Networking and Electronic Highway

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Abstract

In this series of 2 lectures, we present the state-of-the art in networking technology, in particular recent advances in high-speed solutions including ATM and the Internet. In a second part, we also present an overview of the major applications on the emerging information highways, with a particular emphasis on multimedia developments and audio-video services over the Internet.[1]

Keywords: advanced networking, ATM, ADSL, highway, multimedia, Internet, audio-video

1 Introduction

The term “information superhighway” was coined by US Vice-President Al Gore, as a metaphoric reference to an infrastructure transportation program launched many years before by his father. Though the implications of the US Administration initiative for building National Information Infrastructure (NII) are the subject of animated debates, a number of countries have decided to initiate comparable programs. Note that a number of terms are used in the press to refer to information superhighways. They include, ranked by frequency of occurrence : “information highway,” “info highway,” “infobahn,” and “I-way.”

2 From Transmission to ATM

2.1 Transmission

These projects are made possible by the dramatic progress of the underlying networking technology over the past 5 to 10 years. In particular, the development of fiber optics in transmission technology — this is called lightwave transmission — has opened up a new era of communication. Fibers transmit modulated light instead of modulated electrical signals as used over metallic cables. In other words, they carry photons instead of electrons. A single optical fiber today can transmit up to 2500 million bits per second using regular electronics, and support 32 000 concurrent telephone calls. It can also support a thousand compressed digital TV channels of VCR quality at a time. Laboratory fiber systems have recently reached 1000 billion bits per second using wave-length division multiplexing techniques. New ways of exploiting copper-wire twisted pairs have also emerged, leading to performance that was difficult to predict 10 years ago: 100 megabits per second (Mbps) in conventional local area networks (LANs) and before the end of the decade, probably one gigabits per second (Gbps) over short distances.

[1] Parts of this article are reproduced from the book: Understanding Networked Multimedia [1]

2.2 B-ISDN

These developments occurred at a time were efforts were made to turn the concept of Broadband Integrated Services Digital Networks (B-ISDNs) — invented in the mid-80s by the Public Network/Telecommunications Operators (PNOs/PTOs) — to some form of reality. The idea is similar to that of narrow-band ISDN, but extending the range of data rates beyond 2 Mbps. The concept of the service is similar : subscribers can establish a variety of connections, either permanently or on a call-by-call basis through the B-ISDN network. As in N-ISDN, the network only carries digital information. But users benefit from a wide choice of bit rates, from 64 Kbps to about 0.5 Gbps. They may also choose between several levels of service quality. After several years of evaluating competitive options, the International Telecommunication Union (ITU) selected one technology to support the B-ISDN service. The winner was asynchronous transfer mode (ATM).

ATM is one of the most fashionable technologies of the decade. It tries to combine two old telecommunications ideas : the concept, borrowed from telephony, of calls to be set up prior to the use of a communication, and the concept of data packets where the flow of bits is not sent continuously, but has to be chopped into a succession of units called packets in general, and cells in ATM terminology. Apart from bit rates, a major difference with N-ISDN is that ATM does not provide naturally guaranteed transmission rates and transit delay. This is due to the intrinsically statistical nature of the asynchronous mode.

ATM technology can have three levels of usage : as an underlying transport technology used by the PNOs internally, that is invisible to the user ; as a long-distance technology brought into to the user premises ; as a technology used on site by user organizations to build or replace existing local-area networks. Business subscribers are first targeted by B-ISDN services, but individual consumers should come next, especially when applications such as video-on-demand meet a substantial market.

2.3 Cable TV operators

Another significant change in the data communications market is the arrival of cable operators, as competitors to public telecommunications operators. Though the regulations that govern this competition vary from country to country, the impact on the offering of and on the resulting services is considerable. Cable TV operators, whose initial business was to distribute TV, benefit from their past experience of dealing with high-bandwidth transmission media. The initial technology employed by cable TV operators was based on analog transmission over coaxial cables and is usually referred to as CATV. PTOs have an even sounder expertise in high-bandwidth transmission, you will object. True, but generally for their trunk circuits, not really to the home.

As a result, a number of cable TV operators offer to residential users fast access to data networks. As an example, In the USA, an east coast operator offers residential subscribers a permanent 500 Kbps link between a home computer and the Internet for a monthly charge of $ 80. To compare, using conventional modems, the access speed over the regular switched telephone network ranges from 2.4 to 28 Kbps.

