending on the type of the DSL service, the quality of the line and the physical distance from a central office. The DSL types are briefly described in the following text.

Figure 3. Frequency spectrum of the ADSL

- SDSL (Symmetric Digital Subscriber Line) - A technology which can transfer data at up to 2.3 Mbps both downstream and upstream over a single copper twisted-pair line, Figure 4-a. This technology has been typically deployed to small and medium size businesses. Any user that generates as much information as it consumes over the Internet should consider an SDSL solution.

- ADSL (Asymmetric Digital Subscriber Line) - A technology that delivers data faster downstream than upstream. The fastest downstream rate is 8 Mbps, while the fastest upstream rate is 640 Kbps, Figure. 4-b. However, to enable those speeds ADSL requires an installation of a special signal splitter to separate voice and data, as shown in Figure 3. ADSL connections work at distances up to 5.5 km from a central office (CO) over a single copper twisted-pair in a telephone line (the other twisted pair can still be used for POTS voice service). ADSL is the most popular form of DSL for residential users. Today's ADSL technology can automatically adapt its transmission rate to the capability of each line, thus providing the fastest speed possible.
- HDSL (High-bit-rate Digital Subscriber Line) - Delivers up to 1.544 Mbps of data symmetrically (upstream and downstream) over two copper twisted-pairs. Because HDSL provides T1/E1-level speed, telephone companies have been using HDSL for local access to T1/E1 services whenever possible. The operating range of HDSL is limited to 3.7 km, so signal repeaters are installed to extend the service farther away from the CO.
- IDSL (ISDN Digital Subscriber Line) - This form of DSL works over a copper wire that has been provisioned for ISDN. Since this is a repeater technology (repeaters can be installed in the line to boost the signal), it can deliver service up to 11 km from the CO. If someone's business or home is too far from the CO to work with other flavors of DSL, IDSL could be a promising solution. IDSL provides symmetrical speeds of 144 Kbps.
- VDSL (Very High Data Rate Digital Subscriber Line) - A technology that delivers 13 to 52 Mbps downstream and 1.5 to 2.3 Mbps upstream over a single copper twisted-pair. The operating range of VDSL is limited to 300 to 1400 m from the CO. This technology is still in development. VDSL may coexist with ADSL once the latter is more widely deployed.
- G.Lite ("gee'-dot-light") - This is a version of ADSL also known as Universal ADSL. G.Lite is expected to become the most widely installed form of DSL because it is designed to be less complex (splitterless), with less power requirements, making implementation easier and less expensive. G.Lite provides a data rate from 128 Kbps to 1.544 Mbps downstream and from 128 Kbps to 384 Kbps upstream.

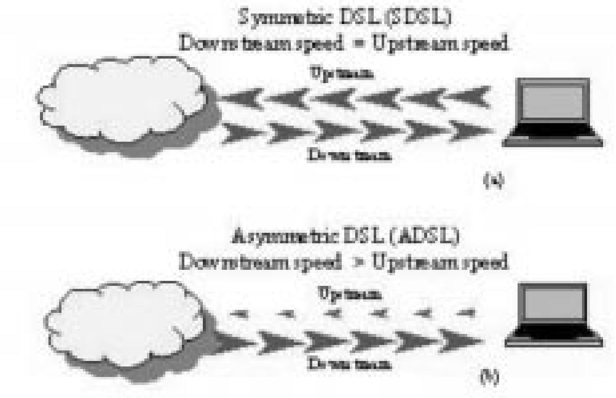

Figure 4. Difference between: (a) symmetric DSL and (b) asymmetric DSL

The frequency spectra for different xDSL technologies are depicted in Figure 5.

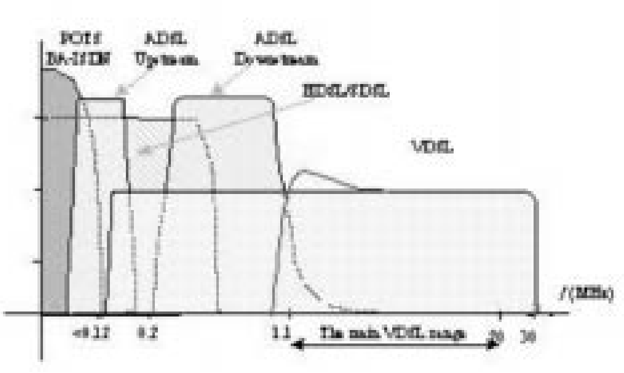

Figure 5. The frequency spectra for different xDSL technologies

DSL equipment

For connecting the end users to DSL service, for instance, the small village medical center and the university hospital, Figure 6 , the following equipment is necessary.

DSLAM (Digital Subscriber Line Access Multiplexer) - network device, usually at a telephone company CO that receives signals from multiple customer DSL connections and puts the signals on a high-speed backbone line. DSLAM enables a phone company to offer business or home users the fastest phone line technology (DSL) with the fastest backbone network technology (ATM).

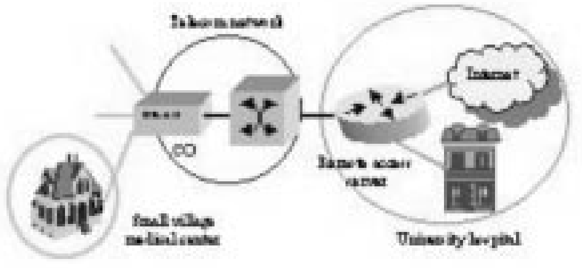

Figure 6. Connection of remote users via DSL

At the end user side a CPE (Customer Premises Equipment) is necessary. The CPE is a generic term for the array of devices connecting the end users to DSL service. The CPE must be interoperable with the DSLAM.

In ADSL installation the splitter and the microfilter are necessary, too. The splitter is a filter placed at both ends of the local loop. It splits the spectrum of a line separating the high-frequency data communications from the low-frequency voice communications. The microfilter is placed on phone lines to filter out the interference between the high and low frequencies.

D. Cellular radio

Cellular radio has been introduced into the existing network to provide coverage of certain geographically bounded areas. The information is transferred between the mobile terminal, base station and the mobile switching office, Figure 7. Base stations cover its cells that are of hexagonal shape.
Cellular radio provides mobile telephony and if used with a personal computer, a suitable modem and appropriate software, it can transmit and receive text, data and video. Some computer manufacturers offer portable computers with interfaces to cellular telephones. An explosive growth of this technology and the large areas covered by mobile phone networks make this technology very suitable for possible telemedicine applications. By using W-CDMA (Wide Band Code Division Multiple Access) standard developed by Japan's NTT DoCoMo, and adopted by the European Technology Standards Institute (ETSI) a wireless high resolution video at 2 Mbps is possible. W-CDMA could allow the base model to be split into point-to-point and multipoint; multipoint would provide video on demand, mobile TV, digital audio information delivery and mobile FM radio. The newest generation of mobile phones, known as 3G (the Third Generation) technology can support the videoconferencing at 300 kbps and achieved 25 frames per second, while it is expectable in the near future to allow video communications at 64 kbps.

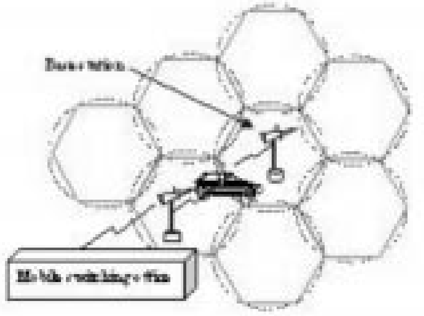

Figure 7. Overlap coverage between the cells

E. The Internet

Internet is a public-accessible, global computer network, which can be used for electronic mail, file transfer and on-line access to information services. The explosive growth of the Internet and the diversity of its services offered make the Internet very attractive for telemedicine applications, although it suffers from several drawbacks and limitations; the lack of confidentiality and the absence of guarantees quality of services are some of them. A popular Internet application is the World Wide Web (WWW), which is an easy to use system of information linked seamlessly from computer to computer by the Internet. Many Web healthcare 'home pages' are access points to medical information databases.
The Internet offers a considerable resource to the practice of telemedicine. This resource is of equal potential value to both industrialized and developing countries. The exponential growth of Internet access and usage means that patients, medical professionals and organizations can jointly benefit from wealth of information and support that is available. There is no doubt that the Internet can play an important role in improving communications and information sharing in the developing world. However, relatively few hospitals in developing countries yet have access to the Internet. Also, searching the WWW can be time consuming if only low-speed network is available.