3. New technologies for the final loop

A wide area networking infrastructure necessary to support the future information highway application may be seen — this is an over-simplified model — as formed of two logical components: the backbone system, generally composed of trunk connections and switching equipment, and the subscriber loop, that is the final link which connect the residential or corporate user. A number of options are proposed by operators or alliances of companies to connect to the home or the enterprise building.

One of them is called the Asymmetric Digital Subscriber Loop (ADSL), ADSL is a digital transmission technology designed to work over the regular two-wire copper telephone loop. It supports in one of its simplest version two VCR-quality channels — that is, a bit rate in the order of 3 Mbps — over about 3.2 km, and one such channel — that is, T1 speed — over 5 km. This is in the direction from the network to the home. The return channel, used for commands and user input, operates at a lower speed. This is why the technology is termed asymmetric. A more elaborated version of this technology allows 8.4 Mbps downstream over a distance of 2.7 km. Many existing pilot information highway projects use this technology. From a user perspective, installing an ADSL access may be viewed as follows: keep the existing telephone cable, unplug the telephone, plug-in the ADSL modem and the audio-video compression device, re-plug the telephone (to get the telephone service), the TV set (to get digital TV channels) and the PC (the get, say, Internet access). All this over the regular telephone wires.

A variant of ADSL, called Very High Rate Digital Subscriber Loop (VDSL) promises to offer up to 50 Mbps downstream and 20 Mbps upstream in the coming years..

The cable TV (CATV) operators may use their cable infrastructure to deliver information highway services. But the simple-tree cable TV infrastructure has to be enhanced with some form of switching capability. Traditionally, cable TV operators have delivered TV to the home over coaxial cables that terminate in set-top devices. This is for the final loop. But in many cases, the internal links, called trunks and supertrunks, have already been converted to fiber technology. In several countries, CATV operators will benefit from their wide coverage. For example, in the USA, their networks pass the door of over 95 % of the homes.

4. The Internet: the 1st step of the Information highway

Pending the deployment of a broadband world-wide Global Information Infrastructure (GII) — but will this ever exist? — a number of envisioned services are already available on the world-wide Internet, mostly through its major application, the World-Wide Web.

Indeed, the combination of the Internet as a universal transport system and the World-Wide Web as a powerful but simple to use technology for the exchange of and access to information of any type, already covers a significant fraction of the GII objectives.

- The Internet — a digital packet-based network of networks — has universal coverage and its technology can satisfy the requirements of companies to attach powerful computer centers over high-speed links as well as those of residential users connecting cheap home personal computers via telephone calls.
- The World-Wide Web allows any person at home or in a company to access remote servers of multimedia information from any regular personal computer. It supports transaction-oriented applications such as teleshopping or telebanking, commerce, interpersonal communications, audio-video cooperative work, or tele-education. It also provides a simple way for any individual to be not only an information consumer, but also an information producer by turning its computer into a server, and making available its reflections, knowledge, or art work.

4.1 Basic Internet features

The Internet is essentially a carrier system: a network of networks, controlled by no single party, and formed of hundreds of network service providers (ISPs) which have agreed to a) use the same technology —that is *IP*, the *Internet Protocol*— and b) interconnect in order to create, as seen from the user, a seamless uniform network. However, the Internet may be increasingly viewed as the universe of connected servers. Thus, the Internet is two things: a carrier system and the "digital library" accessible over it.

The Internet protocols can be classified according to two categories: *carrier* protocols —such as the Internet Protocol itself, link level, routing and service protocols— and *end-to-end* protocols —such as the transport protocols (TCP, UDP) and the application protocols (FTP, HTTP, ...)— which are only understood by the hosts and not by the intermediary routers. The main characteristic of IP is that it provides a *best-effort* delivery *connectionless* service. This means that IP packets can be sent to the network without the prior setting up of a connection, and that all packets are independent. Thus, packets may be dropped by the network, or delivered out of sequence without the end-systems being notified by the network.

4.2 Multicasting in the Internet

In data communications, *broadcasting* means the propagation of information from one source to all potential recipients. *Multicasting* means propagation of information from one source to only a subset of the potential destinations.

How can a network support a multicasting service? This can be achieved in two different ways. First the network can be *inherently* broadcast-capable, that is, using over-the-air transmission (radio, satellite, ...) or a physical medium shared between all stations, each station receiving all packets (Ethernet, Token Ring, FDDI). But other networks do not have this property that a single flow can automatically reach all potential recipients. This is the case of the Internet.