F. Broadband ISDN with the Asynchronous Transfer Mode (ATM)

ISDN has been supposed to solve a lot of problems growing in communications. Unfortunately, ISDN initiated with the BRI or even the PRI (we use the term narrowband ISDN in both cases now) happen to be insufficient to satisfy the needs of the business (or home) world. There was a requirement for transmitting the images and video. So, telecommunication companies have been pushed to create a new technique enabling to transfer data of different, but rather wide spectra. Broadband ISDN was the solution. The real meaning of the B-ISDN should be a network providing us high-bandwidth services. From that standpoint Asynchronous Transfer Mode (ATM) may be recognized as a technique that defines and supports different high-bandwidth services. Becoming very important in nowadays communications, especially in video, the term ATM is sometimes adopted for networks implementing the ATM.
ATM is the first technology that can deliver all types of digital information (data, graphics, voice, video and multimedia) over a common network, which is usually (but not necessary) fiber optic cable. The idea for such a technique appeared in the late sixties (1968) in the Bell Labs, as a way to overcome the inefficiencies of classical Time Division Multiplexing (TDM). Namely, TDM is a synchronous technique invented during the World War II to encrypt the transatlantic radio conversations between Churchill and Roosevelt. By the early 60's, the engineers from Bell Labs had developed the first T1 Channel Banks, that combined 24 digitized voice calls over a 4 wire copper trunk between Bell central office analogue switches. A channel bank sliced a 1.544 Mbps digital signal into 8,000 separate frames, each composed of 24 contiguous bytes. Each byte represented a single telephone call

encoded into a constant bit rate signal of 64 kbps. Channel banks used a byte's fixed position (temporal alignment) in the frame to determine which call it belonged to. In Europe a similar channel bank, named E1, consisting of 32 (30 voice) channels is slightly different enabling 2.048 Mbps per access.

Though TDM is very efficient carrying delay sensitive voice traffic, it wastes bandwidth as individual time slots in the synchronous frame cannot be easily reallocated in the real time between the variable bit rate (which characterize a data transfer, particularly compressed digitized video) and constant bit rate (as voice) traffic sources. By the late sixties, the researchers at Bell Labs had the idea of introducing a label into each cell to identify traffic sources. Thus, a given call would be transported by an asynchronous series of fixed length cells identified by a virtual circuit identifier in the header, instead of being assigned to a fixed time slot. This principle, known later as cell relay, had higher overhead than TDM, but it was better to accommodate bursty traffic as computer's one was. The researchers at Bell Labs called it 'Asynchronous Time Division Multiplexing' while from the middle of 80's, after some improvements and the CCITT (transformed later to ITU-T) recommendations, the term ATM was coined.

Today, the standard switching technique for narrow band (which constitutes mainly voice connections) is using synchronous channels that are switched for the duration of the connection. The switching is realised by combinations of space and time switching stages in exchanges. In synchronous transfer mode (STM) each connection is periodically offered by a fixed-length word, Figure 8-a. This type of transfer mode is base for the Synchronous Digital Hierarchy (SDH) networks which are widely used throughout the world as transport network.

Data applications use packets (information blocks). In a packet transfer mode (ITU-T, I.113), the transmission and switching functions are achieved by packet-oriented techniques (by their address only, without any relation to time), so as to dynamically share resources among different connections, Figure 8-b.

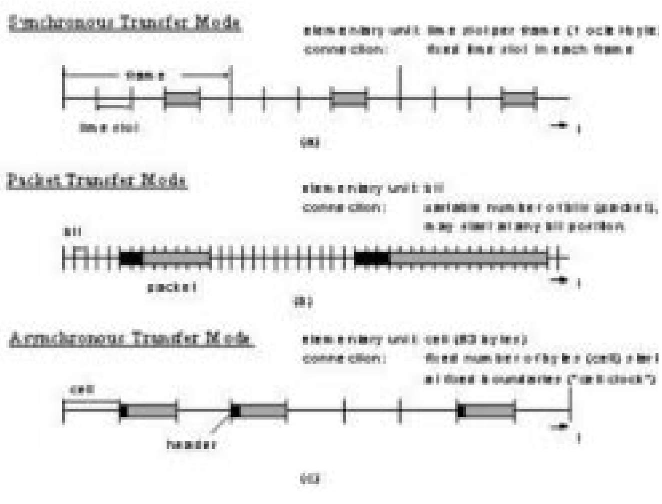

Figure 8. Comparison between the three transfer modes

In the asynchronous transfer mode all information to be transferred is packetized into fixed-size slots called cells and it operates either in connection-oriented or in connectionless mode. These cells have a 48 octet information field and a 5 octet header containing routing and control information. The transfer is asynchronous in the sense that the recurrence of cells containing the information from an individual user is not necessarily periodic, Figure. 8-c.

The ATM technology is targeted to eliminate duplicating of the hardware, as well as the software requirements, too. Thus, a single network should enable higher link efficiency, simplified operations, maintenance, services provisioning, reduced equipment costs and flexible allocation of network resources.

ATM complies with three basic requirements for future services:

Future services require high transmission speed, reaching more than 100 Mbps. It will be used for fast document transmission, fast processor connections or video transmissions.

Many services need a variable transport capacity that can be defined for each connection individually. Depending on the traffic characteristic of the service, continuous as well as packetized information have to be transmitted.

The third requirement follows the variable need for bit rates during the connection. Interactive services have phases with very high bit rates during the information transfer and phases with nearly no information flow during the information processing or view (bursty traffic). Variable bit rate coding is generating different bit rates during the connection as well.

ATM cells can be transported on different transmission systems. The only requirement is that the bit sequence independence is guaranteed meaning that no restrictions on the allowed cell information are present. ITU-T defined two options for the user-network interface, one based on the SDH and the other on pure cell multiplexing.

The source signals are packetized in the terminal - or for conventional terminals in a separate terminal adapter - into ATM-cells. In ATM-Systems (exchanges, multiplexers, concentrators) cells of different connections are statistically multiplexed. Free transmission capacity will be filled with empty cells.

In an ATM-Network, new effects show up which are not known in synchronous networks. For example it takes 6 ms until one cell of a 64 kbps data stream is filled. In the network, these cells are multiplexed with other cells and modified in the switching node. So, additional, not fixed, delays appear. For services with constant bit stream appropriate measures have to be provided.

Furthermore, cells can be lost because of bit error, buffer overflow or by action of the policing function. A lost cell represents the loss of 48 Byte of information. The source coding has to cope with this kind of errors.

One of the main advantages of the ATM technique is that a quality of service (QoS) can be adapted to the users' demands. Although its high quality performances the cost of installing and using an ATM network still is so high as to be prohibitive for most countries, although this may change in the future.

Guaranteed QoS is the great advantage of the ATM and one of the main attributes for introducing such, very complex and rather expensive, technology in communication networks. In the requesting of the connection set up, a certain QoS is specified, too. Depending on the ATM service and QoS capabilities requested, a particular mix of QoS elements are specified (such as cell loss ratio, cell delay, cell delay variation). According to this, ATM switches implement a connection admission control procedure, Figure 9.

Connection admission control (CAC)[CI1] - is the set of actions taken by the network at the call set up phase (or during call re-negotiation phase), within the control part of network nodes, in order to establish whether a virtual channel/virtual path (VC/VP) connection can be accepted or rejected (or a request for re-allocation can be accommodated). Routing is a part of CAC actions (ETSI, TR 101 287).