When a network is not inherently broadcast-capable, it must *replicate* the initial flow at appropriate locations within the network infrastructure. In the Internet, this is known as the *IP Multicast* facility, as defined in RFC 1112. IP multicast relies on a special packet format —the class D format— in which the destination address is not a unique number but rather a reference to a group of hosts. The IP multicast service requires within the network, the presence of *multicast routers* (also called mrouters) capable of recognizing and routing IP multicast packets. There are several routing protocols to propagate group membership from mrouter to mrouter, and route multicast packets: DVMRP, MOSPF, PIM. The creation, deletion and membership of IP multicast groups is dynamic. That is, hosts may join any group at any time without having to refer to some central Internet authority. The protocol which supports this dynamic membership facility is called IGMP (Internet Group Membership Protocol).

Historically, networks good at switching —that is capable of supporting a point-to-point bi-directional service, such as the POTS, X.25, ...— were unable to support broadcasting. Conversely, networks good at broadcasting —radio, satellite, cable-TV— were bad at supporting bi-directional communications. One of the major features of the Internet, unique in the history of telecommunications, is its ability to support both switching and world-wide broadcast services.

4.3 Mbone

Mbone is a physical experimental network based on IP multicast and implemented atop the Internet. It is an overlay network initially implemented only in the form of Unix workstation-based mrouters. Why an overlay? For two main reasons. First, not all routers in the Internet are multicast-capable. Second, not all ISPs are willing their routers to support a multicast service, which implies additional load due to the replications. The major part of the Mbone is a shared, public facility operated voluntarily by organizations interested in the Mbone initiative. Therefore, as all shared public facilities, Mbone has an etiquette to try and prevent abuses of its resources.

At any given instant, several multicast groups can be active over Mbone. How to ensure that anyone can create a group and that two groups are not given the same address, as there is no central authority to allocate on-demand group addresses? In fact, address collisions are possible, but their probability is extremely low if a) the hosts select group addresses randomly and b) only a small fraction of the entire address space — in practice the square root of the total number of addresses — is used at any given instant.

But Mbone is not only a network. It is also increasingly understood as a set of public domain software tools operating over the Mbone infrastructure and supporting audio, video, whiteboard or session directory services.

5. Information highway: the services to residential subscribers

Before describing the modern (audio-video) applications available today over the Internet, let us return for a while to the more ambitious information highway programs. In fact, the nature of the services that should be provided to residential will depend on the evolution of the technology, but will also be dictated by the industrial policies of governments.

The services that may eventually be available to residential users over the superhighway termination links include :

- Conventional telecommunications services

  Plain old telephone service (POTS) and telefax.

- Improved communications between people

  Videophony via existing TV

  Multimedia electronic messaging or on-line forums for residential subscribers.

- Entertainment

  TV broadcasts in analog mode (hybrid approach)

  TV broadcasts in digital mode, often referred to as the “500 channels " program

  High-definition TV broadcasts

  Pay-per-view digital video programs on demand

  Full movies-on-demand with VCR-style control

  Remote computer games possibly multiplayers and with three-dimensional effects.

- Education and culture, general information

  Generalized access to public Internet servers, via simple user interfaces such as WWW

  Customized and multimedia-based learning

Access to cultural information, exploration of museums.

- General information

  Topical or general news on demand (sports, finance, weather, . . .)

  Topical or general static information on demand (administrative data, geographic information, . . .).

- Transactions

  Teleshopping

  Telebanking

  Teleticketing (booking at a distance).

- Advanced presentation of information

  Virtual reality advertising (remote exploration of virtual kitchens, houses, buildings).

Several operators are also planning to integrate regular analog broadcast television channels and digital services over the same access to the home.

6. Requirements of audio and video applications

“*Multimedia is the field concerned with the computer-controlled integration of text, graphics, still and moving images, animation, sounds and any other medium where every type of information can be represented, stored, transmitted and processed digitally*” (excerpt from [1]). Out of these various information types, audio and video —that is continuous media— are those which place the most stringent requirements on the underlying network.

To support audio and video, the two most critical parameters of the quality of service to be provided by the network are the bit rate of the connection and the variation of the transit delay. For audio, the bit rate varies from 5-7 Kbps (in the case of low-quality speech) to 32-64 Kbps for regular telephone-quality and 250 Kbps for CD-quality compressed sound. Motion video requires from 64-100 Kbps for low resolution/very low frame rate to 1.5 Mbps for compressed VCR-quality TV, and in the order of 4-7 Mbps for compressed broadcast-quality TV.