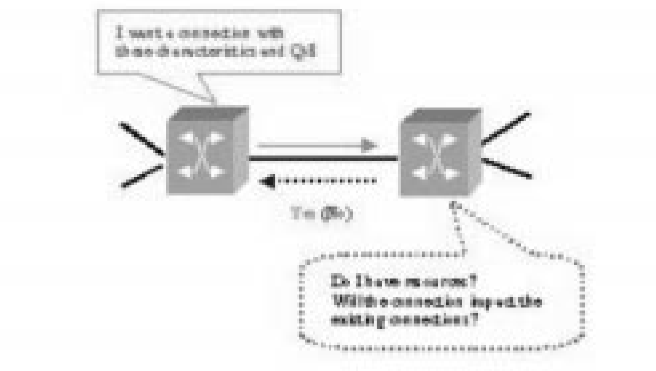

Figure 9. An illustration of the CAC procedure in the ATM

The switch accepts the connection only if violation of current guarantees are not reported. CAC is a local switch function, and it is dependent on the architecture of the switch and local decisions on the strictness of QoS guarantees.

The virtual circuit (VC) routing protocol must ensure that a connection request is routed along a path that leads to the destination and has a high probability of meeting the QoS requested in the connection set up - that is, of traversing switches whose local CAC will not reject the call.

CONNECTION TYPES

In the local area networks (LAN) the users are usually connected via the shared-line bus, Figure. 10-a. For special purpose services the users are connected via leased lines, Figure 10-b. Every customer has its own line (not shared with others) and the connection between the users is possible with a special switch.

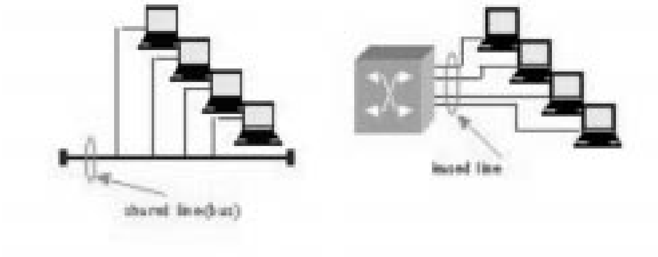

Figure 10. Users connection to: (a) shared line (bus), (b) leased line

The connection between the users can be realized on the point-to-point base, which is the most frequently implemented in video-conferencing, teleconsultation or telediagnosis. The point-to-point protocol (PPP) over ATM, for instance, is ubiquitous for dial-up remote access. It is possible to connect several users either by employing a star topology (by using the common server) or by a direct connection of each-to-each type (a mesh topology), Figure 11. For the teleeducation purposes or video-on-demand a point-to-multipoint connection is very useful. This type of connection can be realized within a network the enabling the contracted QoS.

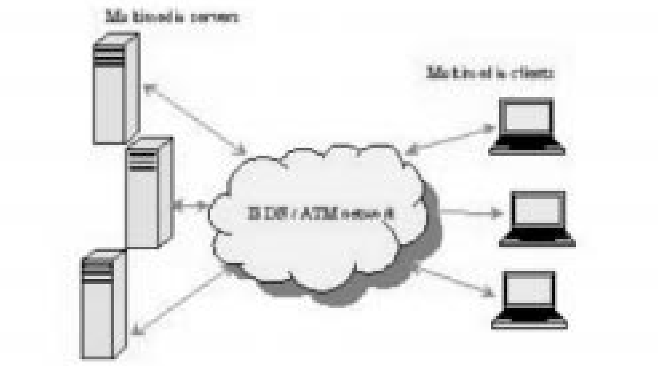

Figure 11. Direct connection: each-to-each

SATELLITE TRANSMISSION IN TELEMEDICINE

Satellite transmission, Figure 12-a , offers large amount of bandwidth, so it is very convenient for multimedia applications. The constraint comes from its rather big price. However, development of satellite earth stations providing relatively small bandwidth and using terminals of very small size (VSAT - Very Small Aperture Terminal), gives the opportunity to have a relatively inexpensive transmission. So, satellite technology can be used to deliver telemedicine services to areas that lack an advanced terrestrial network. Among the space program in the USA and the USSR, the health-care professionals in Canada and Australia began experimenting in the 1960s with satellite technology (as well as with the radio, microwave, and two-way television technology) to link isolated and rural areas to urban medical practices. Today's mobile earth stations are small, portable and they can operate from a variety of power sources, including a car battery. Their independence of fixed telecommunication and power infrastructures facilitates the continuation or restarting of health care after a disaster. Major satellite operators, INTELSAT, EUTELSAT, INMARSAT, etc. mostly use the geostationary orbit (GEO), Figure 12-b. In this orbit, the satellite appears to be stationary when viewed from the Earth. Thus, the Earth station antenna points in a fixed direction.

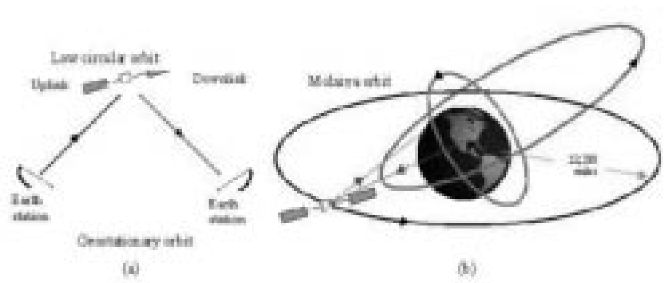

Figure 12. (a) Satellite transmission; (b) Satellite orbit types

Satellite communications are extensively used in fixed voice and data communications both in public and private networks. These networks are based on the use of geostationary satellites. The communication between the satellite and the earth station occurs at microwave frequencies (C and Ku- bands). The diameters of the earth station dish antennas vary mainly between one and ten meters depending on the system and the service. For example a significant fraction of international telephone calls are relayed via the Intelsat system. In addition to international systems there are several regional and national systems. The European EUTELSAT system is a good example of a successful regional system. Even a large international company can implement its own private satellite network for corporate communications. These networks employ VSAT technology and are therefore called VSAT networks.
Satellite TV broadcasting is by far the most well-known sector of the satellite communications. Today the general public is receiving TV programs either via CATV networks or by having its own receiving equipment. In Europe TV programs are relayed from geostationary satellites at Ku band frequencies. The operation started with analogue systems. However, a rapid transition from analogue to digital systems has begun. In Europe, there are already 22 million analogue satellite TV receivers. Within next three years ten million households is expected to be equipped with a digital satellite TV receiver.
Mobile satellite communications started with maritime communications. For this purpose the international Inmarsat organization was created in 1979. During the years the operation extended to aeronautical and land mobile communications. Today operational mobile satellite communication systems are based on the use of geostationary satellites. Early land mobile systems offered only paging and low rate data services. A modern Inmarsat mini-M notebook-size satellite terminal provides compressed voice, G3 facsimile, e-mail and data services and has almost a global coverage.
Satellite navigation is closely related to satellite communications. Today two similar satellite navigation systems, the American Global Positioning System (GPS) and the Russian Global Navigation Satellite System (GLONASS) are operational. Both systems were developed primarily for military purposes. In spite of its military character the GPS system has geared up several applications in the civil market, too. The availability of low-cost navigation receivers has been a key item in the growth of civilian GPS users.

New trends in Satellite transmission

At the moment satellite communications is experiencing a strong change. In the field of mobile satellite communications several new type satellite systems are under development. The most important systems include Iridium, ICO Global Communications, Globalstar and Odyssey. Each of these systems will include many powerful satellites that will orbit the Earth either on LEO (Low Earth Orbit) or MEO (Medium Earth Orbit) orbits, Figure 12-b. They offer voice and low rate data services all-over the globe. The user just needs a small GSM type satellite handset in order to communicate via a satellite.
A strong growth is also foreseen for Ka- band multimedia satellite systems. Several proposals have been made to implement multimedia networks which are able to provide high data rates and direct user access. For instance a computer user will benefit from a high speed access to the Internet. Most of the systems are based on powerful geostationary satellites. The communication between the satellite and the user terminal occurs at 20/30 GHz frequency band. However, also a LEO satellite constellation (Teledesic) comprising 840 satellites has been proposed for this application. Satellites are seen as a major building block of the Global Information Infrastructure.