Another key parameter is the variation of the transit delay, in particular for the transmission of sound. Basically, if packets carrying sound arrive after widely varying transit delays, the only solution to overcome this is for the receiving system to wait a sufficient time, called the offset delay, before the playout, so that most of the delayed packets are given a chance to arrive in time. Incoming packets are stored in a buffer before being played out smoothly after the offset delay. As the technique may add a substantial component to the overall latency, it must be used with care in interactive applications.

7. Internet transport protocols for audio and video

We explained that the Internet provides a best-effort packet delivery service. In practice, loosing a few percent of the packets is often considered as acceptable. The regular *Internet Transport Protocol* (TCP) is designed to recover from such packet losses. TCP uses a mechanism of message numbering and windowing for *flow control*, and relies on retransmissions for *error recovery*. It is connection-oriented —that is, the two end-systems agree to set up an end-to-end connection before the first application message can be sent —

and uses mechanisms such a *“slow-start”* to avoid flooding the network at the beginning of new transfers.

Flow-control, retransmissions and slow-start are all mechanisms which cannot be used with continuous media where the time dependencies between packets of the stream have to be respected. Therefore, audio and video applications use another Internet transport protocol: the *User Datagram Protocol* (UDP), which is connectionless and implements no error detection/recovery mechanism. With UDP, datagrams are unnumbered and sent blindly by the source, regardless of the possible internal losses.

But UDP is not sufficient for audio-video applications. Mechanisms for time-stamping are necessary so that timing relationships between messages can be reconstructed at the receiving end. In addition, the sender may wish to know whether packets are well received by the destination(s), so that it can possibly reduce the bit rate if a too high fraction of the packet stream is lost by the network. These additional facilities are provided by a session protocol. Most applications now use, as session protocol, *the Real-Time Protocol version 2* (RTP2), which is often embedded within the application software itself.

8. Multimedia Internet applications

8.1 Interpersonal Applications

The Internet now supports a complete range of facilities for communications between people, either in the corporate or private environment. This includes telephony, videophony, videoconferencing and, for professional use, integrated tools *for computer supported cooperative work (CSCW).*

Internet *telephony* is particularly attractive for long-distance calls. Numerous products and public-domain implementations are available, including CU-SeeMe, Digiphone, Internet Phone, Netphone, PGPfone, Speak Freely, WebPhone, WebTalk. The main Mbone public-domain software for Internet telephony is *vat*, the *visual audio tool*. With Internet telephony, the quality of the sound may be poor, in particular when sections of the networks are overloaded on the path between the two communicating hosts, but it may also be amazingly good, close to that of the regular telephone conversations. The advantages of Internet telephony are the low incremental cost, the possible integration with email and other Internet applications, the possibility of call recording, play back, voice-mail and multiparty voice conferencing, as well and the improved privacy when encryption is used. The major difficulty lies in interoperability between the various products or implementations. Other drawbacks include the unpredictable sound quality due to changing network conditions, and the fact that no call is possible to powered-off systems.

Videophony is audio-visual communications between two people only, whereas *videoconferencing* is understood as communications between groups of people, either two groups only (biparty videoconferencing) or more than two groups (multiparty). Internet supports both videophony and videoconferencing. A major issue with Internet is: how to implement a *multiparty* session? There are three main categories of implementations currently in use. The first one uses *IP multicast* from the source. Thus, the source generates only one flow, and the IP multicast network looks after the internal replications. Such a solution implies two things. First, the hosts must be capable of generating IP multicast packets —not all systems are; second, a multicast network must exist, either a private one or the Mbone. The Mbone audio and video tools (nv, vat, vic, ivs, ...) use this technique. The second solution relies on *application-level* replicators, that is dedicated hosts which can replicate the flow of data for a particular audio or video application.

Numerous products or public-domain implementations exist today, including Being There, Communique!, CU-SeeMe, DV100, FreeVue, InPerson, InVision, IVS, nv, Picture Window, ShowMe, vic, VidCall. A major advantage of Internet videoconferencing is that, if IP multicast is used, the cost of a session does not increase linearly with the number of participants, as with conventional ISDN-based video-codecs. However, the frame rate has often to be reduced to a couple of frames per second, due to the limited bandwidth between certain pairs of participants. As for telephony, the major issue lies in interoperability between products or implementations. Many differing video encoding and compression schemes are used, including MPEG, native nv, CU-SeeMe, motion JPEG, H.261 and Intra-H.261. However, transcoding software capable of translating on-the-fly between certain pairs of encoding schemes has appeared, the most advanced being the *video-gateway tool* (vgw) from Lawrence Berkeley Laboratory (LBL).