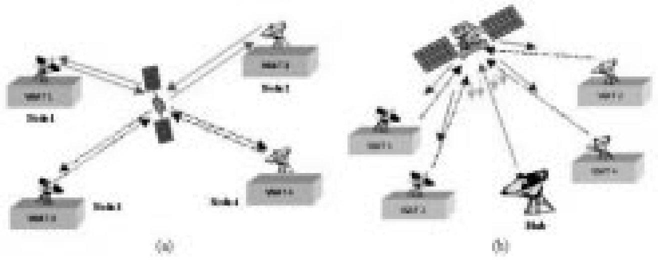

Figure 13. Satellite network of: (a) point-to-point type, (b) point-to-multipoint type

New applications mentioned above require also challenging technical development. On-board digital signal processing and spot beam antennas are of major importance in the novel satellite PCS

(Personal Communications System) systems. In a regenerative satellite the received signal from an earth station is demodulated, processed at baseband frequency and then modulated for transmission back to an earth station. Often baseband signal processing includes switching functions. In this way the satellite becomes a switching board in the sky. Some future mobile satellite communication systems will comprise also inter-satellite links. By the aid of an inter-satellite link a telephone call can be routed directly from one low orbiting satellite to another in order to achieve the shortest link between end users. These inter-satellite links operate either at microwave or at millimeterwave frequencies. In future optical links may become viable in commercial systems, too.

First satellite networks used the point-to-point architecture enabling the transmission with minimal delay between the network nodes, Figure. 13-a. Satellite provided separate bandwidth and power for every particular connection, either as Frequency Division Multiple Access (FDMA), or as Time Division Multiple Access (TDMA). Here a connection between the network nodes is provided as a single hop one. Minimizing satellite delay is a key for voice applications - thus single-hop satellite connections provide the best performance. A single-hop connection minimizes the amount of bandwidth required for a remote-to-remote traffic.

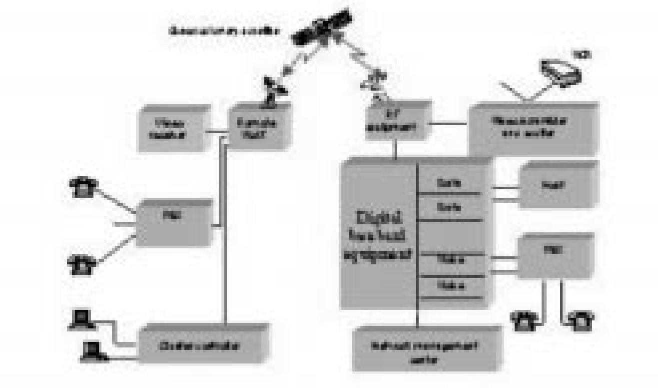

Figure 14. A VSAT network example

Networks with point-to-multipoint architecture, Figure 13-b, realize the connection in two hops. The necessary element in the network is hub station in the center of the star. Protocol support, management and data for every application is realized from the hub.

We should notice that hubs broadcast relatively powerful traffic (range of 64kbps to 2Mbps), we call it outroute transmission. On the other side, remote nodes (VSAT terminals) generate weak traffic - we call it inroutes. Several VSAT nodes share the same inroute frequency, so the traffic is to be spitted in TDMA or CDMA manner.

A typical VSAT network architecture is presented in Figure 14. Regardless the technique employed by the network, it emphasizes the way of connecting the PCs, video receivers, as well as PBX and telephone terminals. The hub represents the complete Earth station with a relatively large antenna. Remote VSATs use small antennas, with small terminal equipment, suitable for outdoor units. The main module in any hub is the baseband equipment performing all of the protocol conversions, multiplexing, modulation and multiple-access functions. Hub is the one that manage the traffic in the VSAT network. The low cost VSAT alternative is depicted in Figure 15, representing the receive-only VSAT system. It enables the reverse transmission by using low-cost landlines.