The large-scale distribution of audio-video is another major class of applications over the Internet. This is a simple extrapolation of audio or videoconferencing based on IP multicast. With this class, the type of communications between parties is asymmetric: there is one particular site —such as a lecture room, a meeting room at the head-quarter of a company— where a particular event is taking place, followed passively or semi-passively by a possibly very large number of sites. The Mbone tools (nv, vat, vic, ivs, wb, vgw) are generally used to support such applications.

8.2 Server-Based Applications

Another class of multimedia applications over the Internet is where the communications no longer involve only people, but rather *people* communicating with computer-based *servers*.

The extreme case of audio-video distribution that we discussed above is pure *broadcasting* where public servers generate flows to anyone interested, without expecting any feedback from the receivers. These sources emulate in practice radio or TV channels. Over regional parts of the Internet operating at very high speed —such as over regional ATM-based pilot infrastructures— these channels may have a quality close to that of broadcast television but generally, the quality is not better than a few frames per second at a quarter the resolution of regular TV.

Users are informed of Internet multicast programs by the *session directory* (sd) tool, a public-domain software supported by LBL. sd is in practice the Internet radio/TV guide. Receivers may select the program (called *session*) they are interested in and a simple click triggers the group membership process and launches the necessary application. Then, the IGMP and DVMRP protocols ensure that the information is propagated over the network. sd can also be used to *create* new sessions in the directory, not only to join existing ones.

Another category of server-based applications is audio- or video-on-demand. There, users access to the server, usually via a World-Wide Web interface, to request a particular audio or video program which is then transmitted to their computer by means of a *dedicated* packet stream. In other words, each recipient requires a private incoming packet stream, even though many users are receiving the same program at the same instant. Clearly, with the current bandwidth limitations of the Internet, this approach can not scale, but it is interesting to validate the concept and part of the technology.

From nearly the inception of the World-Wide Web, servers have supported access to audio or video sequences. Initially, these sequences had to be downloaded first on the client computer before being played out. Now, servers increasingly support access to real-time-transfer audio and video (also called streamed audio-video). In most case, the client browser launches an external software application (called an “helper”) to perform the reception, decoding and playout of the received stream. But clearly the future browsers will integrate these services, thus providing a unique interface to all Internet services including the support of real-time audio and video.

8.3 Technical issues and recent progress

The real-time support of continuous media over the Internet still faces technical challenges, several of them being unresolved to date. However, research in these fields progresses fast and solutions are in sight.

One of these challenges lies in evolving the Mbone into a world-wide and large-scale production network. The routing protocol currently in use, *DVMRP*, relies on mechanisms which can not easily scale up. For example, when a new session starts, an initial broadcast of the flow is triggered, pending a pruning mechanism reduces the distribution tree to only those sections of the network where the flow is actually requested. This process of broadcast followed by pruning is regularly repeated over the lifetime of any active session. Solutions under consideration include *hierarchical routing* —where the Mbone would be divided into sub-regions— and the *Protocol Independent Multicast* (PIM) based on a concept of rendezvous points.

Another major challenge is to improve the quality of service for video and audio applications. One of the problems is to better react to congestion situations. *Appropriate queuing* is one of the techniques which can be used immediately, in conjunction with others to alleviate the problem. This consists for Internet routers of discriminating different traffic flows and giving higher priority to real-time flows. Another approach lies in the use of reservation mechanisms. The *Resource Reservation Protocol* (RSVP) is an Internet Draft which allows the reservation over a path of internal resources for simplex data streams. RSVP is compatible with IP version 4 and 6 and with the existing Internet routing and multicast protocols. Unlike with ATM, Frame Relay or X.25, RSVP does not rely on the setting up of hard connections. It is also *recipient-driven*, which means that this is the destination host, not the source, which decides upon which amount of traffic it is willing/capable of receiving and displaying. RSVP is particularly well suited for multicast streams. Implementations already exist and have been demonstrated at exhibitions.

9. Conclusion

The ambitious political plans to set up an orderly global information infrastructure seems to be overtaken by the rapid, anarchic and largely out-of-control explosion of the Internet. The technology to make the Internet the true information highway of the future is not yet entirely available but it progresses fast. More progress is expected over the coming years for a better support of audio and video. IPv6 —the new version of the IP protocol, which implements a new concept of flows that can be matched to quality of services— will contribute to these improvements. But clearly, all these efforts may remain vain if more raw bandwidth is not allocated to the Internet. When considering the huge amount of bandwidth currently installed—and widely idle— for telephony (how many hours per month do we use our telephone?), allocating a tiny fraction of it to the Internet does not appear impossible. It seems to be a matter of political will.

References

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