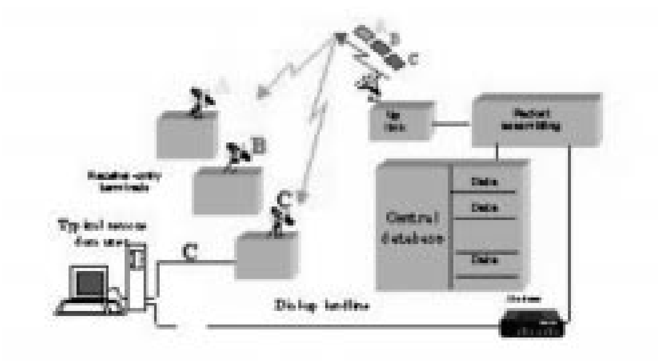

Figure 15. A VSAT network example including the receive-only terminals

CONCLUSION

Telemedicine is multimedia in nature. It sometimes produces very wide spectrum, thus requiring high-speed, low-delay networks. Different communication techniques enable these stringent requirements. Among them broadband ISDN and Internet with new type of protocols guaranteeing quality of service are expected to be used. As satellite VSAT networks are becoming less expensive, they are very often used in telemedicine, especially in northern countries.

REFERENCES

1. Bates R. Broadband Telecommunications Handbook. New York: McGraw-Hill; 2000.
2. Elbert B. The satellite communication application handbook. Boston: Artech House; 1997.
3. Acosta R, Bauer R, Krawczyk R, Reinhart R, Zernic M, Gargione F. Advanced communications technology satellite (ACTS): Four-year system performance. IEEE J. on Sel. Areas in Comm 1999;2(7):193-203.
4. Cox C, Coney T. Advanced communications technology satellite (ACTS) fade compensation protocol impact on very small-aperture terminal bit error rate performance IEEE. J on Sel. Areas in Comm 1999;2(7):173-9.
5. Abramson N. Internet access using VSATs. IEEE Comm. Magazine 2000;38(7):60-8.
6. Leiner B, Cerf V, Clark D, Kahn R, Kleinrock L, Lynch D et al. A brief history of the Internet. http://www.isoc.org/internet-history/brief.htm
7. Fowl H, Leland W. Local area network traffic characteristics with implications

for broadband network congestion management. IEEE J. on Sel. Areas in Comm 1991;9(7);1139-49.

8. Giordano S, Perazzini G, Russo F. Multimedia experiments at the University of Pisa: from videoconference to random fractals. In :Proc. INET '95, Honolulu (Hawaii), June 26-28, 1995:543-50.
9. Mertzanis I, Sfikas G, Tafazolli R, Evans B. Protocol architectures for satellite ATM broadband networks. IEEE Comm. Magazine 1999;37(3):46-54.
10. Reljin B, Reljin I. Multimedia: The impact on the teletraffic. In: Mastorakis N, Ed. Advances in Physics, Electronics and Signal Processing Applications. Danvers, MA: World Scientific and Engineering Society Press; 2000. p. 366-73.
11. McLeod D, Neumann U, Nikias C, Sawchuk A. Integrated media systems. IEEE Signal Processing Mag 1999;16(1):33-43.
12. Rao K, Bojkovic Z. Packet Video Communications over ATM Networks. NJ: Prentice-Hall; 1999.
13. Rosenberg J, Kraut R, Gomez L, Alan Buzzard C. Multimedia communications for users. IEEE Comm. Mag 1992;30(5):30-6.
14. McLeod D, Neumann U, Nikias C, Sawchuk A. Integrated media systems. IEEE Signal Processing Mag 1999;16(1):33-43.
15. Rao K, Bojkovic Z. Packet Video Communications over ATM Networks. NJ: Prentice-Hall; 1999.
16. Rosenberg R. Kraut, L. Gomez, C. Alan Buzzard. Multimedia communications for users. IEEE Comm. Mag 1992;30(5):20-36.